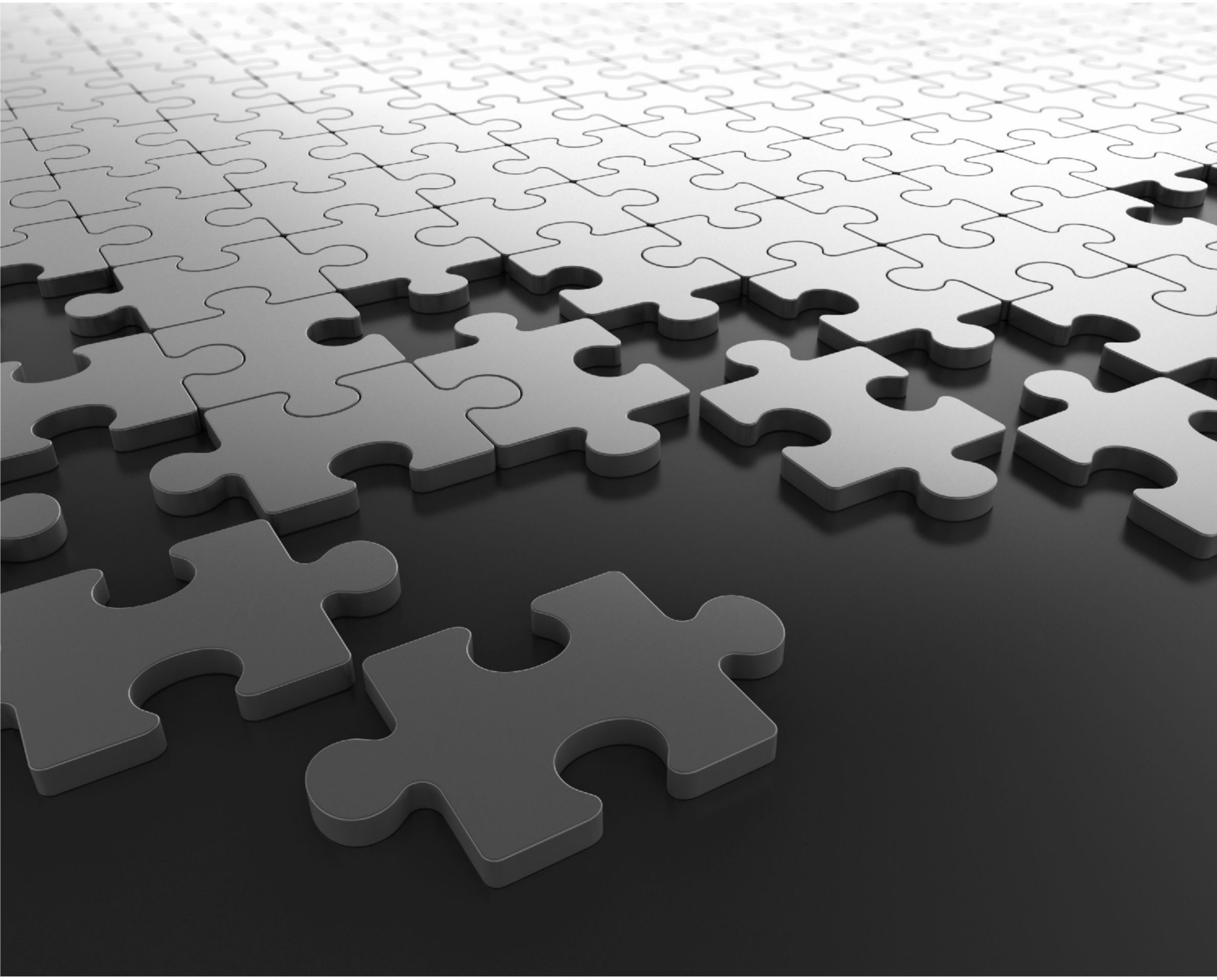
ANANT UCS
System Manual



MATRIX
TELECOM SOLUTIONS

ANANT UCS

Unified Communication Server

System Manual

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This is a general documentation of the product. The product may not support all the features and facilities described in the documentation.

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CHAPTER 1

Introduction

Welcome

Thank you for choosing Matrix ANANT UCS! We hope you will make optimum use of this intelligent and versatile Unified Communication Server. Please read this document carefully to get acquainted with the product before installing and operating it.

About this System Manual

This System Manual includes the description of ANANT UCS and step-by-step instructions for configuring and accessing different features and facilities offered by it. It also provides essential guidelines for accessing, maintaining and monitoring the Unified Communication Server.

This document also aims to provide important information and instructions related to ANANT so that you get properly acquainted with the Unified Communication Server and can use it efficiently and effectively.

The installation and deployment procedure of ANANT Unified Communication Server is documented in **ANANT UCS Installation Guide**. Similarly, the instructions and guidelines for setting up and operating ANANT UCS in hotels and health care establishments is documented in **ANANT UCS Hospitality System Manual**. You can download the documents from https://www.matrixcomsec.com/support/telecom-product-manuals/

Intended Audience

This System Manual is aimed at:

- **System Engineers**, who will install, maintain and support the server. System Engineers are persons who customize the system configuration to meet the requirements of the organization/users. It is assumed that they are experienced in installing the Unified Communication Server, are familiar with various technical terms and functions associated with it. The SE must have undergone training in configuring ANANT UCS.

  No one, other than the System Engineer is permitted to make any alterations to the configuration of ANANT UCS.

- **System Administrators**, are the people administering the server. Generally an operator/receptionist in an organization, or the staff manning the reception or front desk area of the establishment are selected as System Administrators.

  It is assumed that the System Administrators have some previous experience in administering the system and its Terminals and Consoles. The System Administrators are not expected to install and configure the

Unified Communication Server, but only the routine jobs and features that are specific to them like generating SMDR reports, Setting report filters, Setting Alarms, reminders, etc.

- **Users**, are people/organizations who will use the features and facilities of ANANT UCS. They may be personnel executives, include personnel of small and medium businesses, large enterprises, front desk and service staff of Hotels/Motels, hospitals, and other commercial and public organizations/institutions.

Organization of this Document

This system manual is broadly divided into the following:

- **Introduction:** This chapter provides an overview of this document, its purpose, intended audience, organization, terms and conventions used to present information and instructions.
- **Product Overview:** This chapter provides the description of the server features and facilities, supported licenses and the different interfaces supported by the server.
- **Connecting SIP Extensions:** This chapter gives step-by-step instructions for connecting the SIP extensions in the network.
- **Configuring ANANT UCS Settings:** This chapter contains description of how to configure ANANT UCS.
- **Configuring Extensions:** This chapter provides detailed description of how to configure the various extensions supported by ANANT UCS. This chapter also provides the instructions for configuring the basic parameters in ANANT.
- **Configuring Trunks:** This chapter provides detailed description of how to configure the SIP Trunks.
- **Configuring Voice Mail system:** This chapter provides detailed description of how to configure the Voice Mail System.
- **Features and Facilities:** This chapter describes in detail, each feature and facility offered by ANANT UCS. This includes a description of the feature/facility, how it works, and how to configure the feature/facility.

  The feature description is arranged alphabetically by Feature Name to make it easy for you to locate the description you want to look up.
- **Voice Mail Features:** This chapter describes in detail, the voice mail features offered by ANANT.
- **System Maintenance:** This chapter provides instructions for back-up, generating reports and debugging.
- **Status:** This chapter describes the status of the System, Network, SIP Trunks, SIP Extensions and other related parameters.

How to Read this System Manual

This System Manual is organized in such a way that you will find all the information you need easily and quickly.

You may use the **Table of Contents** and the **Index** to navigate through this document to the relevant topic or information you want to look up.

Cross-references are provided in blue font with hyperlinks. You can look up the source by clicking the links.

Instructions

The instructions in this document are written in a numbered, step-by-step format, as follows. Each step, its outcome and indication/notification, wherever they occur, have been described.

Access Codes

Access codes are strings of digits dialed by an extension to

- Call another extension, Department Group
- Grab a trunk line
- Use a Feature, like Call Transfer, Call Forward.

The Access Codes provided in the instructions throughout this document, are default access codes. It is possible to change the Access Codes according to user requirement and preferences. Verify with the Installer/System Engineer, if the default Access Codes have been changed, and use the access codes configured by the System Engineer. For more information, read the topic "Access Codes" in this document.

Notices

The following symbols have been used for notices to draw your attention to important points.

Important: *to indicate something that requires your special attention or to remind you of something you might need to do when you are using the system.*

Caution: *to indicate an action or condition that is likely to result in malfunction or damage to the system or your property.*

Warning: *to indicate a hazard or an action that will cause damage to the system and or cause bodily harm to the user.*

Tip: *to indicate a helpful hint giving you an alternative way to operate the system or carry out a procedure, or use a feature more efficiently.*

Terminology used in this System Manual

The technical terms and Acronyms used in this Manual are standard terms, commonly used in the telecommunications and data communications industry. Considering the broad group of intended users of this manual, wherever possible, use of jargon has been avoided.

Acronyms have been defined in the text and a list of the same is appended.

Some of the terms specific to this Manual that you will encounter are defined below:

The words '**UCS**', **'ANANT UCS'**, **'Server'** and **'System'** are used interchangeably and synonymously to mean ANANT Unified Communication Server.

- **SIP Phone/ IP Phone:** These words are used interchangeably and synonymously to mean IP Phones connected to ANANT UCS. A SIP Phone/ IP Phone is a phone that uses the VoIP Technologies for placing and transmitting calls over an IP network, such as the Internet.

- **SIP Extensions:** Any Proprietary IP Phone of Matrix (SPARSH VP330, SPARSH VP310, SPARSH VP 510, SPARSH VP248, SPARSH VP710, SPARSH VP210) or Matrix UC Client (Matrix VARTA WIN200, Matrix VARTA ADR100, Matrix VARTA AMP100), a Standard SIP Phone, a Soft Phone or an Analog Terminal Adapter — registered with the server, from which you can make/receive calls to any extension or external number.

- **SIP Trunking:** VoIP or streaming media service based on Session Initiation Protocol (SIP) by which Internet Telephony Initiation Protocols (ITSPs) deliver telephonic services and Unified communications to the customers equipped with SIP-based System.

- **Extension:** It is the soft port of the system on which an IP Phone is registered (hard or soft).

- **Mobile Extension:** A mobile/landline phone used as a remote extension of ANANT UCS. You can access all the features of an extension of ANANT UCS from the mobile/landline phone.

- **Station:** same as extension.

- **Service Provider:** The providers of Internet, that is, Internet Telephony Service Providers (ITSP).

- **Called party/Callee:** The person to whom the call is made.

- **Calling party/Caller:** The person who makes a call.

- **Enterprise Application/Features:** Pertaining to the general and special telephone and call management features required by business establishments, public and private organizations.

- **Internal numbers:** same as extension numbers.

- **External Numbers:** Numbers of parties/individuals outside the UCS network. The unique number string given to subscribers of PSTN, PLMN, ITSP, etc.

- **Internal Calls:** Calls made from and received by one extension to another extension of the ANANT UCS.

- **External Calls:** Calls made by users of ANANT UCS to subscribers of PSTN, PLMN, ITSPs, etc.

- **ITSP:** An Internet service provider that provides the facility of SIP Trunking over VoIP Technology.

- **Hospitality Application/ Features:** Pertaining to the special telephone and guest/patient management features required by accommodation establishments like hotels and hospitals.

Using this Manual, we hope, you will be able to make optimum use of this feature packed Unified Communication Server. If you encounter any technical problems, please contact your Dealer/reseller or the Matrix Customer Care.

CHAPTER 2

Product Overview

Today's dynamic enterprise environment not only requires a solution that provides seamless connectivity across the globe, but also eye's upon a solution that is apt for their diversified workforce. Moreover, the enterprises also look for a solution that provides the UC experience to both, the in-office workforce as well as the mobile workforce.

With the growing competition, round-the-clock connectivity and flexible endpoints is the need of an hour of the mobile workforce for consistent in-office and remote work experience.

This increasing need of flexible work places thrives the need for a smarter collaborative communication solution. To accomplish the collaborative need of Modern Enterprises, Matrix offers ANANT Unified Communication Server.

ANANT UCS is a Business Unified Communication that aims to bring the individuals located at diverse location, on a common communication platform for the real-time communication and collaboration.

ANANT UCS unifies all the communication networks and devices to provide individuals the flexibility of accessing the calls, messaging and voicemails from any of the devices irrespective of their location, thus making it an ideal solution for Small, Medium and Large Businesses, Hotel-Motel Industries, Hospitals, Manufacturing Units, Retail Chains and Financial Institutions.

With Collaboration and Mobility as the key aspect of ANANT UCS, we are moving a step ahead to meet the day-to-to-day communicative needs of all the modern enterprises.

Benefits of ANANT Unified Communication Server

Seamless Connectivity with the Globe: ANANT UCS allows individuals located at any geographical location to communicate and collaborative seamlessly. Individuals can use the UC features like Instant Messaging, Call Management, Smart Directory and Video Calling to connect with their remote peers.

Enhanced Proficiency and Productivity: ANANT UCS offers collaboration, communication, messaging and mobility for both Small as well as Large Enterprises, thus allowing individuals to connect with their colleagues within seconds. This allows them to work collaboratively with each other, increasing the overall proficiency and productivity of an organization.

Economical and Smart Investment: The key features like Auto Attendant, Presence, SMDR Storage with a large capacity, that normally require additional investment in most of the other brands, are in-built in ANANT.

Other Intelligent features like CLI/DDI based Routing and Dial by Name ensure efficient call management and prompt response to the callers. The advanced features like Least Cost Routing and Call Budgeting helps to reduce the communication cost and enhance the productivity. Further, the routing features like Fixed or Least Cost Routing can also be selected to route the outgoing calls.

ANANT UCS can also work as an adjunct to your existing telephony infrastructure, as a Gateway, saving the cost of equipment replacement, giving you the IP Connectivity and a host of intelligent features, thus making ANANT UCS an economical and smart investment.

Durability and Reliability: A reliable server is the key requirement of all the modern organizations and to fulfill this paramount need of the organizations, ANANT UCS supports redundancy. Redundancy is the solution to any kind of unplanned system downtime and system failures.

ANANT UCS supports redundancy between the two servers — Primary Server and Backup Server. When the Primary Server fails, the Backup Server takes over the control and becomes active, thus providing uninterrupted connectivity with the globe.

Universal Connectivity via SIP Trunking: In this diversified technological era, organizations generally seek for a communication solution with a SIP Interface. Keeping the same in mind, ANANT UCS is designed with a provision of SIP Trunking. The support of SIP Trunk Services in ANANT, allows organizations to establish universal connectivity at desired time and space.

Communication and Collaboration using SIP Extensions: SIP Extensions are the first and foremost choice of most of the modern organizations. To meet this requirement, ANANT UCS supports SIP Extensions. You can registered any proprietary IP Phones of Matrix (SPARSH VP330, SPARSH VP310, SPARSH VP510, SPARSH VP248, SPARSH VP710, SPARSH VP210) or Matrix UC Clients (VARTA WIN200, VARTA ADR100, VARTA AMP100) or Standard SIP Phones or Softphone or Analog Terminal Adapter with ANANT UCS as per your requirement.

Flexibility and Mobility: The tight integration of the enterprise features of ANANT with the Matrix UC Clients - VARTA ADR100, VARTA AMP100 and VARTA WIN200 allows the mobile workforces to connect with their colleagues across the world. The Mobile workforces can use any Wi-Fi or cellular data networks to stay connected with business communication while working from office, home or traveling.

Listed below are the important UC Features and Key Features of ANANT UCS:

UC Features

- Auto-attendant with configurable call-flow
- Presence based call controlling (make call/reject call)
- Smart Directory access using VARTA WIN200, VARTA ADR100 and VARTA AMP100 for easy and quick access to the extensions and other contacts
- Mobile and Remote workers support
- Set/View Presence
- Video Calling
- IM using VARTA WIN200, VARTA ADR100 and VARTA AMP100
- IM using third party SIP Phones/soft clients

Key Features

- Account Codes
- Auto Black List IP Address for Web access, VoIP access, etc.
- Call Billing
- Class of Service
- CLI Based Routing
- Closed User Group
- Simultaneous conferences with 3 parties
- Emergency Call Detection and Reporting
- Firmware and Configuration file upload using browser

- Internal SMDR Report buffer
- Least Cost Routing
- Logical Partitioning for restricting toll bypass
- Priority
- Syslog for Logs and Debugs
- System Activity Log
- System Fault Log
- Toll Control
- Trusted IP Address to avoid unauthorized access
- Voice Mail
- Web-based Programming, Jeeves

The Interfaces

ANANT UCS supports the following interfaces.

VoIP Interface

The Voice-over-IP (VoIP) Interface connects ANANT UCS with the internet or the private IP network. It enables ANANT UCS to handle incoming and outgoing calls over the IP network. It offers great flexibility and mobility to the users for making or receiving calls across the world.

The VoIP Interface supports UC Clients, SIP Extensions and SIP Trunks. With SIP Trunks, users can make IP calls using the SIP Server of the Internet Telephony Service Providers (ITSPs).

The Registrar Server of ANANT allows any SIP enabled device like a UC/SIP Soft Client or an IP-Phone to register with it and function as the 'SIP Extension' of ANANT. The SIP Extension users can make and receive calls to any extension user of ANANT as well as any external numbers over VoIP. With SIP Extensions, organizations can communicate and stay connected at the lowest cost without any geographical restrictions.

The VoIP Interface supports adaptive jitter buffer for reducing delay and improving speech quality.

The key features of the VoIP Interface are:

- Up to 99 SIP Trunks - for Proxy as well as Peer-to-Peer (non-Proxy) calls
- Up to 5000 SIP Extensions
- Upto 1024 DRTP/ RTP Relay Audio Calls
- Upto 102 DRTP/ RTP Relay Video Calls
- Upto 1024[1] simultaneous Transcoding calls
- Selectable Network Assignment (Connection Type) - Static IP and DHCP
- Selectable DNS - Automatic and Static
- Dynamic DNS for SIP devices in public network
- STUN
- TCP and UDP NAT Keep Alive
- VLAN
- Symmetric RTP Selection
- Send CLI Option for outgoing calls
- Selectable DTMF - RTP (RFC 2833), SIP Info
- Flash Detection using SIP INFO and RFC2833
- Broad Voice Codec Selection: G.729 AB, iLBC - 30 ms, iLBC - 20 ms, G. 711 μ-Law and G. 711 A-Law, G.722, G.723[2]
- Quality of Service - SIP DiffServe/ToS, Voice DiffServe/ToS, Video DiffServe/ToS for UC Clients
- Quality of Service - SIP DiffServe/ToS, RTP DiffServe/ToS
- Voice Mail Subscription on SIP Extensions
- Busy Lamp Field Subscription for SIP Trunks
- Upto 10 Call Appearances on SIP Extensions
- Registration of SIP Extensions from 3 different locations and Shared Call Appearance
- Sharing the same SIP Extension number among a maximum of 3 different SIP devices using Share Call Appearance functionality.

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1. *The maximum number of Transcoding calls supported in the system depends on the server configuration and system load. To know more, refer to the topic Technical Specifications of ANANT UCS in ANANT UCS Installation Guide.*
2. *The option to select G.723 as one of the voice codec is allowed only when the RTP mode is set as RTP Relay or DRTP.*

SIP Trunks

ANANT UCS supports a maximum of 99 SIP Trunks, allowing you to subscribe to as many as 99 different Internet Telephony Service Providers (ITSP) or peer-to-peer connectivity with multiple end SIP devices.

SIP Extensions

ANANT UCS supports 5000 SIP Extensions. Any SIP-enabled device like an IP Phone, a Softphone, Analog Terminal Adapter can be registered and function as the 'SIP Extension' of ANANT UCS.

SIP Extension users can make and receive calls from and to other extensions of System and external numbers over SIP trunks. You can also connect the Standard and Extended IP Phones offered by Matrix as SIP Extensions.

A SIP Extension can be registered with ANANT UCS from three different locations. This helps the organization to overcome geographical distances and reduce call costs.

Voice Mail System (VMS)

ANANT UCS application supports Voice Mail System to provide mailbox facility to all its extensions users. The Voice Mail System also forms the basis of other features like Conversation Recording and Call Taping.

Each Mailbox has the capacity of storing 10,000 voice messages. The maximum size of each Mailbox is 60,000 minutes. By default, the size of each Mailbox is set to 5 minutes. The maximum Message Length for each Mailbox is 3600 seconds. By default, the Maximum Message Length for each Mailbox is set to 120 seconds.

The key Voice Mail and Auto Attendant features supported by ANANT UCS are:

- Configurable Mailbox Size
- Configurable Message Length
- Welcome Greetings according to the time of the day
- Different voice greetings for different time zones
- Special greetings for holidays
- Five call transfer types: none, blind, wait for ring, wait for answer, and screened
- Dial by extension
- Dial by name
- Personalized greetings for each mailbox
- Individual mailbox size
- Call forward to voice mail
- Message forwarding
- Distribution lists
- Broadcast message
- Message Wait Notification
- Redirecting messages
- Message Wait Notification via Email and Call

ANANT UCS's Voice Mail System also forms the basis of other features like:

- Conversation Recording
- Call Taping
- Voice-guided Wake-up Calls and Reminders
- Message Wait Notification
- Call Transfer to Mailbox
- Call Forward to Voice Mail
- Department Calls - Mailbox for Department Groups

Ethernet Ports

ANANT UCS supports the WAN as well as the LAN interface.

The WAN Port is provided to connect:

- a Switch/Hub/Router/Modem.
- the public network over a Router/Modem. Any user on the public network can be registered as SIP Extension through the WAN Port.
- set up and run software applications such as PMS and CAS on any PC.
- generate Station Message Detail Record (SMDR) Reports on any PC.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

The LAN Port is provided to connect:

- the system to a PC or a private area network.
- the private area network to register SIP extensions through the LAN Port.
- set up and run software applications such as PMS and CAS on any PC.
- generate Station Message Detail Record (SMDR) Reports on any PC.
- capture "System Activity Log", "System Fault Log" and Hotel Motel Activity Log.

However, the provision of a WAN and a LAN Interface purely depends on the number of the Ethernet Ports present on the Bare Machine on which ANANT is installed.

If you want the provision of a LAN as well as a WAN Interface, purchase the Integrated Hardware and Software Solution or install ANANT UCS on a Bare Machine having multiple Ethernet Ports. To know the different solution types offered by MATRIX and/or how to install ANANT on a Bare Machine, refer to the *ANANT UCS Installation Guide*.

If you install ANANT UCS on a Bare Machine having a single Ethernet Port, the provision of only WAN Interface will be provided.

To know more about the LAN and WAN Interface, refer to "Configuring Network Parameters".

This document is written considering ANANT UCS has a provision of both the WAN as well as the LAN Interface.

Compatibility Version of Clients

Compatibility is an essential element for interactions between the Server and the clients.

The following table lists, version of the clients compatible with ANANT UCS.

Clients	Version
VARTA ADR100	V01R02
VARTA AMP100	V01R03
VARTA WIN200	V01R05
SPARSH VP248	V5R22
SPARSH VP310	V1R11
SPARSH VP330	V1R11
SPARSH VP510	V1R9
EXTENDED SPARSH VP710	V1R3
SPARSH VP210	V1R1

It is recommended to use the above mentioned client version with ANANT UCS for the overall functionality.

Licenses Supported in ANANT UCS

License Management is an important aspect of ANANT UCS. A License Dongle is required for availing the features and facilities of the Unified Communication Server.

If you have purchased the Integrated Hardware and Software Solution, then connect this License Dongle to the Dell Server.

If you have purchased only the Software Solution, then connect this License Dongle to the Host Machine, on which you have installed ANANT.

To know more refer to ANANT UCS Installation Guide.

Contact your respective Channel Partner for the License Dongle.

Make sure you do not remove the License Dongle, after connecting it to the Dell Server/Host Machine.

ANANT UCS offers the following licenses:

- ANANT UCS Platform License
- IP Subscribers License
- VMS Channels License
- Conference Participants License
- VARTA Essential Users License
- VARTA Professional Users License
- VARTA Collaboration Users License
- Redundancy Users License
- Hospitality Users License
- E911 Users License
- Hospitality PMS Users License
- ANANT Application Upgrade Package License

Hospitality related licenses are supported in Firmware Version V2.1 or later.

By default, no license is activated in the system. However, when you purchase the ANANT UCS Platform License, some licenses are pre-activated, by default.

ANANT UCS Platform License

ANANT UCS Platform License is required for accessing the features and facilities of ANANT.

By default, this license allows you to configure and register 10 SIP Extensions. However, you can configure and register a maximum of 5000 SIP Extensions with ANANT. To configure and register additional SIP Extensions, you need to purchase and activate the IP Subscribe License. To know about IP Subscriber refer “IP Subscribers License”.

If you have not activated the ANANT UCS Platform License, you can still configure and register 5000 SIP Extensions with ANANT. However, the system will disconnect all the connected calls[3] (internal or external, incoming or outgoing) after 60 seconds.

IP Subscribers License

This license permits the configuration and registration of SIP Extensions with ANANT UCS.

By default, 10 IP Subscribers licenses are pre-activated when you purchase the **ANANT UCS Platform License**. However, you can configure and register a maximum of 5000 SIP Extensions with ANANT. To configure and register the additional SIP Extensions, you need to purchase and activate the **IP Subscribers** License. To know the different types of **IP Subscribers** license, see "Supported Licenses".

To register a Standard SIP Phone/Extended IP Hard Phone with ANANT, you only need to purchase an **IP Subscribers** License.

To register a UC VARTA Client/ Extended SPARSH VP710 with ANANT, you need to purchase both the IP Subscribers as well as the VARTA Licenses.

VMS Channels

VMS Channels license is required for allowing the VMS calls. The system allows simultaneous VMS calls as per the license only. ANANT UCS supports a maximum of 64 VMS Channels.

By default, 4 VMS channels are pre-activated when you purchase **ANANT UCS Platform License**. If you require more channels, you must purchase and activate the channel license according to your requirement. To know the license details, see "Supported Licenses"

If you have not activated the **ANANT UCS Platform License**, no VMS channels will be available.

Conference Participant License

By default, **ANANT UCS Platform License** allows you to conduct a conference with a maximum of 8 participants. You can either conduct a single conference with maximum 8 participants or multiple conferences with total 8 participants overall.

As per your enterprise requirements you may require additional participants, hence ANANT UCS offers the Conference Participants license.

To know the license details, refer "Supported Licenses".

To know about the different types of conferences, see"Conference-3 Party","Conference-Multiparty" and "Conference Dial-In"

If you have not activated the **ANANT UCS Platform License**, no conference channels will be available.

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3. *Connected calls means where speech is connected between calling party and called party even if the called party port is not matured.*

For Raid feature to function, you must have a conference license.

Redundancy Users License

With Redundancy Users license, you can increase the system reliability by providing uninterrupted communication.

For redundancy, you must install ANANT on two identical servers having same configuration — Primary Server and Backup Server.

Make sure Redundancy Users license is activated in the Primary Server in which you have the other licenses activated as well. This allows the Primary Server to share its licenses with the Backup Server during redundancy. In other words, you can use all the licensed features and functionalities that were supported by the Primary Server. However, you can use these licenses for a limited period of time.

To know more about configuring the redundancy parameters, refer to “Configuring Redundancy Parameters”.

When you activate the Redundancy Users License, the system will restart.

By default, 10 Redundancy Users Licenses are pre-activated when you purchase the **ANANT UCS Platform License**.

Redundancy will be supported only when the number of Redundancy Users License activated in the system is greater than or equal to the number of activated IP Subscribers License.

Consider a scenario, in which 15 IP Subscribers Licenses and 10 Redundancy Users Licenses are activated in the system. In this case, redundancy will not be supported as the number of Redundancy Users Licenses activated in the system is less than the number of IP Subscribers Licenses. To support redundancy, you must purchase and activate the Redundancy Users Licenses accordingly. To know the license details, see “Supported Licenses”

To know more about Redundancy feature, refer to “Redundancy”.

Hospitality Users License

Hospitality Users License provides a set of special telephonic and guest/patient management features for hospitality and accommodation establishments like hotels and hospitals, which ANANT UCS supports when it is deployed in a hotel or hospital. When you activate the Hospitality Users license, the following features will be activated:

- Room Shift
- Check-In, Check-Out
- Change Room Occupancy Status
- Floor Service
- Change Room Clean Status
- Front Desk User GUI (Web Interface)

By default, no Hospitality Users License are pre-activated when you purchase the **ANANT UCS Platform License**. Hospitality Users License will be applicable only when the number of Hospitality Users Licenses activated in the system is equal to or greater than the number of IP Subscribers Licenses.

Consider a scenario, in which 15 IP Subscribers Licenses and 10 Hospitality Users Licenses are activated in the system. In this case, features and facilities related to the Hospitality License will not be provided as the number of

Hospitality Users Licenses activated in the system is less than the number of IP Subscribers Licenses. To attain the features and facilities related to Hospitality, you must purchase and activate the Hospitality Users Licenses accordingly. To know the license details, see “Supported Licenses”.

E911 Users License

ANANT UCS supports the E911 Users license. You will be able to dial the Emergency number 911 from the system only if you have activated this license.

By default, no E911 Users License are pre-activated when you purchase the **ANANT UCS Platform License**. E911 Users License will be applicable only when the number of E911 Users Licenses activated in the system is equal to or greater than the number of IP Subscribers Licenses.

Consider a scenario, in which 15 IP Subscribers Licenses and 10 E911 Users Licenses are activated in the system. In this case, the features related to the E911 License will not be provided as the number of E911 Users Licenses activated in the system is less than the number of IP Subscribers Licenses. To attain the features related to this license, you must purchase and activate the E911 Users Licenses accordingly. To know the license details, see “Supported Licenses”.

PMS Users License

ANANT UCS supports interface for PMS, the application software commonly used by hotels to manage their administrative functions. The PMS Interface supports proprietary PMS protocols of Matrix (Matrix PMS Type 1 and Type 2), Micros Opera and Softbrands Extended Starlight. To be able to select any of these PMS Protocols, you must purchase and activate the PMS Users License.

By default, no PMS Users License are pre-activated when you purchase the **ANANT UCS Platform License**. PMS Users License will be applicable only when the number of PMS Users Licenses activated in the system is equal to or greater than the number of IP Subscribers Licenses.

Consider a scenario, in which 15 IP Subscribers Licenses and 10 PMS Users Licenses are activated in the system. In this case, the features related to the PMS will not be provided as the number of PMS Users Licenses activated in the system is less than the number of IP Subscribers Licenses. To attain the features related to this license, you must purchase and activate the PMS Users Licenses accordingly. To know the license details, see “Supported Licenses”.

License for the Hospitality module is a pre-requisite for PMS.

ANANT AUP License

ANANT Application Upgrade Package (AUP) License is required for upgrading all the firmware packages released for the next one year from the date of activation of this license.

After activating ANANT UCS Platform License, you can upgrade all the firmware packages released within the next one year from the date of activation of this license. These firmware up-gradations are free of cost.

However, if you wish to upgrade the firmware packages released after one year from the activation of this license, you need to purchase and activate the ANANT AUP License.

Let us understand this with the help of an example:

If you have activated ANANT UCS Platform License on Jan 2018 and a new firmware is released in May 2018. You can upgrade the ANANT UCS with this firmware free of cost. However, if a firmware is released in May 2019 and you wish to upgrade this firmware, you need to activate the ANANT AUP License.

To take the above example further, if a new firmware is released in Jan 2020, and you wish to upgrade this firmware, you must renew ANANT AUP License twice.

To know the license details, refer “Supported Licenses”.

MATRIX VARTA User Licenses

ANANT UCS supports three types of user licenses for VARTA Users and Extended SPARSH VP710 Users — VARTA Essential Users, VARTA Professional Users and VARTA Collaboration Users.

By default, 5 VARTA Collaboration License are pre-activated when you purchase **ANANT UCS Platform License**. Thus allowing you to configure and register 5 VARTA clients/ Extended SPARSH VP710 Phones with ANANT. If you want to configure and register more VARTA Clients/ Extended SPARSH VP710 Phones, you must purchase and activate the additional VARTA licenses as per your requirement. To know the license details, see “Supported Licenses”.

To register a VARTA Client/ Extended SPARSH VP710 with ANANT, you need to purchase both **IP Subscribers** and **VARTA** Licenses.

Following table lists the features which will be supported in MATRIX VARTA WIN200 / VARTA ADR100 / VARTA AMP100/ Extended SPARSH VP710 when you activate the respective license.

Features	VARTA Essential Users			VARTA Professional Users			VARTA Collaboration Users		
	WIN200	ADR 100/ Extended SPARSH VP710	AMP100	WIN200	ADR100/ Extended SPARSH VP710	AMP100	WIN200	ADR100/ Extended SPARSH VP710	AMP100
Making Calls	✓	✓	✓	✓	✓	✓	✓	✓	✓
Receiving Calls	✓	✓	✓	✓	✓	✓	✓	✓	✓
Hold	✓	✓	✓	✓	✓	✓	✓	✓	✓
Transfer	✓	✓	✓	✓	✓	✓	✓	✓	✓
Blind Transfer	✓	✓	✓	✓	✓	✓	✓	✓	✓
One Touch Transfer	✓	✓	✓	✓	✓	✓	✓	✓	✓
3-Party Audio Conference	✓	✓	✓	✓	✓	✓	✓	✓	✓
Video Call	✓	✓	✓	✓	✓	✓	✓	✓	✓
Intercom	✓	✓	✓	✓	✓	✓	✓	✓	✓
Voicemail	✓	✓	✓	✓	✓	✓	✓	✓	✓
Call Forward	✓	✓	✓	✓	✓	✓	✓	✓	✓
Do Not Disturb	✓	✓	✓	✓	✓	✓	✓	✓	✓
Presence	✓	✓	✓	✓	✓	✓	✓	✓	✓
IM and SMS	✓	✓	✓	✓	✓	✓	✓	✓	✓
Favorites	✓	✓	✓	✓	✓	✓	✓	✓	✓
Global Directory Access	✓	✓	✓	✓	✓	✓	✓	✓	✓
All Menu Features	✓	✓	✓	✓	✓	✓	✓	✓	✓
All Call Features	✓	✓	✓	✓	✓	✓	✓	✓	✓
Hotkeys	✓	✗	✗	✓	✗	✗	✓	✗	✗
Multiparty Audio Conference	✗	✗	✗	✓	✓	✓	✓	✓	✓
Handover	✗	✗	✗	✓	✓	✓	✓	✓	✓
Drag and Drop Transfer	✗	✗	✗	✓	✗	✗	✓	✗	✗
Drag and Drop Conference	✗	✗	✗	✓	✗	✗	✓	✗	✗
Contact Grouping	✗	✗	✗	✓	✗	✗	✓	✗	✗
BLF Subscription	✗	✗	✗	✓	✓	✓	✓	✓	✓
DSS Soft Keys	✗	✗	✗	✓	✗	✗	✓	✗	✗
DSS Soft Keys for Mobile users	✗	✗	✗	✗	✓	✓	✗	✓	✓
Call Transfer to other user's Voicemail (Blind Transfer to VMS)	✗	✗	✗	✓	✗	✗	✓	✗	✗
Click to Call	✗	✗	✗	✓	✗	✗	✓	✗	✗

Refer **Matrix VARTA WIN200 User Guide**, **Matrix ADR100 User Guide**, **Matrix AMP100 User Guide**, **Extended SPARSH VP710 User Guide** to know more.

Supported Licenses

Refer to the table below to know the name of the respective licenses you need to activate for each feature.

License Name	Description
Platform License	
ANANT UCS PLATFORM	Unified Communication Server License for an Enterprise.

License Name	Description
IP SUBSCRIBER License	
ANANT IPSUB10	License for 10 IP Subscribers to create 10 SIP Users.
ANANT IPSUB50	License for 50 IP Subscribers to create 50 SIP Users.
ANANT IPSUB100	License for 100 IP Subscribers to create 100 SIP Users.
ANANT IPSUB500	License for 500 IP Subscribers to create 500 SIP Users.
VARTA License	
ANANT VARTA USER5E	License for 5 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER10E	License for 10 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER50E	License for 50 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER100E	License for 100 VARTA UC Users with Essential features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER5P	License for 5 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER10P	License for 10 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER50P	License for 50 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER100P	License for 100 VARTA UC Users with Professional features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER5C	License for 5 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER10C	License for 10 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER50C	License for 50 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
ANANT VARTA USER100C	License for 100 VARTA UC Users with Collaboration features. This license is required to register Android/iOS/Windows UC Clients.
REDUNDANCY USERS License	
ANANT REDUNDANCY USER10	License for 10 Redundancy Users.
ANANT REDUNDANCY USER50	License for 50 Redundancy Users.
ANANT REDUNDANCY USER100	License for 100 Redundancy Users.
ANANT REDUNDANCY USER500	License for 500 Redundancy Users.
VMS CHANNEL License	
ANANT VMS CHNL4	License for activating 4 additional VMS Channels.
ANANT VMS CHNL16	License for activating 16 additional VMS Channels.
ANANT VMS CHNL32	License for activating 32 additional VMS Channels.
CONFERENCE PARTICIPANT License	

License Name	Description
ANANT CONF8	License for permitting 8 additional participants in a conference.
ANANT CONF16	License for permitting 16 additional participants in a conference.
ANANT CONF32	License for permitting 32 additional participants in a conference.
SOFTWARE UPGRADE	
ANANT AUP	License for upgrading the System Firmware. Validity of this license is one year.
HOSPITALITY License	
ANANT HOSPITALITY USER10	License for 10 Hospitality Users.
ANANT HOSPITALITY USER50	License for 50 Hospitality Users.
ANANT HOSPITALITY USER100	License for 100 Hospitality Users.
ANANT HOSPITALITY USER500	License for 500 Hospitality Users.
ANANT PMS USER10	License for 10 PMS Users.
ANANT PMS USER50	License for 50 PMS Users.
ANANT PMS USER100	License for 100 PMS Users.
ANANT PMS USER500	License for 500 PMS Users.
E911 Users License	
ANANT HOSPITALITY E911 USER10	License for 10 E911 Users.
ANANT HOSPITALITY E911 USER50	License for 50 E911 Users.
ANANT HOSPITALITY E911 USER100	License for 100 E911 Users.
ANANT HOSPITALITY E911 USER500	License for 500 E911 Users.

ANANT UCS keeps a track of the major events and activities associated with the License Key[4] and AUP License. The Server sends a notification email to you, whenever a major event or activity is observed related to License Key and AUP License. The notification email informs you about this event thus helping you in analyzing and taking the required action.

For Example: You have activated ANANT AUP License in the Server and the same is about to expire in few days. In this case, ANANT keeps a track of the activation and expiry date of this license and informs you the same via notification email. The notification email prompts you to activate the required license before the expiry of this license.

The notifications will be sent to specific email Ids only. To know more, refer to “System Log Notification”.

To know how to activate the above licenses, refer to “License Management”.

4. *A License Key is used for activating various licenses in ANANT.*

System Access Level

ANANT UCS can be configured using the following tools:

- Web-based graphic user interface (GUI), Jeeves
- Matrix proprietary IP Phone (specific parameters only)

Configuring using Web-based GUI: Jeeves

ANANT UCS can be configured using an interactive and intuitive Web Graphic User Interface, Jeeves. Jeeves is the proprietary web-based configuration software of Matrix.

Jeeves allows system configuration at three levels:

- System Engineer
- System Administrator
- Front Desk

A distinct set of features and facilities can be configured at each of these levels. The accessibility to each level is secured by a password.

System Engineer

You can configure the entire system from the System Engineer level. To be able to do this, you must log into Jeeves as System Engineer.

Only the System Engineer, who installs, configures and maintains the system is allowed to access this level. Hence access to the SE level is protected by means of a password, referred as SE Password.

System Administrator

You can configure certain features for the extension users from the System Administrator level. You can also keep a track of various system reports such as System Activity Logs Reports, System Fault Logs Reports and Station Message Detail Recording Reports from this level.

To be able to do this, you must log into Jeeves as System Administrator. The access to the SA mode is protected by means of a password, referred as SA password.

Front Desk

The Front Desk Mode is relevant only for the Hospitality Industry.

This mode is meant for the personnel at the Front Desk of the Hotels/Motels, allowing them access to and operation of the hospitality features of ANANT UCS, for example: Check-In/Out of guests, Changing Room Occupancy and Clean Status, setting Call Budgets for guests, setting Wake-up calls, Reminders, Do Not Disturb for guests, printing Call Reports and Hotel reports, and several others.

- *It is possible to configure ANANT UCS from any location using Jeeves. You can use Jeeves to configure the system On-site (where it is installed) and Off-site, from a remote location.*
- *It is possible for four users to simultaneously log into the System Engineer level of Jeeves.*

Configuring using Matrix proprietary IP Phone

You can configure some of the important parameters and/or set/ cancel certain features from an Extended IP Phone. This can be achieved by dialing certain command strings from any of the Extended IP Phone (SPARSH VP330, SPARSH VP310, SPARSH VP510, SPARSH VP248, SPARSH VP710, SPARSH VP210), connected as a SIP extension of the system.

This kind of system configuration using an Extended IP Phone is useful in scenarios, where you cannot access the Jeeves of the system.

The system can be configured using an Extended IP Phone at two modes:

- System Engineer
- System Administrator

System Engineer

You can configure some of the basic network and debug parameters from the System Engineer Mode by dialing the relevant command strings from an Extended IP Phone.

SE Commands

These are number strings to be dialed by the System Engineer on entering the System Engineer Mode from an Extended IP Phone. The access to the System Engineer Mode from an extension is protected by means of a password.

Contact the Matrix Technical Support Team for the list of SE Commands.

System Administrator

You can set/cancel certain features for the extensions from the System Administrator Mode by dialing the relevant command strings from an Extended IP Phone.

You can also capture and print various system activity logs and reports from the System Administrator Mode.

SA Commands

These are number strings to be dialed by the System Administrator on entering the System Administrator Mode from an Extended IP Phone. The access to the System Administrator Mode from an extension is protected by means of a password.

For the list of SA Commands supported on your Extended IP Phone, refer *"SA Commands".*

It is recommended to configure the system using the web-based configuration software, Jeeves. To know how to configure the system using Jeeves, refer to **"Configuring ANANT UCS".**

CHAPTER 3

Connecting SIP Extensions

ANANT UCS supports up to 5000 SIP/UC Users. SIP/UC Users can make and receive calls to any extension user of the system and to external numbers.

You may connect/register any SIP enabled device — a Matrix UC Client (VARTA WIN200, VARTA ADR100, VARTA AMP100), Matrix IP Phone (SPARSH VP330, SPARSH VP310, SPARSH VP510, SPARSH VP248, SPARSH VP710, SPARSH VP210), Standard SIP Phone, Soft Phone, Analog Terminal Adapter — as the SIP User of the system.

The Matrix UC Clients also offer UC functionalities in addition to the SIP functionalities.

You may also connect/register the following as SIP Extensions of the system:

- Connect SPARSH VP248, the Extended IP Phone. For instructions, see *“Connecting SPARSH VP248 as Extended SIP Extension”*.
- Connect SPARSH VP310, the Extended IP Phone. For instructions, see *“Connecting SPARSH VP310 as Extended SIP Extension”*.
- Connect SPARSH VP330, the Touch Screen Extended IP Phone. For instruction,*“Connecting SPARSH VP330 as Extended SIP Extension”*.
- Connect SPARSH VP510, the Premium IP Phone. For instruction, *“Connecting SPARSH VP510 as Extended SIP Extension”*.
- Connect Extended SPARSH VP210, the Entry Level IP Phone. For instruction*“Connecting SPARSH VP210 as Extended SIP Extension”*.
- Connect Extended SPARSH VP710, the Smart Video IP Phone. For instruction *“Connecting Extended SPARSH VP710 as Extended SIP Extension”*.

You can register the following UC Clients as SIP Users of the system:

- Matrix VARTA WIN200, Unified Communication Client for Windows. For instructions, refer to the *VARTA WIN200* User Guide.
- Matrix Mobile UC Clients, as given below:
    - Matrix VARTA AMP100, the Mobile UC Client for iPhones. For instructions, refer to the *VARTA AMP100* User Guide.

- Matrix VARTA ADR100, the Mobile UC Client for Android Smartphones. For instructions, refer to the *VARTA ADR100* User Guide.

Refer to “ANANT UCS Features Supported in Terminals” to know the features supported in each client.

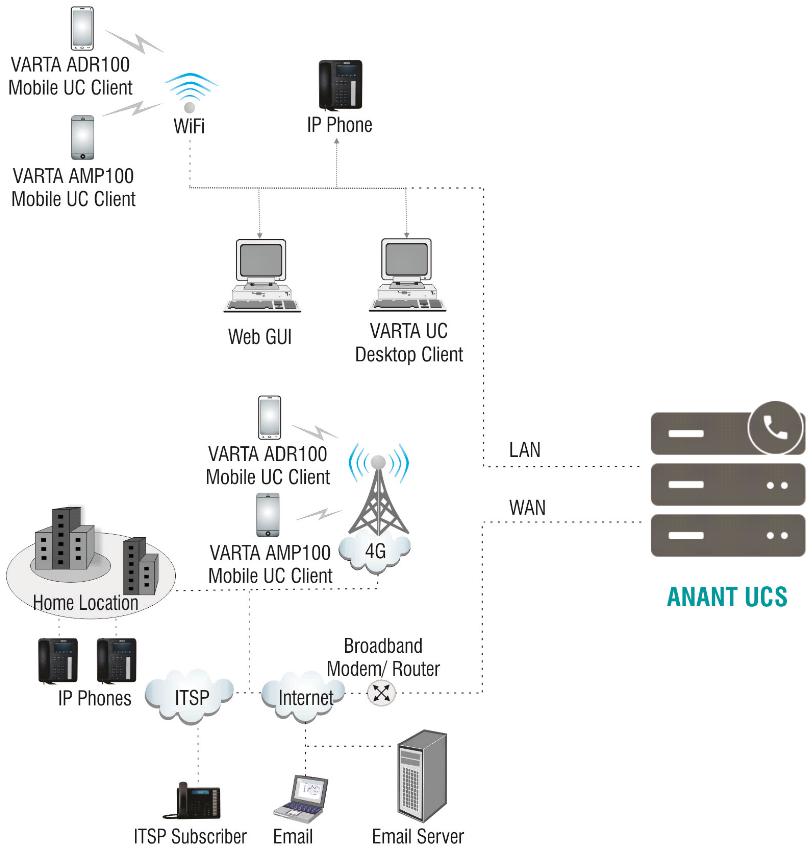
The SIP Users may be registered over ***WAN*** or over ***LAN*** according to your preference and your IP network installation scenario. Extended SIP Phones and UC Clients can be registered with ANANT UCS using IPv4 Addresses only.

You can register the same SIP User from three different locations.

If you register the Extended IP Phone outside the Region/Country selected for ANANT UCS, the time and Time Zone dependant features, such as Alarms, Reminders, Time Zone Display of the phone at each location will operate according to the Real Time Clock of ANANT UCS. Also, Access Codes and Emergency Numbers will work according to the Region/Country selected for ANANT UCS.

- Connect the Extended IP Phone, or any Standard IP Phone to the LAN Switch.
- Register any SIP device (Extended IP Phone/ Soft clients or Standard IP Phone) on the public network as SIP Extension.
- When you register the Matrix Extended IP Phone with the system, the WAN/LAN port is used for Auto Configuration as well for Registration of the Extended IP Phones.
- When you register a SIP device other than the Matrix Extended IP Phone on the public network as SIP Extension, do the following:
  - In this SIP device configure the following:
    - the Registrar Server Address of ANANT UCS
    - the Registrar Server Port
    - the SIP ID
    - Authentication ID and Password
  - Configure Port Forwarding for the WAN Port of ANANT UCS on the Router.

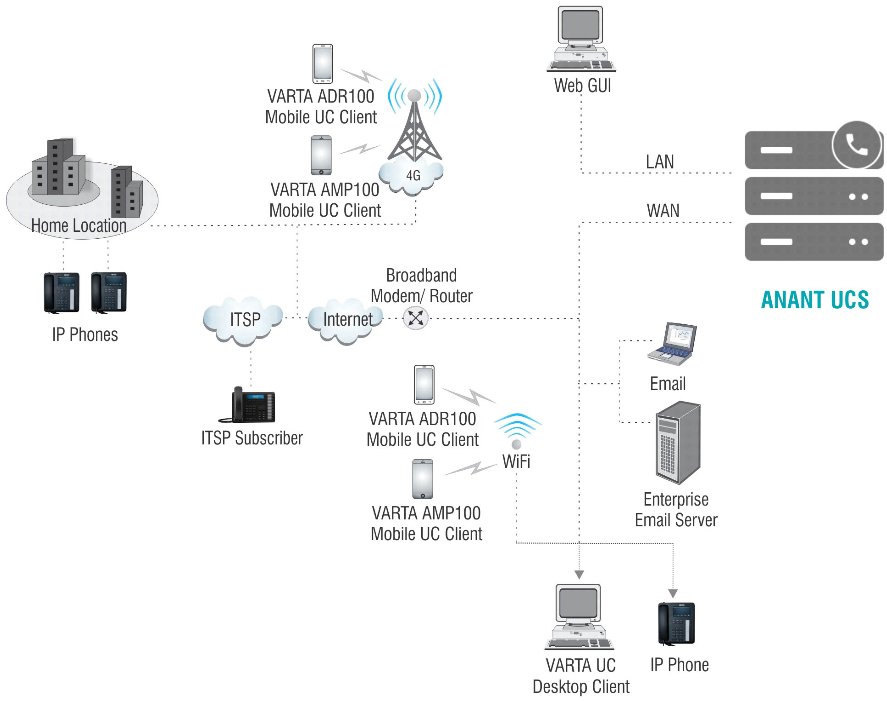
If WAN Port of ANANT UCS is connected to a ***Public Network***,



Connecting ANANT UCS to the Public IP Network

- Connect the Matrix VARTA WIN200, Extended IP Phone, or any Standard SIP device to the LAN Switch.
- Register any SIP device (Matrix VARTA UC Clients, Extended IP phone or Standard SIP phone) on the public network as SIP extension.

If WAN Port of ANANT UCS is connected to a ***Private Network***,



Connecting ANANT UCS to the Private IP Network

- Connect Matrix VARTA WIN200, Extended IP Phones or Standard SIP Phones to the LAN Switch.
- You may also register any SIP device (Matrix VARTA UC Clients, Extended IP Phone or Standard SIP phone) on the public network as SIP Extension.

  When you register the Matrix Extended IP Phone with ANANT UCS, configure **Port Forwarding** for the **WAN port of the system** on the Router. The WAN Port is used for Auto Configuration of the Extended IP Phones.

Connecting SPARSH VP248 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to ANANT UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as **'String'** and configure the LAN or WAN IP Address /Domain Name of ANANT UCS and SPARSH Port in the format **"IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Login to Jeeves. For instructions, read the topic "Configuring ANANT UCS".

- Assign SIP User ID (will work as an extension number) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP User ID you assigned to the Extended IP Phone, you must configure the necessary parameters in ANANT UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

Now, follow the steps described below to install the Extended IP Phone. The instructions are common for all models of the SPARSH VP248. For the purpose of illustration, the premium model, SPARSH VP248P, has been used.

1. Unpack the SPARSH VP248 box and verify package contents.

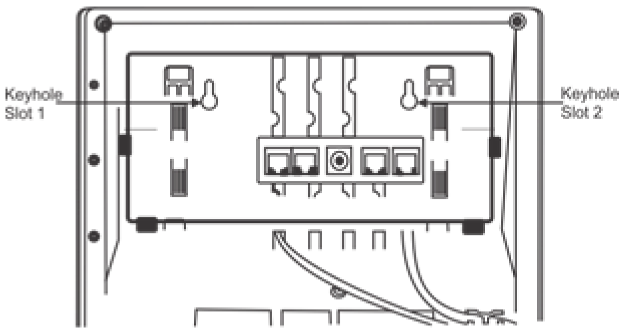
2. Mount the phone on a desk or wall at a location convenient to you.

   - When mounting the phone on the wall,

     - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.

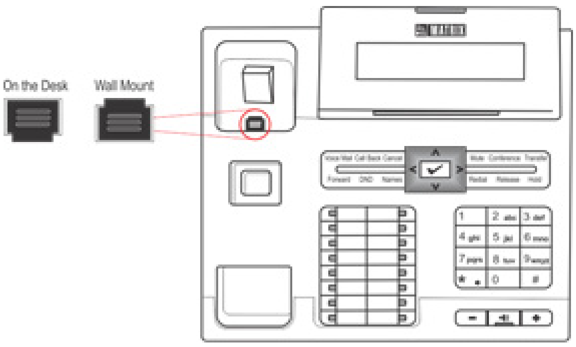
     - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.

     - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.



     - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.

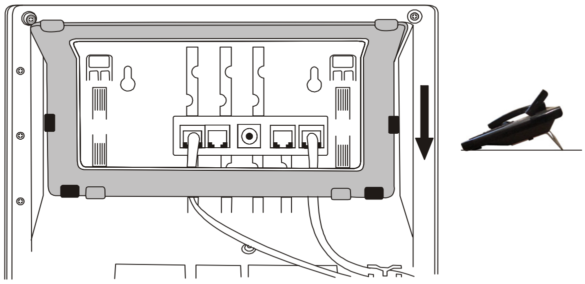
     - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



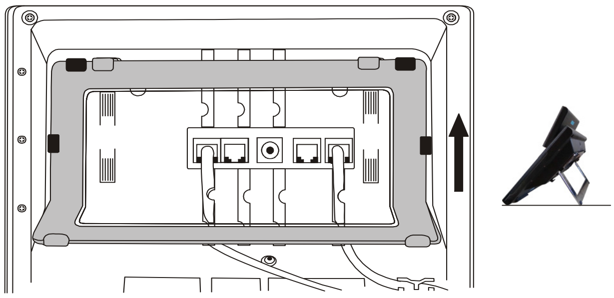
   - When you mount the phone on a desk,

- You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 30° Angle



Foot Stand attached at 50° Angle



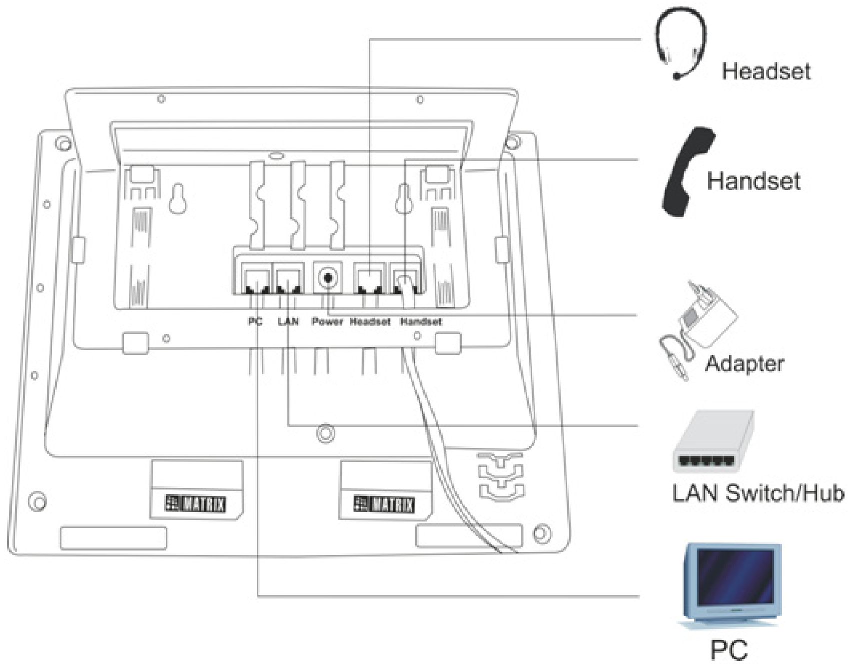
If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

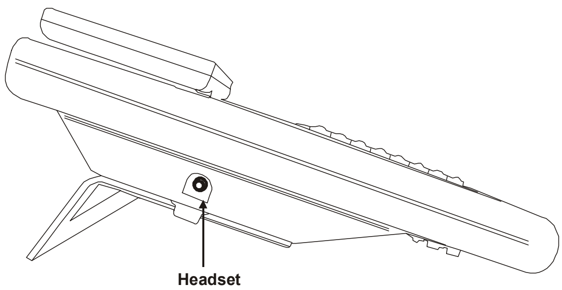
3. Connect the Handset to the Phone body.

    - Plug the long straightened end of the phone cord into the handset jack at the bottom of the phone marked with the handset symbol.

- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

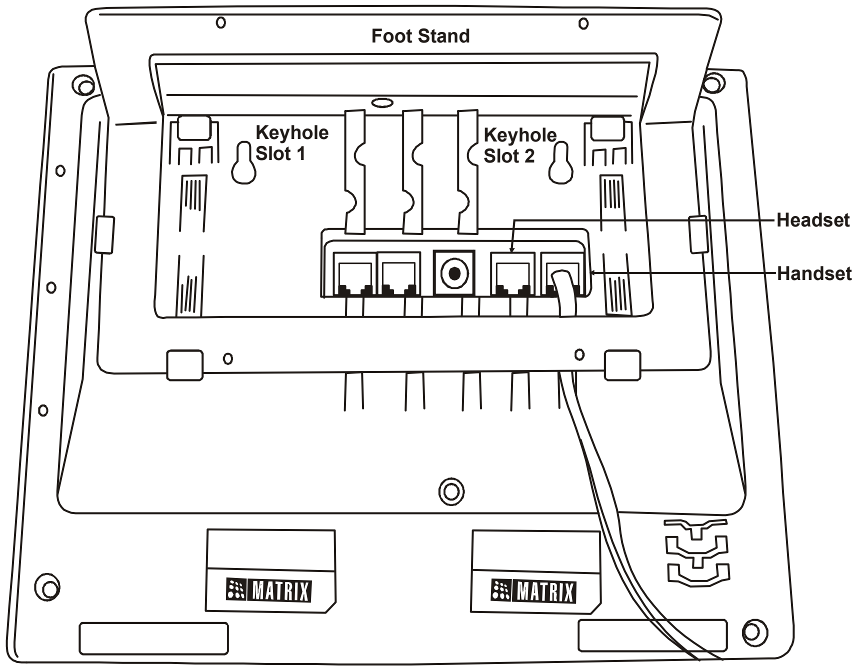


4. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 2.5 mm single connector into the headset jack headset jack with the symbol  on the left side panel of the phone, as illustrated in the figure below.



OR

You may plug a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol , as illustrated in the figure below.



5. Connect the LAN Port of SPARSH VP248 to the LAN Switch/Hub or a Router, according to your installation scenario.

6. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

7. Plug the connector of the Power Adapter in to the power jack at the back of the phone[5]. Use only the adapter provided with the phone to prevent any damages that may arise from the use of other adapters. If you want to use Power over Ethernet (PoE), ensure that your LAN supports PoE. Supply power through an 802.3af connection on the LAN Port of the phone. Do not connect the Adapter!

8. Plug the Power Adapter into a power outlet.

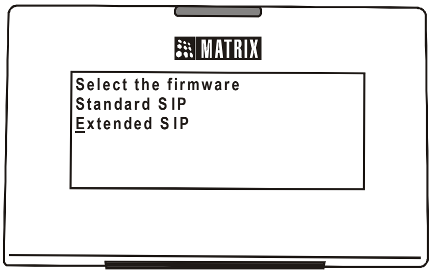
9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

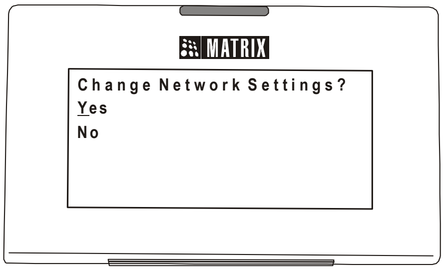
   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and booting message appears.
   - As soon as the ‘Loading...’ message appears on the phone display, press **#** key.
   - Select the firmware **Extended - IP Phone**. Move the cursor by pressing the DOWN navigation key **V**.

5. *The SPARSH VP248 does not have a power switch.*

- When the cursor is placed under the Extended IP Phone, press Enter key.



- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.



- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from ANANT UCS.
- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with ANANT UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone by accessing the Local Menu of the phone. To move the cursor and scroll through the menu and submenu options, use the following touch sense navigation keys on your phone.

- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.

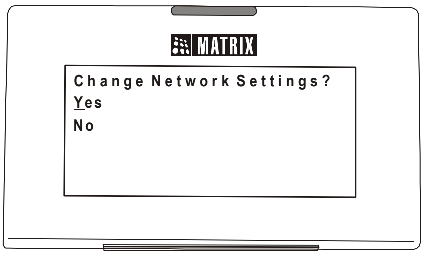
- The Back key < to move the cursor one character and to return from the submenu to the main menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

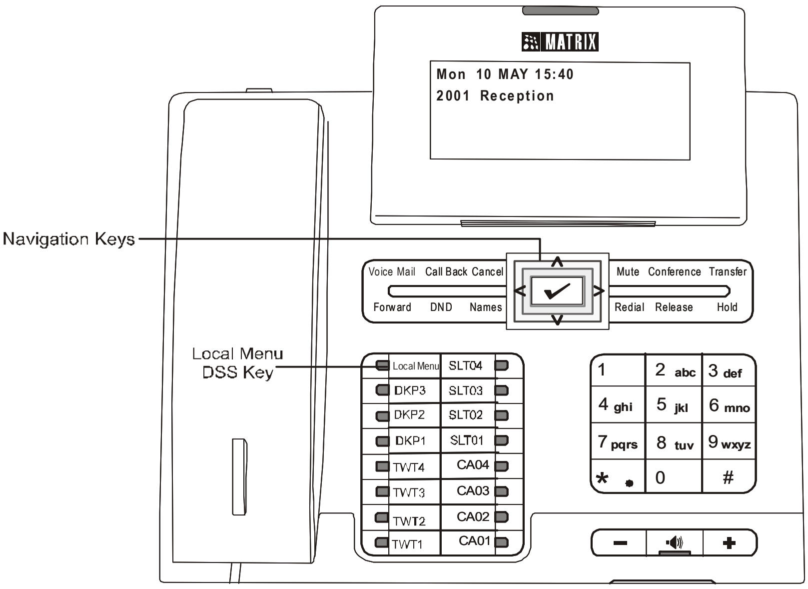


You must press the Enter Key to select Yes and access network settings.

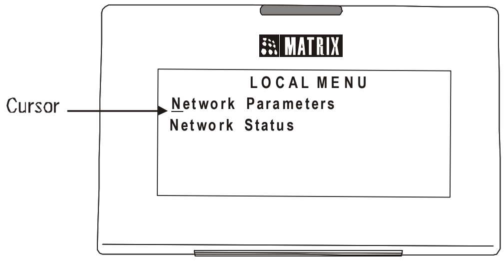
2. When the phone is making Network discovery, downloading configuration files, attempting registration.

You must press the Enter Key ✔ to access network settings,

3. When the phone is in idle state. You must press the DSS key assigned to 'Local Menu'.



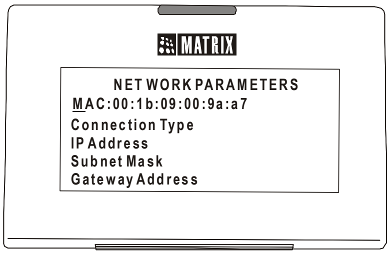
4. When you press the Local Menu DSS Key (in idle state) or when you press the Enter key during any process, the Local Menu appears on your phone display.



You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.
- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

> - *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
> - *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of ANANT UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Server Port

- Enter the SPARSH Port of ANANT UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[6].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	**Traffic Type**
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.

6. The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.

- Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
- Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
- Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.

- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
  - To configure PC VLAN/COS, select **Enable?**.
  - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
  - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see "Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone", under *PCAP Trace*.

When you change the Network Settings, the phone will restart.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?**.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

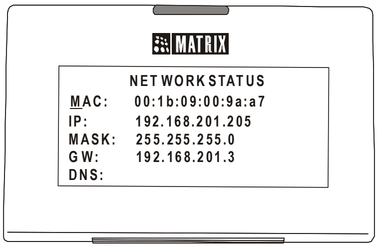
You need to configure the following 802.1x Authentication parameters:

- Select **Enable?**.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

Viewing Network Status

- In the Local Menu of the phone, place the cursor on Network Status and press the Enter key.

- The Network Status submenu appears.



Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

- **MAC:** This is the MAC Address of the phone.
- **IP:** The current IP Address assigned to the phone.
- **MASK:** The current Subnet mask assigned to the phone.
- **GW:** The current Gateway IP Address assigned to the phone.
- **DNS:** The Domain Name Server address assigned to the phone.
- **S. ADD:** The LAN or WAN IP Address / Domain Name of ANANT UCS.
- **S. PORT:** The SPARSH Port ANANT UCS.
- **DOMAIN:** The Domain Name assigned to the phone.
- **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
- **FIRM:** The version of the current Firmware of the phone.
- **UBOOT**: The UBOOT release date.
- **KERNEL**: The KERNEL release date.

Connecting SPARSH VP310 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to ANANT UCS:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of ANANT UCS and SPARSH Port in the format "**IP_Address:Port**" in your DHCP Server as per your installation scenario.*

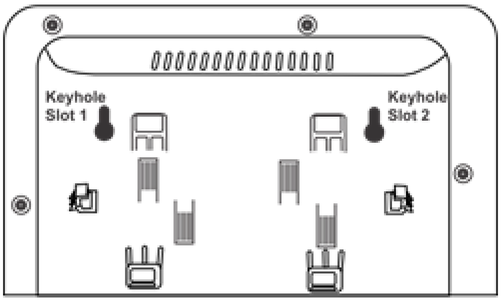
- Log in to Jeeves. For instructions, read the topic "Configuring ANANT UCS".

- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

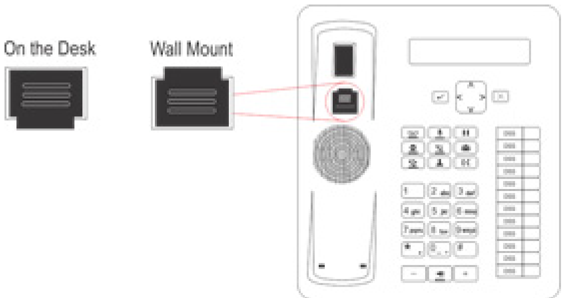
  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in ANANT UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

Now, follow the steps described below to install the Extended IP Phone.

1. Unpack the SPARSH VP310 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP310 on a wall,
   - Use the mounting template to drill holes of appropriate size and distance.
   - Fix the screw grips in the holes you drilled.
   - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP310. The screws should protrude from the wall to fit into the Keyhole Slots.

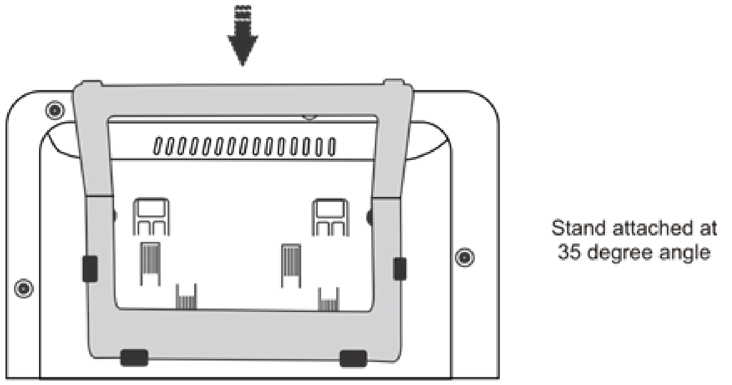


   - Now, mount the phone with the screws fitting into the Keyhole Slot.
   - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

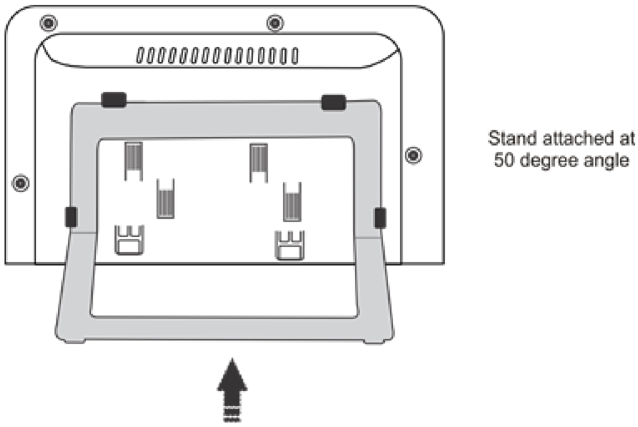


- When you mount the phone on a desk,
  - You can attach the Foot Stand in two ways as illustrated in the following.

Foot Stand attached at 35° Angle

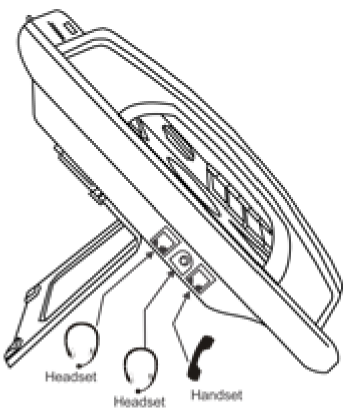


Foot Stand attached at 50° Angle



If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



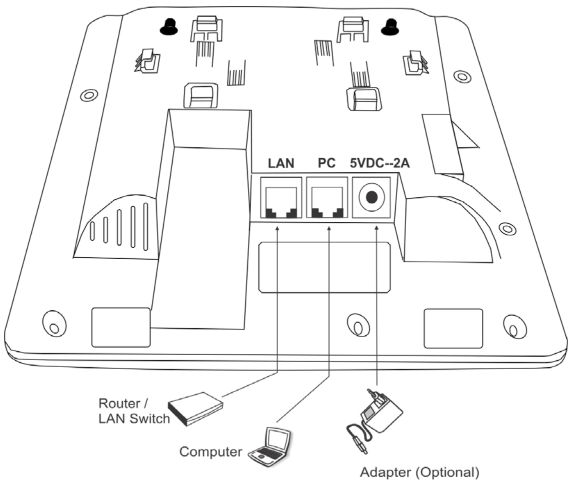
4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol  .
- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol  on the left side panel of the phone, as illustrated in the figure above.

OR

You may also plug in a headset with RJ9 connector into the headset port on the left side panel of the phone, marked with the symbol  .



6. Connect the LAN Port of SPARSH VP310 to the LAN Switch/Hub or a Router, according to your installation scenario using the Ethernet cable.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port of the phone to the LAN Port of the computer.

8. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN switch(IEEE802.3af complaint).

   If you do not want to use PoE, plug in the connector of the adapter into the power jack(DC Jack) with the label 5VDC-2A at the bottom of the phone. Plug in the power adapter into a power outlet

   The IP phone should be used with Matrix original power adapter (5V/2A) only. The use of any third-party power adapter may cause damage to the phone.

   If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

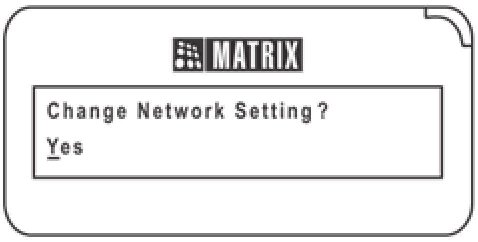
9. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
   - The LCD display will light up and the booting message appears.

- Then the 'Loading...' message appears on the phone display.

- The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
- After loading the firmware, the phone will prompt you to change Network settings.



- Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press the Enter key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See *"Network Settings" at the end of this topic.*

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from ANANT UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with ANANT UCS.
- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key ∨ when the phone is in idle state. To move the cursor and scroll through the menu and submenu options, use the following navigation keys on your phone.

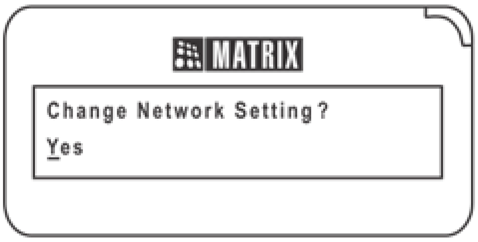
- The Enter key ✓ to make a selection or to complete an action.
- The Up key ∧ to move up the Menu.
- The Down key ∨ move down the Menu.
- The Forward key > move the cursor one character.
- The Back key < to move the cursor one character and to return from the submenu to the main menu.
- The Cancel key ✕ to exit a menu.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.



   You must press the Enter key ✓ to select Yes and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

   You must press the Down key V to access network settings.

3. When the phone is in idle state. You must press the Down key V to access the Network Settings.

Configuring Network Parameters

- When the phone is in idle state. You must press the Down key V to access the Network Settings.
- Press Enter key to select Network Parameters.
- The Network Parameters submenu appears.
- Use the Down/Up key to reach the desired network parameter and press Enter key to select and change the settings.
- You can configure all network parameters described below, except the MAC Address.

> - *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press * (Star) key.*
> - *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Cancel key. The digit to the left of the cursor will be deleted. If the cursor is to the extreme left and you press the Cancel key, you will go to the previous menu.*

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

  If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

  For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

  If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.
  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.
  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of ANANT UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of ANANT UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.
The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[7].

The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

7. *The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.*

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
  - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
  - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
  - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
  - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
  - To configure PC VLAN/COS, select **Enable?**.
  - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
  - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and for instructions on how to use PCAP Trace on the phone, see “Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone”, under *PCAP Trace*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key ∨ to access the Network Settings.

- Again press Down key ∨ to select Network Status and press the Enter key ✔.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of ANANT UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of ANANT UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
  - **FIRM:** The version of the current Firmware of the phone.
  - **UBOOT**: The UBOOT release date.
  - **KERNEL**: The KERNEL release date.

Connecting SPARSH VP330 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix SPARSH VP330 to ANANT UCS:

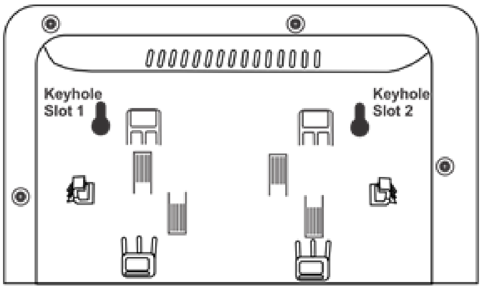
- Decide the location where you want to place SPARSH VP330 within your LAN.
- By Default, in SPARSH VP330, the Connection Type selected is DHCP.
- If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN IP Address /Domain Name of

ANANT UCS and SPARSH Port in the format “**IP_Address:Port”** in your LAN DHCP Server as per your installation scenario.

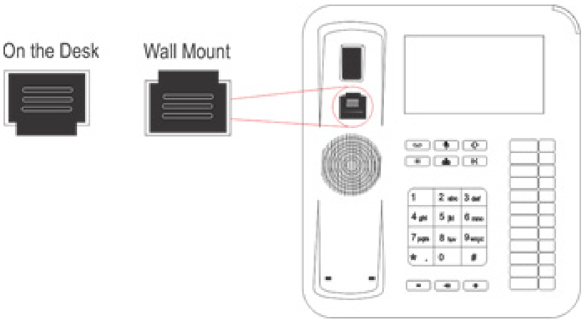
- Login to Jeeves. For instructions, read the topic “Configuring ANANT UCS”.
- You must configure the necessary parameters in ANANT UCS so that SPARSH VP330 can register as a SIP Extension. For instructions, see “Configuring Matrix SPARSH VP330”.

Now, follow the steps described below to install SPARSH VP330.

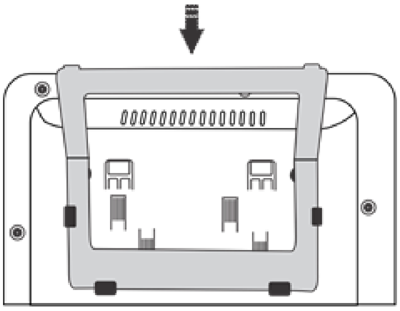
1. Unpack the SPARSH VP330 box and verify package contents.
2. Mount the phone on a desk or wall at a location convenient to you.
   - When mounting the phone on the wall,
     - Use the mounting template to drill holes of appropriate size and distance. Fix the screw grips in the holes you drilled.
     - Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2.
     - Use wall plugs, if required, to fix the screws. Leave the screw heads protruding from the wall to fit into the Keyholes.

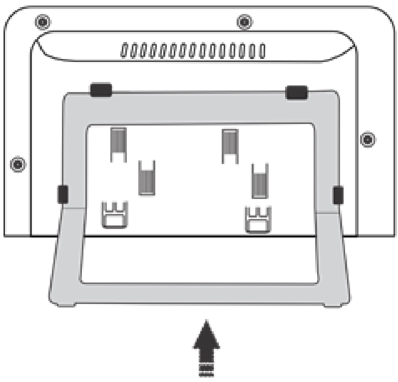


     - Now, mount the phone on the wall, with the screws fitting into the Keyhole slots.
     - Reverse the handset wall mount tab to make sure the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.

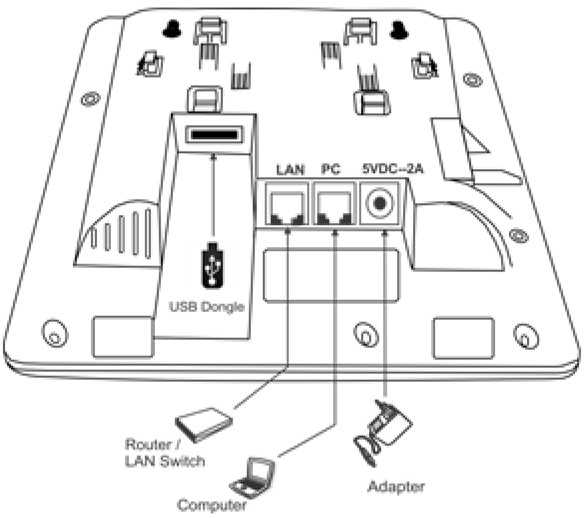


3. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **35° Angle** or at **50° Angle**.





- If you attach the Foot Stand at 50°, the phone will be placed in an almost upright position on your desk.
- Decide which of these positions would work for you best and accordingly attach the Foot Stand.



4. Connect the Handset to the Phone body.

- Plug the long straightened end of the phone cord into the handset jack on the left side panel of the phone marked with the handset symbol.
- Plug the other (short straight) end of the phone cord into the jack at the bottom of the handset.

5. If you want to use a Headset (not supplied) with your phone, you may plug a headset with a 3.5 mm single connector into the headset jack headset jack with the symbol on the left side panel of the phone.

   OR

   You may plug a headset with an RJ9 connector into the headset port on the side panel of the phone, marked with the symbol .

6. Connect the LAN Port of SPARSH VP330 to the IP Network — A Router or LAN Switch — using the Ethernet Cable.
   OR
   Connect the Wi-Fi USB Adapter into the USB port of the phone.

You can purchase the Wi-Fi USB Adapter from Matrix as an optional peripheral device to support wireless network.

7. To connect your phone to a computer on your desk, use an Ethernet cable (not supplied with this phone) to connect the PC Port at the bottom of the phone to the LAN Port of the computer.

8. Plug the connector of the Power Adapter in to the power jack at the back of the phone[8]. Use only the adapter provided with the phone to prevent any damages that may arise from the use of other adapters.

   If you want to use Power over Ethernet (PoE), ensure that your LAN supports PoE. Supply power through an 802.3af connection on the LAN Port of the phone. In this case you need not connect the Power Adapter.

9. Plug the Power Adapter into a power outlet.

   If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

10. Switch ON power supply.

    When you power the phone, the boot process will be initiated in the following sequence.

    - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
    - The LCD display will light up and booting message appears.
    - While loading the application then the loading message appears on the phone display,
    - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

8. *The SPARSH VP330 does not have a power switch.*

If you want to change the Network Settings/Server Settings or want to use Wi-Fi for connectivity, press Settings .

Refer to the SPARSH VP330 User Guide, for detailed instructions:

- *To change the Network Settings of the phone and configure the network parameters.*
- *To use Wi-Fi for connectivity and configure its parameters.*

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from ANANT UCS.
- On successful download of all configuration files, the phone attempts to register with ANANT UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in ANANT UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP330 User Guide*** to know more.

Connecting SPARSH VP510 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to the system when used with ANANT UCS application:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN Port IP Address /Domain Name and SPARSH Port in the format "**IP_Address:Port"** in your DHCP Server as per your installation scenario.*

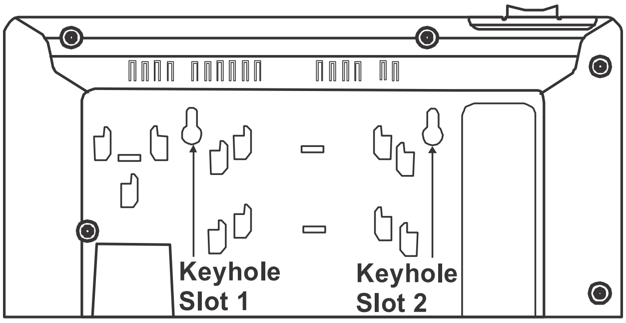
- Login to Jeeves. For instructions, read the topic "Configuring ANANT UCS".
- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in ANANT UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

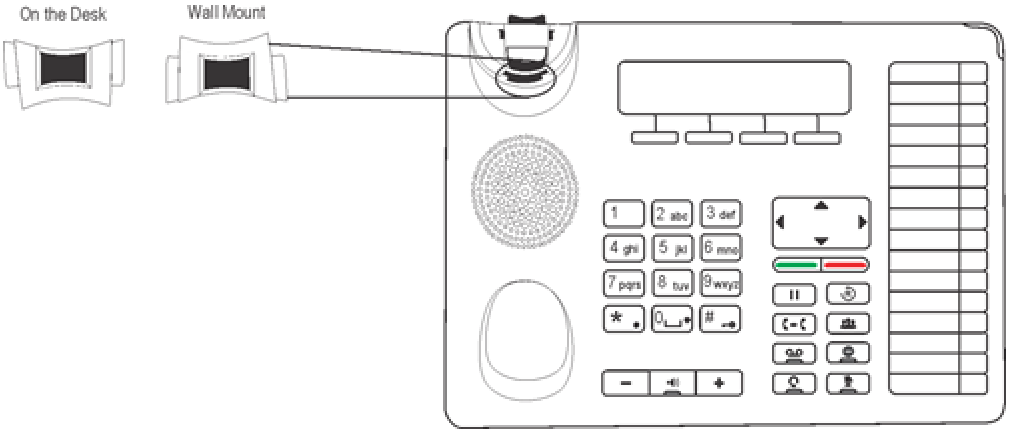
Now, follow the steps described below to install the Extended IP Phone:

1. Unpack the SPARSH VP510 box and verify package contents.
2. You can mount the phone on a wall or on the desk.
3. When you mount SPARSH VP510 on a wall,

- Use the mounting template to drill holed of appropriate size and distance.
- Fix the screw grips in the holes you drilled.
- Fix two screws in the holes on the wall, ensuring that they are aligned with the Keyhole Slots 1 and 2 of SPARSH VP510. The screws should protrude from the wall to fit into the Keyhole Slots.



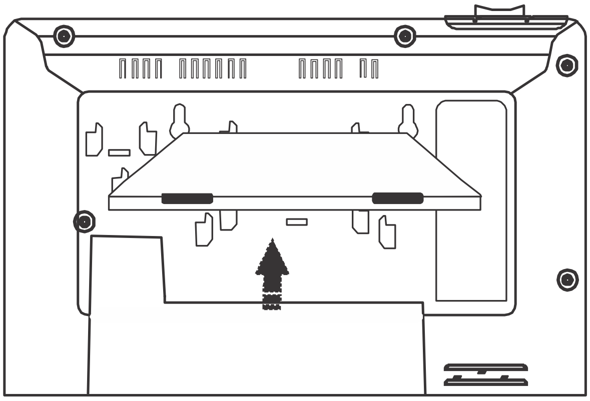
- Now, mount the phone with the screws into the Keyhole Slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the slot.



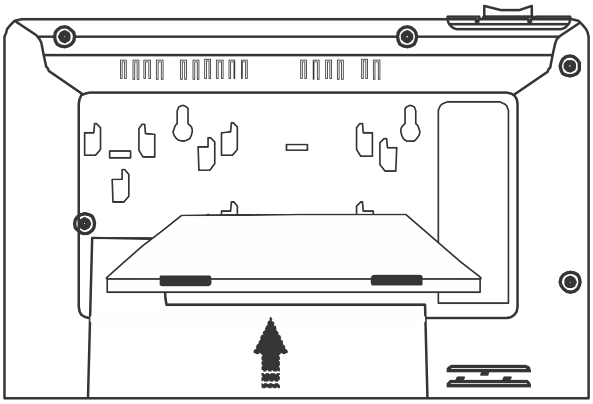
If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

- When you mount the phone on a desk,

- You can attach the Foot Stand in the following ways — at an angle of 45 degrees or 55 degrees



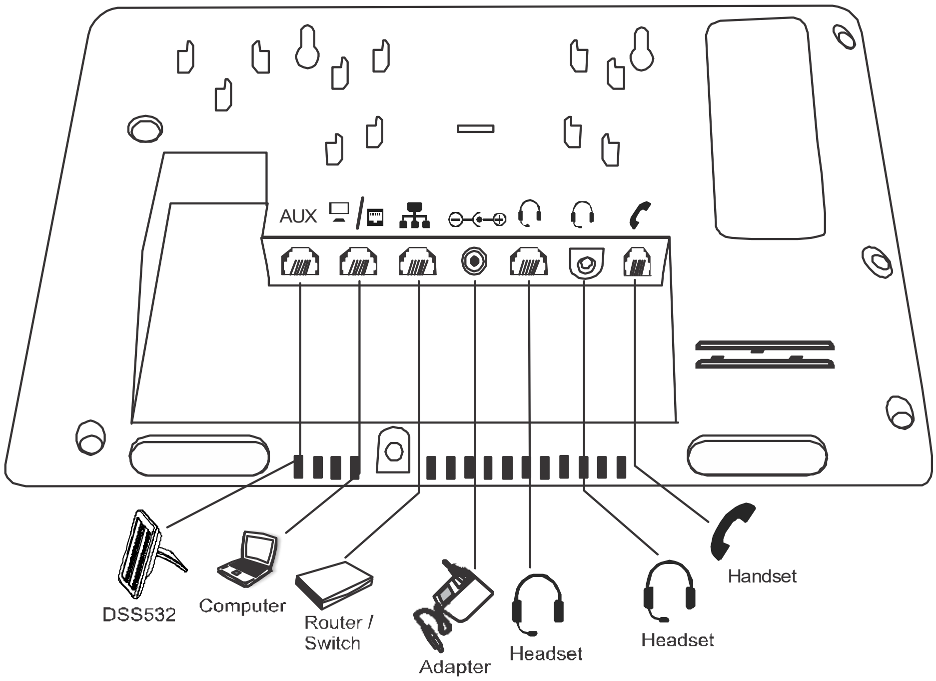
Stand attached at
45 degree angle



Stand attached at
55 degree angle

- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

Refer to the diagram below for connectivity.



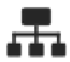
4. Connect the Handset.

- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol  .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

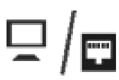
5. Connect the Headset (not supplied by Matrix).

- To use a Headset (not supplied with the phone), plug any standard stereo headset with 3.5mm single connector into the headset audio jack at the bottom of the phone, marked with the symbol .
  OR
  You may also plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .

6. Connect to the IP Network.

- Plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone, marked with the symbol  and the other end to the IP Network — A Router or LAN Switch.

7. Connect a PC to the Phone.

- Plug one end of the Ethernet Cable into the PC Port at the bottom of the phone, marked with the symbol  and the other end into the LAN Port of your PC/LAN Switch.

8. Connect DSS532 with the Phone.

    - To connect DSS532 with the phone, plug one end of the RJ11 cable into the AUX Port of the phone and the other end into the IN Port of the DSS532. For installation, see "Installing DSS532 with SPARSH VP510".

9. Connect the Power Supply.

    - Plug the connector of the Power Adapter in to the power jack at the back of the phone, marked with the symbol ⊖–(•–⊕ . Use only the adapter provided with the phone to prevent any damages that may arise from the use of other adapters.

      If you want to use Power over Ethernet (PoE), ensure that your LAN supports PoE. Supply power through an 802.3af connection on the LAN Port of the phone. In this case you need not connect the Power Adapter.

10. Plug the Power Adapter into a power outlet.

    If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

11. Switch ON power supply.

    When you power the phone, the boot process will be initiated in the following sequence.

    - All keys with LED, including the Speaker key, and the Ringer LED, will glow.
    - The LCD display will light up and the booting message appears.
    - Then the 'Loading...' message appears on the phone display.
    - The phone will start loading the Extended IP Phone Firmware. It will display current firmware being loaded.
    - After loading the firmware, the phone will prompt you to change Network settings.
    - Wait for a few seconds.

If you want to change the Network Settings or Server Settings, press Yes key. Detailed instructions for changing the Network Settings of the phone are provided at the end of this topic. See "Network Settings" at the end of this topic.

- The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

  On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from ANANT UCS.

- As the phone downloads the configuration files, the file names will appear one by one.
- On successful download of all configuration files, the phone attempts to register with ANANT UCS.

- On successful registration, the phone will display the current day, date and time, the extension number and name assigned to the Extended IP Phone.

Network Settings

You can change the network settings of the Extended IP Phone. Press the Down key V when the phone is in idle state.

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

The cursor is a non-blinking underscore that appears under the first letter of the first option in the menu. To make a selection in the menu, you must move the cursor in the desired direction using the Up, Down, Forward and Back key. When the cursor is at the desired position, press Enter key to make a selection.

Accessing Network Settings

You can access the Network Settings of the Extended IP Phone in any of the following stages:

1. During start-up, when the phone prompts you to change the network settings after loading the firmware.

   You must press **Yes** key and access network settings.

2. When the phone is making Network discovery, downloading configuration files, attempting registration.

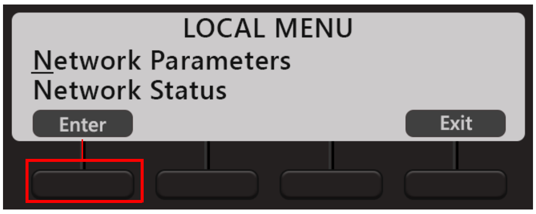
   You must press the **Menu** key to access network settings.

3. When the phone is in idle state, press the Down key V .

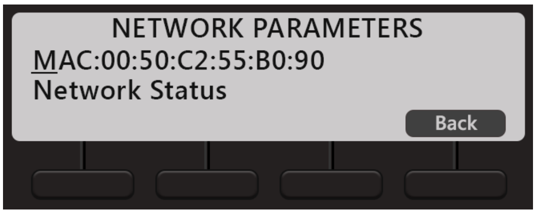
You can configure Network Parameters and view Network status from the Local Menu.

Configuring Network Parameters

- In the Local Menu of the phone, select Network Parameters by pressing the Enter Key.



- The Network Parameters submenu appears.



- Use the Down/Up key to reach the desired network parameter and press Enter key to select. Change the settings as per your requirements.
- Press **Save** key, to save the changes you make.
- You can configure all network parameters described below, except the MAC Address.

- *To enter a dot in the editable fields — IP Address, Subnet Mask, Gateway Address, DNS Address, Server Address — press* * *(Star) key.*
- *If you want to clear a single digit/character, move the cursor to the right of the digit/character you wish to clear. Now press the Delete key. The digit to the left of the cursor will be deleted.*

Before you start configuring the Network Parameters, get acquainted with following context keys:

Context Keys	Description
Enter/OK	To select a particular parameter
Save	To save the changes
Back	To move a step backwards without saving the changes
Delete	To delete previous characters from the cursor position
2Ab/123	2Ab - Alphanumeric Mode 123 - Numeric Mode

Connection Type

- Select the Connection Type as DHCP, PPPoE or Static according to the IP Addressing scheme of your network.

If you select DHCP or PPPoE, the phone will be assigned IP Address, Subnet Mask and Gateway Address, DNS Address, Server Address, automatically by the DHCP/PPPoE server.

For PPPoE Connection Type, you must configure the PPPoE User ID and Password provided by the Internet Service Provider.

If you select Static, you must assign the IP Address, Subnet Mask and Gateway Address to the phone.

IP Address

- If you select Static as Connection Type, enter the static IP Address to be assigned to the phone.

  Enter the desired Static IP Address by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Subnet Mask

- If you select Static as Connection Type, enter the Subnet Mask to be applied on the phone by pressing the digit keys.

  To enter the dot/period in the IP Address, press the Star '*' key.

Gateway Address

- If you select Static as Connection Type, enter the Gateway Address here. This is the IP Address of the LAN Port of the Router.

DNS Server

- If you select Static as Connection Type, select the DNS Server option **Static** and configure the DNS Address.

- If you select DHCP or PPPoE as Connection Type and your Internet Service Provider provides DNS Address, select the DNS Server option **Automatic**. However, if your Internet Service Provider does not provide DNS Address, select **Static** and configure the DNS Address.

DNS Address

- If you select DNS Server as Static, enter the DNS Address here.

  To enter dot/period in the IP Address, press the Star '*' key.

DNS Domain Name

- If you select DNS Server as Static, enter the DNS Domain Name here. DNS Domain Name is optional.

PPPoE User ID

- If you have selected PPPoE as Connection Type, you must enter the User ID provided to you by your Internet Service Provider.

PPPoE Password

- This is the password provided by your Internet Service Provider for the PPPoE User ID. If you have selected PPPoE as Connection Type, you must enter the password provided by your Internet Service provider here.

PPPoE Service Name

- If your Internet Service Provider has provided a Service Name, enter the Service Name here. If your Internet Service Provider has not provided a Service Name, do not configure this parameter.

Primary Server Address

- The system works as the Auto Configuration Server for the phone. Enter the LAN or WAN IP Address/ Domain Name of ANANT UCS here. Default: blank. The phone sends the request for configuration files to this Server Address.

  If you have selected DHCP as Connection Type, the phone will get the Primary Server Address and Port automatically from the DHCP Server. For this, use ***DHCP option 224*** and ***Data Type*** as '***String***' to provide Server Address and Port from the DHCP Server.

  For PPPoE and Static Connection Types, you need to enter the Server Address.

Primary Server Port

- Enter the SPARSH Port of ANANT UCS here. The phone sends the request for configuration files to this port.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

Secondary Server Address

- If required, you can also configure the Secondary Sever Address as a fallback option. If the registration with the Primary Server fails the phone will send the registration and configuration requests to the Secondary Server Address. Speech-cut or unclear speech may be observed during on-going mature calls.

Secondary Server Port

- Enter the Secondary Server Port. The phone sends the request for configuration files to this port if the Primary Server fails.

  Valid range of the port is: 80 or 1025–65535. Default: 80.

VLAN Setting

If your phone is connected to a virtual LAN, you need to configure VLAN Settings.

To enable the VLAN switch to correctly route packets generated by the phone and the computers (on the LAN) to each other, the packets must be tagged with a VLAN header.

The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[9]. The meaning of CoS bits with respect to traffic type is as follows:

CoS	Traffic Type
0	Best Effort
1	Background
2	Spare

9. *The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), that is, better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.*

CoS	Traffic Type
3	Excellent Effort
4	Controlled Load
5	Video
6	Voice
7	Network Control

- Select **Phone VLAN/COS** to add VLAN header to the packets generated by the phone, and add VLAN header to the packets relayed from the PC to its LAN port (packets generated by the PC connected to its PC port).
    - To configure Phone VLAN/COS, select **Enable?**. The VLAN ID will be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc.). Default: Disabled.
    - Select **VLAN ID** and enter the VLAN ID that you have assigned to the VLAN in which the IP Phones are connected. Valid range: 0-4094. Default: 1.
    - Select **SIP CoS** and define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3
    - Select **RTP CoS** and define the CoS (priority) bits in all RTP packets. Valid range: 0-7. Default: 6.
- Select **PC/VLAN CoS** to add VLAN header to all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.
    - To configure PC VLAN/COS, select **Enable?**.
    - Select **VLAN ID** and enter the same ID as you have assigned to the VLAN in which the computers are connected. Valid range: 0-4094. Default: 1.
    - Select **CoS** and define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

PCAP

To capture packets sent and received from and by the phone for monitoring and troubleshooting, you can enable PCAP on the phone. The phone captures up to 1 MB of packets. For more information and instructions on how to use PCAP Trace on the phone, refer to the *EON510_SPARSH VP510 User Guide*.

MAC Cloning

If you require cloning of the MAC Address, configure the following:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- In **Enter Clone MAC Address**, enter the address you wish to clone.

802.1x Authentication

If you want to restrict unauthorized clients from connecting to your LAN, you need to enable 802.1x Authentication. Using EAP MD5 protocol the PC connected to the LAN port of the IP Phone is first authenticated and then it gets connected to LAN.

You need to configure the following 802.1x Authentication parameters:

- Select **Enable?** and press the Enter key. Select Yes to enable.
- Enter the 802.1x Authentication **Identity** provided by you network administrator.
- Enter the 802.1x Authentication **MD5 Password** associated with identity provided by your network administrator.

  When you change the Network Settings, the phone will restart.

Viewing Network Status

- When the phone is in idle state. You must press the Down key V to access the Network Settings.
- Again press Down key V to select Network Status and press the Enter key.

  Use the Down/Up key to view the status of the various network parameters. The status of the following parameters appear on your display as you scroll.

  - **MAC:** This is the MAC Address of the phone.
  - **IP:** The current IP Address assigned to the phone.
  - **MASK:** The current Subnet mask assigned to the phone.
  - **GW:** The current Gateway IP Address assigned to the phone.
  - **DNS:** The Domain Name Server address assigned to the phone.
  - **Active Server:** This displays the Server that is active — Primary, Secondary — with which the phone is currently registered.
  - **S. ADD:** This displays the IP address of the Active Server. It may be the LAN or WAN IP Address / Domain Name of ANANT UCS or the Secondary Server IP Address (if configured) or any Fallback Server.
  - **S. PORT:** This displays the port of the Active Server. It may be the SPARSH Port of ANAN UCS or the Secondary Server Port (if configured) or the Fallback Server Port.
  - **DOMAIN:** The Domain Name assigned to the phone.
  - **802.1x Authentication:** The 802.1x authentication status is displayed—Success, Failure, Authenticating or Disabled.
  - **FIRM:** The version of the current Firmware of the phone.
  - **UBOOT**: The UBOOT release date.
  - **KERNEL**: The KERNEL release date.

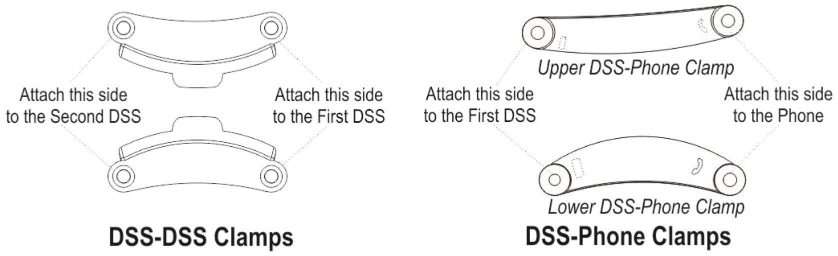
Refer to the ***EON510_SPARSH VP510 User Guide*** to know more.

Installing DSS532 with SPARSH VP510

Once you have installed SPARSH VP510 with ANANT UCS, you can install the DSS532 by following the steps given below:

1. Unpack the box and verify the package contents[10].

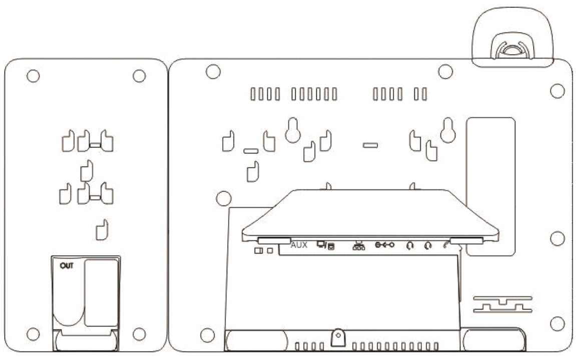
2. Four clamps are provided with the phone — 2 DSS-Phone Clamps and 2 DSS-DSS Clamp.



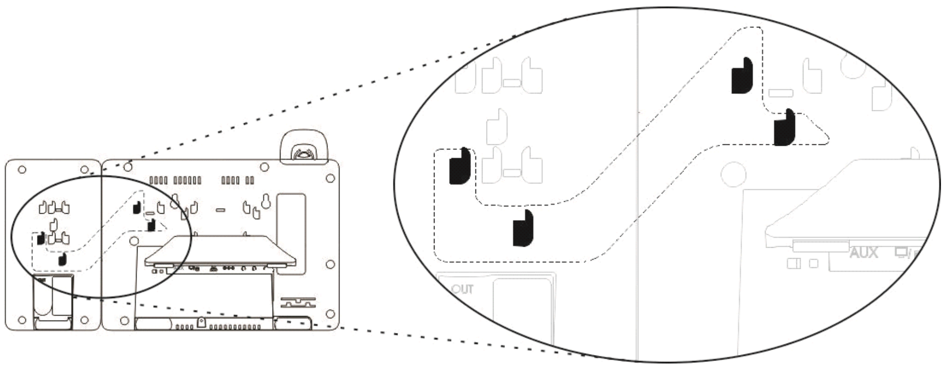
Connecting the First DSS532

Connecting the Extender

3. Turn the phone upside down on the table and place the inverted DSS532 adjacent to it.

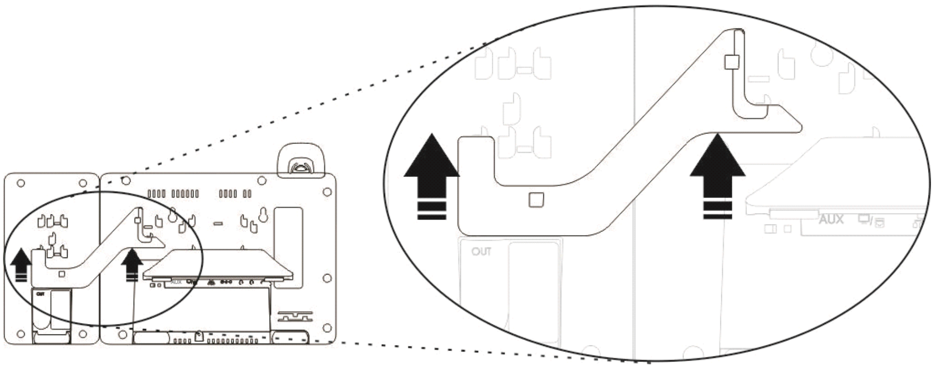


4. To attach the DSS532 with the phone, place the DSS Extender as illustrated below.

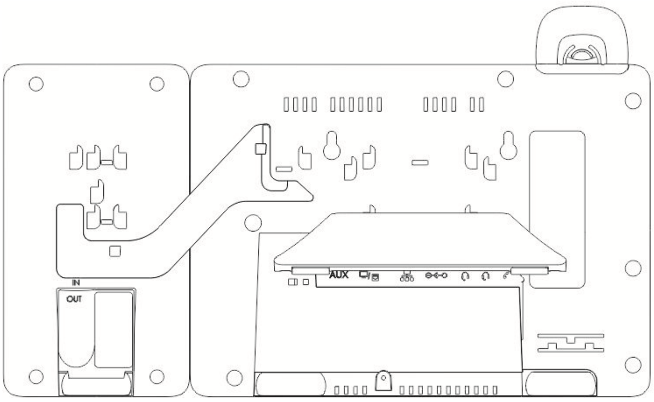


10. *See "Packing List" of Appendix topic.*

5. Insert the hooks on the Extender into the slots provided on the phone and the DSS532.



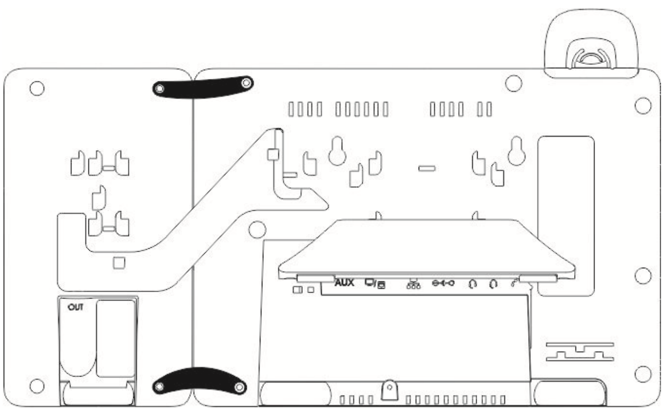
6. Firmly slide the DSS Extender upwards to lock them in place.



Attaching the Clamps

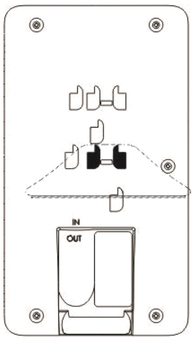
7. Now attach the clamps. To do so,

- Remove the screws to attach the clamps.
- Place the DSS-Phone Clamps between the DSS532 and the phone.
- Insert the screws back to fix the clamps.

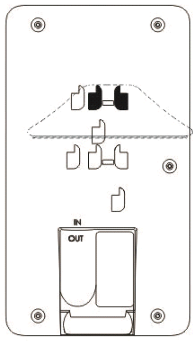


Attaching the Footstand

8. You can mount the DSS532 with the phone on the desk at two angles — **45 degrees** or **55 degrees** by attaching the Foot Stand.



Stand attached at 45 degree angle

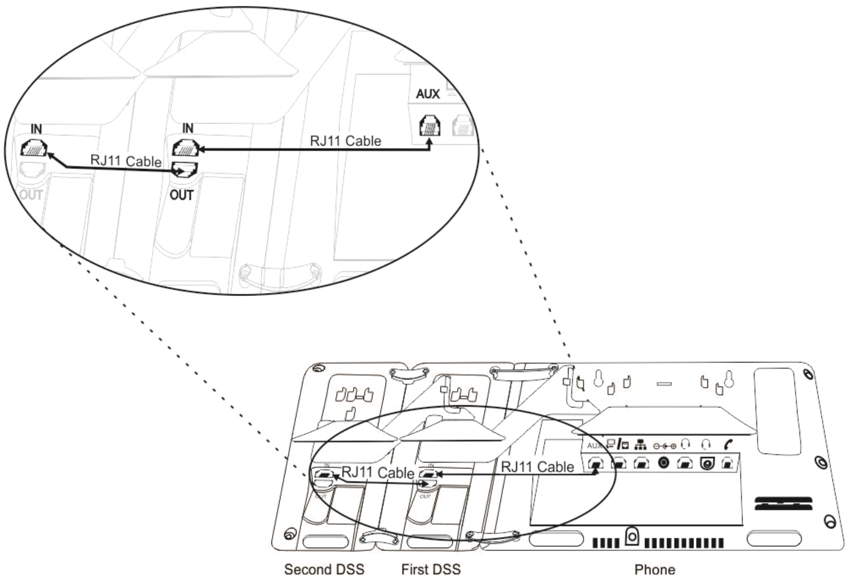


Stand attached at 55 degree angle

Make sure both, the DSS532 and phone are mounted at the same angle.

Connecting the Cables

9. To connect the DSS532 with phone, plug one end of RJ11 Cable into **Auxiliary(AUX) Port** of the phone and the other end into the **IN Port** of the DSS532.

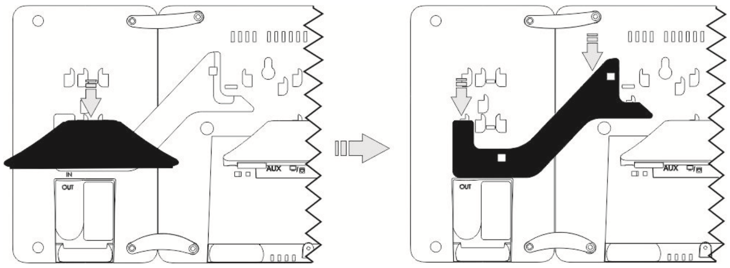


Connecting Multiple DSS532

Remove the Foot Stand

10. Remove the Foot Stand of attached DSS532. To do so,

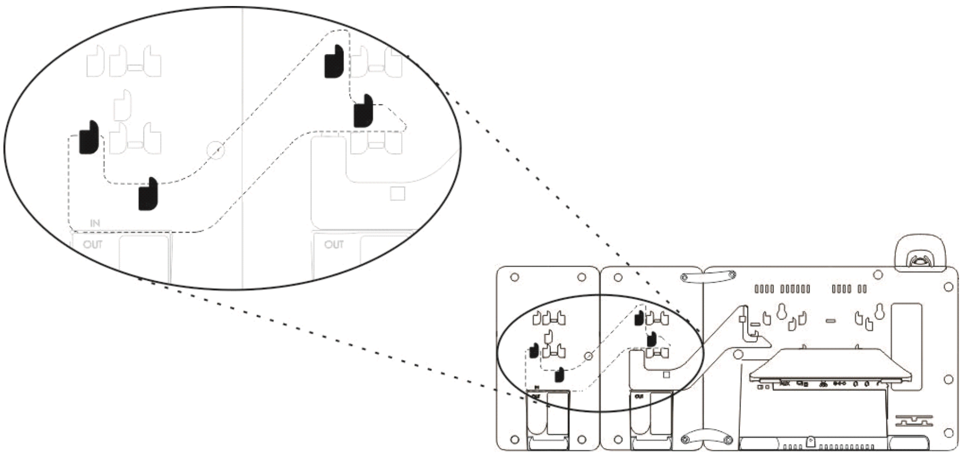
- Firmly slide the Foot Stand of the attached DSS532 downward to unlock.
- Now, slide down the attached DSS Extender in downward direction.



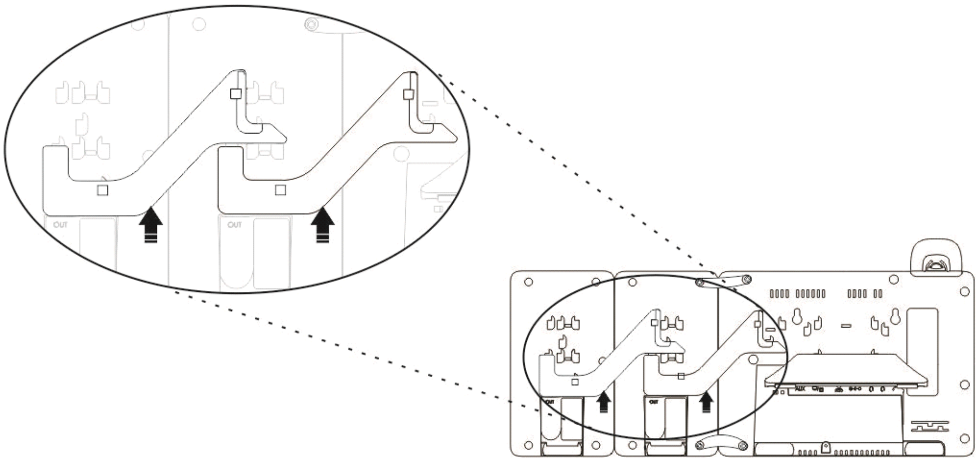
Attach the second DSS Extender

11. To attach the second DSS Extender,

    - Place another inverted DSS532 adjacent to the existing assembly.
    - Place the DSS Extender as illustrated in the diagram below.
    - Insert the hooks on the Extender into the slots provided on both the DSS532.

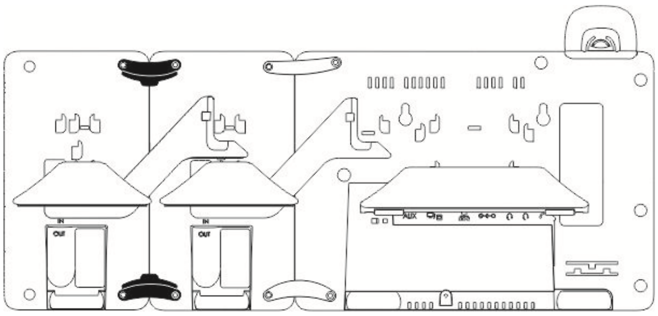


12. Firmly slide both the DSS Extenders upward consecutively (attach the second extender first followed by the existing one attached to the phone) and lock them in place.



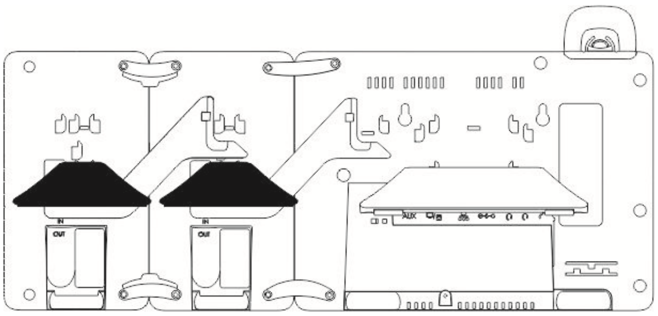
Attach the Clamps

13. Attach the DSS-DSS Clamps between both the DSS532.



Attach the Foot Stand

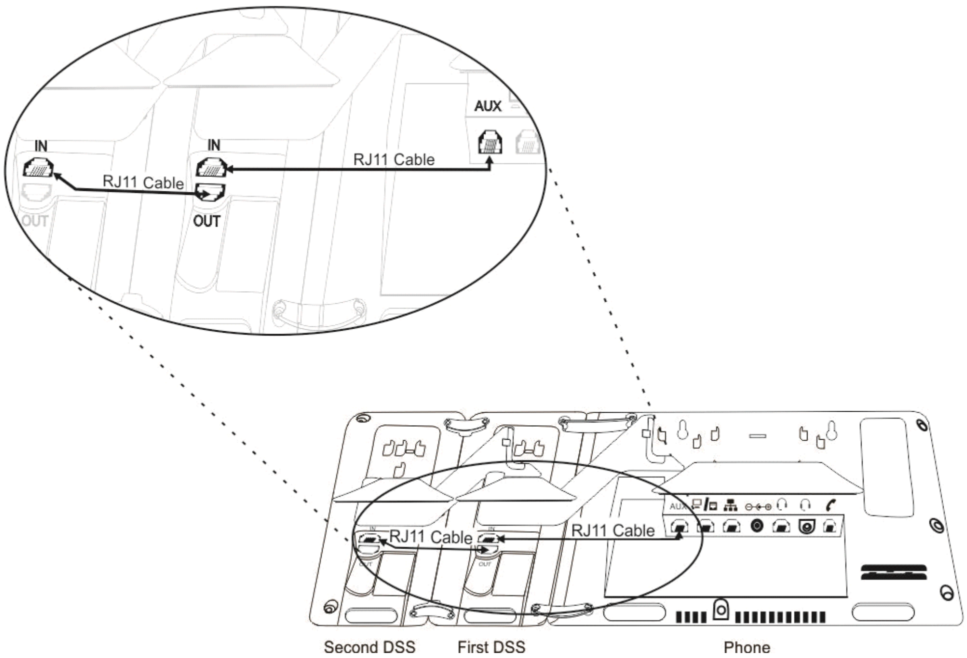
14. Attach the Foot Stand of both the DSS532.



Make sure both, the DSS532 and the phone are mounted at the same angle.

Connect the second DSS532 to the existing assembly

15. Plug one end of the RJ11 Cable into the OUT Port of the existing DSS532 (already connected with the phone) and the other end into the IN Port of the second DSS532.



You can install a maximum of four DSS532 with a phone.

16. After you have connected the DSS532 with the phone, you can configure the DSS Keys. For instructions, see “Programming DSS Console Keys”.

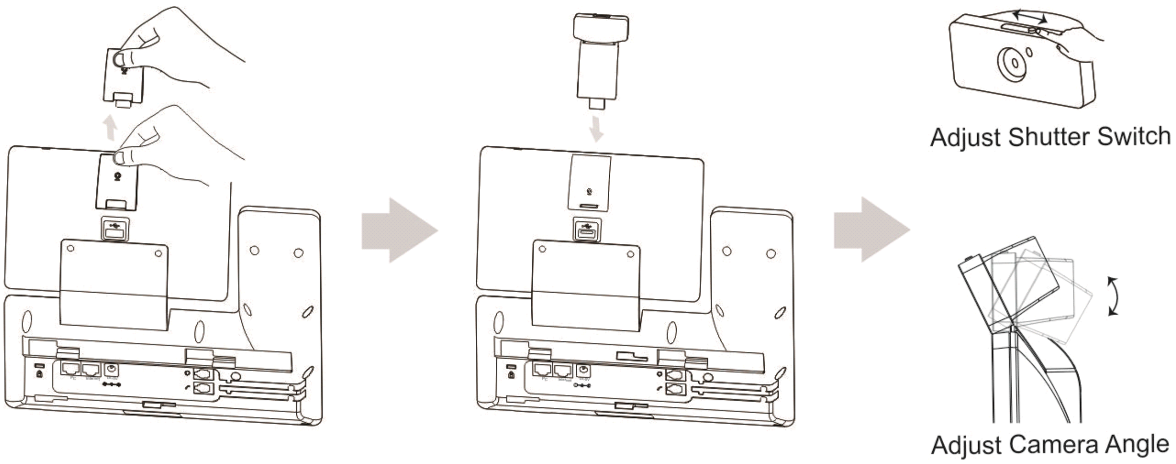
Connecting Extended SPARSH VP710 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended SPARSH VP710 to ANANT UCS:

- Decide the location where you want to place Matrix Extended SPARSH VP710 within your LAN.
- Log in to Jeeves. For instructions, read the topic “Configuring ANANT UCS”.
- You must configure the necessary parameters in ANANT UCS so that Extended SPARSH VP710 can register as a SIP Extension. For instructions, see “Configuring Matrix Extended SPARSH VP710”.

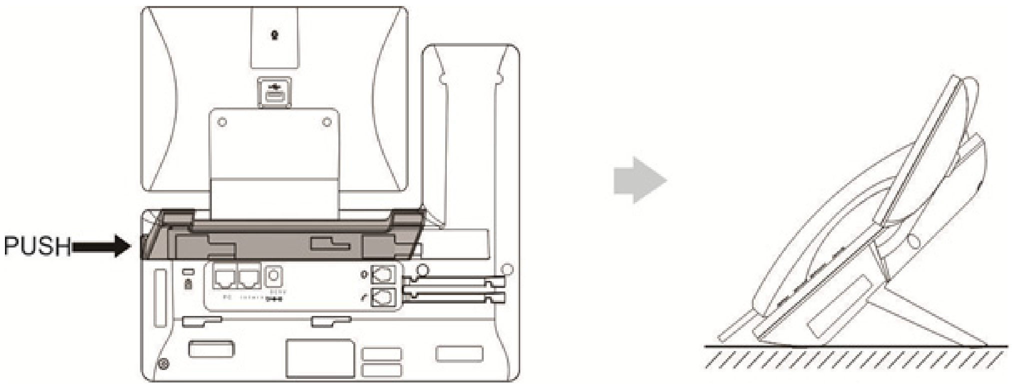
Now, follow the steps described below to install Extended SPARSH VP710.

1. **Inserting the camera**

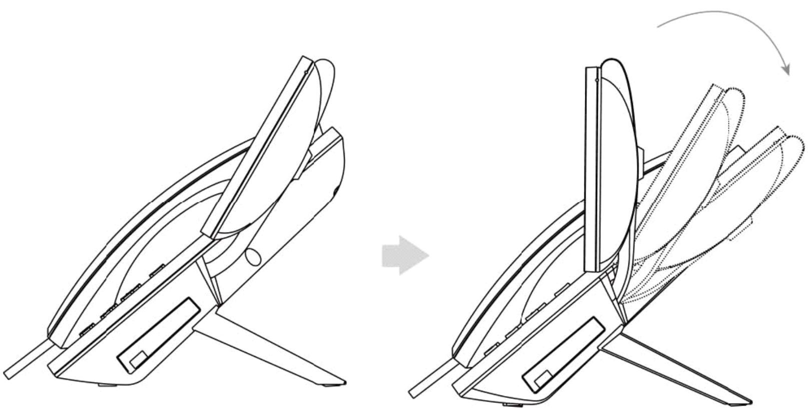


It is recommended to use only the Matrix original Camera, supplied with the IP Phone for video calling. The use of any third-party camera may cause damage to the phone. Damages to the phone caused by using third-party camera is not covered by Matrix warranty.

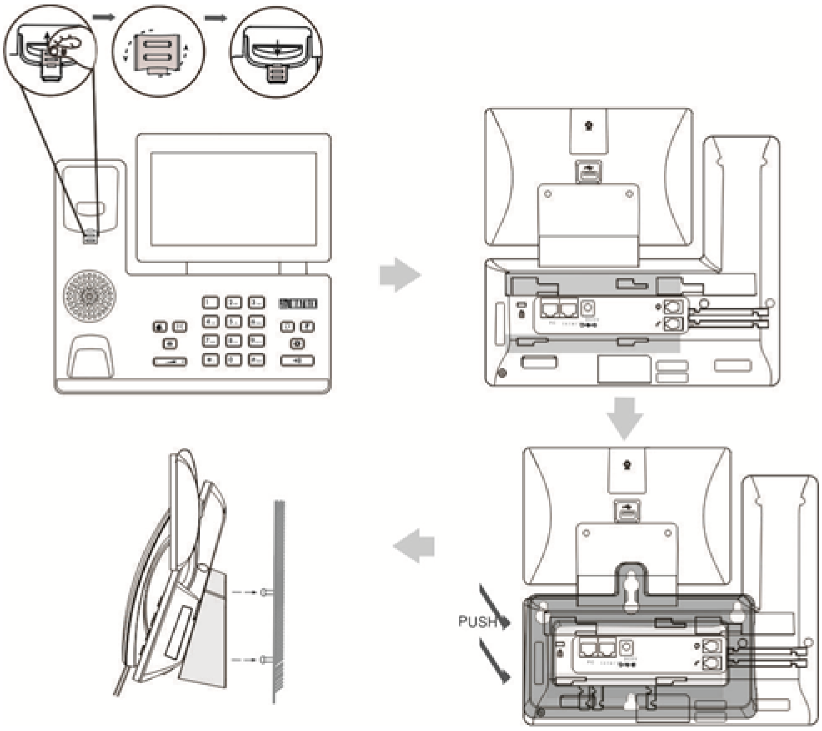
2. **Attaching the stand**



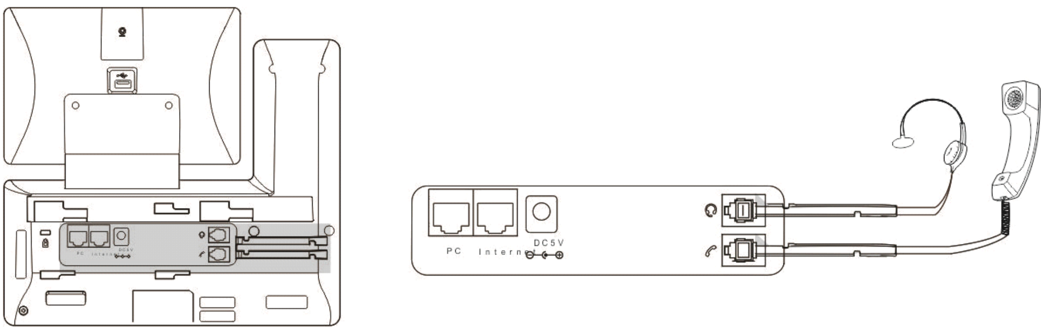
3. **Adjusting the angle of the touch screen.**



4. **Attaching the optional wall mounting bracket**

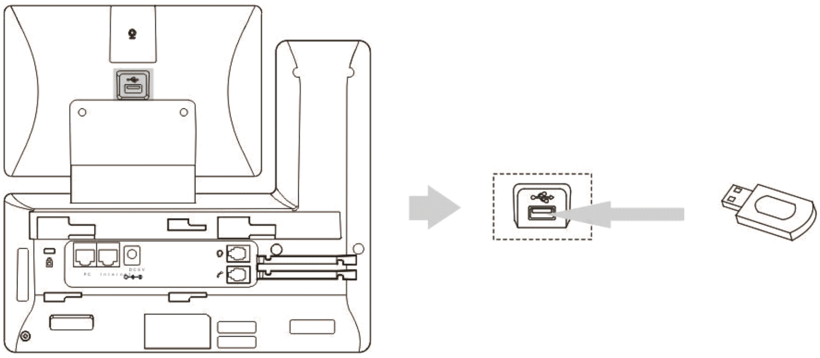


5. **Connect the handset and optional headset.**



A headset is not included in the packaging contents. Contact your dealer/reseller for more information.

6. **Connect the optional USB Flash drive.**



7. **Connect the network and power.**

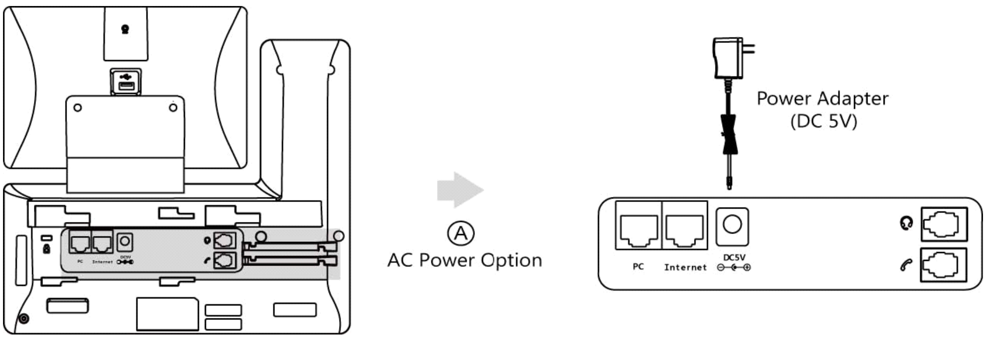
There are two options to connect the power and the network.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.

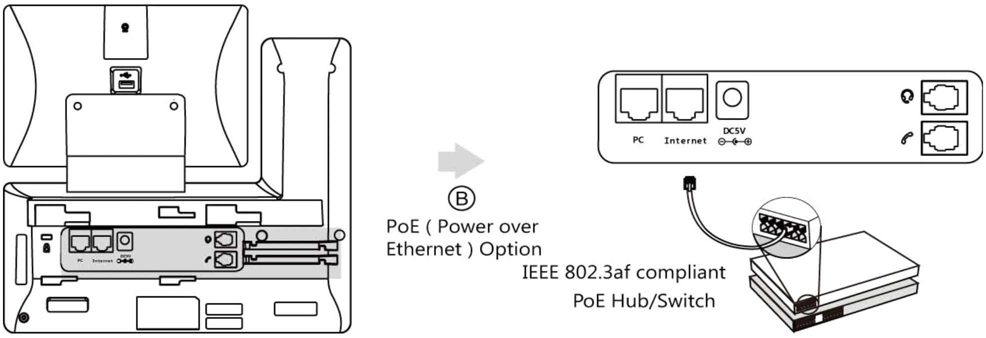


Power over Ethernet (PoE)

With the included or a regular Ethernet cable, the IP Phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

- Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



If in-line power switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

Do not unplug or remove power while the phone is updating firmware.

After the IP Phone is assembled and connected to the power supply, it automatically begins the initialization process.

During this process, the IP Phone displays the start up screen "Welcome Initializing…please wait".

Once the IP Phone is initialized, it displays two different phone modes:
- Standard SIP
- Extended SIP

- Select Extended SIP, to operate the IP Phone in the extended mode. As soon as you select this mode, the booting process initiates again and the start up screen displays "Welcome Initializing…please wait". After the IP Phone is initialized, it attempts to contact a DHCP Server in your network to obtain valid IPv4 network settings (example: IP address, Subnet Mask, Gateway address, DNS address).You need to configure the basic network parameters of the IP Phone manually, if these are not provided by the DHCP Server or if your network does not support DHCP.

Refer to the *EXTENDED SPARSH VP710 User Guide*, for detailed instructions:

- To change the Network Settings of the phone and configure the network parameters.
- To use Wi-Fi for connectivity and configure its parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration (when DHCP is selected) to download the configuration files from ANANT UCS.
- On successful download of all configuration files, the phone attempts to register with ANANT UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in ANANT UCS have been correctly configured as per your installation scenario.

Connecting SPARSH VP210 as Extended SIP Extension

You are recommended to complete the following steps before connecting the Matrix Extended IP Phone to the system when used with ANANT UCS application:

- Decide the location of the Extended IP Phone, whether within the same network or outside, according to your installation scenario.

*If you want to use the **DHCP Server** for assigning IP Address to the Extended IP Phone, select **DHCP option 224** and **Data Type** as '**String**' and program the LAN or WAN Port IP Address /Domain Name and SPARSH Port in the format "**IP_Address:Port"** in your DHCP Server as per your installation scenario.*

- Login to Jeeves. For instructions, read the topic "Configuring ANANT UCS".

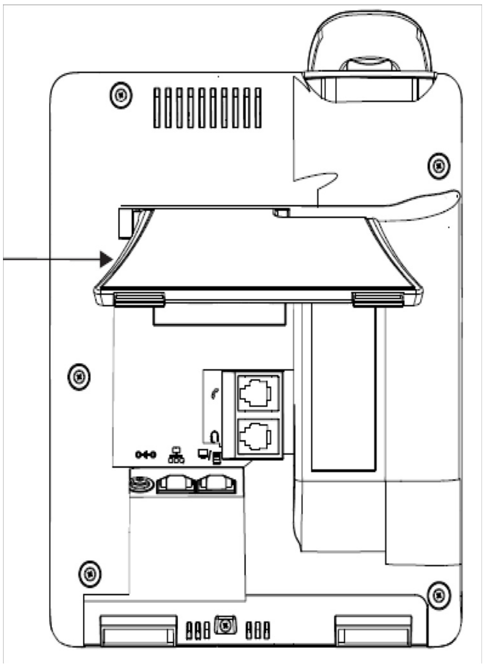
- Assign an extension number (**SIP ID**) to the Extended IP Phone. For instructions on assigning SIP ID, see "Configuring SIP Extensions".

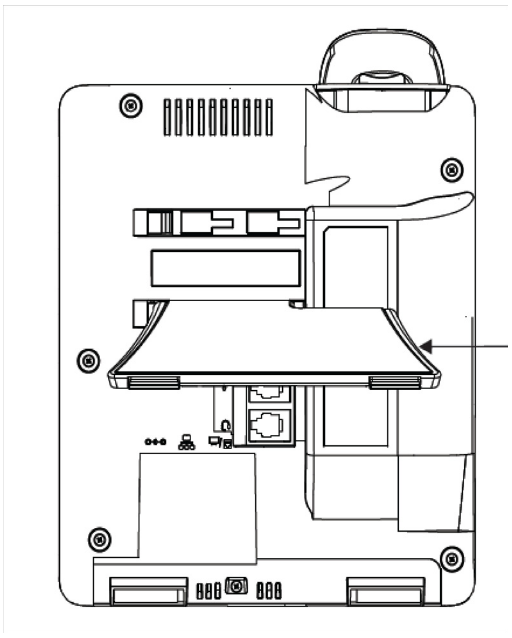
  For the SIP extension number you assigned to the Extended IP Phone, you must configure the necessary parameters in ANANT UCS so that Extended IP Phone can register as a SIP Extension. For instructions, see the topic "Configuring SIP Extension Settings as per the Extended Phone Type" under *Configuring SIP Extensions*.

Now, follow the steps described below to install SPARSH VP210.

1. Unpack the SPARSH VP210 box and verify package contents.

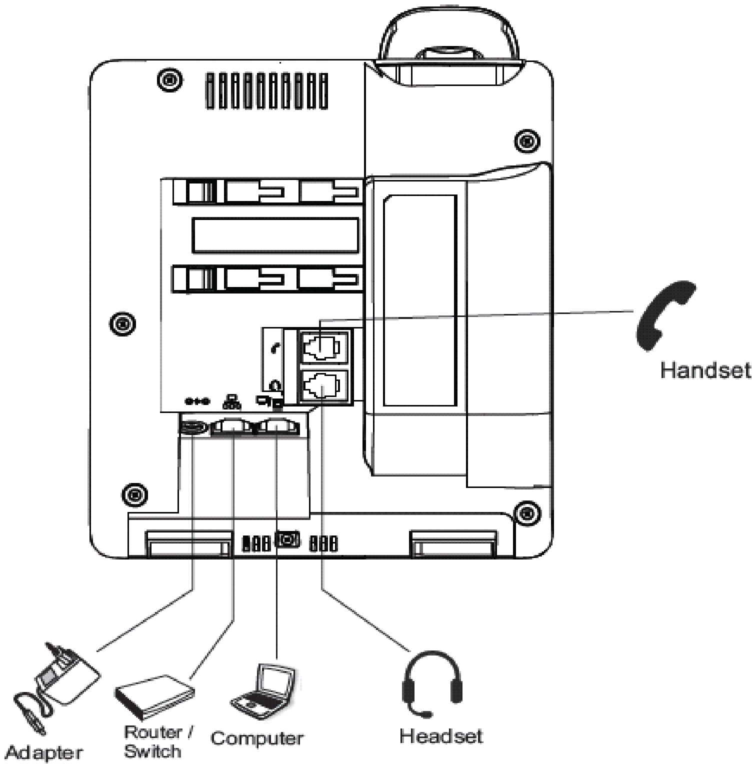
2. When you mount the phone on a desk, you can attach the Foot Stand in two ways at **45° Angle** or at **55° Angle**.





- Decide which of these positions would work for you best and accordingly attach the Foot Stand.

  Refer to the diagram below for connectivity.



3. Connect the Handset to the Phone body.

- Plug the long straightened end of the Spring Cord into the handset jack at the bottom of the phone, marked with the handset symbol .
- Plug the other (short straight) end of the Spring Cord into the jack at the bottom of the handset.

4. If you want to use a Headset (not supplied) with your phone, You may plug in a headset with an RJ9 connector into the headset port at the bottom of the phone, marked with the symbol .
5. To connect the LAN, Port , plug one end of the Ethernet Cable into the LAN Port at the bottom of the phone marked with the symbol and the other end to the IP Network — A Router or LAN Switch.
6. To connect your phone to a computer on your desk, plug one end of the Ethernet Cable (not supplied with this phone) into the PC Port at the bottom of the phone, marked with the symbol / and the other end into the LAN Port of your PC/LAN Switch.
7. It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). In this case you need not connect the Power Adapter.

   If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone, marked with the symbol . Plug in the Power Adapter into a power outlet.

If both the power options, that is, PoE as well as Power Adapter are available to the phone, then the phone will derive power from the PoE enabled LAN Switch.

The IP phone should be used with Matrix original power adapter (5V/0.6A) only. The use of any third-party power adapter may cause damage to the phone.

8. Switch ON power supply.

   When you power the phone, the boot process will be initiated in the following sequence.

   - The LCD display will light up and booting message appears.
   - While loading the application then the loading message appears on the phone display.
   - The phone makes DHCP Discovery and fetches its IP Address and Server Address from the DHCP Server.

If you want to change the Server Settings, press Settings.

Refer to the SPARSH VP210 (Extended) User Guide, for detailed instructions, to change the Network Settings of the phone and configure the network parameters.

- On getting the IP Address and Server Address, the phone initiates Auto Configuration to download the configuration files from ANANT UCS.
- On successful download of all configuration files, the phone attempts to register with ANANT UCS.
- On successful registration, the Home screen appears.

The phone will register successfully, only if the SIP Extension parameters in ANANT UCS have been correctly configured as per your installation scenario.

Refer to the ***SPARSH VP210 (Extended) User Guide*** to know more.

CHAPTER 4

Configuring ANANT UCS

This chapter provides essential information and instructions for configuring ANANT UCS, after you have completed the installation and deployment procedure. To know about the installation and deployment of ANANT, refer to ANANT UCS Installation Guide.

ANANT Unified Communication Server can be configured using an interactive and intuitive Web Graphic User Interface, Jeeves.

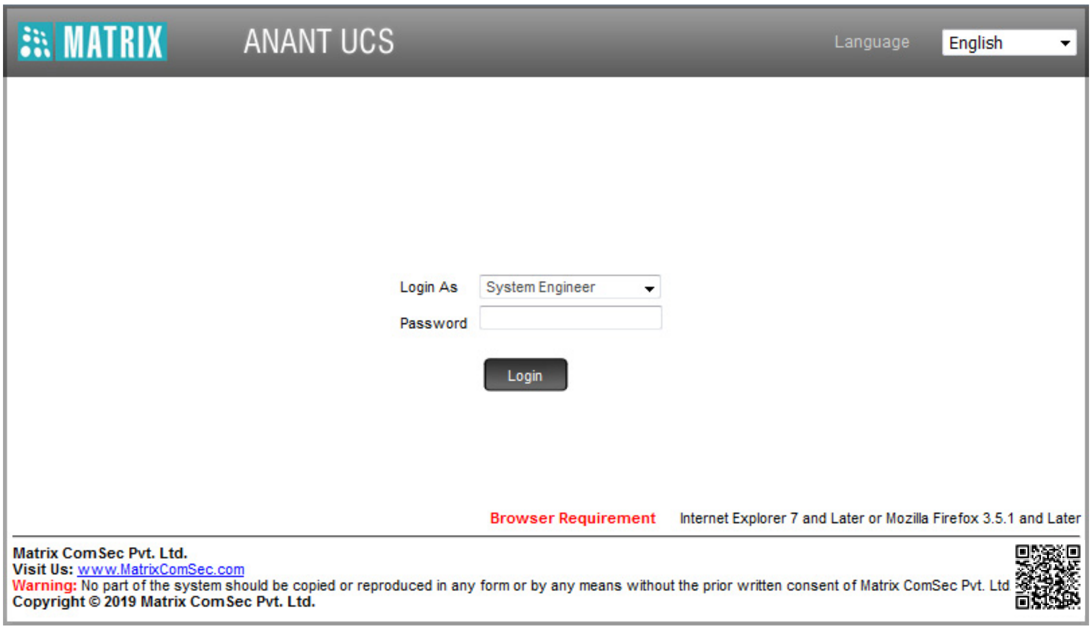
Due to security concerns, the default system settings have been changed. If you have purchased a new system with Firmware later than V2.2, the new default settings will be applied automatically. Refer to “Modified default parameter values for Firmwares later than V2.2”. With these default setting the incoming calls will be placed on the system but outgoing calls (except calls between extensions) will not be routed. Hence, you must change the settings as per your installation requirement.

If you are upgrading the system, refer to “After updating Firmware later than V2.2” and “Modified default parameter values for Firmwares later than V2.2”.

Jeeves is the proprietary web-based configuration software of Matrix. Follow the steps below for configuring ANANT UCS.

- Open the web browser Internet Explorer 7 and Later or Mozilla Firefox 3.5.1 and Later on a (Standalone or LAN PC). To know more, refer to ANANT UCS Installation Guide.
- Enter the default IP address 192.168.1.100 of ANANT UCS in the address bar of the browser.

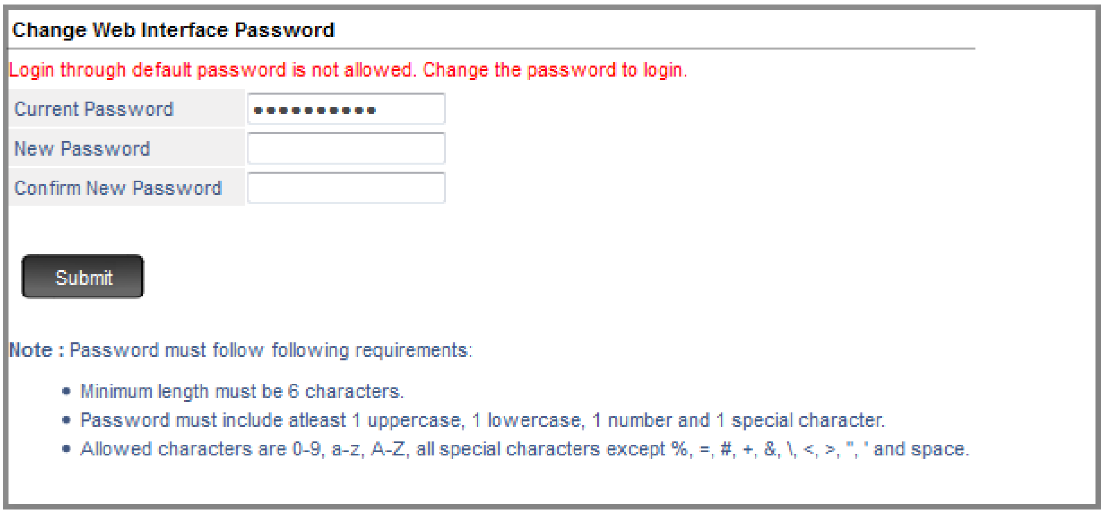
The Login Page of ANANT UCS opens.



Before you start configuring the system, if you wish to view or download the ANANT UCS Quick Start or any other related documents, you can click or scan the QR Code present on the login page of Jeeves.

- In **Login As** select **System Engineer**.
- In **Password**, enter the default SE password, 1234.
- Click **Login**.

You will be prompted to change the default SE Password for accessing Jeeves.



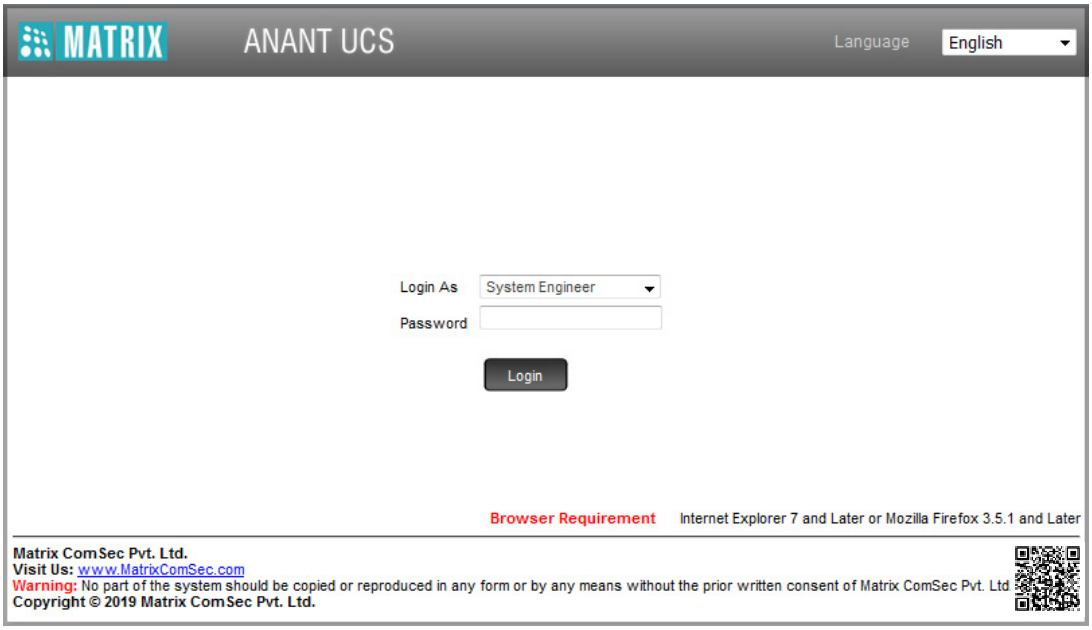
- In **Current Password**, enter the default SE Password, 1234.
- In **New Password**, enter the desired Password.

All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ’ , Double Quote " and Space) are allowed. The new password must be:
• a minimum of 6 characters to a maximum of 12 characters.
• include atleast one upper-case, one lower-case, one number and one special character.

- In **Confirm New Password**, re-enter the new password to confirm.

- Click **Submit**.

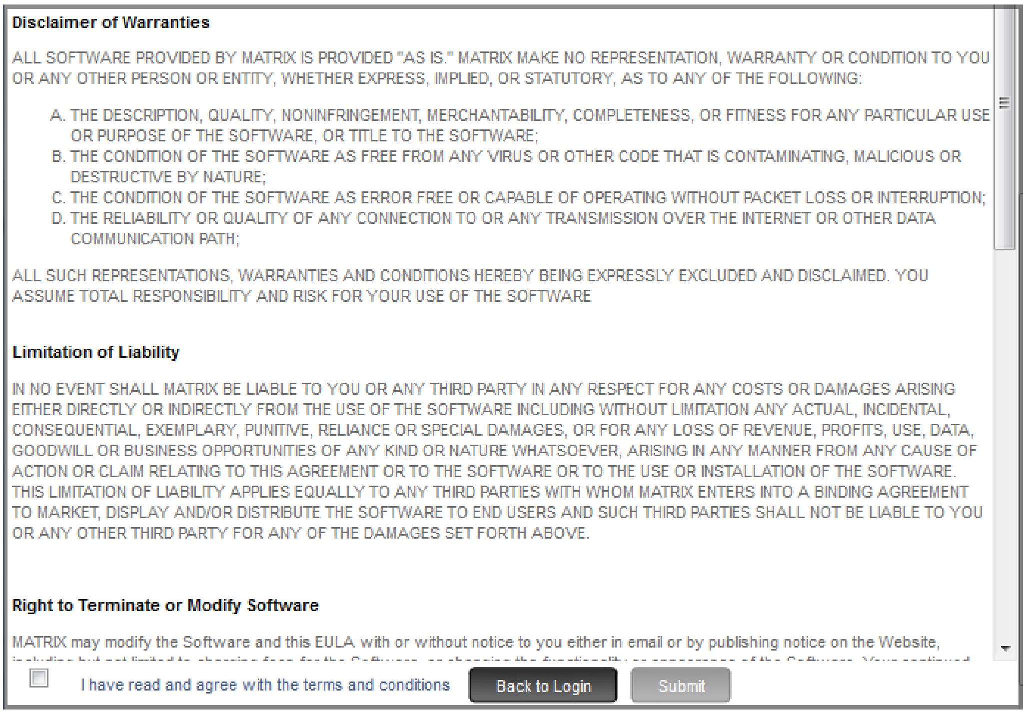
The default SE Password for accessing Web User Interface can be changed using Jeeves only.
You will be re-directed to the Login page again.



- Now, in **Login As** select **System Engineer** and in Password enter the new password.

- Click **Login**.

The MATRIX SOFTWARE END USER LICENSE AGREEMENT window opens.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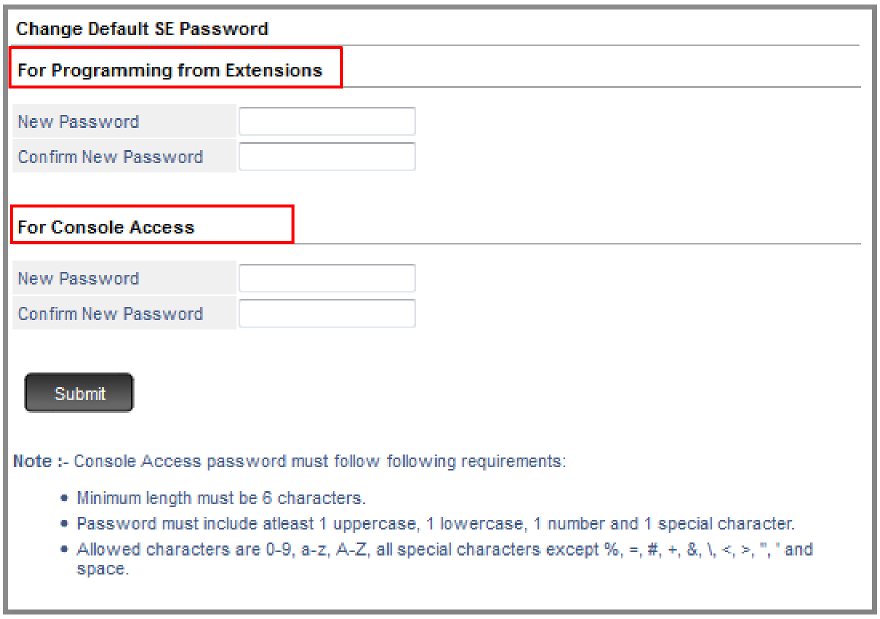
I have read and agree with the terms and conditions | Back to Login | Submit

- Select **I have read and agree with the terms and conditions** and click Submit.

  You will be prompted to change the default SE Password for Programming from Extensions & accessing Console.


Change Default SE Password

For Programming from Extensions

New Password

Confirm New Password

For Console Access

New Password

Confirm New Password

Submit

Note :- Console Access password must follow following requirements:

- Minimum length must be 6 characters.
- Password must include atleast 1 uppercase, 1 lowercase, 1 number and 1 special character.
- Allowed characters are 0-9, a-z, A-Z, all special characters except %, =, #, +, &, \, <, >, ", ' and space.

Under For Programming from Extensions

- Enter the **New Password.** The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.

- In **Confirm New Password**, re-enter the new password to confirm.

You cannot set 1234 as the New SE Password for Programming from Extensions.

Under Console Access

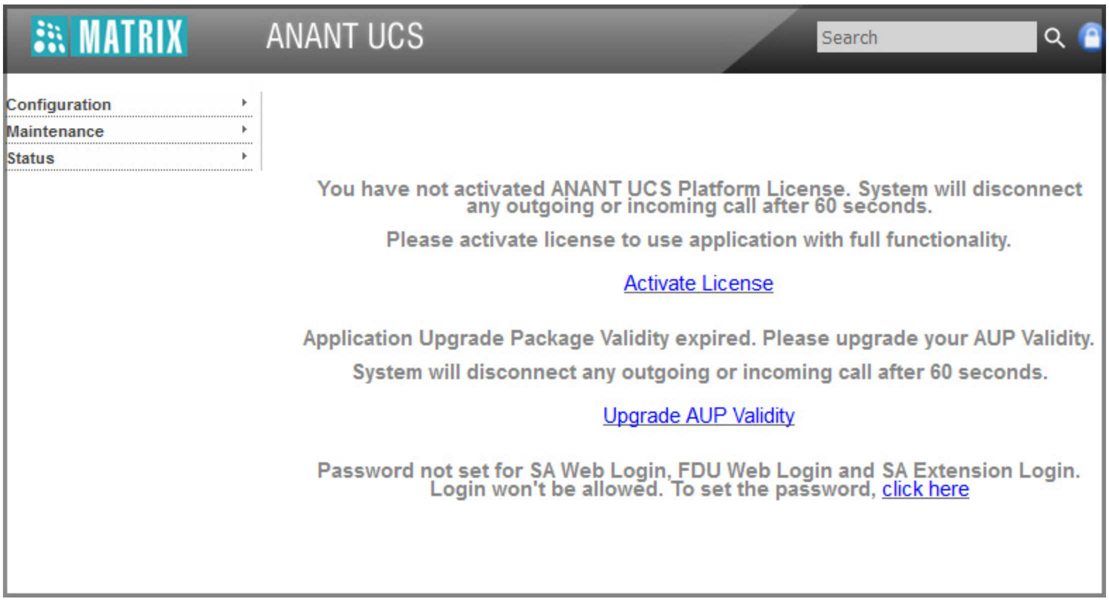
- Enter the **New Password**.

  All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ' , Double Quote " and Space) are allowed. The new password must be:
  - a minimum of 6 characters to a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.

- In **Confirm New Password**, re-enter the new password to confirm.

- Click **Submit**.

  The Welcome page opens.

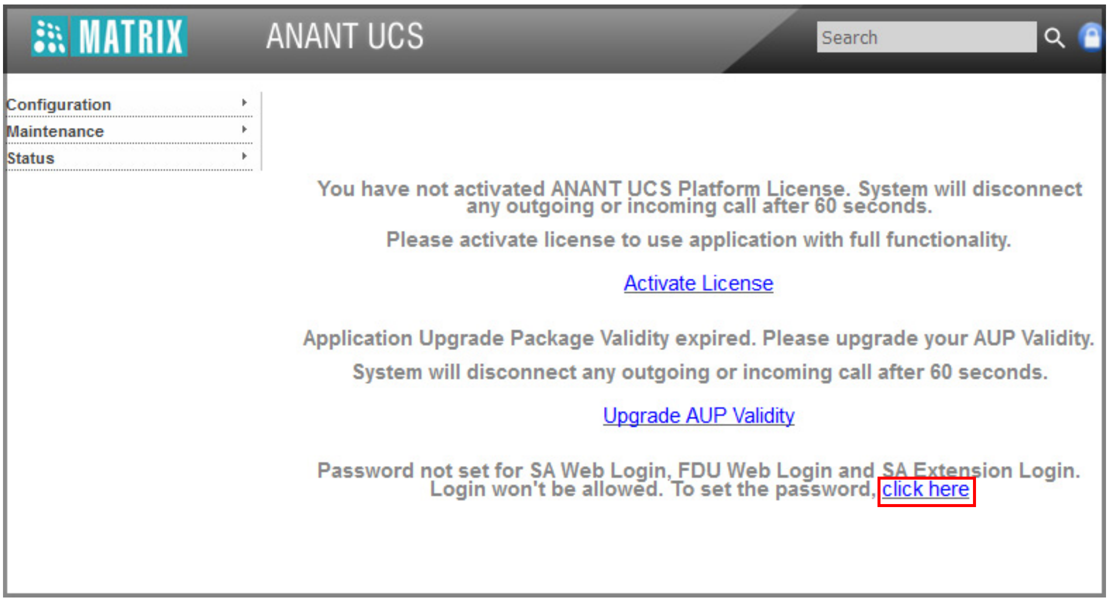


As this password is meant for restricting access to the SE mode, we strongly recommend you to:
- *Keep the password confidential.*
- *Select a complex password that cannot be easily guessed.*
- *Change the password regularly. See "System Security".*
- *In case you forget the SE password, contact the Matrix Technical Support Team. To know more, refer "Forgot the SE Password?".*
- *Do not use the **"Remember Password"** property of your Web Browser.*

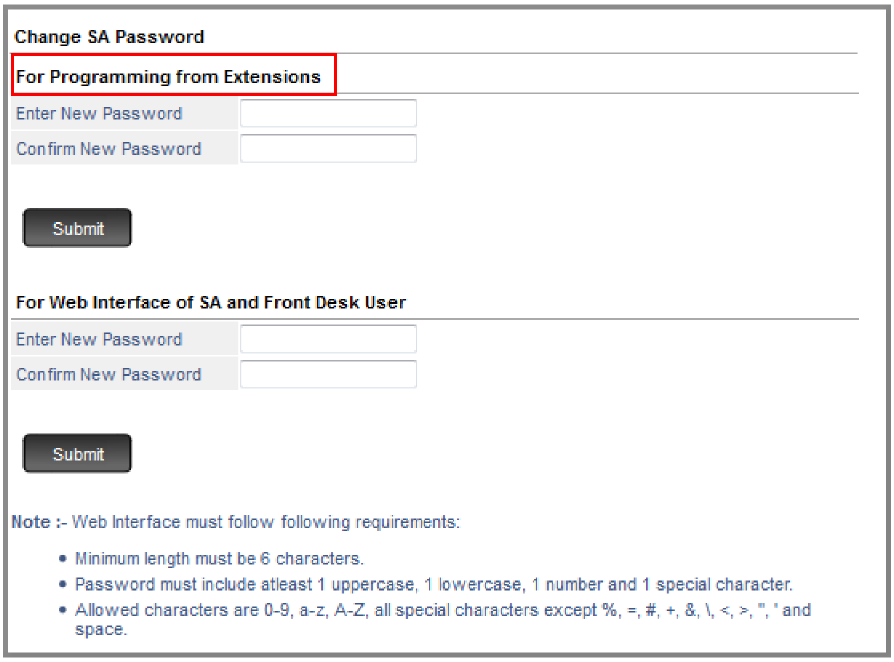
- *Each login session into the SE Mode is set to 60 minutes by default. So, the login session will expire at the end of 60 minutes. You can change the login session according to your preference by changing **Web Interface Logout Timer.** To do so, refer "System Timers and Counts".*

- *It is possible for four users to simultaneously log into the System Engineer Mode of Jeeves.*

You can now set the password for **SA Web Login**, **FDU Web Login** and **SA Extension Login**, click on the link.



The Change SA Password window opens.

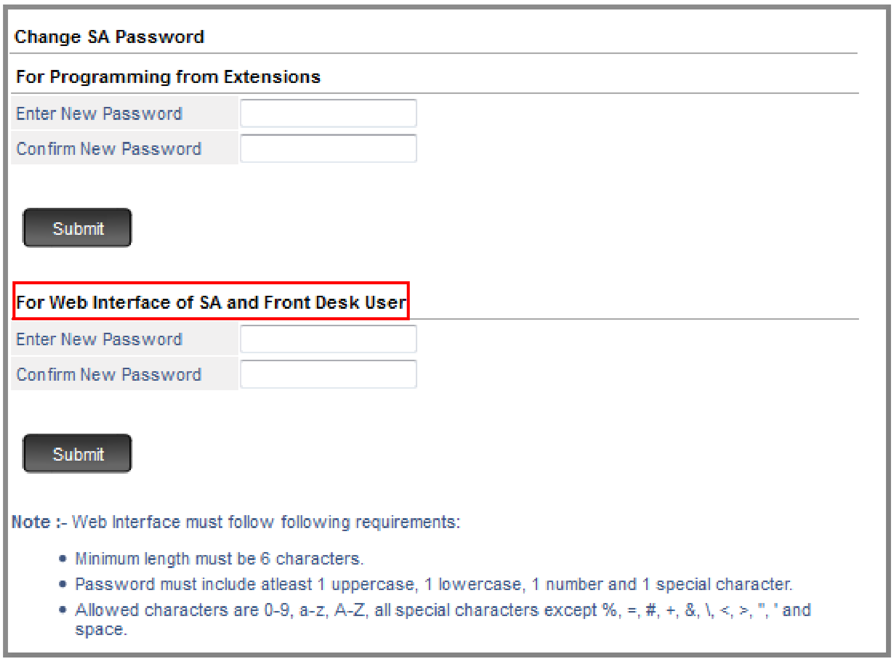


Under For Programming from Extensions

- Enter the **New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit**.

1111 cannot be set as the New SA Password for Programming from Extensions.

Under For Web Interface of SA and Front Desk User,



- **Enter New Password**.

  All ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **'** , Double Quote **"** and **Space**) are allowed. The new password must be:
  - a minimum of 6 characters to a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.

- In **Confirm New Password**, re-enter the new password to confirm.

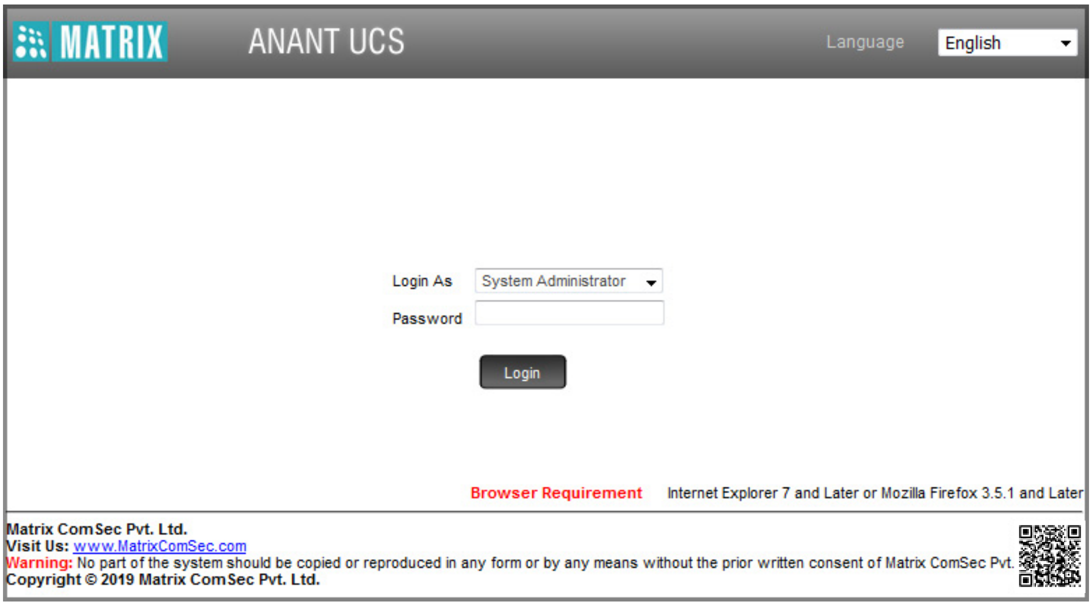
- Click **Submit**.

Log out of the SE mode to enter the SA mode.

The SA password is meant for restricting access to the SA mode, we strongly recommend you to:
- *Keep the password confidential.*
- *Select a complex password that cannot be easily guessed.*
- *Change the password regularly. See "System Security".*
- *Not to use the* ***"Remember Password"*** *property of your Web Browser.*

You can log into the SA mode through Jeeves or from extensions only after you have set the password from SE mode. The password can be set using Jeeves only.

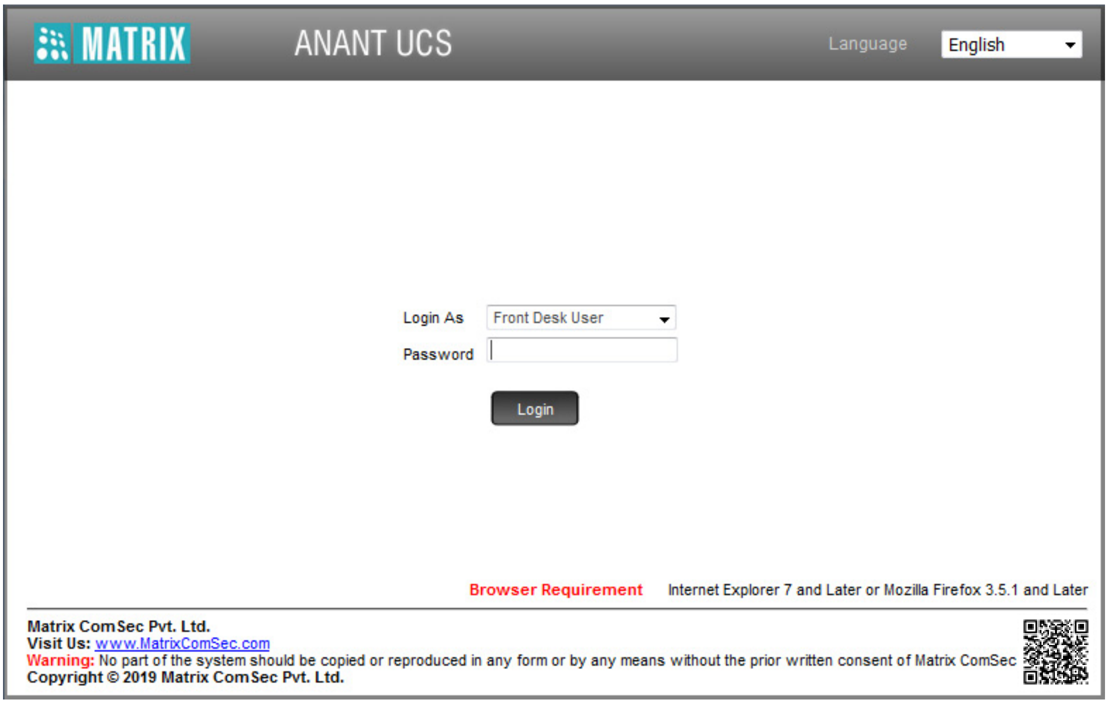
Once the SA password is set, you can log into the SA level through Jeeves.



- On the **Login** page, in **Login As** select **System Administrator**.
- In **Password**, enter the new SA password.
- Click **Login**.

Similarly, you can log into the FDU mode.

To enter the FDU mode,



- On the **Login** page, in **Login as** select **Front Desk User**.

- In **Password**, enter the FDU password.

  Click **Login**.

To provide additional security,

- *the SE and SA Password for accessing Web User Interface will be valid for 90 days only and you will not be able to login with the existing password after 90 days. You will be prompted to change the password.*
- *if you enter a wrong password for five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the "System Activity Log".*

- *If you select the language on the Welcome page or on any of the Login pages, it is valid for the current session only. The default language will be applied on next login.*

- *When you select "Configuring Region" for the country in which ANANT UCS is being installed, the system will load the country-specific default settings and automatically select the local language of the country. This default local language will be applied for every login session, unless you select another language as the default local language.*

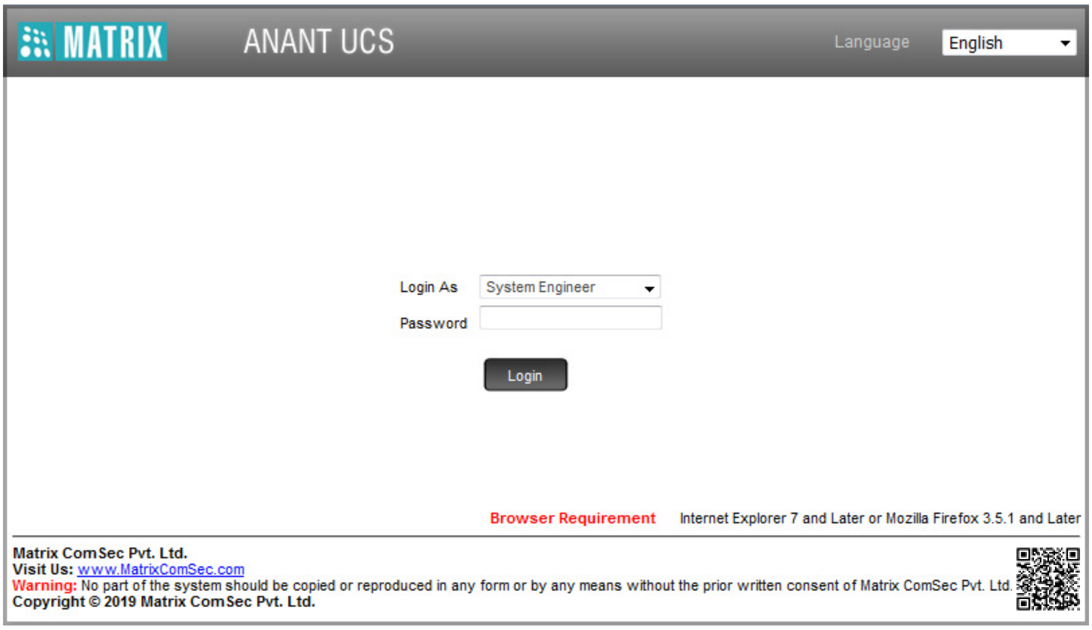
- *The default local language set on selecting the Region can also be changed from the "System Parameters" page of Jeeves.*

Using Configuration

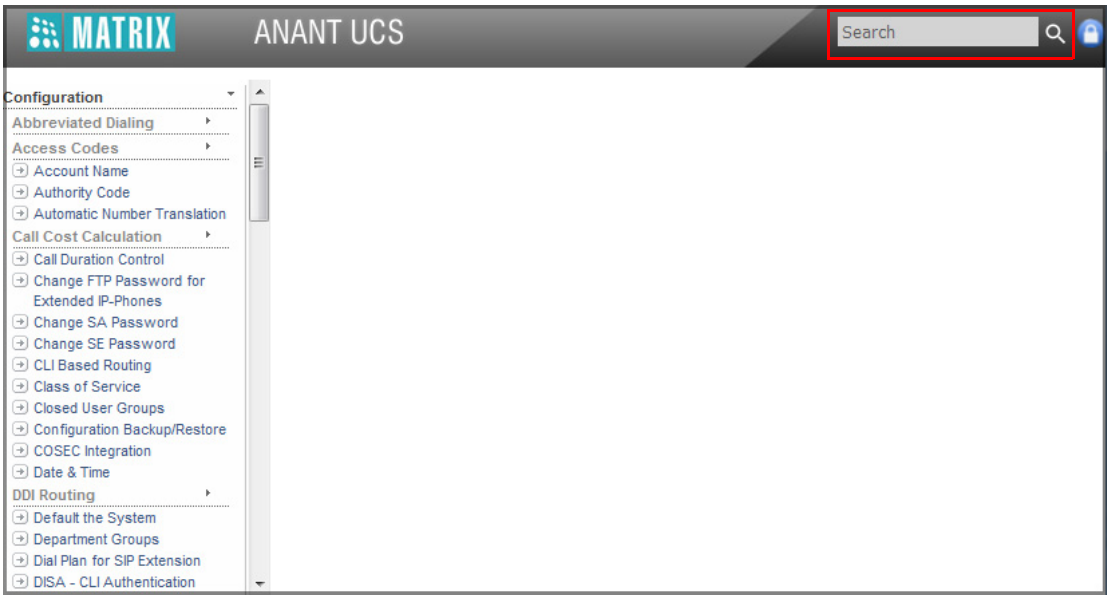
The Configuration allows configuration of all configurable parameters of the system.

You can configure all the parameters using Jeeves.

To configure the Configuration parameters,



- You must log into Jeeves via System Engineer Login.
- Click **Configuration** to configure the desired parameters.



- You can also use **Search** bar to search for the desired parameter page you wish to configure.

To search a parameter,

- Enter the key word in the **Search** bar.

  A list of the pages containing the matching search results will be displayed.

- Click on the desired link displayed in the search list.

  The page opens. Configure the parameters.

The maximum number of search results that will be displayed are 25.

The system does not support search of user configured data. For example Feature Access Code, Extension number, Name, SIP ID etc.

Configuring Region

ANANT UCS is a versatile system that can operate anywhere in the world, meeting the diverse customer requirements worldwide.

To speed up the process of system configuration, ANANT UCS is supplied with pre-defined values for the system and feature settings, referred to as "Default Settings". These default values are loaded when the system is installed and are sufficient for getting the system into operation. However, users may alter or customize the Default Settings to match their exact requirement.

ANANT UCS provides Default Settings to match country/region-specific requirements of users around the world. The system is designed to work efficiently in any country with these default settings.

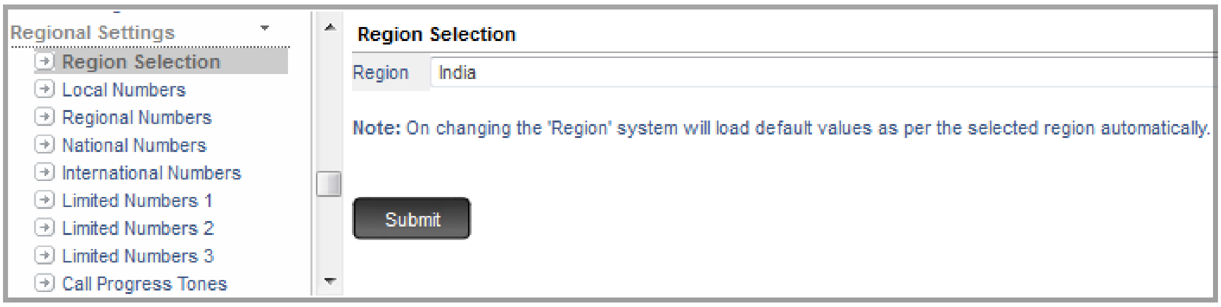
To load the country-specific Default Settings, users must select the Region that is, the country in which the system is installed.

Certain countries are divided into various regions. If you select only a different region in the same country the DST and Date and Time Settings will only change as per the selected region. The other parameters are country-specific.

India is selected as the default Region. So, if you are installing ANANT UCS in a country other than India, change the Region.

Changing Region

- Login as System Engineer.
- Under **Configuration**, click **Regional Settings**.
- Click **Region Selection**.



- In the **Region** list, select the country where the system is installed.
- Click **Submit**.

Configuring Network Parameters

ANANT UCS has a WAN as well as a LAN Port. However, the provision of the WAN as well as the LAN Port purely depends on the number of Ethernet Ports present on the Bare Machine on which ANANT is installed.

When ANANT is installed on a Bare Machine having a single Ethernet Port, the provision of only the WAN Port is provided. If you want the provision of both the WAN and the LAN Ports, you must install ANANT on a Bare machine having multiple Ethernet Ports.

You can install ANANT UCS in a Public IP Network or in a Private Network, behind a NAT Router.

When ANANT UCS is installed in a Public IP Network,

- the WAN Port is connected to a Broadband Router/Modem
- Public IP is assigned to the WAN Port
- the LAN Port is connected to a Switch/Hub to which SIP devices are connected

When the ANANT UCS is installed in a Private Network, behind a NAT Router,

- the WAN Port is connected to the Switch/Hub
- Private IP is assigned to the WAN Port
- SIP devices within the LAN can get registered with the system

You must configure the Network Parameters, depending on your installation scenario.

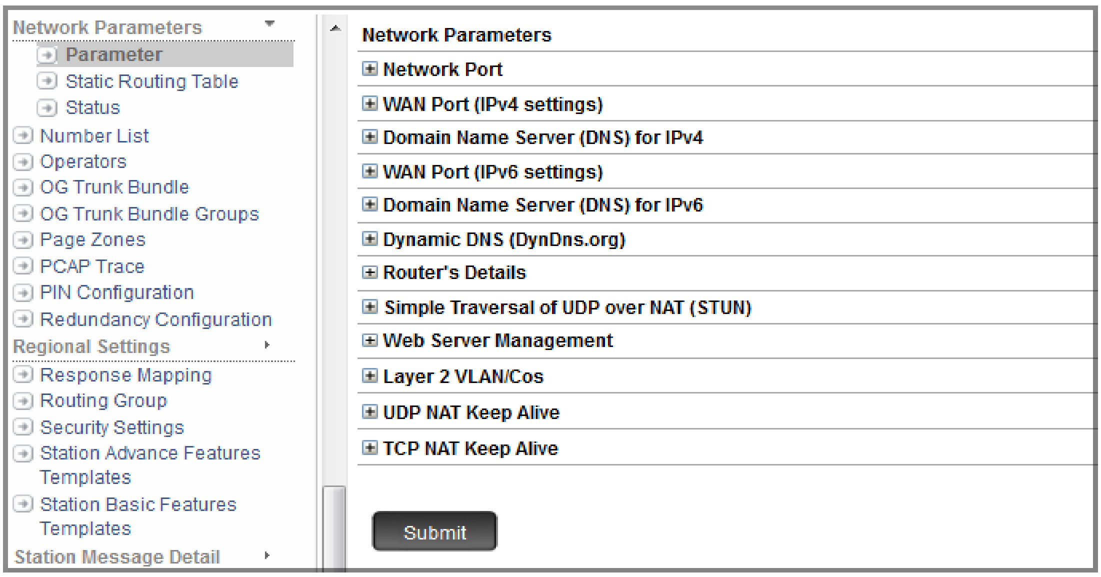
This topic is written considering ANANT UCS has a provision of both the WAN as well as the LAN Port.

Configuring Network Parameters

- Login as System Engineer.

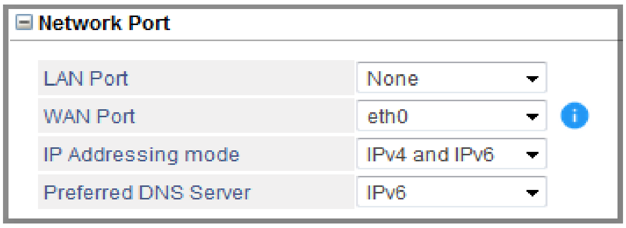
- Under **Configuration**, click **Network Parameters**.

- Click **Parameter.**



Network Port

Click **Network Port** to expand.



- **LAN Port:** Select the Ethernet Port you want the system to use as a LAN Port. By default, LAN Port is not defined. After defining the LAN Port, you can hover the mouse over the **Information** icon to know the mac address and the status of the selected Ethernet Port.

- **WAN Port:** Select the Ethernet Port you want the system to use as a WAN Port. By default, WAN Port is defined. You can hover the mouse over the **Information** icon to know the mac address and the status of the defined Ethernet Port.

  You cannot assign the functionality of both the LAN and the WAN Port to a single Ethernet Port.

Following are the parameters in which the option to select the LAN Interface will be provided only when the LAN Port is defined in the System.

- ***Redundancy On** in Redundancy Configuration*
- ***Registrar Server Address** in SIP Extension Settings*
- ***Internal Registrar Server Address** in SIP Extension Settings*
- ***Internal Server Address** in Auto Sign-In Parameters*
- ***Source Port IP Address** in SIP Trunk Parameters*
- ***Interface** in PCAP Trace*

Moreover, when a defined LAN Port is set to None, the system will change the LAN Interface to the WAN Interface automatically in the above listed parameters.

*For example: You have selected the option **Use LAN Port IP Address** in **Registrar Server Address** for registering the IP Phones. Now, when the LAN Port is set to None, the system will automatically set the option **Use WAN Port IP Address** in **Registrar Server Address**. However, your IP Phones will get de-registered. To register the IP Phones again, you must update the IP Address of the Server in the IP Phones manually and change your existing network deployment, if required.*

*Similarly, if you have configured Redundancy on the LAN Port and due to some reason, when LAN Port is set as None, the redundancy configuration will automatically fallback on the WAN Port. However, you have to change the IP Address configured in the parameters **Primary Server-Redundancy WAN IP Address** and **Backup Server-Redundancy WAN IP Address** in both the Servers for the working of redundancy. For more clarity, refer to* *"Configuring Redundancy Parameters".*

When the LAN/ WAN Port is displayed as Blank, it means that the Ethernet port configured as LAN/ WAN is not detected.In such cases, you must check the Ethernet port and change the configuration, if required.

- **IP Addressing mode:** Select the IP version you want the system to use. You may select — IPv4 only or IPv4 and IPv6. Default: IPv4 only.

If you select IPv4 only, you can configure the IPv4 parameters only.

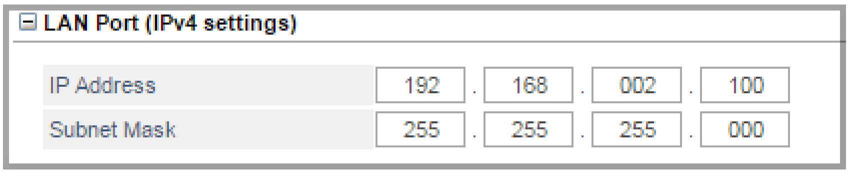
If you select IPv4 and IPv6, you can configure both IPv4 and IPv6 parameters.

- **Preferred DNS Server:** If you select IPv4 and IPv6 as the IP Addressing mode, you must select the Preferred DNS Server — IPv4 or IPv6. Default: IPv4

LAN Port (IPv4 settings)

Make sure that the Redundancy LAN IP Address configured in “Configuring Redundancy Parameters” and LAN Port IP Address are in the same subnet if you wish to use Redundancy.

Click **LAN Port (IPv4 settings)** to expand.



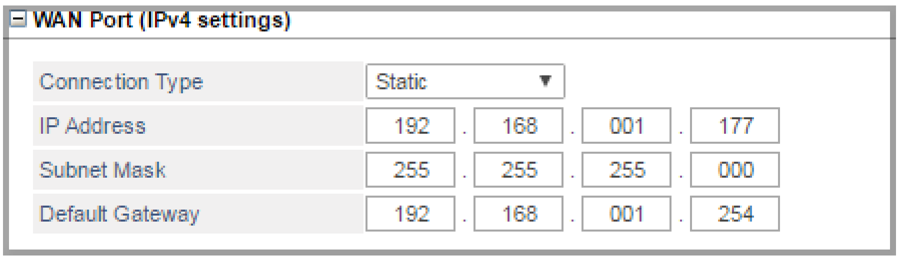
- **IP Address:** Enter the IP Address to be assigned to the LAN Port. The default IP Address is 192.168.002.100. You can assign only Static IP to the LAN Port.

- **Subnet Mask:** Enter the Subnet Mask to be assigned to the LAN Port. The default Subnet Mask is 255.255.255.0

WAN Port (IPv4 settings)

Make sure that the Redundancy WAN IP Address configured in “Configuring Redundancy Parameters” and WAN Port IP Address are in the same subnet if you wish to use Redundancy.

Click **WAN Port (IPv4 settings)** to expand.



- **Connection Type:** Select the appropriate Connection Type for the WAN port, according to the IP Addressing scheme of your installation scenario. Consult your Network Administrator in this regard. Default: Static.

  - **Static:** Select this option if the connection type is Static. When you select this option, you must:
    - assign an IP Address to the WAN Port.
    - change the Subnet mask of the WAN Port as appropriate.
    - configure the Router's LAN Interface IP Address as the Gateway IP Address.
    - Configure the DNS Address/Domain Name provided by your ISP or ask your LAN Administrator for the DNS Address and Domain Name.

  - **DHCP:** Select this option if the connection type is DHCP. As the DHCP Server will automatically assign IP Address, Subnet Mask, Gateway Address to the WAN Port, you need not configure any of these.

- **IP Address:** If you have selected 'Static' as the Connection Type, the default IP is 192.168.001.100.

  If you have selected DHCP as the Connection Type, the IP Address will be assigned by the DHCP server.

- **Subnet Mask:** You must enter the Subnet Mask, only if you have selected 'Static' as the Connection Type.

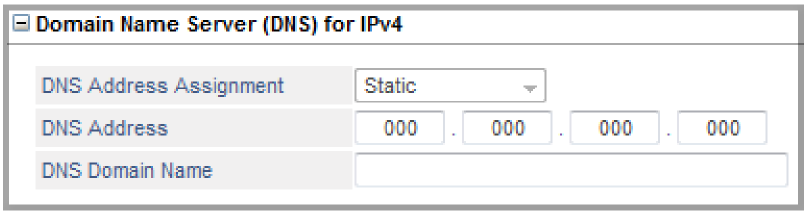
  If you have selected DHCP as the Connection Type, the Subnet Mask will be assigned by the DHCP server.

- **Default Gateway:** You must enter the Gateway IP Address, only if you have selected 'Static' as the Connection Type.

  If you have selected DHCP as the Connection Type, the Gateway IP Address will be assigned by the DHCP server.

Domain Name Server (DNS) for IPv4

Click **Domain Name Server (DNS) for IPv4** to expand.



- Configure the following DNS Connection settings for the WAN Port:

  - **DNS Address Assignment:** If you have selected 'Static' as your network Connection Type (IP Addressing), you can select only 'Static' as the DNS Address Assignment.

    If you have selected DHCP as your network Connection Type, and the DHCP server provides DNS Address, set the DNS Address Assignment to 'Auto'. If the DHCP server does not provide DNS Address, set DNS Address Assignment as 'Static' and configure the DNS Server Address provided by your ISP.

  - **DNS Address:** This field will be editable only if you have selected DNS Address Assignment as 'Static'. Enter the DNS Address here.

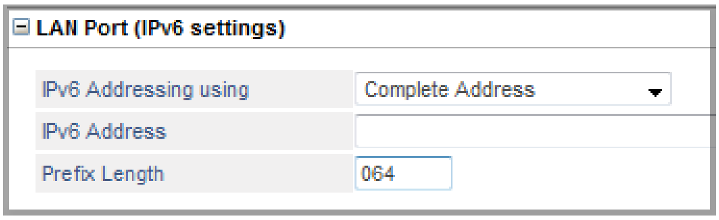
    If you have selected DNS Address Assignment as 'Auto', the DNS Address will be assigned by the DHCP server.

  - **DNS Domain Name:** Configure the DNS Domain Name if provided by your ITSP/LAN Administrator. Otherwise, keep it blank. The Domain Name may be a maximum of 40 characters. Default: Blank.

LAN Port (IPv6 settings)

Make sure that the Redundancy LAN IP Address configured in "Configuring Redundancy Parameters" and LAN Port IP Address are in the same subnet if you wish to use Redundancy.

Click **LAN Port (IPv6 settings)** to expand.



- **IPv6 Addressing using:** You can select — Complete Address or Prefix. Default: Complete Address.

  If you select Complete Address,

  - Configure the **IPv6 Address** and the **Prefix Length**. The IP Address configured will be considered as the complete IPv6 address.

    For example: 2001:0:3238:DFE1:63::FEFB

    The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

    Valid Range of the IPv6 Address is A to F, a to f, 0 to 9,:(colon). It can be a maximum of 39 characters. Default: Blank.

    The Prefix Length range is from 1 to 128 bits. Default: Blank.

  If you select Prefix,

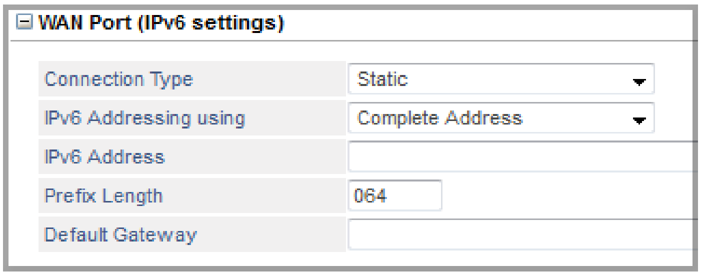
  - Configure the **IPv6 Prefix**. The system will consider the configured value as 64 bit Prefix of the IPv6 Address. Then the system will generate the complete IPv6 Address from it. Default: Blank.

    Valid characters 0 to 9, a to f, A to F and : (colon). It can be a maximum of 21 characters.

WAN Port (IPv6 Settings)

Make sure that the Redundancy WAN IP Address configured in "Configuring Redundancy Parameters" and WAN Port IP Address are in the same subnet if you wish to use Redundancy.

Click **WAN Port (IPv6 Settings)** to expand.



- **IPv6 Connection Type:** By Default, Static is defined as the Connection Type for the WAN port.
  - **IPv6 Addressing using:** You can select — Complete Address or Prefix.

    If you select Complete Address,

    - Configure the **IPv6 Address** and the **Prefix Length**. The IP Address configured will be considered as the complete IPv6 Address.

      The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).

      Valid Range of the IPv6 Address is A to F, a to f, 0 to 9,:(colon). It can be a maximum of 39 characters. Default: Blank.

      The Prefix Length range is from 1 to 128 bits. Default: Blank.

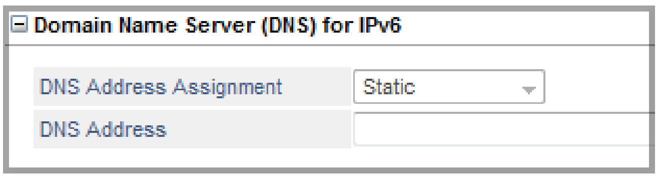
    If you select Prefix,

    - Configure the **IPv6 Prefix**. The system will consider the configured value as 64 bit Prefix of the IPv6 Address. Then the system will generate the complete IPv6 Address from it. Default: Blank.

      Valid characters 0 to 9, a to f, A to F and : (colon). It can be a maximum of 21 characters.
  - **Default Gateway:** Configure the Gateway IP Address for the WAN Port. It can be a maximum of 39 characters.

Domain Name Server (DNS) for IPv6

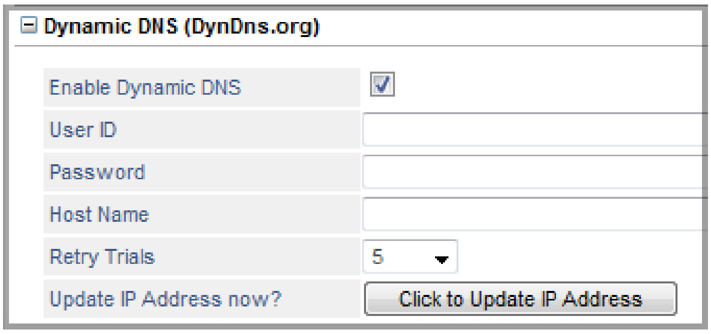
Click **Domain Name Server (DNS) for IPv6** to expand.



- Configure the following DNS Connection settings for the WAN Port:
  - **DNS Address Assignment:** By default, Static is defined as the DNS Address Assignment.
  - **DNS Address:** Enter the DNS Address here. The DNS Address can be a maximum of 39 characters.

Dynamic DNS (DynDns.org)

Click **Dynamic DNS (DynDns.org)** to expand.



- **Dynamic DNS (DynDNS.org):** This parameter is applicable only when you are going to configure the SIP Extensions.

  When the WAN port is assigned dynamic IP Address using DHCP, SIP-enabled devices registered with the system as SIP Extensions need to change their configuration whenever a new IP Address is assigned to the WAN port. Dynamic DNS resolves this.

  ANANT UCS supports Dynamic DNS Server client of the Service Provider Dynamic DNS.org.

  If you want to use the DNS Service of DynDNS.org, configure these parameters:

  - **Enable Dynamic DNS:** If you have taken the services of DynDNS.org, you must enable this check box. Default: Disabled.

  - **User ID:** Enter the User ID created by you with DynDNS.org here. A maximum of 40 characters, including all ASCII characters are allowed. Default: Blank.

  - **Password:** Enter the Password created by you for your User ID with DynDNS.org here. The password may be not more than 24 characters long. Default: Blank.

  - **Host Name:** Enter the Host Name created by you with DynDNS.org here. A maximum of 40 characters. All ASCII characters except < > and " (double quote) are allowed.Default: Blank.

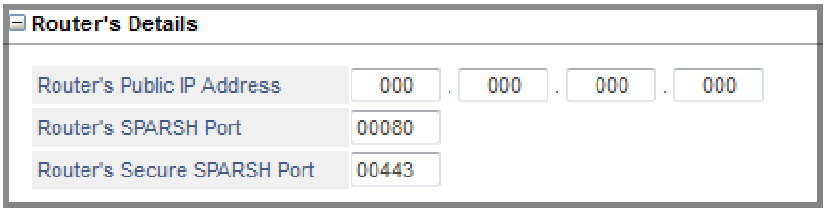
  - **Retry Trials:** This count defines the number of attempts that the system should make to send the IP Address Update Request to the Dynamic DNS Server. The Retry Count may be set from 1 to 9. By default the count is set to 1. Default: 5.

  - **Update IP Address Now?:** Click the **Click to Update IP Address** button**,** if you want to update the IP Address in the DDNS server at any point of time. Default: Disabled.

    This option is useful if you have not enabled the option Update IP Address during each Power ON. You can update the IP Address in the DDNS server, whenever required.

Router's Details

Click **Router's Public IP Address** to expand.



- **Router's Public IP Address:** This parameter is of relevance if the WAN port of the system is located behind a NAT Router and configuration requests are to be forwarded to the public internet.

  Router's Public IP Address is also used for web access and auto configuration of the phones. It specifies the fixed IP Address of your NAT router required for NAT Traversal in SIP messages.

- **Router's SPARSH Port:** Enter the Router's Port mapped with the SPARSH Port of the system. This allows auto configuration of the external VARTA clients (VARTA ADR100/ VARTA AMP100) using the Auto Sign-In Email. If you want the VARTA clients to auto configure with the system, you must enter the Router's SPARSH port value as the Server Port along with the Server Address in the VARTA ADR100/ VARTA AMP100.

  Valid range: 80 or any value ranging from 1025 to 65535. Default: 80.

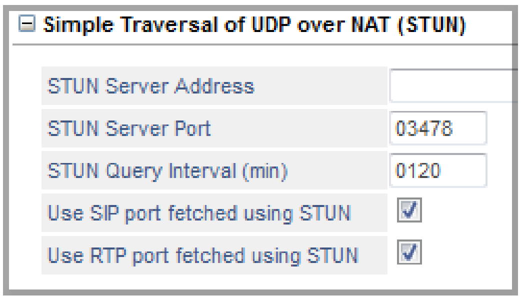
- **Router's Secure SPARSH Port:** Enter the Router's Port mapped with the SPARSH Secure Port of the system. This allows auto configuration of the external VARTA clients (VARTA ADR100/ VARTA AMP100) using the Auto Sign-In Email. If you want the VARTA clients to auto configure with the system using a secure protocol, you must enter the Router's Secure SPARSH port value as the Server Port along with the Server Address in the VARTA ADR100/ VARTA AMP100.

  Valid range: 443 or any value ranging from 1025 to 65535. Default: 443.

- *Make sure you select the Use Router/STUN's IP Address option as the Registrar Server Address, when you configure the SIP Extensions.*
- *You can also use STUN as an alternative to the Router's Public IP Address as NAT Traversal mechanism. Ask your Network Administrator about the NAT Traversal mechanism that suits best for your voice network and configure this parameter.*

Simple Traversal of UDP through NATs (STUN)

Click **Simple Traversal of UDP through NATs (STUN)** to expand.



- **Simple Traversal of UDP through NAT (STUN):** This parameter is to be configured only if the WAN port is located behind a NAT Router and SIP Messages need to be forwarded to the public internet.
  Simple Traversal of UDP through NAT (STUN) specifies the mechanism required for NAT traversal in SIP messages. The STUN Server facilitates traversing through most NATs, except symmetric NATs. If your router has symmetric NAT, do not configure this parameter. If your router as asymmetric NAT, configure the following STUN parameters:

  - **STUN Server Address:** Enter the STUN Server Address, a maximum of 40 characters. Default: Blank.

  - **STUN Server Port:** Enter the Listening Port of the STUN Server. The valid range for this field is from 1025-65535. The default STUN Port is 03478.

  - **STUN Query Interval (min)**: This is the time interval between each STUN query made for the Public IP Address of the NAT Router. The range of this interval is from 0001 to 9999 minutes. Default: 120 minutes.

  - **Use SIP Port fetched using STUN:** By default, this check box is selected (enabled), to allow SIP Port Number to be fetched using STUN in the SIP message. Clear this check box, if you are using Port-Forwarding in the Router for SIP messages.

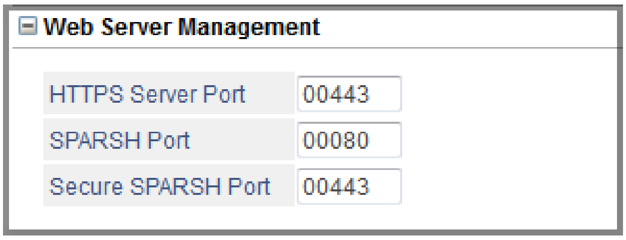
  - **Use RTP Port fetched using STUN:** By default, this check box is selected (enabled), to allow RTP Port Number to be fetched using STUN in the SIP message. Clear this check box, if you are using Port-Forwarding in the Router for SIP messages.

  - *You also need to select 'Use IP Address fetched using STUN" option as the Source Port IP Address in SIP Trunk Parameters and select the same as the Registrar Server Address, when you configure SIP Extensions.*

  - *Since STUN does not work with symmetric NAT, as an alternative to STUN you can use the Router's Public IP Address as NAT Traversal mechanism. Ask your Network Administrator about the NAT Traversal mechanism that suits best for your voice network and program this parameter.*

Web Server Management

Click **Web Server Management** to expand.



- **HTTPS Server Port**: Enter the HTTPS Port number. Valid range: 443 or any value ranging from 1025 to 60000. Default: 443.

- **SPARSH Port:** Enter the SPARSH port number. The system will listen for the configuration request of the Extended IP Phones/ VARTA clients/ Standard SIP Phones on this port. If you want any Extended IP Phones/ VARTA clients/ Standard SIP IP Phones to auto configure with the system, you must configure the SPARSH port value as the *Server Port* along with the Server Address in the Extended IP Phones/ VARTA clients/ Standard SIP IP Phones.

  Valid range: 80 or any value ranging from 1025 to 60000. Default: 80.

- **Secure SPARSH Port:** Enter the Secure SPARSH port number if you want to auto configure the VARTA AMP100/VARTA ADR100/ VARTA WIN200 with the server using a secure protocol. The system will listen for the configuration request from the VARTA clients on this port. You must also configure the Secure SPARSH port value as the *Server Port* along with the Server Address in the VARTA clients.

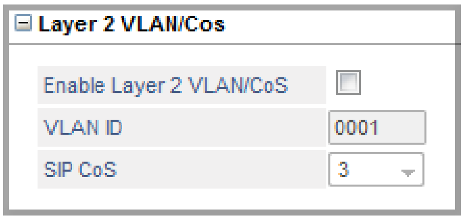
  Valid range: 443 or any value ranging from 1025 to 60000. Default: 443.

When the system is set to default, the port values of HTTPS Port, SPARSH Port, Secure SPARSH Port are not set to default.

- *We recommend you to connect ANANT UCS behind a router/firewall to avoid attacks such as ping flood, DoS etc.*
- *If ANANT UCS is connected on the public internet, the system is prone to attacks, hence to ensure system security we recommend you to change the value of the default ports.*

Layer 2 VLAN/Cos

Click **Layer 2 VLAN/Cos** to expand.



- **Layer 2 VLAN/CoS:** This parameter is to be configured if the WAN port is to be connected in VLAN network.

  This parameter enables ANANT UCS to add VLAN header to the packets generated by it. The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic[11].

  VLAN Tag is applied on all packets generated by system (SIP, RTP, DNS, ARP, etc.), whereas CoS bits are applied only for SIP and RTP packets generated by system.

  The corresponding meaning of CoS bits with respect to traffic type is as follows:

COS	Traffic Type
0	Best Effort
1	Background
2	Spare
3	Excellent Effort
4	Controlled Load
5	Video

11. *The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), better handling of data that pass over a network, thereby resulting in greater reliability and quality. Quality of Service (QoS) on Layer2 is referred to as Class of Service (CoS) which is defined by IEEE 802.1P.*

COS	Traffic Type
6	Voice
7	Network Control

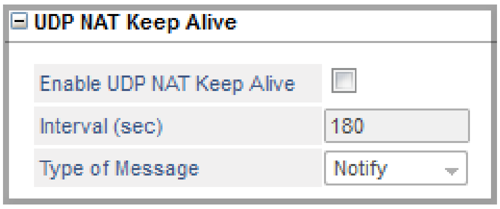
- **Enable Layer 2 VLAN/CoS:** Select this check box, if you want all packets generated by the system (SIP, RTP, DNS, ARP, etc.) to be tagged with VLAN ID as configured. The CoS bits as configured for SIP and RTP packets will be included in the VLAN header. Default: Disabled.

- **VLAN ID:** Consult your network administrator and configure the VLAN ID. The valid range for this is from 0 - 4094. Default: 1.

- **SIP CoS:** Define the CoS (priority) bits in all SIP packets. The range of CoS bits is from 0 to 7. Default: 3.

UDP NAT Keep Alive

Click **UDP NAT Keep Alive** to expand.



- **UDP NAT Keep Alive:** This parameter is to be configured when the WAN port is connected behind a NAT router[12] and SIP messages are transported over UDP. UDP NAT Keep Alive messages must be sent to refresh the UDP binding in the NAT router.

  - **Enable UDP NAT Keep Alive:** Enable this check box to send UDP NAT Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - **Interval (sec):** Select Time period after which the WAN Port should send UDP NAT Keep Alive messages. This time period should be less than the UDP Binding Timer of the router. The valid range is 001-999 seconds. Default: 180 seconds.

  - **Type of Message:** Select the type of message type to be sent when UDP NAT Keep Alive is enabled. Select either REGISTER or NOTIFY. Default: NOTIFY.

TCP NAT Keep Alive

Click **TCP NAT Keep Alive** to expand.

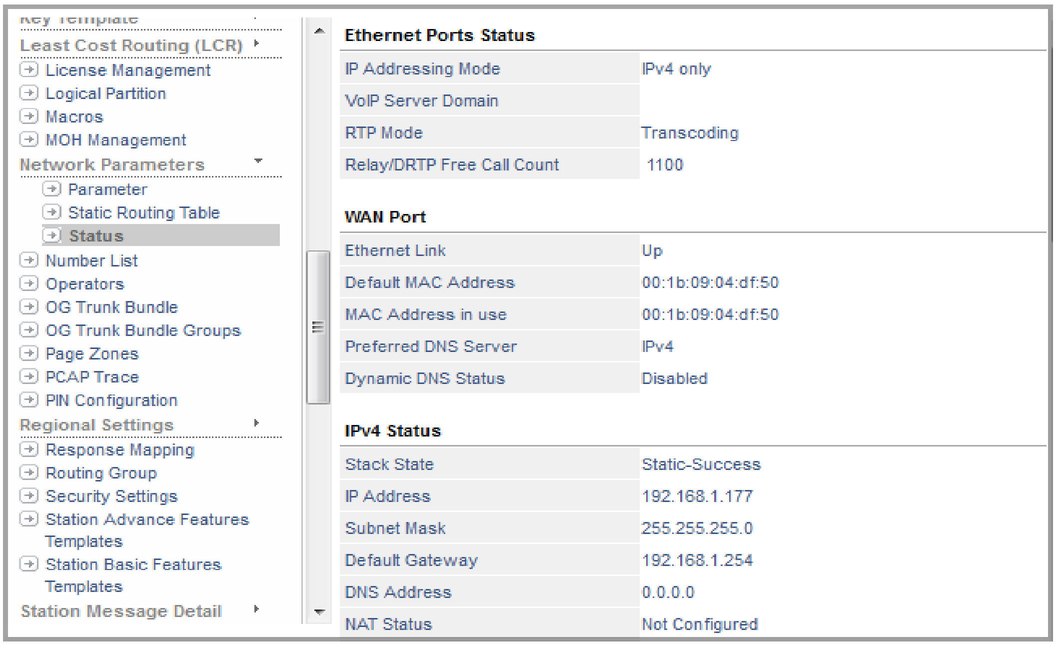
12. *Network Address Traversal (NAT) allows multiple hosts in the network to share the single public routable IP address. Means all the hosts in the private network shall be identified by single public IP address in the global IP cloud.*

- **TCP NAT Keep Alive:** This parameter is to configured when the WAN Port is connected behind a NAT router and SIP messages are transported over TCP. TCP NAT Keep Alive messages must be sent to refresh the TCP binding in the NAT router.
    - **Enable TCP NAT Keep Alive:** Enable this check box to send TCP NAT Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.
    - **Interval:** Select Time period after which the WAN Port should send TCP NAT Keep Alive messages. This time period should be less than the Binding Timer of the router. The valid range is 0001-9999 seconds. Default: 120 seconds.

Viewing Network Parameters

You can view the status of WAN Port in Jeeves. To do this,

- Login as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Status**.



It displays the statuses of the various IPv6 and IPv4 parameters for the LAN and WAN port.

When the system receives the IP Address from the DHCP Server, the system performs DAD (Duplicate Address Detection). If DAD fails due to conflict in IP Address, the respective network parameters (WAN or LAN) need to be re-initialized.

To re-initialize the **WAN** parameters, click the **IPv4 Network Reinitialization** or **IPv6 Network Reinitialization** button under WAN. Similarly, to re-initialize the **LAN** parameters, click the **IPv4 Network Reinitialization** or **IPv6 Network Reinitialization** button.

*You can also view the Network Status from the **Status** link. To view, click the Network link under Status.*

Configuring Redundancy Parameters

In this topic we will explain how to configure the Redundancy parameters. To know about the Redundancy feature and how it works, refer to *"Redundancy"*.

How to configure

For this feature to work,

- You must have two servers with same specifications.
- Activate the Redundancy Users License in only one server (preferably the one with other licensed features). To know about Redundancy Users License and how to activate it, refer to *"Licenses Supported in ANANT UCS"* and *"License Management"* respectively.
- Virtual IP Address for the LAN/WAN port must be configured in the Network Parameters. The IP Address configured in the LAN/WAN in the Network Parameters is known as Virtual IP. To know more refer to *"Configuring Network Parameters"*.

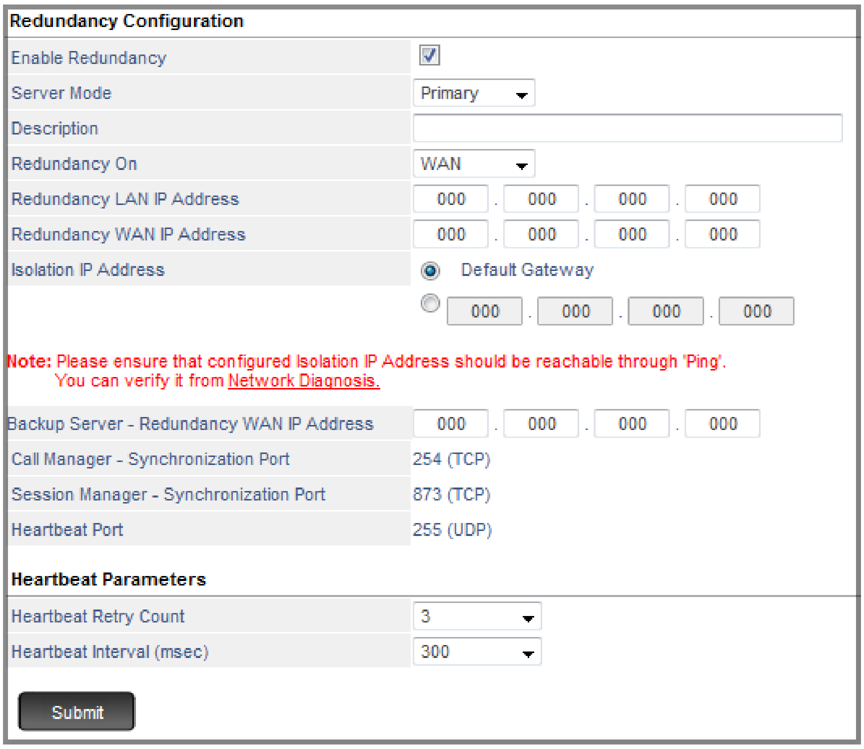
  Virtual IP Address is used for communication. This IP Address is assigned to the server which is in the active state. Make sure that the Virtual IP is configured in all the clients — IP Phones and VARTA UC Clients.

  Similarly, you can connect the Backup Server.
- It is recommended to configure the Primary Server first and then configure the Backup Server. To know more, refer to *"Configuring the Servers"*.

ANANT UCS supports redundancy only when the Connection Type configured for the WAN Port is Static, if WAN is selected as the ***Redundancy On*** *Option.*

Configuring the Servers

- Login as System Engineer.
- Under **Configuration,** click **Redundancy Configuration**.



Redundancy Configuration

- **Enable Redundancy:** By default, this check box is disabled. To use Redundancy feature, select the check box.
- **Server Mode:** To configure the first system as Primary Server, select **Primary.** This is the main server that responds to the client requests. Default: Primary.

If you are configuring this parameter using the Backup Server IP Address,

- *Select the **Server Mode** as **Backup**. This is the secondary server that responds to the client requests when the Primary Server is unavailable. It stays in sync with the Primary Server such that when the Primary Server fails, the Backup Server has latest configuration and call information at the time of redundancy.*

*If you have configured the same **Server Mode** in both the systems, then Redundancy will not be supported.*

- **Description:** Enter the location of the server. The description may consist of 24 characters (maximum). For example, enter R&D.
- **Redundancy On:** Select the desired network — **LAN** or **WAN**, on which you want the redundancy to be applied. Default: **LAN**.
- **Redundancy LAN IP Address:** Enter the IP Address assigned to the LAN Port of the system.

- **Redundancy WAN IP Address:** Enter the IP Address assigned to the WAN Port of the system.

*Make sure that the **Redundancy LAN IP Address** and **LAN Port IP Address** configured in "LAN Port (IPv4 settings)" in Network Parameters are in the same subnet if you wish to use Redundancy. Similarly, this is also valid for the WAN IP Address.*

- **Isolation IP Address:** This is the IP Address that is used by the Standby System to check whether it isolated or not before applying redundancy.

  When you select radio button of the Default Gateway, then the system uses the Gateway Address configured in WAN Port of Network Parameters.

  To assign a different IP address as the Isolation IP address, select the radio button and manually enter the desired IP address.

- *Make sure that the IP address to be configured as Isolation IP address is reachable through 'Ping'. You can check the status of this IP using "Network Diagnosis".*
- *The Isolation IP address configured should be same for both Primary and Backup Server.*

- **Backup Server - Redundancy LAN IP Address:** This is the IP address of the LAN port interface connected to the Backup Server on which redundancy will be applied. This parameter is displayed when you select **Server Mode** as Primary. Similarly, this is also valid for the WAN IP Address.

Primary Server - Redundancy LAN IP Address: *This is the IP address of the LAN port interface connected to the Primary Server on which redundancy will be applied. This parameter is displayed when you select **Server Mode** as Backup. Similarly, this is also valid for the WAN IP Address.*

- **Call Manager - Synchronization Port:** This port is responsible for synchronization between the Primary Server and Backup Server. The Active server sends configuration to the Standby server through this port. The value of this port is 254 and it is fixed.

- **Session Manager - Synchronization Port:** This port is responsible for synchronization between the Primary Server and Backup Server. The Active server sends configuration to the Standby server through this port. The value of this port is 873 and is fixed.

- **Heartbeat Port:** The Standby server periodically checks the availability of the Active server through this port. The value of this port is 255 and is fixed.

Heartbeat Parameters

- **Heartbeat Retry Count:** The Standby server checks the availability of the Active server using Heartbeat requests. The number configured as the **Heartbeat Retry Count,** defines the number of attempts that need to be made by the Standby server to check the availability of the Active server. When no response is received from the Active server and the count expires, the Standby server becomes Active. The Heartbeat Retry Count may be set from 3 to 9. Default: 3.

- **Heartbeat Interval (msec):** This is the time interval after which the Standby server retries to fetch Heartbeat response from the Active server. Valid Range is 100 to 1000 millisecond. Default: 300 millisecond.

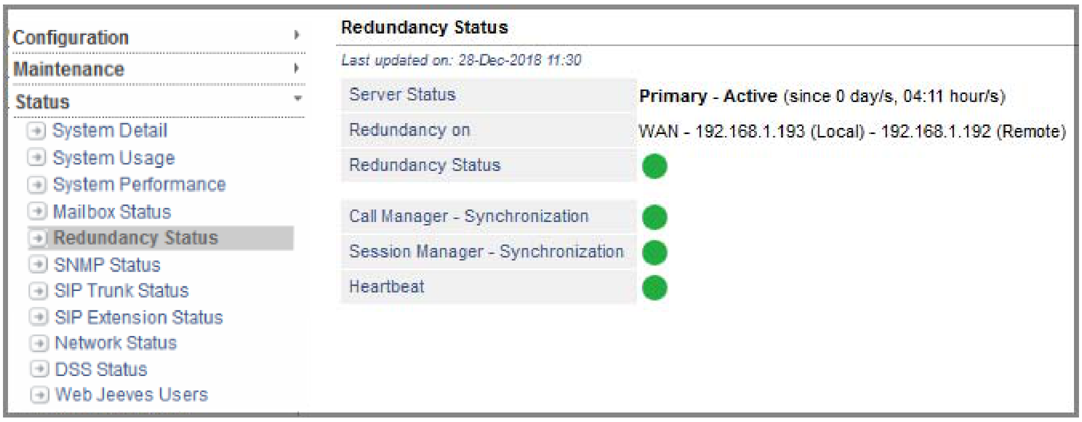
- *Make sure that **Backup Server-Redundancy LAN IP address** is same as **Redundancy LAN IP address** of Backup Server.*
- *Similarly, if you are configuring the parameters in the Backup Server, make sure the **Primary Server-Redundancy LAN IP address** is same as **Redundancy LAN IP address** of Primary Server.*
- *When you submit the page after configuring, the system will restart to activate redundancy.*

- Now, connect the LAN port of the Primary Server to LAN port of the Backup Server.
- Connect the WAN port to a switch/router for communication with the client devices.

Viewing Redundancy Status

To view the Redundancy status,

- Login as System Engineer.
- Under **Status**, click **Redundancy Status**.



- **Server Status:** Displays current systems mode, status and uptime.
- **Redundancy on:** This displays the port through which redundancy occurs and the WAN/LAN IP address of the Active system and Standby system.
- **Redundancy Status:** This displays:
  - Green dot when redundancy is aptly configured and Backup Server is in synchronization.
  - Red dot when the two systems are disconnected.
  - Orange dot when the system is in Standby state.
- **Call Manager - Synchronization, Session Manager - Synchronization, Heartbeat:** When there is any problem during the synchronization, the respective status will be displayed.

The Redundancy status is also displayed in the top banner,

- It displays the current **Server Mode** — Primary or Backup and the **Status** — Active or Standby depending upon the availability of the server.

Configuring System Pre-requisites

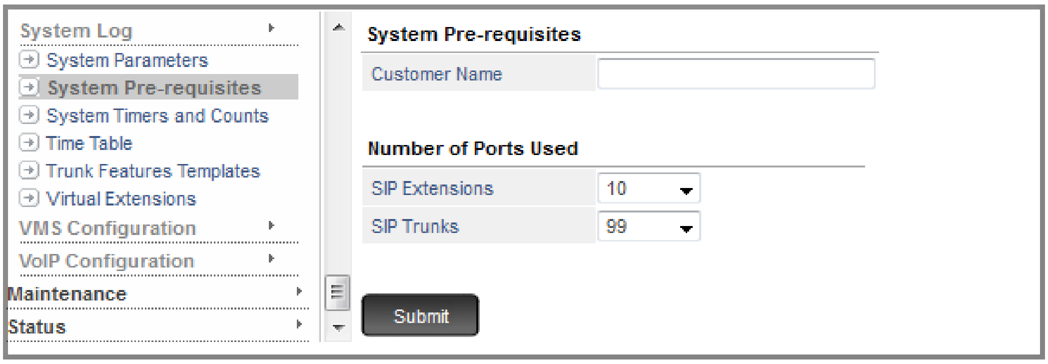
This displays the System resources (number of trunk and extensions supported) of ANANT UCS. It is quite common for users to utilize the system resources below its capacity, especially when they begin using a new system.

To make the task of configuring easier for such users, ANANT UCS allows you to specify the number of trunks and extensions you want to configure. Accordingly all the relevant pages of Jeeves will show only as many trunk and extension that you have specified, instead of showing all the trunks and extensions supported by ANANT.

To be able to do this, you must configure the **System Pre-requisites**.

Defining System Pre-requisites

- Login as System Engineer.
- Under **Configuration**, click **System Pre-requisites.**



- Configure the following parameters:
  - **Customer Name:** You can assign the name of the enterprise/organization that is using ANANT UCS as the Customer Name. The Customer Name may contain up to 80 characters. You may enter the address of organization/enterprise along with the name.

    The Customer Name you assign will appear on the various System Reports generated and printed by the ANANT UCS.
  - **Number of Ports Used:** Define the number of ports to be used for each Port Type — SIP Extensions and SIP Trunks in the respective boxes.
  - Click **Submit**.

CHAPTER 5

Configuring Extensions

ANANT UCS supports the following types of extensions:

- **SIP Extensions:** Any SIP-enabled device like Matrix VARTA UC Client, an IP Phone, a Softphone or a Wi-Fi mobile handset can be registered with the ANANT and function as the 'SIP Extension' of the ANANT UCS.

  SIP Extensions users can make and receive calls to any extension user of the ANANT UCS as well as any external numbers over VoIP, depending on the “Logical Partition” configured in the System.

  ANANT UCS supports 5000 SIP Extensions.

- **Virtual Extensions**. For more information see “Virtual Extension”.

Templates for Configuring Extensions

To make the task of configuring extensions easy, ANANT UCS offers the following Templates:

- **SIP Hardware Template** - for SIP Extensions (and SIP Trunks) only. See “SIP Hardware Template”.
- **Station Basic Feature Template** - for SIP Extensions and Virtual Extensions. See “Station Basic Feature Template”.
- **Station Advanced Feature Template** - for SIP Extensions and Virtual Extensions. See “Station Advanced Feature Template”.

You can use these templates to configure the extensions which are to be assigned the same set of features at one go, saving your effort and time.

The features in these templates are assigned default values that fulfill the requirements of a very broad user base. The Templates may be customized as per user requirements and applied to the extensions.

Before you start the configuration of the extensions, please read the description of the templates and how to customize the templates according to user requirements.

Station Basic Feature Template

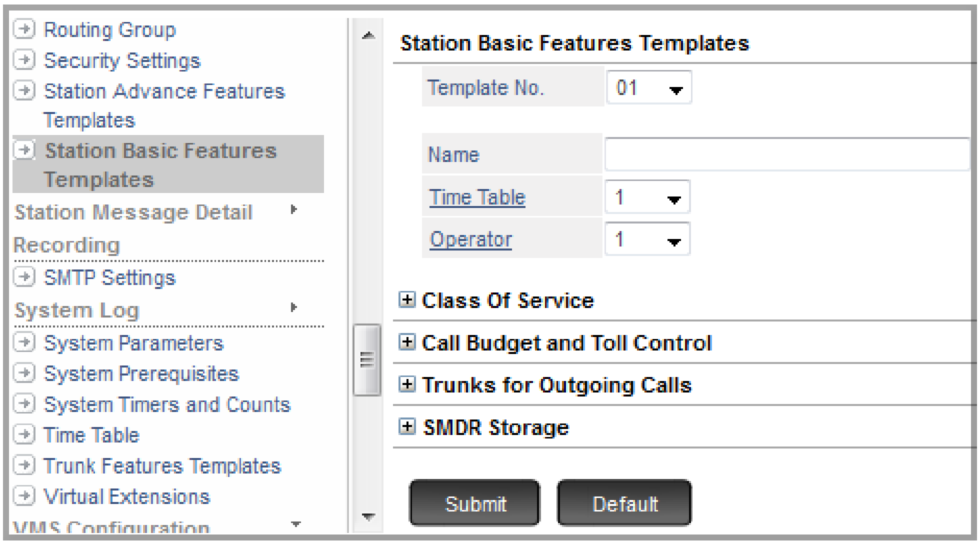
The Station Basic Feature Template is a set of general features that completely define the basic behavior of an extension. Instead of configuring each extension individually, the Station Basic Feature template makes it possible to group together extensions that are to be assigned the same set of features, prepare a Station Basic Feature Template with the common set of features and apply it on these extensions.

ANANT UCS offers 50 such Station Basic Feature Templates. A Station Basic Feature Template is assigned to all the types of extensions, namely SIP Extensions and Virtual Extensions.

These templates have commonly used values, but can be customized as per the requirement and applied on the extensions.

Station Basic Feature Template Parameters

The Station Basic Feature Template contains the following features:



- **Template Number:** Select the **Template Number**, from 1 to 50 that you wish to configure.
- **Name:** Configure the **Name** you wish to assign to the Template you selected.
- **Time Table:** Select the desired **Time Table** number from 1 to 8.

  A Time Table is a schedule for three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week.

  Certain features of ANANT UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Access, among others, require the extension to behave differently in each Time Zone[13].

  So, a Time Table is assigned to extensions defining the Time Zones for the entire week, so that the system can execute the Time Zone-dependent features and facilities according to the Time Table.

13. *For example, incoming calls are to be routed to the security personnel extension, instead of the Operator when the office is closed (non-working hours), or certain features in the Class of Service are to be allowed only during working hours, or access to outgoing long distance calls are to be denied during non-working hours, or the extension must play a different greeting message to the callers during break hours and holidays (non-working hours).*

There are 8 different Time Table templates to select from. By default, Time Table 1 is assigned to all Station Basic Feature Templates.

The System Time Table has Monday to Saturday as working days with working hours set as 9:00 to 18:00 and break hours set as 13:00 to 14:00. Sunday is a non-working day with working and break hours set as 00:00.

You may also customize the default Time Table 1 OR customize and assign a different Time Table to the Station Basic Feature Template. To customize the required Time Table, Click the Time Table link. For detailed instructions, refer to "Time Tables".

- **Operator:** Define the **Operator** for the extensions on which the Station Basic Feature Template is applied.

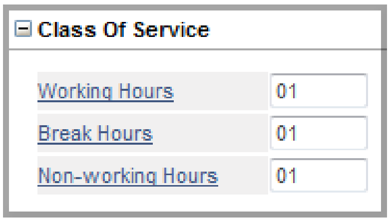
  The system supports multiple Operators. In each Time Zone one of the 20 Operators can be configured.

  Operator can be a single extension or a group of extensions, so that call management is more efficient. For instance certain extensions may be assigned Operator 1, certain others Operator 2 and the rest may be assigned Operator 3.

  By default, Operator 1 is assigned to all Station Basic Feature Templates. If you want to assign different extensions to different Operators, you must configure a separate Station Basic Feature Template with a different Operator for each extension group.

  For detailed instructions, refer to "Configuring 'Operator'".

Class of Service (COS)



Class of Service (COS) defines the set features of the system that the extension is to be allowed access to.

- Click **Class of Service** to expand.

  The set of features assigned to the extension users may differ as per their requirements. Some group of extensions may need the ability to forward calls to a cell phone, and others may have no need to make calls outside the office.

  Similarly, certain features may be required during working hours, but not during break or non-working hours.

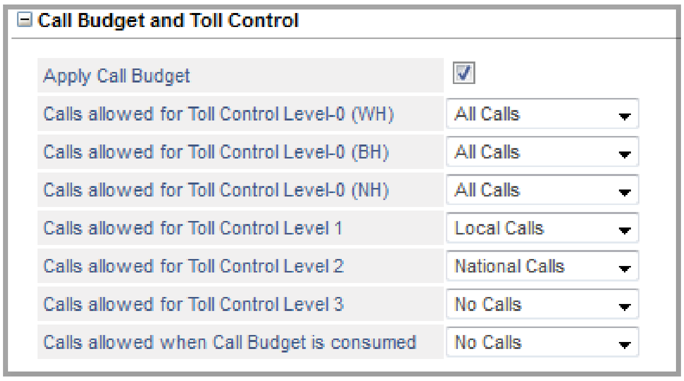
  It is possible to assign a different Class of Service to different extensions according to their feature requirements as well as according to the Time Zones.

  By default Class of Service group 01 is assigned to the Station Basic Feature Templates for all Time Zones —.Working Hours, Break Hours or Non-working Hours. If you want to assign a different COS for each Time Zone, you must customize the COS group first and then assign the number of the COS group in the Station Basic Feature Template.

Refer the topic “Class of Service (CoS)” to know more and for instructions on how to enable or disable a feature in a COS group.

- Click **Submit**.

Call Budget and Toll Control



Call Budget

The Call Budget feature will allot a 'budget' limit for outgoing calls made by extensions on which the Station Basic Feature Template is applied.

- Click **Call Budget and Toll Control** to expand.
- **Apply Call Budget:** Select this check box to enable the Call Budget feature. The Call Budget feature will allot a 'budget' limit for outgoing calls made by extensions on which the Station Basic Feature Template is applied. Refer “Call Budget on Extension” for more details.

Toll Control

The Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to extensions according to the time of the day, during working hours (WH), break hours (BH) and non-working hours (NH). For each Time Zone, you may define the calling permission to be allowed to extensions by selecting the Type of Call Privilege.

- **Calls allowed for Toll Control Level 0 (WH):** This Toll Control Level allows you to define the Call Privilege (calling permission) allowed to an extension during Working Hours. You can select: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls, Limited Calls.
- **Calls allowed for Toll Control Level 0 (BH):** This Toll Control Level allows you to define the Call Privilege (calling permission) allowed to an extension during Break Hours. You can select: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls, Limited Calls.
- **Calls allowed for Toll Control Level 0 (NH):** This Toll Control Level allows you to define the Call Privilege (calling permission) allowed to an extension during Non-Working Hours. You can select: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls, Limited Calls.

The Toll Control levels on this page are based on the allowed and denied number lists of Local, Regional, National, International and Limited Call numbers you configured in “Toll Control”.

Toll Control for Dynamic Lock

Dynamic Lock allows extension users to change the Toll Control Levels (Calling Permissions) of their extensions on their own by dialing a code. ANANT supports Toll Control Levels 0 to 3 for Dynamic Lock.

For each Toll Control Level from 0 to 3, you must assign 'Call Privilege'[14]. For each Call Privilege, you need to configure the corresponding number strings to be allowed and number strings to be denied. See “Dynamic Lock” to know more about this feature.

- **Calls allowed for Toll Control Level 1:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to an extension, regardless of Time Zone. By default Toll Control Level 1 is set to No Calls.
- **Calls allowed for Toll Control Level 2:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to an extension, regardless of Time Zone. By default Toll Control Level 2 is set to No Calls.
- **Calls allowed for Toll Control Level 3:** This Toll Control Level allows you to define the Call Privilege (calling permission) to be allowed to an extension, regardless of Time Zone. By default Toll Control Level 3 is set to No Calls.

The Lock Levels on this page are based on the allowed and denied number lists of Local, Regional, National, International and Limited Call numbers you configured in “Toll Control”.

- **Calls allowed when Call Budget is consumed:** You to define the Call Privilege (calling permission) to be allowed to an extension, after the assigned budget has been consumed. By default Calls allowed when Call Budget is consumed is set to No Calls.
- Click **Submit.**

Trunks for Outgoing Calls

⊟ Trunks for Outgoing Calls

TAC	OG Trunk Bundle Group - WH	OG Trunk Bundle Group - BH	OG Trunk Bundle Group - NH
TAC-1	01	01	01
TAC-2	01	01	01
TAC-3	01	01	01
TAC-4	01	01	01
TAC-5	01	01	01
TAC-6	01	01	01

Outgoing calls (to external numbers) are made by dialing Trunk Access Codes (TAC).

For each TAC, you need to select the Outgoing Trunks. All external calls made by dialing a particular TAC will be routed through the outgoing trunks you selected for that TAC.

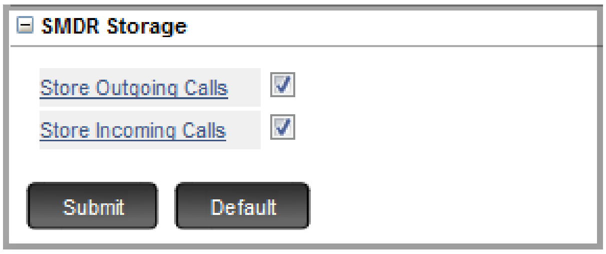
- Click **Trunks for Outgoing Calls** to expand.

14. *The Call Privilege types are: No Calls, Local Calls, Regional Calls, National Calls, International Calls and Limited Calls.*

- **OG-Trunk Bundle Group -WH:** This is the Outgoing Trunk Bundle Group to be allotted to the extension for Working Hours. The extension will be allowed to make outgoing calls through the trunks in this group.
- **OG-Trunk Bundle Group - BH:** This is the Outgoing Trunk Bundle Group to be allotted to the extension for Break Hours.
- **OG-Trunk Bundle Group - NH:** This is the Outgoing Trunk Bundle Group to be allotted to the extension for Non-Working Hours.

  Refer the topic “OG Trunk Bundle Group” for more details.
- Click **Submit**.

SMDR Storage

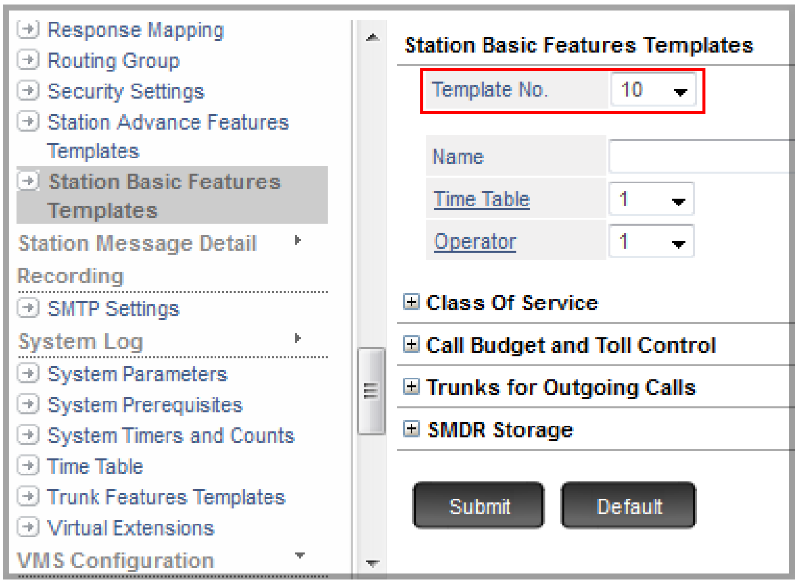


The Station Message Detail Recording (SMDR) feature of ANANT enables you to record the details of Incoming (IC) and Outgoing (OG) calls made from/to all its extensions. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See “Station Message Detail Recording (SMDR)” to know more.

- Click **SMDR Storage** to expand.
- **Store Outgoing Calls:** You can enable or disable the storage of call details - Station Message Detail Records - of Outgoing Calls made from the extensions on which the Station Basic Feature Template is applied. Select the check box to enable and clear it to disable. Refer the topic “Station Message Detail Recording-Storage” for more details.
- **Store Incoming Calls:** You can enable or disable the storage of call details - Station Message Detail Records - of Incoming Calls landing on the extensions on which the Station Basic Feature Template is applied. Select the check box to enable and clear it to disable. Refer the topic “Station Message Detail Recording-Storage” for more details.
- Click **Submit**.

Customizing Station Basic Feature Template using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Station Basic Feature Templates**.
- Select a Template number you wish to customize, for example Template 10.

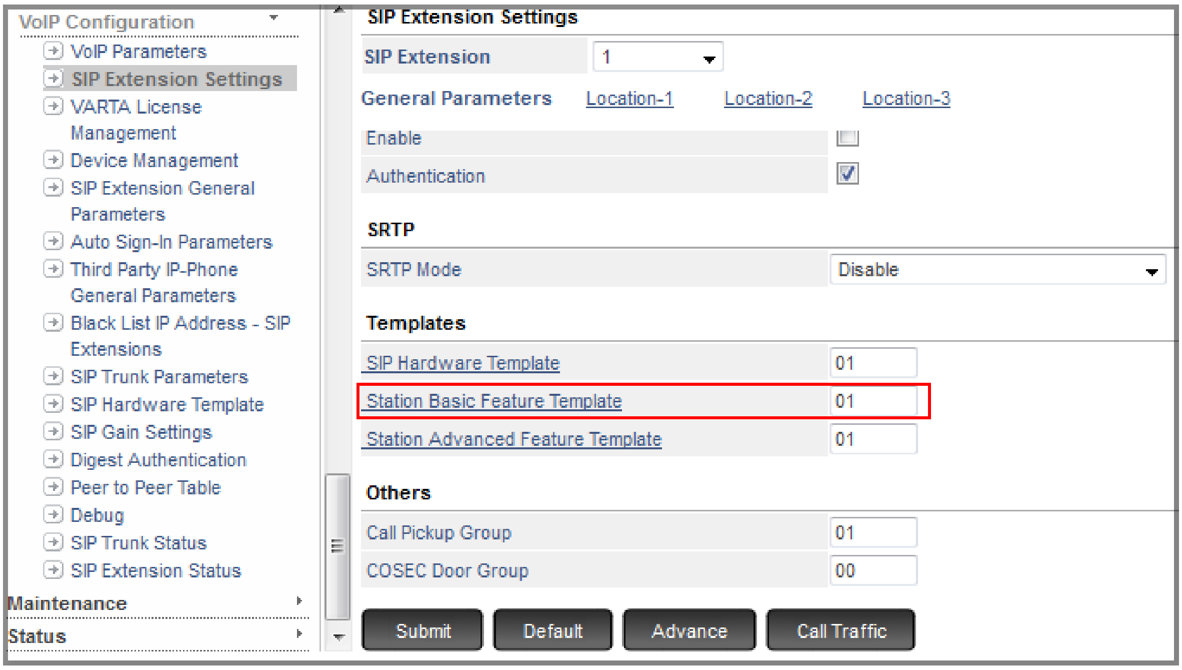


- Change the values of the Station Basic Feature Template parameters as desired.
- Click **Submit** to save your changes.
- Now, apply this Station Basic Feature Template 10 on the SIP / Virtual Extensions.

To apply the customized Station Basic Feature Template on SIP Extensions,

- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.

- Go to the SIP Extensions, for example SIP Extension 1, to which this Station Basic Feature Template is to be assigned, and enter the Station Basic Feature Template number.



- Click **Submit.**

- Repeat the same steps to customize another Station Basic Feature Template and apply it on the desired extensions.

Station Advanced Feature Template

The Station Advanced Feature Template is a set of advanced extension features, to be applied on extensions to support features like CLIP, Call Duration Control, Storage of Internal SMDR, Call Taping, Alarm Notification, Floor Service, Walk-In Class of Service, etc.

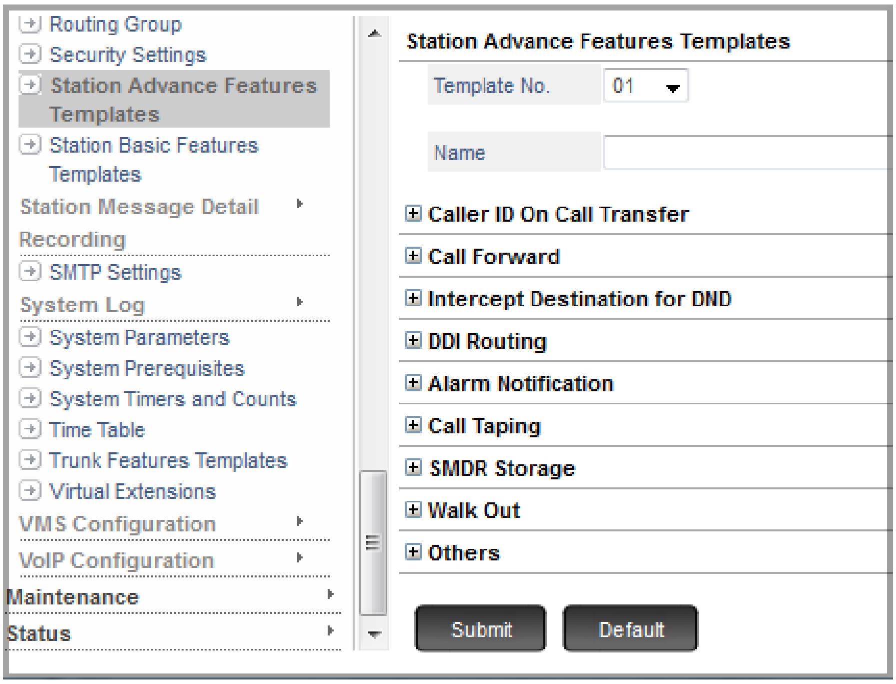
Instead of configuring each extension individually, the Station Advanced Feature makes it possible to group together extensions that are to be assigned the same set of features, prepare a Station Advanced Feature Template with the common set of features and apply it on these extensions.

ANANT UCS offers 50 such Station Advanced Feature Templates. A Station Advanced Feature Template is assigned to SIP extension and Virtual extensions.

These templates have commonly used values, but can be customized as per the requirement and applied on the extensions.

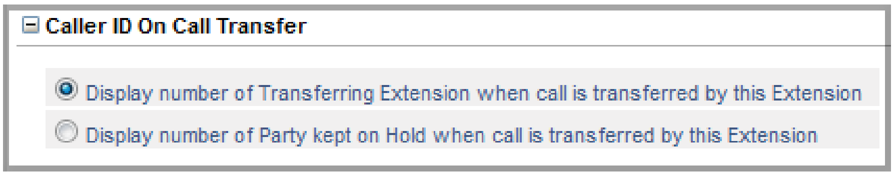
Station Advanced Feature Template Parameters

The Advanced Feature Template comprises the following features on extensions:



- **Template Number:** Select the **Template Number**, from 1 to 50 that you wish to configure.
- **Name:** Configure the **Name** you wish to assign to the Template you selected.

Caller ID On Call Transfer

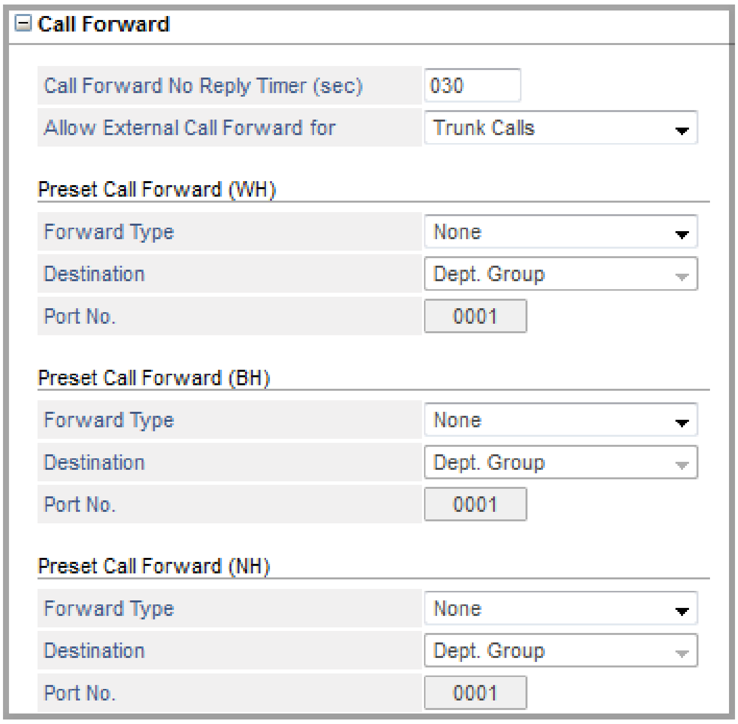


This parameter is related to the CLIP feature. It allows you to choose whether the system should display the CLI of the ‘Held Party’ or the CLI of the ‘Transferring Party’ to the transfer destination extension while the call is being transferred.

Refer the feature description for “Calling Line Identification and Presentation (CLIP)” to know more.

- Click **Caller ID on Call Transfer** to expand.
- Select the radio button of the desired option:
  - Display number of Transferring Extension when call is transferred by this Extension.
  - Display number of Party kept on Hold when call is transferred by this Extension.
- Click **Submit.**

Call Forward



- Click **Call Forward** to expand.
- **Call Forward No Reply Timer (sec):** Set the **Call Forward No Reply Timer (sec)** to the desired value, if required. The range of this timer is 001-255. Default: 030 seconds.

  Call Forward No Reply Timer signifies the duration for which the system will wait for an extension to answer an incoming call, before forwarding the call to the configured destination as Call Forward-No Reply. By default the Timer is set to 30 seconds. This timer is applicable for both Call Forward and Preset Call Forward. Refer the feature description for "Call Forward" and "Preset Call Forward" to know more.
- **Allow External Call Forward for:** Select the type of calls to be forwarded if Calls are forwarded to an external number. You may select from the following options:
  - Internal Calls
  - Trunk Calls
  - Internal + Trunk Calls

    Default: Trunk calls.

  This parameter is relevant for the features "Call Forward", "Mobility Extension" and "Call Forward-When Not Registered".
- **Preset Call Forward (WH):** Select the **Forward Type** for Preset Call Forward (WH). You may select:
  - None
  - When Busy
  - When No Reply
  - When Busy or No Reply

    Default: None

- **Destination:** Select the **Destination** — SIP Extension, Vocie Mail or Dept Group — to which the calls are to be forwarded for Preset Call Forward (WH). Default: 21.

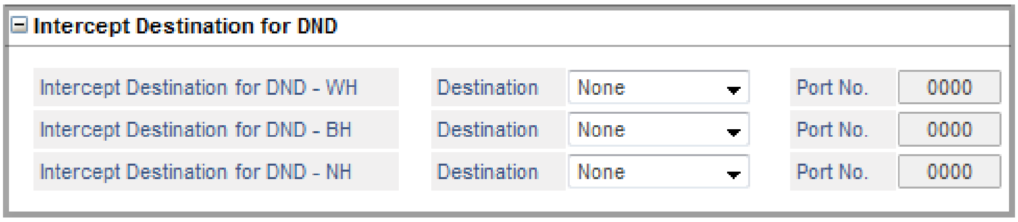
- **Port Number:** Enter the Software Port Number of the SIP Extension/ Departmental Group, if you have selected the **Destination** as SIP Extension/ Dept Group.

  Follow the instructions mentioned above to configure **Preset Call Forward (BH)** and **Preset Call Forward (NH)**.

  See "Preset Call Forward" for more details.

- Click **Submit**.

Intercept Destination for DND



When the DND set extension users want their calls to be attended by someone even if DND is set, they must select an Intercept Destination. Incoming calls landing on extension that has set DND will be routed to the Intercept Destination. This destination can be the users own mailbox or another extension. See "Do Not Disturb (DND)" for more details.

- Select **Intercept Destination for DND** to expand.

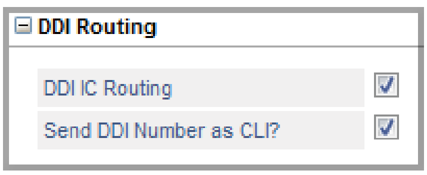
- **Intercept Destination for DND - WH:** Select the **Destination** for Intercept Destination for DND - WH. You may select:
  - None
  - Voice Mail
  - SIP Extension
    Default: None

  If you select the **Destination** as a SIP Extension, enter the desired Port Number (**Port No.**).

  Similarly, you may select **Intercept Destination for DND - BH** and **Intercept Destination for DND - NH**.

- Click **Submit**.

DDI Routing



DDI IC Routing is useful in organizations where the system is connected and each extension user needs to be assigned a separate direct number. When there is an incoming call on a specific DDI number, it is routed directly to the extension user who is assigned the number without the intervention of the operator.

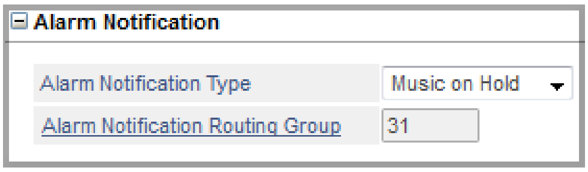
- Select **Intercept Destination for DND** to expand.
- **DDI IC Routing:** By default this check box is selected, that is it is enabled. The system will land incoming calls with the same DDI number on the extension user who is assigned this number. Default: Enabled.

  Clear the check box, if you want to disable.
- **Send DDI Number as CLI?** By default this check box is selected, that is it is enabled. The system will send the DDI number as CLI when outgoing calls are made by the extension user who is assigned this number. Default: Enabled.

  Clear the check box, if you want to disable.

  Refer the topics “Direct Dialing-In (DDI)” to know more.
- Click **Submit**.

Alarm Notification



Using Alarm Notification you can notifying the extension user about the Alarm call.

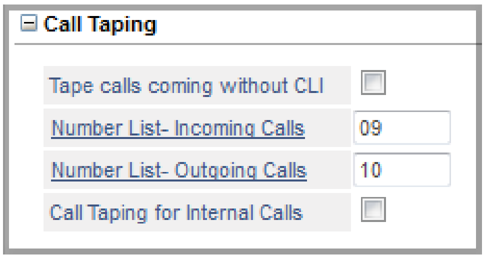
- Click **Alarm Notification** to expand.
- **Alarm Notification Type:** ANANT supports the following types of Alarm Notifications:
  - **Music-on-Hold:** Extension users will be played music-on-hold when they answer the alarm call.
  - **Voice Mail:** The Extension users will be played the message recorded in the VMS, when they answer the alarm call.
  - **Routing Group:** Extension users will be connected to the extensions configured in the 'Alarm Notification Group'. For this you must configure a Routing Group.
- **Alarm Notification Routing Group:** Configure this only if you have selected *Routing Group* as the *Alarm Notification Type*. In **Alarm Notification Routing Group**, enter the number of the Routing Group you have configured for Alarm Calls.

  By default Routing Group 31 is assigned as the Alarm Notification Routing Group. If the same Routing Group is to be assigned to all extensions, click the link *Alarm Notification Routing Group*. Select members (extensions) in this routing group. Save your changes by clicking *Submit*.

  You can configure a different Routing Group repeating these steps. Make sure to enter the number of the Routing Group you configured.

  Refer the feature description for “Alarms” to know more.
- Click **Submit**.

Call Taping



To use the "Call Taping" feature on the extension,

- Click **Call Taping** to expand.
    - **Tape Calls coming without CLI:** Select this check box, if you want incoming calls without CLI to be taped. By default it is disabled.
    - **Number List-Incoming Calls:** Assign a Number List containing numbers of Incoming Calls that must be taped. You must first configure the Number List. By default, Number List 09 is assigned for incoming calls.
    - **Number List-Outgoing Calls:** Assign a Number List containing numbers of Outgoing Calls that must be taped. You must first configure the Number List. By default, Number List 10 is assigned also for outgoing calls.

      If Number list 10 is already used for another application, prepare a different number list and assign it to the template.
    - **Call Taping for Internal Calls:** Select this check box if you want to allow Call Taping of internal calls made and received by the extension.
- Click **Submit**.

SMDR Storage

The Station Message Detail Recording (SMDR) feature of ANANT enables you to record the details of Internal calls made from/to all its extensions. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See "Station Message Detail Recording (SMDR)" to know more.

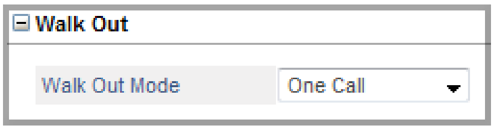
- Click **SMDR Storage** to expand options.
- **Internal Calls Storage:** Select the type of internal calls to be stored. You can select from the following options:
    - **Made/received by this extension**: The system will store all calls made to and received from this extension.
    - **Made by this extension**: The system will store outgoing calls made from this extension.

- **Received by this extension**: The system will store only incoming calls from other extensions.

- **Never**: The system will not store internal calls.

- Click **Submit**.

Walk Out



This parameter is related to the feature Walk-In Class of Service. ANANT UCS offers two types of Walk-In:

- One-Call per Walk-In, whereby the user is automatically logged out after a call.

- Walk-In until Logout, whereby the user remains logged on until s/he manually walks out or a second user walks into the same extension.

To know more about this feature, refer "Walk-In Class of Service".

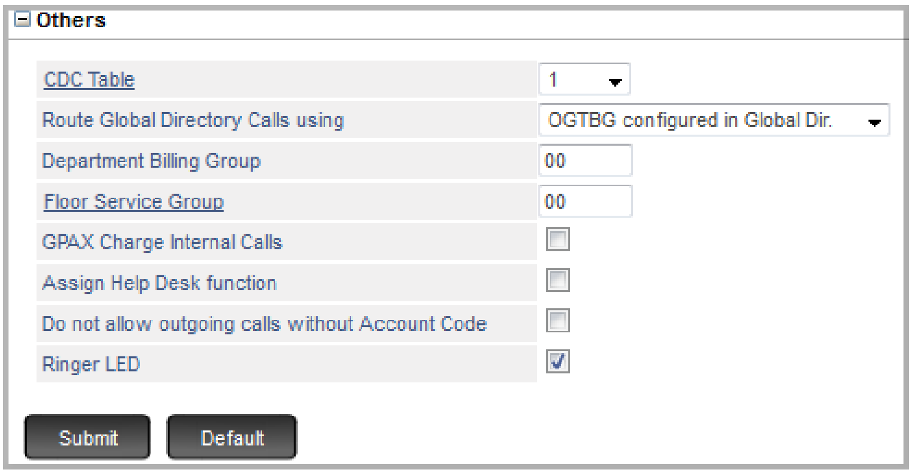
- Click **Walk Out** to expand.

- **Walk-Out Mode:** You must select the Walk-Out mode for the extension.

  - **One Call:** Select this option, if you want to assign One-Call per Walk-In to the extension.
  - **Multiple Calls:** Select this option, if you want to assign Walk-In until Logout to the extension.

- Click **Submit**.

Others



- Click **Others** to expand.

- **CDC Table:** This parameter needs to be configured if you have enabled the "Call Duration Control (CDC)" feature on the extension. The system will check the Call Duration Control (CDC) Table applied to the extension to implement this feature on the extension. So, you must first configure the CDC Table and select the number of the CDC Table you have configured.

You can configure 8 different CDC tables. By default, CDC Table No. 1 is assigned to all extensions. If CDC is to be applied on extensions of the ANANT UCS, simply enter the default CDC Table No. 1.
To do this, click the link CDC Table to open the page. Configure the CDC Table parameters and Submit to save your settings. Now return to the Station Advanced Feature Template and enter the number of the CDC Table you configured in the CDC Table of the template.

Refer the feature description “Call Duration Control (CDC)” to know more and for instructions on creating CDC Tables.

- **Route Global Directory Calls using**: This parameter decides the OGTBG to be used to route the Global Directory number. You may select either **OGTBG configured in the Global Dir.** or any other **OGTBG from 01 to 32**.

- **Department Billing Group:** This parameter enables you to know the total cost of the calls made by a particular group of extensions. This parameter is used as a one of the filters for printing SMDR Reports namely, *Print outgoing calls department group wise*. To be able to use this filter, you must assign the extensions to a Department Bill Group. You can create as many as 99 different Department Bill Groups. Enter the number of the Bill Group you want to assign the extensions here. By default, no Routing Group is assigned to Floor Service in the Template ('00').

- **Floor Service Group:** This parameter is related to the Floor Service feature. Floor Service can be floor-wise or centralized. Floor Service requires you to configure the Routing Groups as landing destinations for extension calls.

  Configure the Floor Service (Routing) Group first and enter this Floor Service (Routing) Group number here. There are 96 different Routing Groups that can be configured as Floor Service Groups. By default, no Routing Group is assigned to Floor Service in the Template ('00').

  To know more about this feature, refer the feature description for “Floor Service”.

  Calls from the extension will land on the Floor Service (Routing) Group you have assigned here.

- **GPAX - Charge Internal Calls:** This parameter is related to the GPAX application. If the extension is configured as a GPAX user, select this check box for billing internal calls made by the extension. When it is enabled, the system will record all calls made from the extension in the Station Message Detail Record-Outgoing buffer. If it is disabled the calls will not be billed and will be recorded in the Station Message Detail Record - Internal buffer as an internal call.

- **Assign Help Desk function:** Select this check box, if you want to define the extension as “Help Desk”. When it is enabled, Auto Call Back will be automatically set whenever this extension is found busy.

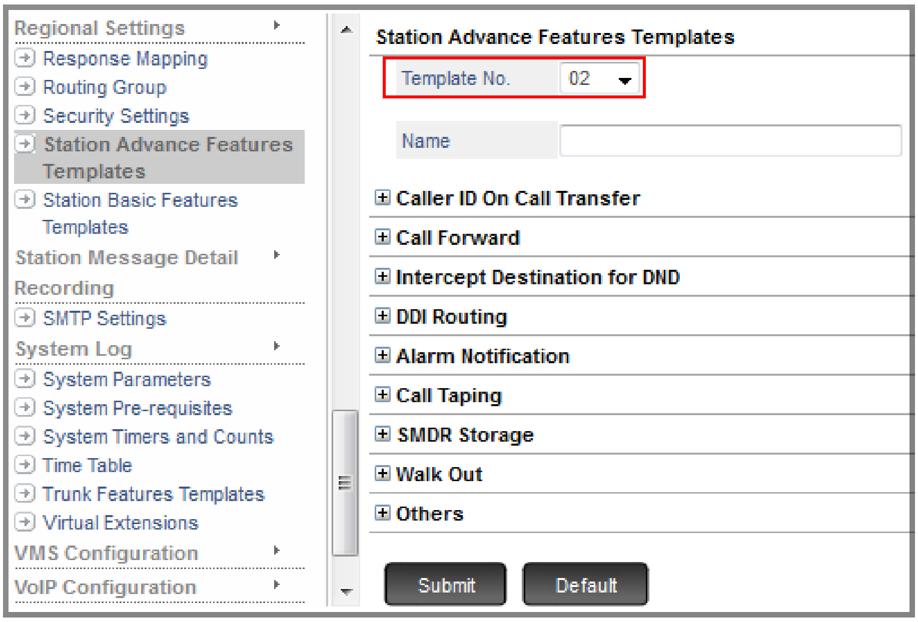
- **Do not allow outgoing calls without Account Code:** Select this check box to apply Forced Account Code on the extensions. When it is enabled, the system will allow the extension user to dial an external number only after entering the Account Code. To know more refer to “Account Codes”.

- **Ringer LED:** This parameter decides whether the Ringer LED should glow or not on the desired extensions for incoming calls and as missed call notification. By default, the LED will continue to blink until the missed call log is read or the call is answered/disconnected. Disable this check box if you do not want the Ringer LED to glow.

- Click **Submit.**

Customizing Station Advanced Feature Template using Jeeves

- Login as System Engineer.
- Under **Configuration**, click **Station Advanced Feature Template**.

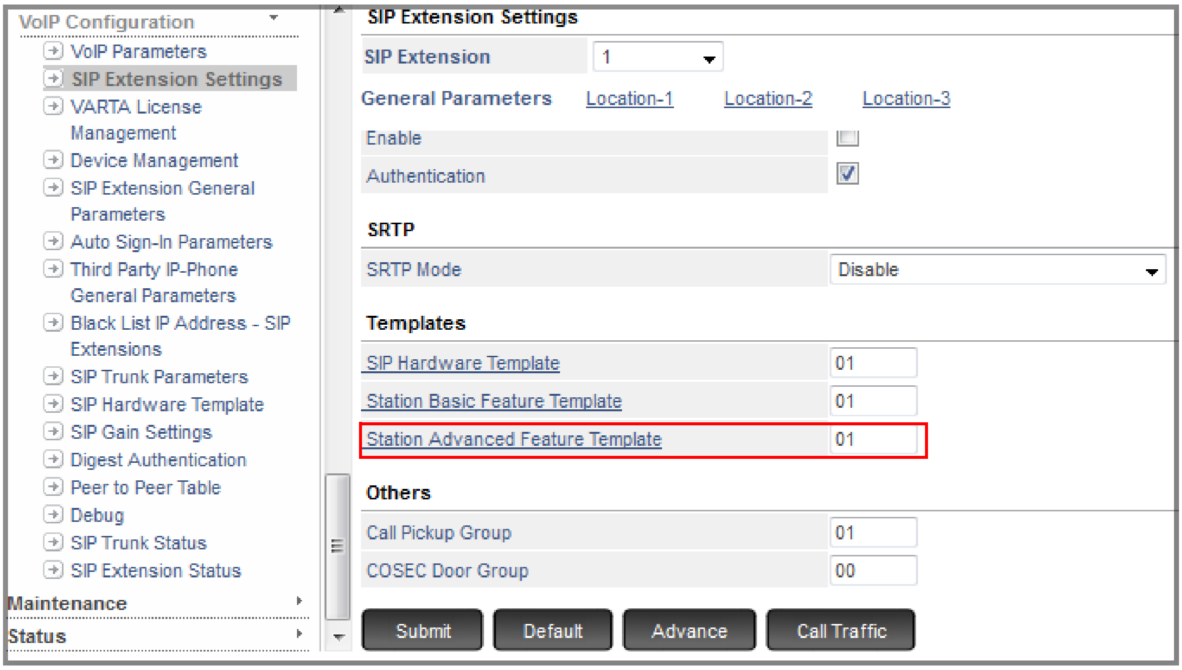


- Select a Template number you wish to customize, for example Template 02.
- Change the values of the Station Advanced Feature Template parameters as desired.
- Click **Submit** to save your changes.
- Now, apply this Template 02 on the SIP and Virtual Extensions.

To apply the customized template on SIP Extensions,

- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.

- Go to the SIP Extensions, for example SIP Extension 1, to which this Template is to be assigned and enter the Template number.



- Click **Submit.**

- Repeat the same steps to customize another template and apply it on the extension ports.

Configuring SIP Extensions

ANANT UCS supports 5000 SIP extensions. You can:

- Connect SPARSH VP248, the Extended IP Phone by Matrix.
- Connect SPARSH VP330, the Touch Screen Extended IP Phones by Matrix.
- Connect SPARSH VP310, the Executive IP Phone by Matrix.
- Connect SPARSH VP510, the Premium IP Phone by Matrix.
- Connect SPARSH VP210, the Entry Level IP Phone by Matrix.
- Connect Extended SPARSH VP710, the Smart Video IP Phone by Matrix.
- Register the Matrix VARTA ADR100 and VARTA AMP100 UC Clients for Mobile with ANANT UCS.
- Register the MATRIX VARTA WIN200 Desktop UC Client with ANANT UCS.
- Connect Matrix Standard SIP Phones.
- Connect any other Standard SIP phone or SIP enabled device, such as an IP Phone, a Soft phone, an Analog Terminal Adapter.

  To know about the Extended IP Phones, Mobile and Desktop UC Clients, see *"Extended IP Phone/VARTA UC Client - Operation"*

  .

ANANT UCS supports interoperability with the Standard IP Phones. For the list of IP Phones on which various features of ANANT UCS have been tested, see "ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.

SIP Extensions can make and receive calls to any extension user of ANANT UCS as well as to external numbers, depending on the *"Logical Partition"* configured in the System.

SIP Extensions (IP Subscribers) is a licensed feature. Decide the number of IP Subscribers you will require and purchase the license accordingly. Refer the topic "Licenses Supported in ANANT UCS" to know more.

You can register a SIP Extension at three different locations as a single SIP Extension for Call Forking.

Configuring SIP Extensions

You need to configure the following parameters for SIP Extensions:

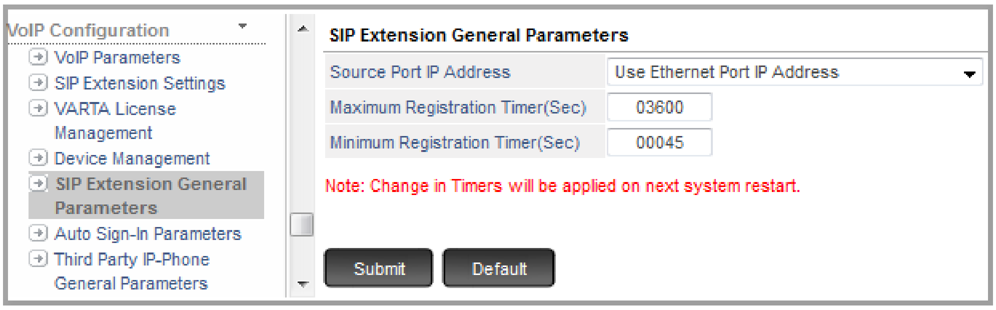
- **SIP Extension General Parameters**, see *"Configuring SIP Extension General Parameters"*.

- **SIP Extension Settings**, you can configure the SIP extensions either one by one or by using bulk configuration. See *"Configuring SIP Extension Settings as per the Extended Phone Type"* to know how to configure extensions one-by-one and *"Configuring SIP Extensions using Bulk Configuration"* to know how to configure extensions using bulk configuration.

- **Voice Mail Settings,** if you want to provide mailbox to the SIP extensions. See *"Extension Voice Mail Settings"*.

Configuring SIP Extension General Parameters

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension General Parameters**.



- Set the following SIP Extension General Parameters, as required.
  - As the **Source Port IP Address,** select the NAT Traversal mechanism for SIP messages from the following options:
    - **Use Ethernet Port IP Address:** Select this option, if your system is not located behind a NAT Router.
    - **Use IP Address fetched using STUN:** Select this option, if your system is located behind a NAT Router. Make sure you configure the **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".
    - **Use Router's Public IP Address:** Select this option, if your system is located behind a NAT Router. Make sure you configure the **Router's Public IP Address** in Network Parameters. For details, see "Configuring Network Parameters".
  - You may set the **Maximum Registration Timer (sec)** as required. This is the Maximum Expiry Timer, which the system will accept in the REGISTER request received. If the value of Maximum Expiry Timer received in the REGISTER request is greater than the value you have set here, the system will send the value you have set in the SIP message. The same timer is used for handling SUBSCRIBE requests. The valid range of this timer is from 10 to 99999 seconds. By default, it is set to 3600 seconds.
  - You may set the **Minimum Registration Timer (sec)**, as required. This is the Minimum Expiry Timer, which the User Agent should send in its REGISTER request. If the expiry value in the REGISTER message is less than this value, the request will be rejected. The valid range of this timer is from 10 to 99999 seconds. By default, it is set to 45 seconds.

The Timers will be applicable only after System Restart.

- Click **Submit**.

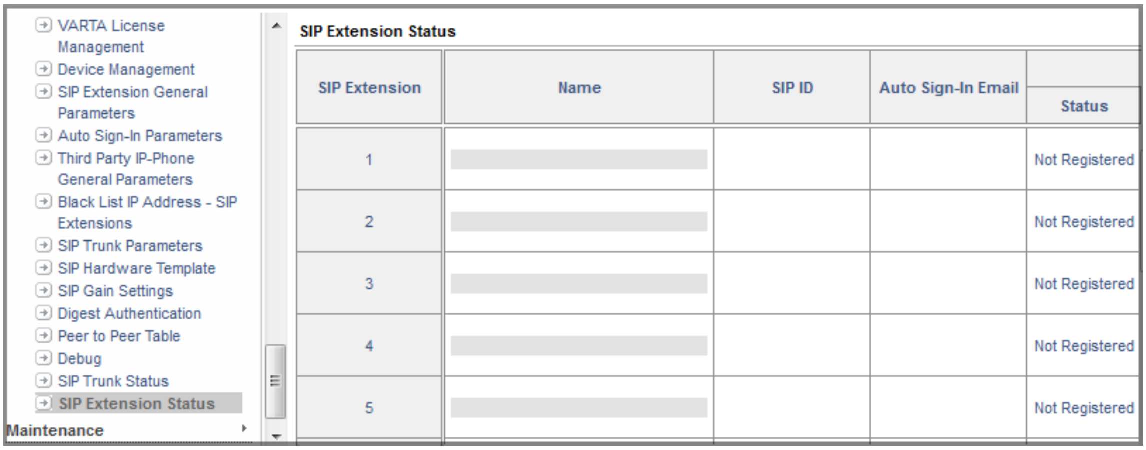
Configuring SIP Extension Settings as per the Extended Phone Type

- If you have registered the Matrix Mobile UC Clients as SIP Extensions, for configuration instructions see *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.
- If you have registered the MATRIX VARTA WIN200 Desktop UC Client as SIP Extensions, for configuration instructions see *"Configuring Matrix VARTA WIN200 UC Client"*.
- If you have connected the Matrix SPARSH VP248 as SIP Extensions, for configuration instructions see *"Configuring Matrix SPARSH VP248"*.
- If you have connected the Matrix SPARSH VP330 as SIP Extensions, for configuration instructions see *"Configuring Matrix SPARSH VP330"*.
- If you have connected the Matrix SPARSH VP310 as SIP Extensions, for configuration instructions see *"Configuring Matrix SPARSH VP310"*.
- If you have connected the Matrix SPARSH VP510 as SIP Extensions, for configuration instructions see *"Configuring Matrix SPARSH VP510"*.
- If you have connected the Matrix SPARSH VP210 as SIP Extensions, for configuration instructions see *"Configuring Matrix SPARSH VP210"*.
- If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710"*.
- If you have connected Standard SIP Phones or SIP enabled devices as SIP Extensions, for configuration instructions see *"Configuring Standard SIP Phones"*.

Viewing SIP Extension Status

You can view the Status of SIP Extension, whenever required. To do this,

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Status**.



- The SIP Extension Status page will open and display the following for each SIP Extension,
    - SIP Extension number
    - Name of the SIP extension
    - SIP ID assigned to the SIP Extension
    - Status of Auto Sign-In Email
    - Registration status; whether the SIP Extension is registered or not.
    - Contact 1
    - Contact 2
    - Contact 3

*You can also view the SIP Extension Status from the **Status** link. To view, click the SIP Extension Status link under Status.*

Configuring Matrix SPARSH VP248

SPARSH VP248[15], the proprietary SIP-based IP Phone for ANANT UCS, supplied by Matrix, is a feature-rich phone, providing voice communication over IP network. To know the list of features supported, refer to *“ANANT UCS Features Supported in Terminals”*.

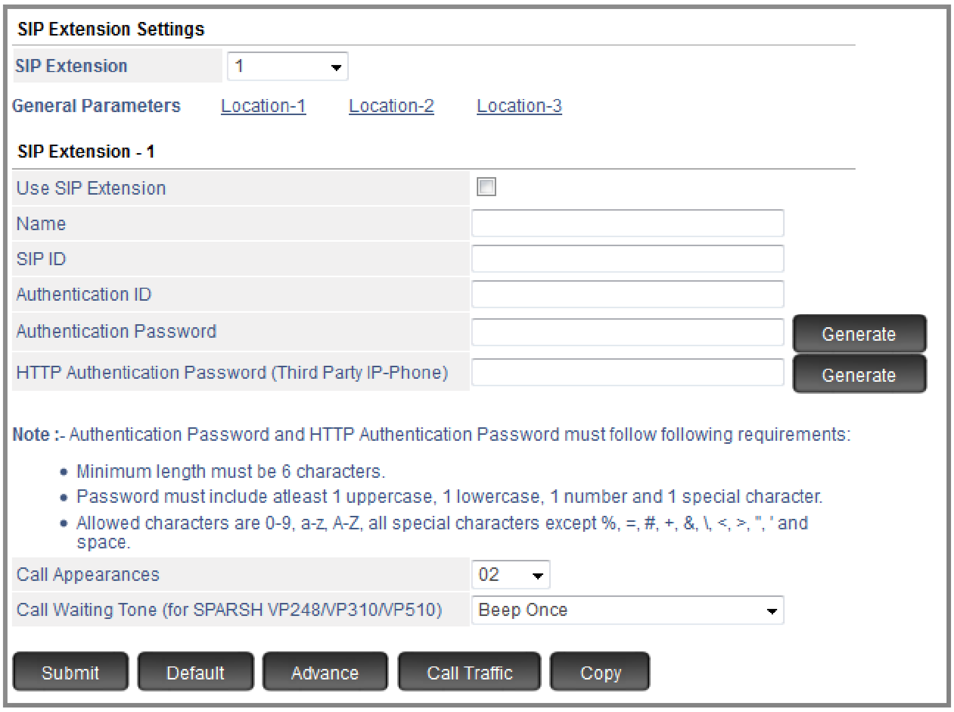
For instructions on how to use SPARSH VP248, refer to the common *EON48_310_SPARSH VP248_310 User Guide.*

To be able to use SPARSH VP248 - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see *“Configuring SIP Extension General Parameters”*.
- SIP Extension Settings, see *“Configuring SIP Extension Settings”*
- Extended IP Phone Settings, see *“Configuring Matrix Extended Phone Settings”*
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See *“Extension Voice Mail Settings”*.

Configuring SIP Extension Settings

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

15. *ANANT UCS supports only IPv4 Addresses for registering SPARSH VP248.*

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In **Name**, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- In **Call Appearances**, define the maximum number[16] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once

  However, when a ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

***Auto Sign-In parameters — Send Configuration Mail** and **Mail Status,** are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - INVITE Request
  - SUBSCRIBE Request

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:

  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.

  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

16. *The calls that are routed through the system will depend on the number of transcoding channels available.*

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and other related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click **SIP Hardware Template**.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic “SIP Hardware Template” to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01. The “Station Basic Feature Template” has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension (“Class of Service (CoS)”, “Toll Control”, “OG Trunk Bundle Group”, etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Template**.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic “Station Basic Feature Template” to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The “Station Advanced Feature Template” has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

- Under **Configuration**, click **Station Advanced Feature Template**.
- Select the Template number, for example 02, and customize this template.
- Click **Submit** and return to the 'SIP Extension Settings' page.
- In the **Station Advanced Feature Template** field, select the number of the template you customized.
- Click **Submit**.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

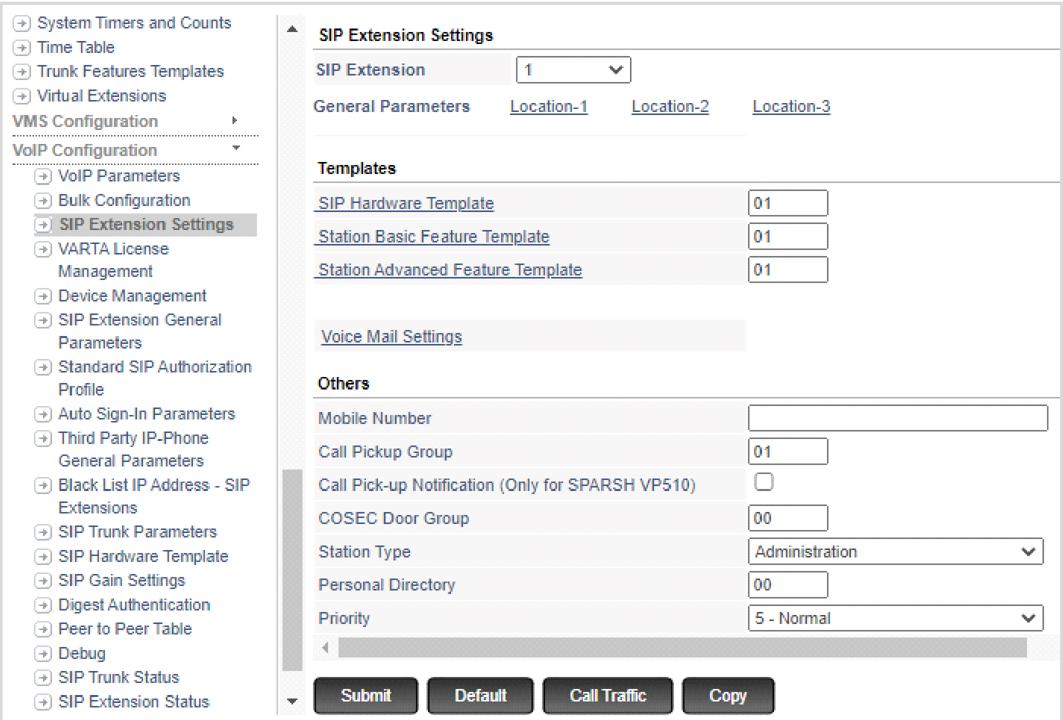
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (CoS)". Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.**

  If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the SIP Extension users and by the System Engineer. Refer the topic “Abbreviated Dialing” for instructions on configuring the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal.

  Each extension of UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended IP Phones/UC Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones/UC Clients —SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330, SPARSH VP210, Extended SPARSH VP710, VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP248 is connected at Location 1, 2 and 3.

If you want to use more than one SPARSH VP248 Extended IP Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

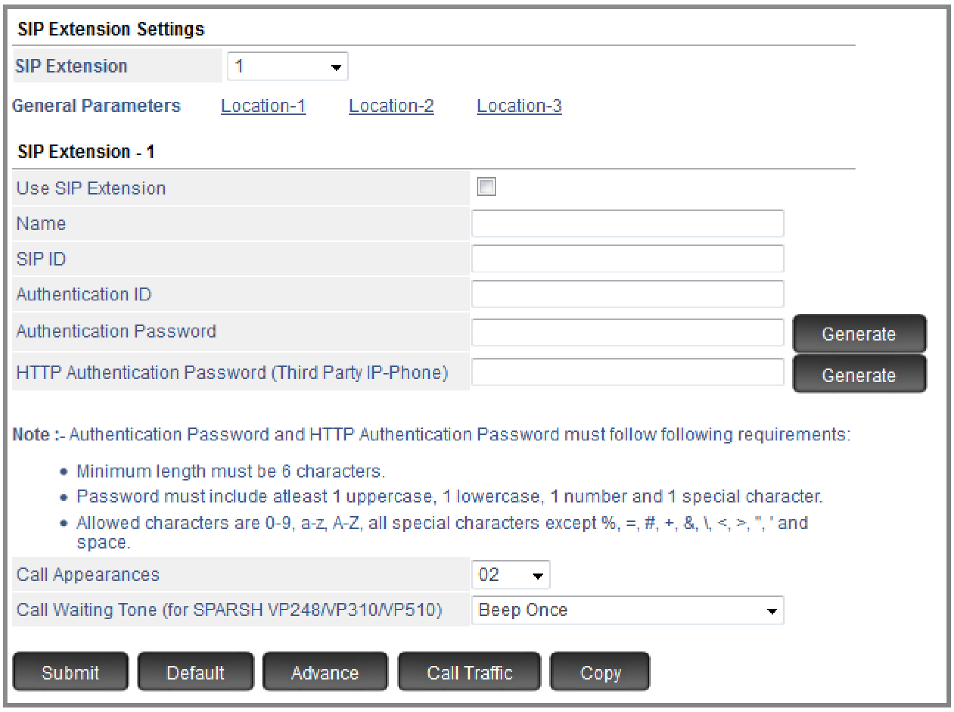
If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710"*.

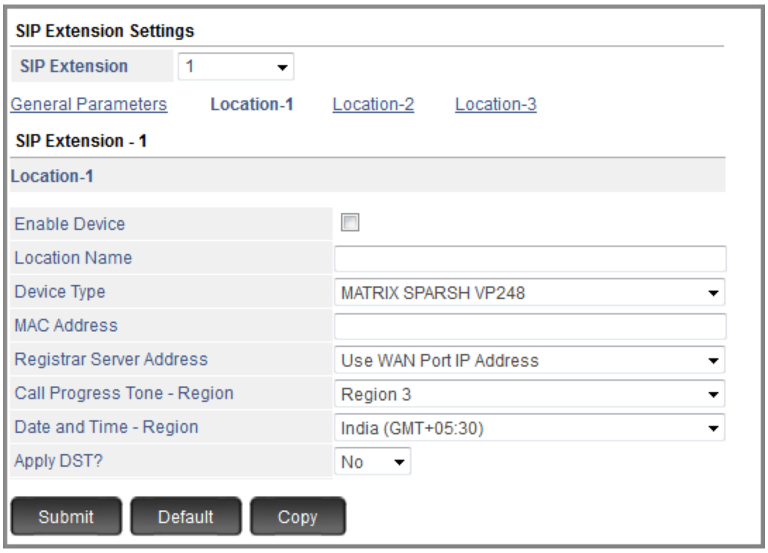
If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP248** as the **Device Type** at this location.

- Enter the **MAC Address**[17] of the SPARSH VP248 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

ANANT UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by configured MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP248 with the SIP Registrar of ANANT UCS, according to your installation scenario:

    - If the SPARSH VP248 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

    - If the SPARSH VP248 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

    - If the SPARSH VP248 is connected in the (Global) Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

      Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see *"Configuring Network Parameters"*.

    - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

      By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP248 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See *"Call Progress Tones"* to know more.

- To display the Date and Time of the country where the SPARSH VP248 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to Yes. Default: No.

  The Daylight Saving Time convention followed in the country/region you selected will be automatically applied. The SPARSH VP248 will change its date and time settings according to the DST convention of the selected country/region.

- Select the **Trunk CLIP Pattern** for the SPARSH VP248. This is the type of Calling Line Presentation on the phone for incoming calls from trunks. You can select any of these options:

    - **Name Only** (only the name of the caller will be displayed).
    - **Number Only** (only the number of the caller will be displayed).
    - **Number + Name** (both the name and the number of the caller will be displayed).

  Default: Number + Name.

---

17. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

- Select the **Display Language** for the SPARSH VP248. Default: English.

  ANANT UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP248. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- *SIP Extension users can change the language by accessing and navigating through the phone menu.*
- *The SA can change the Language by logging into the SA Jeeves.*

- Select a **Ringer Mode** for the phone from the four options:

  - Ring immediately (it rings immediately as a fresh calls lands on the phone).
  - Ring if idle (rings only if the phone is idle).
  - Ring after delay (if the call is still not answered).
  - Silent.

  Default: Ring Immediate.

  If you select *Ring after delay* as Ringer Mode, set the **Ring Delay Timer (sec)**, if required, to the desired value.

  The Ring Delay Timer is the time in seconds the system waits on receiving a call before ringing on the phone. The range of this timer is 0 to 99 seconds. Default: 10 seconds.
- If you want to enable *Ringer Auto Acknowledge* mode, set the **Acknowledge Timer** (sec) to the desired value.

  The Ringer Auto Acknowledge mode determines when to stop the ring on the phone. There are two options for Ringer Auto Acknowledge:
  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the phone after a delay, the Acknowledge Timer must be configured. The range of this timer is 00 to 99 seconds. Default: 00 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- To assign the Ring Destination for the SPARSH VP248, select the desired destination for **Play Ring on**. You may choose

  - **Speakerphone:** The ring will be played on the Speakerphone.
  - **Headset:** The ring will be played on the Headset.

  Default: Speakerphone.

  When you select the Headset as the destination, make sure that you selected the *Headset Connected?* check box, connect a Headset to the SPARSH VP248.

- Select the desired **Ring Tune** according to your/SPARSH VP248 user's preference. Default: 1.

- Set the **Ringer Volume** to the desired level, from 0 to 7, according to your preference. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the handset of the SPARSH VP248, set the **Handset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of incoming speech (Receive Gain) on the handset of the SPARSH VP248, set the **Handset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the headset of the SPARSH VP248, set the **Headset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Receive Gain) on the headset of the SPARSH VP248, set the **Headset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To use a Headset with the SPARSH VP248, select the **Headset Connected?** check box. Default: Disabled.

  Make sure that you connect a Headset to the SPARSH VP248, if you enable this option.

- Select the **Auto Answer** check box to enable this feature on the SPARSH VP248. Default: Disabled.

  When you set the “Auto Answer” feature on the SPARSH VP248, the phone goes OFF-Hook automatically after a preset period of time, without the extension user having to pick up the handset or press the speaker or headset key. When you enable Auto Answer, you must configure the Auto Answer Timer.

- If you have enabled Auto Answer on the phone, set the **Auto Answer Timer (sec)** to the desired value.

  This timer defines the time in seconds that the SPARSH VP248 should wait before going OFF-Hook to auto answer a call. The range of this timer is 1 to 9 seconds. Default: 1 second.

- Adjust the Backlight brightness of the phone’s LCD display, by setting the **LCD Backlight Level** to the desired value, from 1 to 4. Default: 3.

- Set the **Back Light Off Timer (sec)** to the desired value, if required, from 000 to 999 seconds. Default: 10 seconds.

- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See “Customizing Extended IP Phone Templates” for more details.

  OR

- You can personalize the key map of the SPARSH VP248 for this location. To do so,

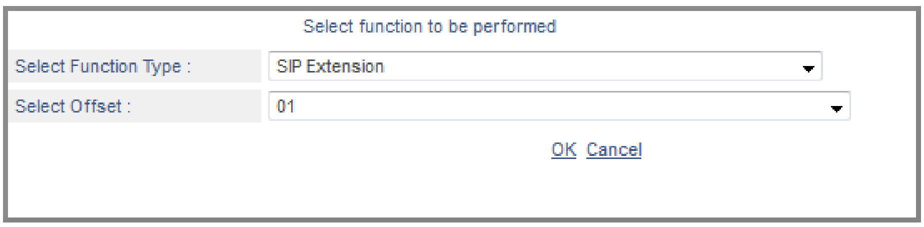
  - Select **Personalized** as the **Key Template** option.

- Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.

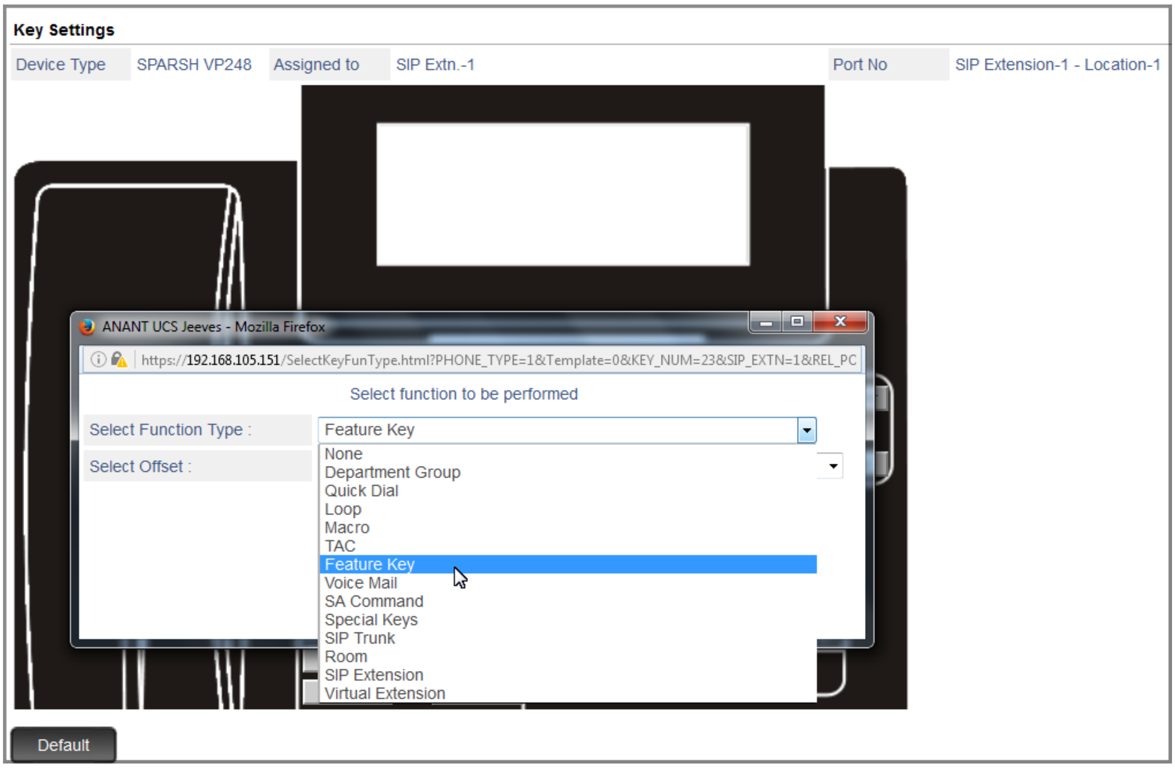
- Click the key you want to configure. For example, **SIPEXTN 1**.

  The **Functions to be Performed** by the key opens in a new window.



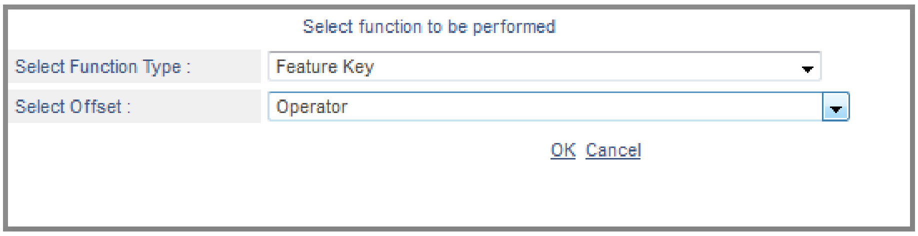
- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

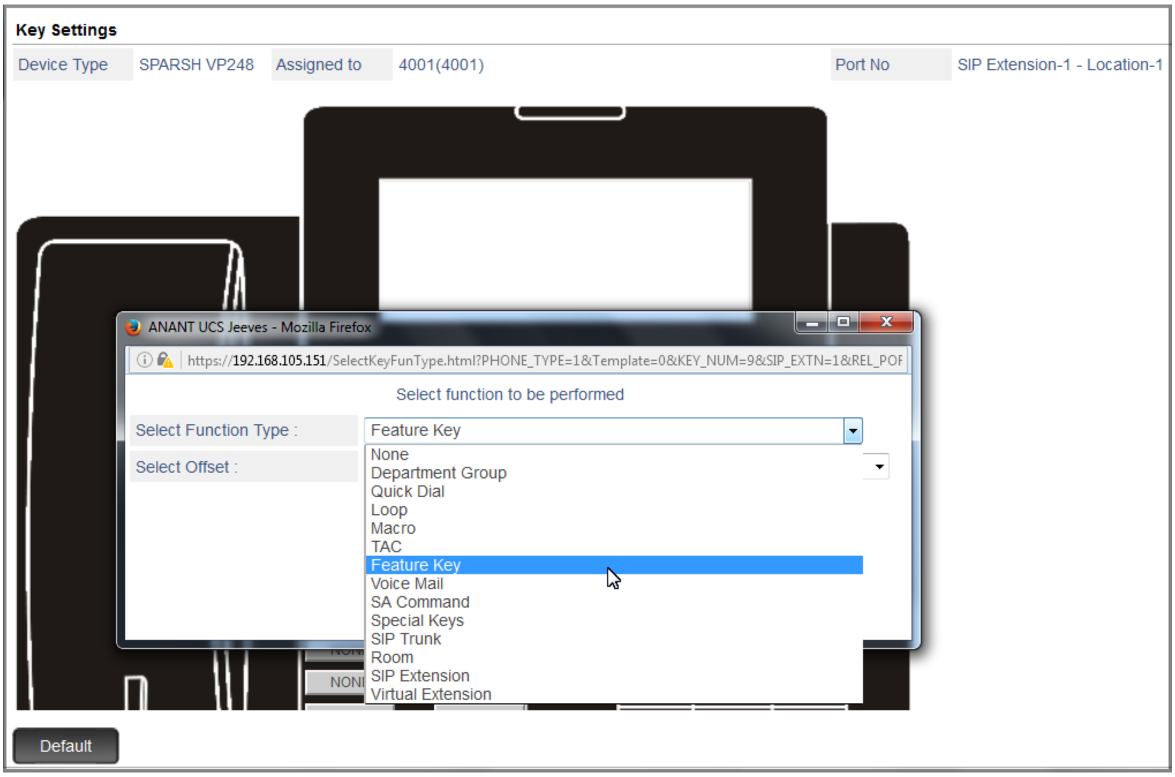


From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

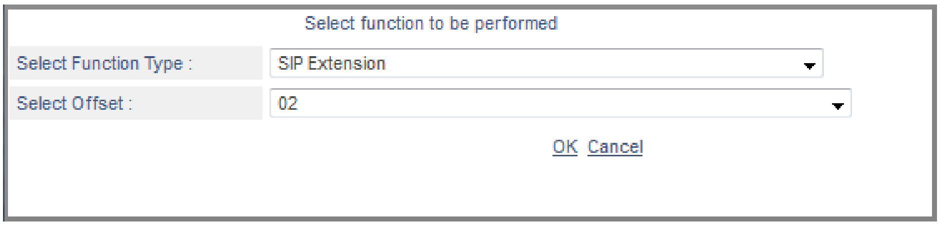
- Select **Operator** from the list of features in the **Select Offset** box.
- Click **OK**.



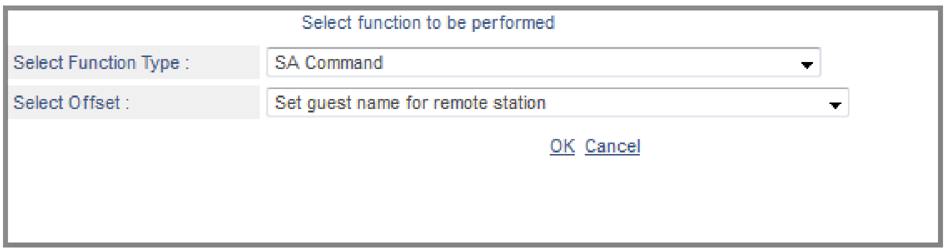
The *Operator* feature appears on the key label.



- To take a second example, if you want to assign **Remote DND** to the key currently assigned **SIPEXTN 2** key, click the key.

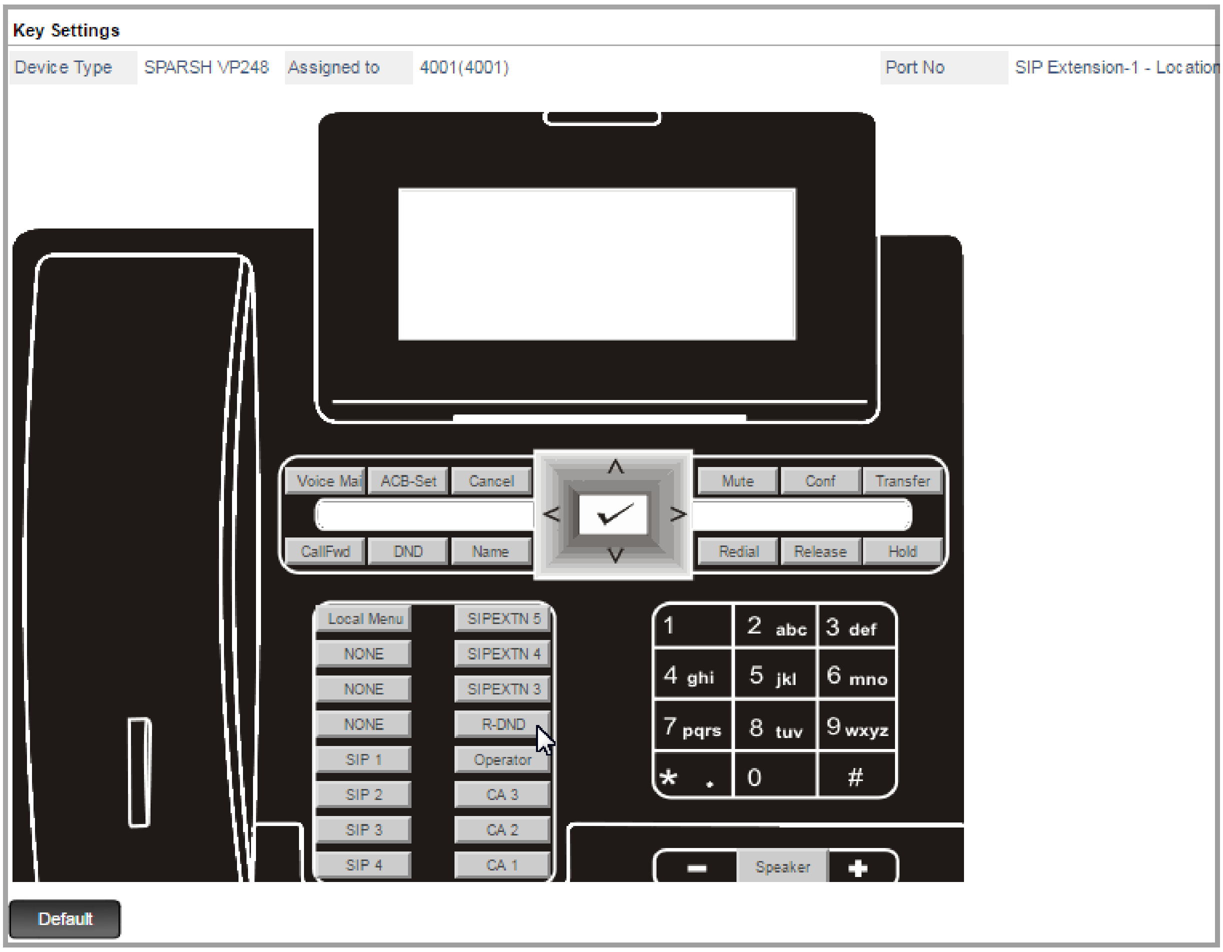


- In the **Select Function Type** list box, select the option **SA Command**, as Remote DND is a System Administrator (SA) Command.



- In the **Select Offset** box, select the option **Set DND for remote station**.

- Click **OK**. The box closes. Remote DND feature will appear in abbreviated form as *R-DND* on the key label.



- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a SIP extension, select **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **1** or any other trunk number as Offset.

  Click **OK**, each time you select a Function Type and Offset in the dialog box.

  You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- If you assign/re-assign functions to the following keys, the Phone will restart:
  - Speaker
  - Headset

- Ringer
- Acknowledge
- Local Menu

RTP Port

- Define **RTP Port**:

  - **RTP Listening Port:** This is the port on which the SPARSH VP248 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP248 is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the **Enable NAT Keep Alive** check box to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for

Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP248, configure Debug parameters:
  - Select the **Enable Debug?** check box. Default: disabled.

    When the Debug check box is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.
  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).
  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

    You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of 'Call's is required, you can select this option, and disable all others.
- Click **Submit** to save settings.
- If you have completed the configuration of the SPARSH VP248 Settings at Location 1, follow the same steps as described above to configure the SPARSH VP248 at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP248 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will restart automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password

- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Call Progress Tone
- Date and Time
- Apply DST?
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of ANANT UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP310

SPARSH VP310, the Executive IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. It functions like SPARSH VP248, the proprietary SIP-based IP Phone for ANANT UCS, of Matrix. To know the list of features supported, refer to *“ANANT UCS Features Supported in Terminals”*.

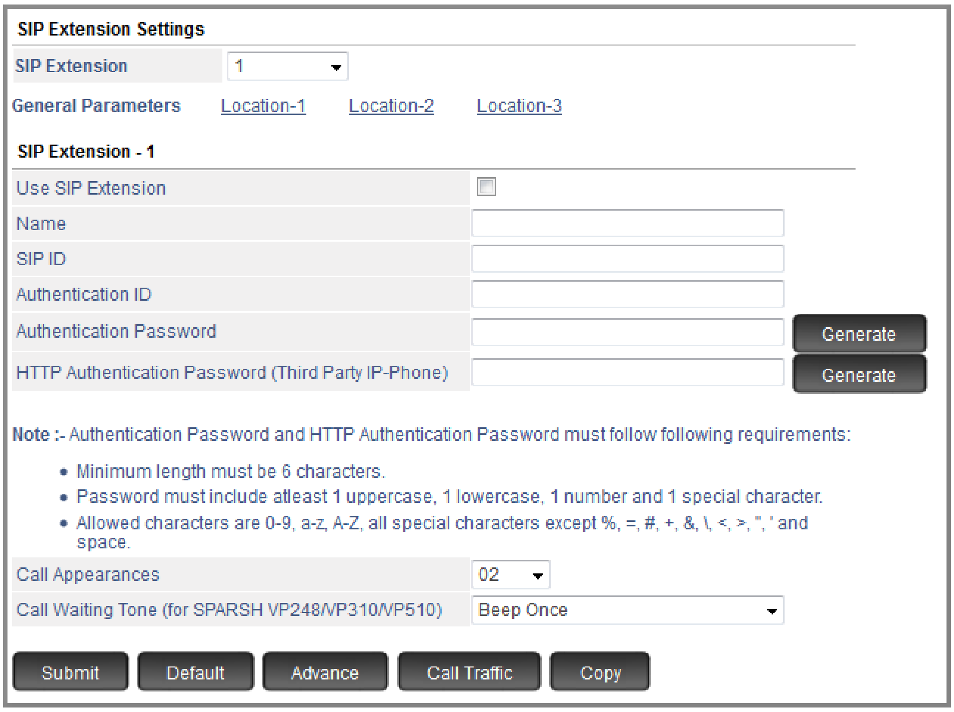
For instructions on how to use SPARSH VP310, refer to the common *EON48_310_SPARSH VP248_310 User Guide.*

To be able to use SPARSH VP310[18] - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see *“Configuring SIP Extension General Parameters”*.
- SIP Extension Settings, see *“Configuring SIP Extension Settings”*.
- Extended IP Phone Settings, see *“Configuring Matrix Extended Phone Settings”*.
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See *“Extension Voice Mail Settings”*.

Configuring SIP Extension Settings

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

18. *ANANT UCS supports only IPv4 Addresses for registering SPARSH VP310.*

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

ANANT UCS supports IPv4 Addresses only for registering Extended IP Phones.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension.

However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

- In **Call Appearances**, define the maximum number[19] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the **Call Waiting Tone (for SPARSH VP248/VP310/ VP510)** beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once

  However, when a ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

__Auto Sign-In parameters — Send Configuration Mail__ and __Mail Status,__ are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:
  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.
  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

19. *The calls that are routed through the system will depend on the number of transcoding channels available.*

- **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.
  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click **SIP Hardware Template**.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01. The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (CoS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Template**.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

- Under **Configuration**, click **Station Advanced Feature Template**.
- Select the Template number, for example 02, and customize this template.
- Click **Submit** and return to the 'SIP Extension Settings' page.
- In the **Station Advanced Feature Template** field, select the number of the template you customized.
- Click **Submit** to save changes.

Also see the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required. Default: 01

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

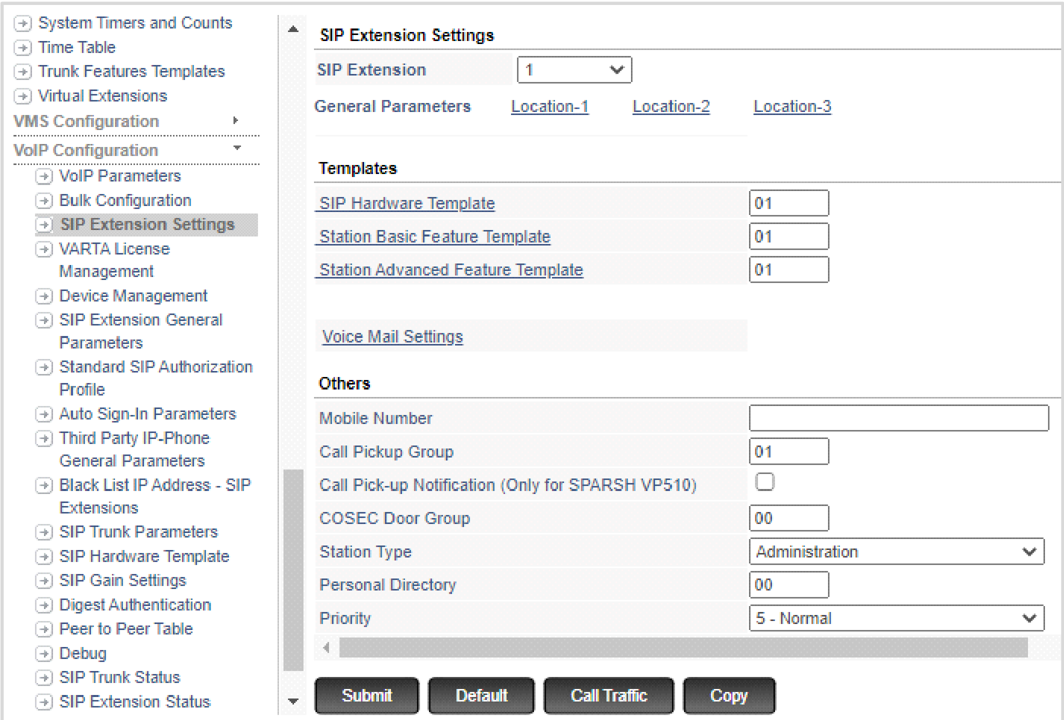
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to “Call Pick Up” and “Class of Service (CoS)”. Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.**

  If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on configuring the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended IP Phones/Soft Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones—SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330, SPARSH VP210, Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP310 is connected at Location 1, 2 and 3.

If you want to use more than one SPARSH VP310 Extended IP Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP248 at any of the locations, refer to *"Configuring Matrix Extended Phone Settings"*.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

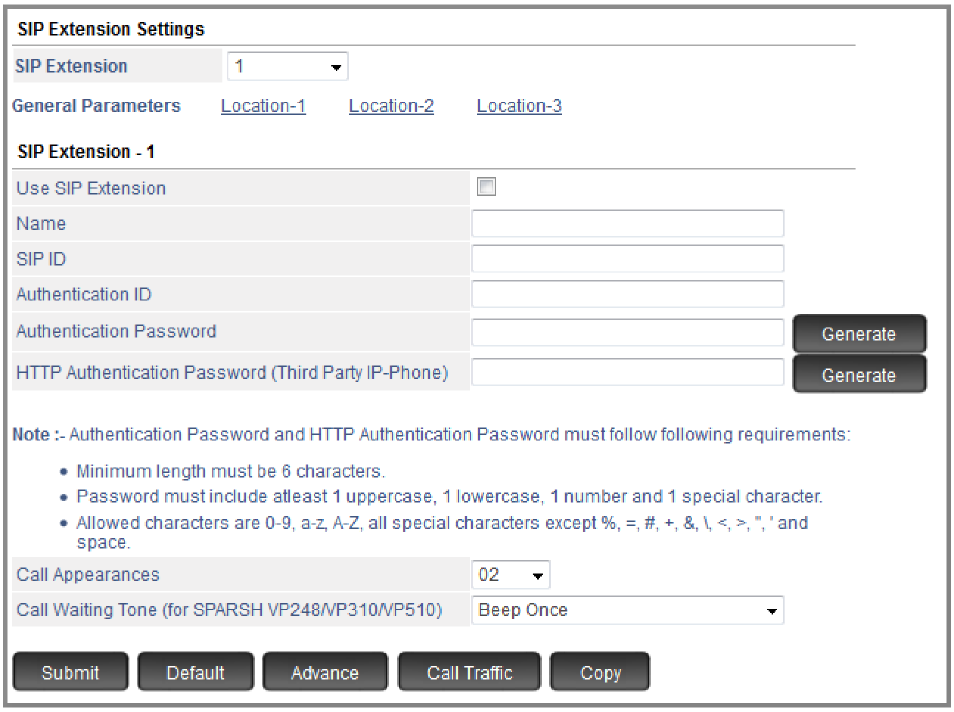
If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710"*.

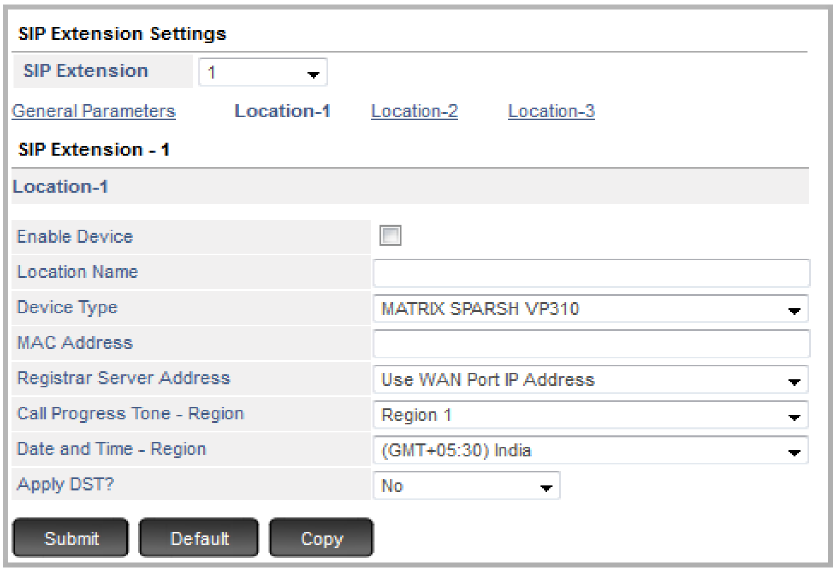
If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.
- Select **MATRIX SPARSH VP310** as the **Device Type** at this location.

- Enter the **MAC Address**[20] of the SPARSH VP310 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

ANANT UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the configured MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP310 with the SIP Registrar of ANANT UCS, according to your installation scenario:
  - If the SPARSH VP310 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.
  - If the SPARSH VP310 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.
  - If the SPARSH VP310 is connected in the Global Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".
  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

    By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.
- To set the call progress tone generation standards of the country where the SPARSH VP310 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.
- To display the Date and Time of the country where the SPARSH VP310 is installed, select the **Date and Time - Region**. Default: (GMT+5:30) India.
- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of ANANT UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  ANANT UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to "Daylight Saving Time (DST)". To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in "Default Settings".

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20. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

When you select **Manual** as the DST option, the Real Time Clock of ANANT UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

- In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

- In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

  If you select **'Date-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Date**: Select the date on which DST begins (1-31).
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
  - **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

  Similarly, in the **DST End** configure the desired DST End Time details.

  If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
  - **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
  - **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

  Similarly, in **DST End** configure the DST End Time details.

  Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- Select the **Trunk CLIP Pattern** for the SPARSH VP310. This is the type of Calling Line Presentation on the phone for incoming calls from trunks. You can select any of these options:

  - **Name Only** (only the name of the caller will be displayed).
  - **Number Only** (only the number of the caller will be displayed).
  - **Number + Name** (both the name and the number of the caller will be displayed).

  Default: Number + Name.

- Select the **Display Language** for the SPARSH VP310. Default: English.

  ANANT UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP310. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- *SIP Extension users can change the language by accessing and navigating through the phone menu.*
- *The SA can change the Language by logging into the SA Jeeves.*

- Select a **Ringer Mode** for the phone from the four options:

  - Ring immediately (it rings immediately as a fresh calls lands on the phone).
  - Ring if idle (rings only if the phone is idle).
  - Ring after delay (if the call is still not answered).
  - Silent.

  Default: Ring Immediate.

  If you selected *Ring after delay* as Ringer Mode, set the **Ring Delay Timer (sec)**, if required, to the desired value.

  The Ring Delay Timer is the time in seconds the system waits on receiving a call before ringing on the phone. The range of this timer is 0 to 99 seconds. Default: 10 seconds.

- If you want to enable *Ringer Auto Acknowledge* mode, set the **Acknowledge Timer** (sec) to the desired value.

  The Ringer Auto Acknowledge mode determines when to stop the ring on the phone. There are two options for Ringer Auto Acknowledge:
  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the phone after a delay, the Acknowledge Timer must be configured. The range of this timer is 00 to 99 seconds. Default: 00 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- To assign the Ring Destination for the SPARSH VP310, select the desired destination for **Play Ring on**. You may choose
  - **Speakerphone:** The ring will be played on the Speakerphone.
  - **Headset:** The ring will be played on the Headset.

  Default: Speakerphone.

  When you select the Headset as the destination, make sure that you enable the *Headset Connected?* check box. Connect a Headset to the SPARSH VP310.

- Set the **Ringer Volume** to the desired level, from 0 to 7, according to your preference. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the handset of the SPARSH VP310, set the **Handset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of incoming speech (Receive Gain) on the handset of the SPARSH VP310, set the **Handset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the headset of the SPARSH VP310, set the **Headset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Receive Gain) on the headset of the SPARSH VP310, set the **Headset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To use a Headset with the SPARSH VP310, select the **Headset Connected?** check box**.** Default: Disabled.

  Make sure that you connect a Headset to the SPARSH VP310, if you enable this option.

- Select the **Auto Answer** check box to enable this feature on the SPARSH VP310. Default: Disabled.

  When you set the "Auto Answer" feature on the SPARSH VP310, the phone goes OFF-Hook automatically after a preset period of time, without the extension user having to pick up the handset or press the speaker or headset key. When you enable Auto Answer, you must configure the Auto Answer Timer.
- If you have enabled Auto Answer on the phone, set the **Auto Answer Timer (sec)** to the desired value.

  This timer defines the time in seconds that the SPARSH VP310 should wait before going OFF-Hook to auto answer a call. The range of this timer is 1 to 9 seconds. Default: 1 second.

- Adjust the Backlight brightness of the phone's LCD display, by setting the **LCD Backlight Level** to the desired value, from 1 to 4. Default: 3.

- Set the **Back Light Off Timer (sec)** to the desired value, if required, from 000 to 999 seconds. Default: 10 seconds.

- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates" for more details.

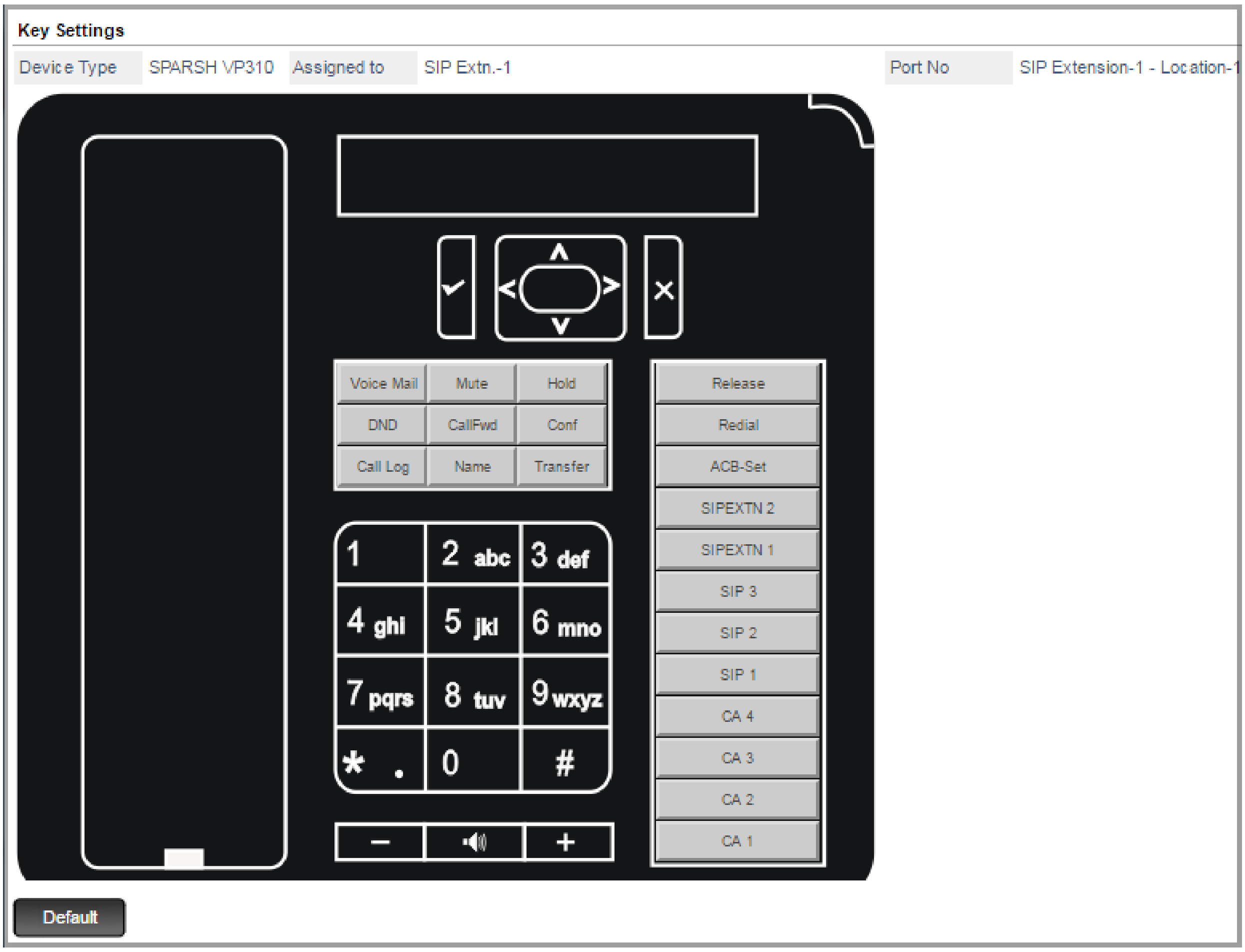
  OR

- You can personalize the key map of the SPARSH VP310 for this location. To do so,

  - Select **Personalized** as the **Key Template** option.

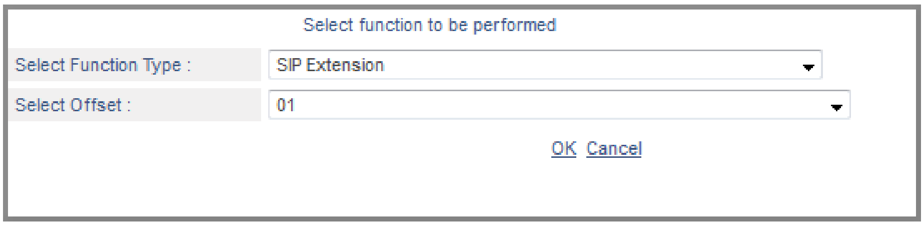
- Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.



- Click the key you want to configure. For example, **CA 1**.

  The **Functions to be Performed** by the key opens in a new window.

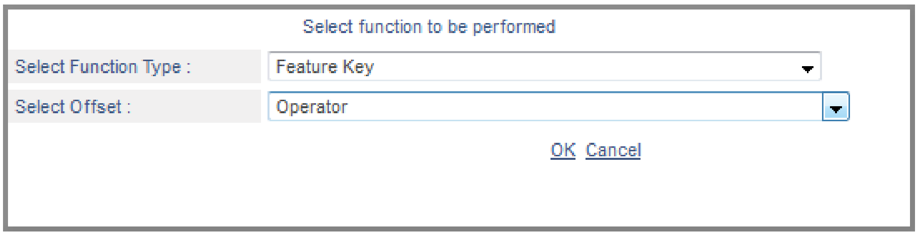


- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

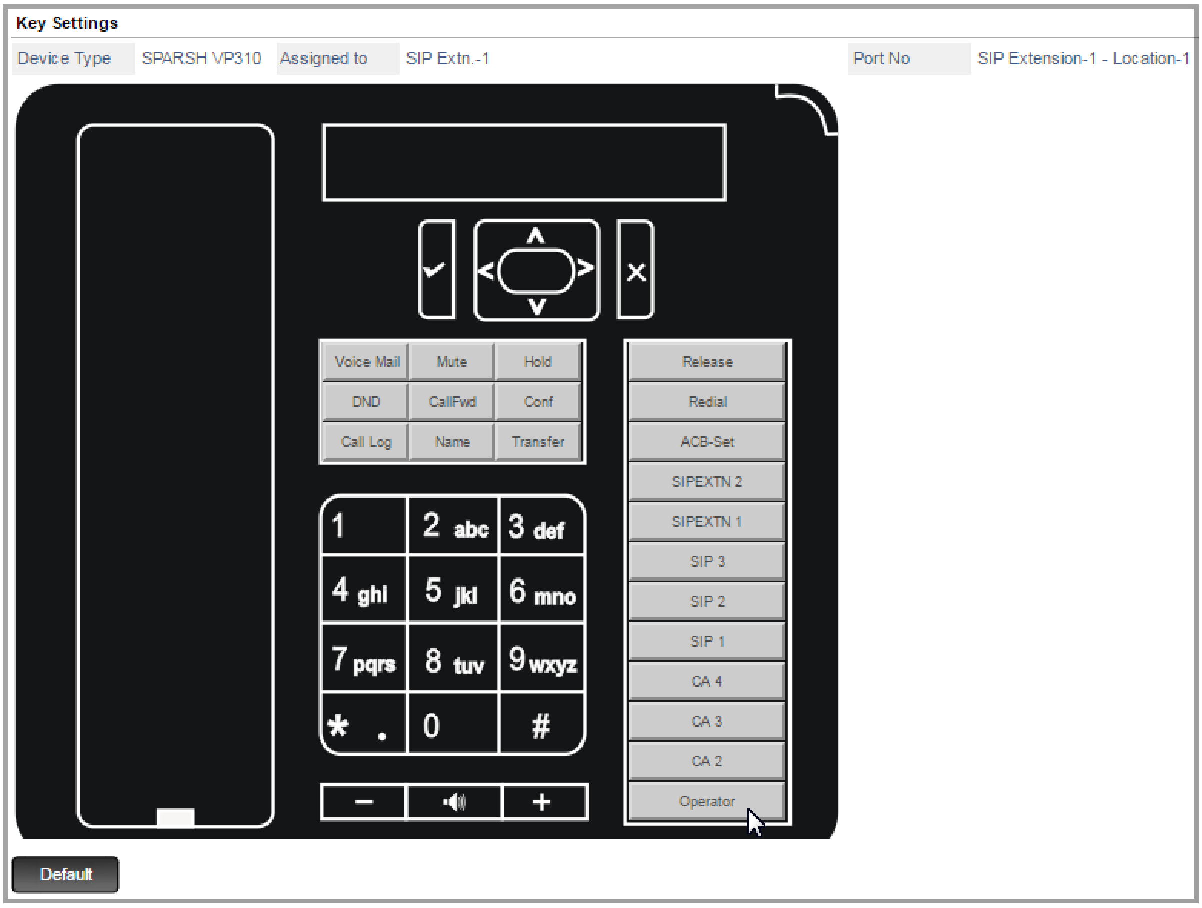
The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

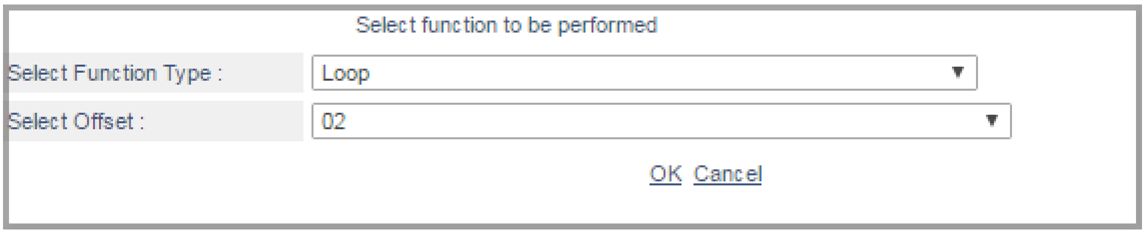
- Select **Operator** from the list of features in the **Select Offset** box.
- Click **OK**.



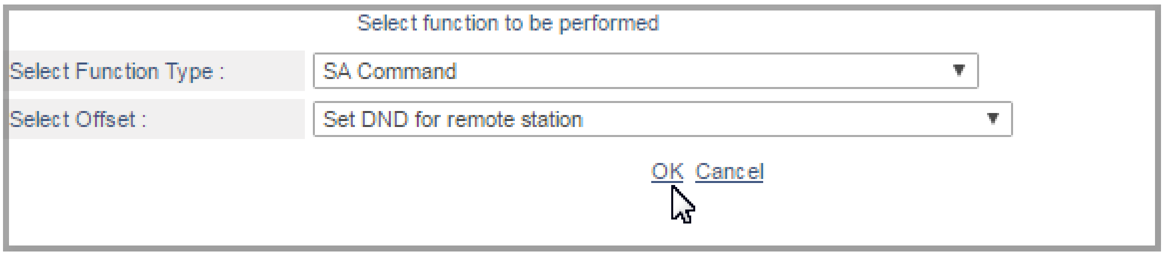
The *Operator* feature appears on the key label.



- To take a second example, if you want to assign **Remote DND** to the key currently assigned **CA 2** key, click the key.

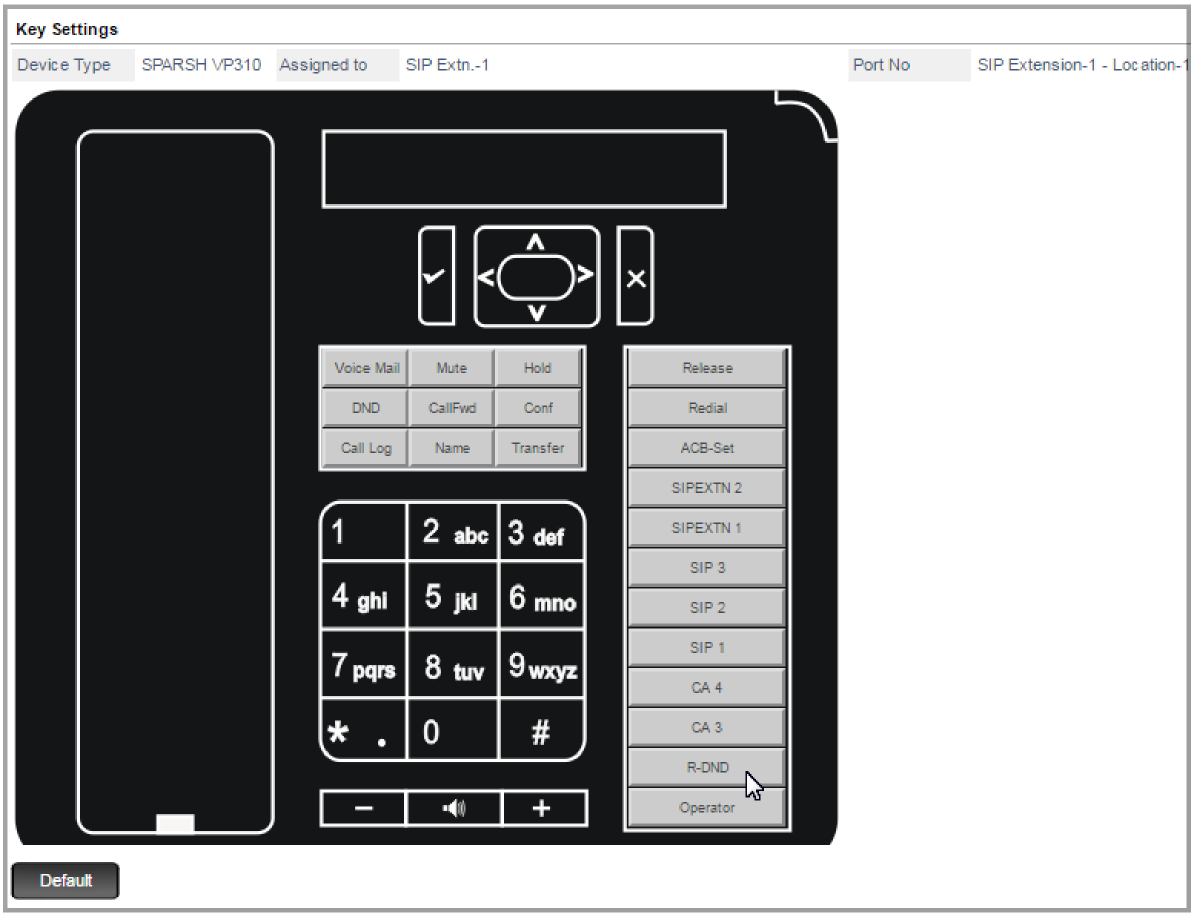


- In the **Select Function Type** list box, select the option **SA Command**, as Remote DND is a System Administrator (SA) Command.



- In the **Select Offset** box, select the option **Set DND for remote station**.

- Click **OK**. The box closes. Remote DND feature will appear in abbreviated form as *R-DND* on the key label.



- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a SIP extension, select **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **01** or any other trunk number as Offset.

  Click **OK**, each time you select a Function Type and Offset in the dialog box.

  You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- If you assign/re-assign functions to the following keys, the Phone will restart:
  - Speaker
  - Headset
  - Ringer Acknowledge

- Local Menu

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

- *If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.*
- *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.*

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

If you select this check box, make sure you have selected SRTP Mode as Forced or Optional in the General Parameters under SIP Extension Settings.

RTP Port

- Define **RTP Port**:
  - **RTP Listening Port:** This is the port on which the SPARSH VP310 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:
  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP310 is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.
  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:
  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

- **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

- **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP310, configure Debug parameters:
  - Select the **Enable Debug?** check box. Default: disabled.

    When the Debug check box is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.

  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

    You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of 'Call's is required, you can select this option, and disable all others.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP310 Settings at Location 1, follow the same steps as described above to configure the SPARSH VP310 at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP310 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will restart automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Call Progress Tone
- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of ANANT UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters in Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP330

SPARSH VP330 is the proprietary Extended IP Phone with graphical touch-screen user interface, supplied by Matrix. The feature-rich SIP based phone support most of the features and function, to know the list of features supported, refer to *"ANANT UCS Features Supported in Terminals"*.

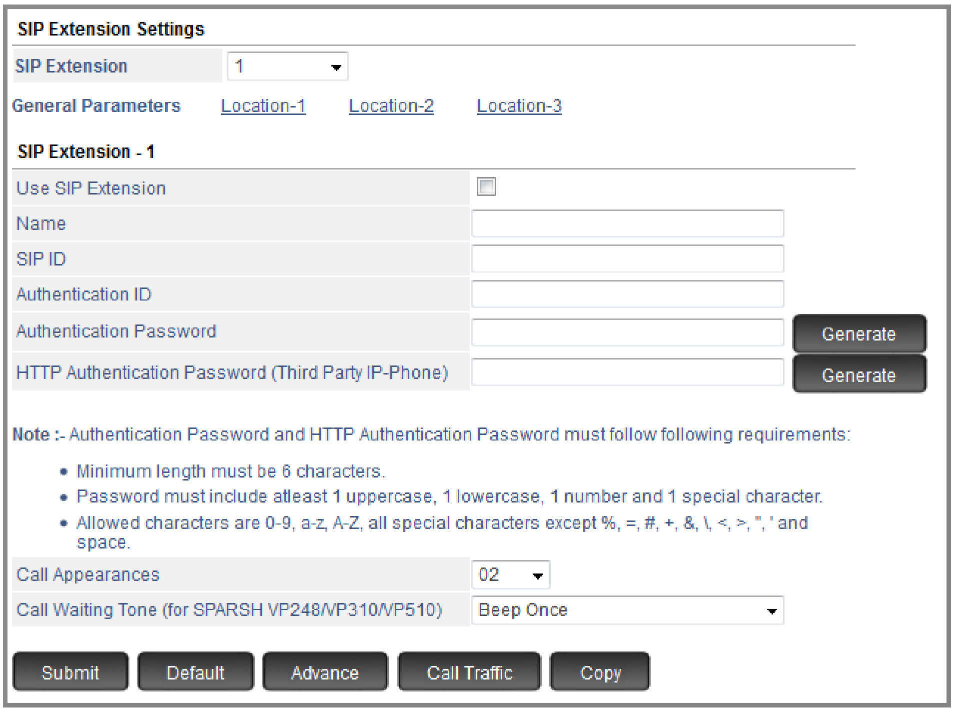
For detailed product information and operation instructions, refer to the *SPARSH VP330 User Guide.*

To be able to use SPARSH VP330[21] as SIP Extensions, you must configure the following:

- SIP Extension General Parameters, see *"Configuring SIP Extension General Parameters"*.
- SIP Extension Settings, see *"Configuring SIP Extension Settings"*
- Extended Phone Settings, see *"Configuring Matrix Extended Phone Settings"*
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See *"Extension Voice Mail Settings"*.

Configuring SIP Extension Settings

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

- You may select the **SIP Extension** number you want to configure.

21. *ANANT UCS supports only IPv4 Addresses for registering SPARSH VP330.*

The parameters of the SIP Extension number you selected will appear on this page.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In **Name**, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension.You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, it must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension.

However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

- In **Call Appearances**, define the maximum number[22] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

***Auto Sign-In parameters — Send Configuration Mail** and **Mail Status,** are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:
  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.
  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.
  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

22. *The calls that are routed through the system will depend on the number of transcoding channels available.*

If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

- Under **VoIP Configuration**, click **SIP Hardware Template**.
- Select a Template number, for example 02.
- Customize Template number 02 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01.The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (CoS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Template**.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click **Station Advanced Feature Template**.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit** to save changes.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

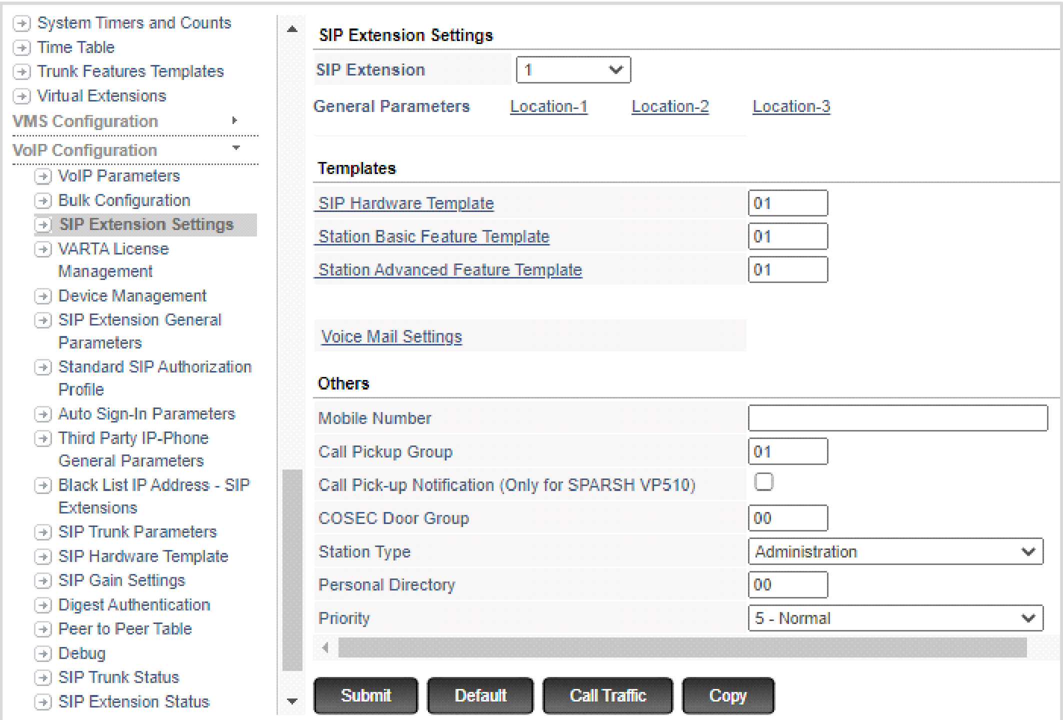
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (CoS)". Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.** If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features “Abbreviated Dialing” and “Dial By Name”.

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the System Engineer. Refer the topic “Abbreviated Dialing” for instructions on configuring the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal.

  Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description “Priority”.

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended Phones/Soft Clients at three different locations as a single SIP Extension. You can connect the same or different types of Extended Phones/Soft Clients —SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330, SPARSH VP210, Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, Matrix VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP330 is connected at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to *"Configuring Matrix Extended Phone Settings"* in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

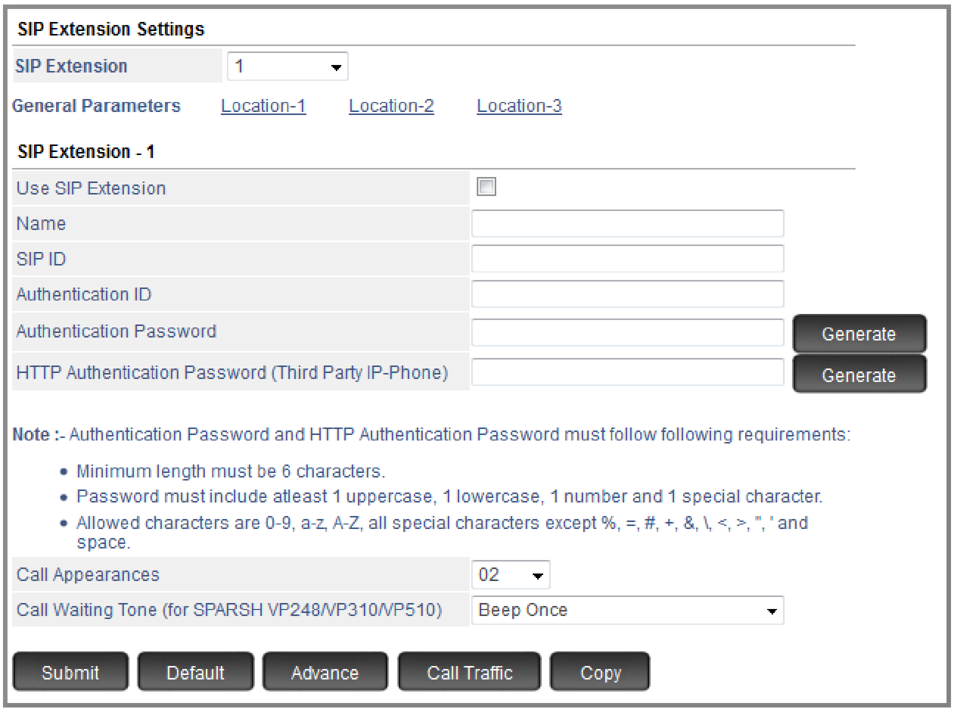
If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710"*.

If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.

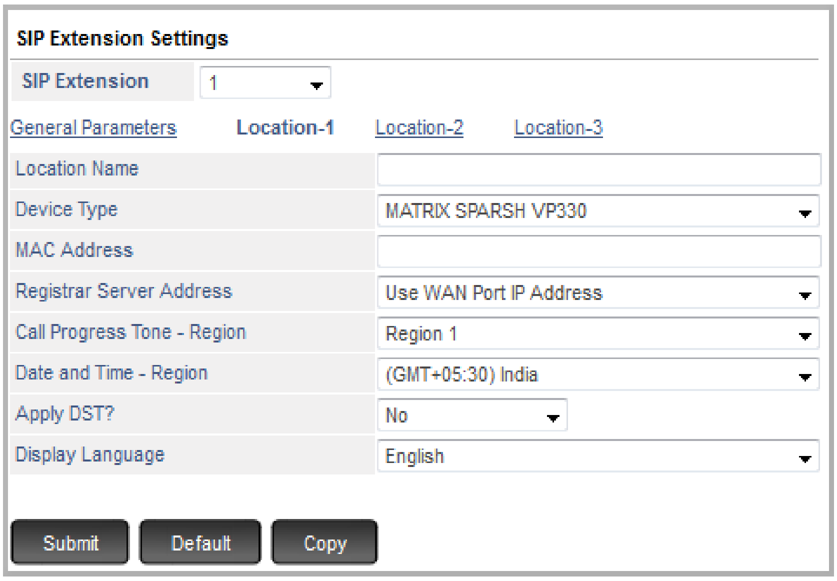
If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

If you want to use more than one SPARSH VP330 Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP330** as the **Device Type** at this location.

- Enter the **MAC Address**[23] of the SPARSH VP330 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

ANANT UCS validates the SPARSH VP330 on the basis of the MAC Address, and provides configuration on validation.

As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the configured MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP330 with the SIP Registrar of ANANT UCS, according to your installation scenario:
  - If the SPARSH VP330 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.
  - If the SPARSH VP330 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.
  - If the SPARSH VP330 is connected in the Global Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".
  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

    By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.
- To set the call progress tone generation standards of the country where the SPARSH VP330 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See "Call Progress Tones" to know more.
- To display the Date and Time of the country where the SPARSH VP330 is installed, select the **Date and Time - Region**. Default: (GMT+5:30) India.
- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of ANANT UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  ANANT UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to "Daylight Saving Time (DST)". To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in "Default Settings".

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23. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

When you select **Manual** as the DST option, the Real Time Clock of ANANT UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

- In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

- In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

  If you select **'Date-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Date**: Select the date on which DST begins (1-31).
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
  - **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

  Similarly, in the **DST End** configure the desired DST End Time details.

  If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
  - **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
  - **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

  Similarly, in **DST End** configure the DST End Time details.

  Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- Select the **Display Language** for the SPARSH VP330. Default: English.

  ANANT UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP330. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- *SIP Extension users can change the language by accessing and navigating through the phone menu.*
- *The SA can change the Language by logging into the SA Jeeves.*

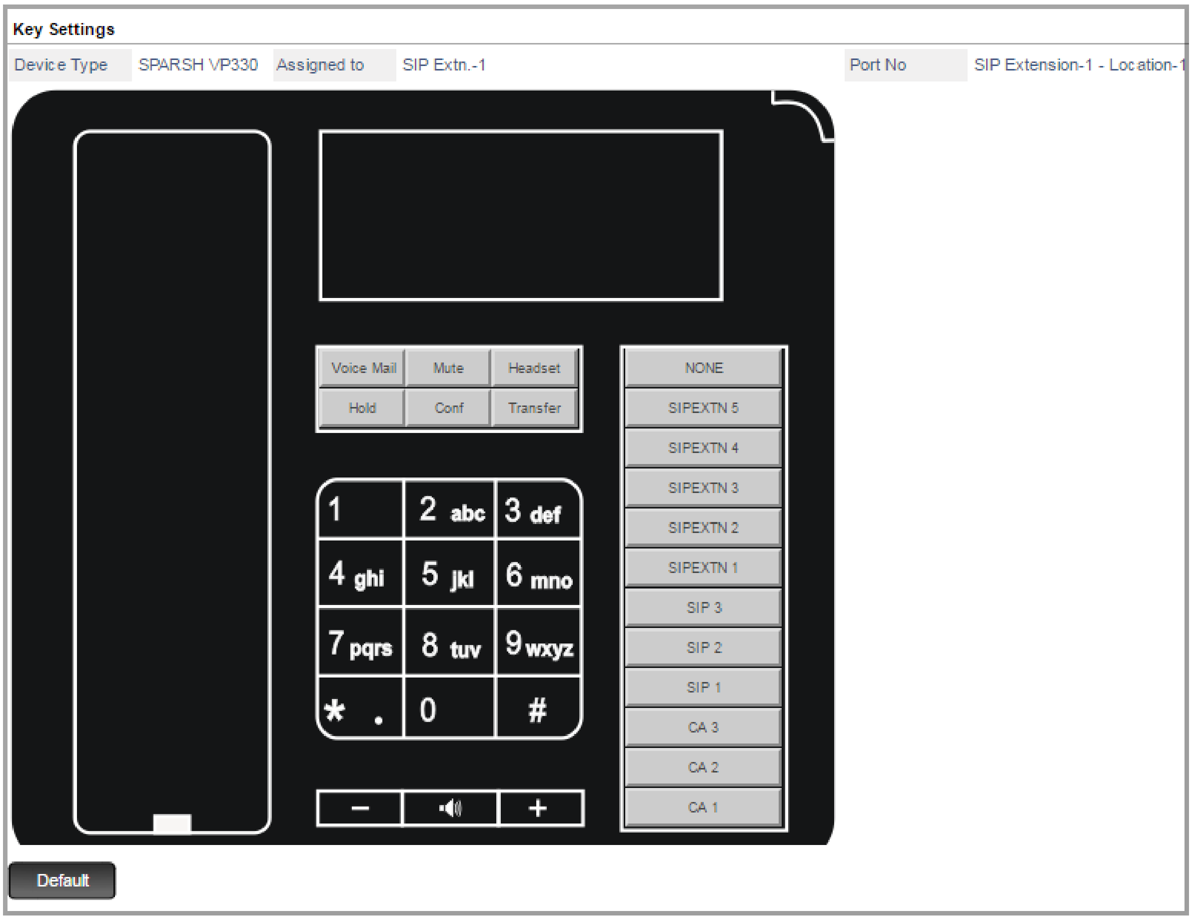
DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates" for more details.

  OR

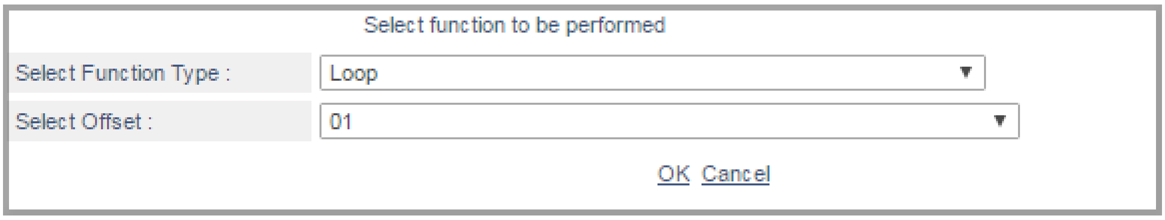
- You can personalize the key map of the SPARSH VP330 for this location. To do so,
  - Select **Personalized** as the **Key Template** option.
  - Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.



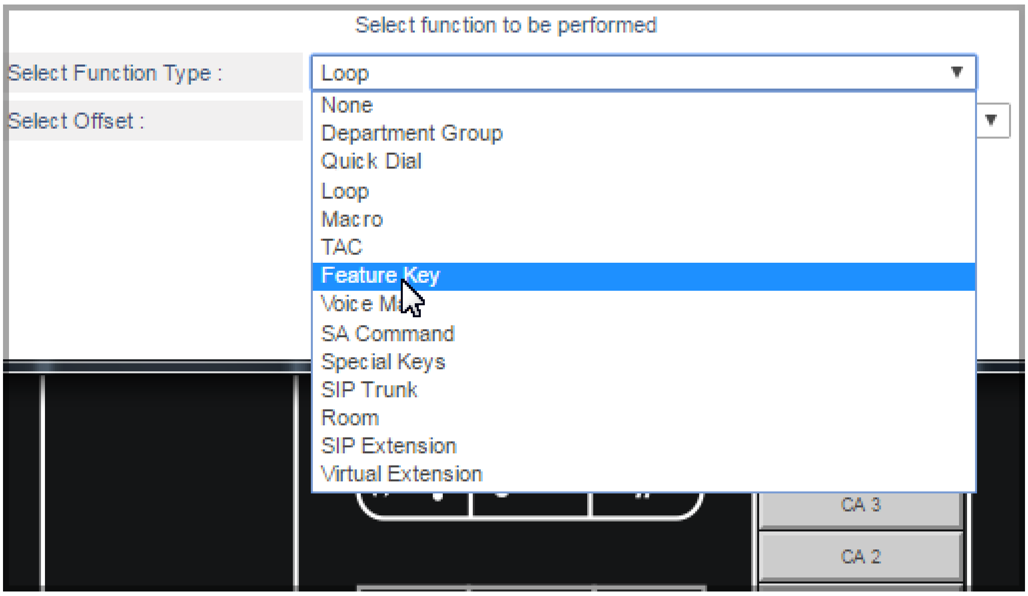
- Click the key you want to configure. For example, **CA 1**.

  The **Functions to be Performed** by the key opens in a new window.



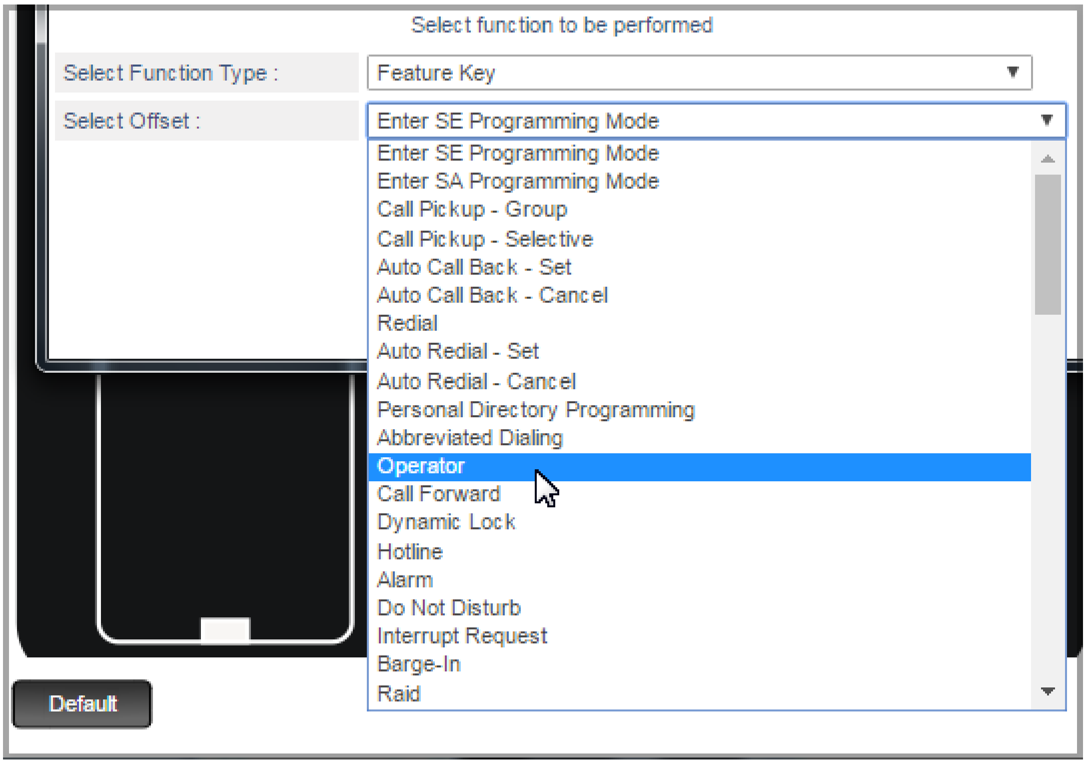
- In the **Select Function Type**, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

  The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

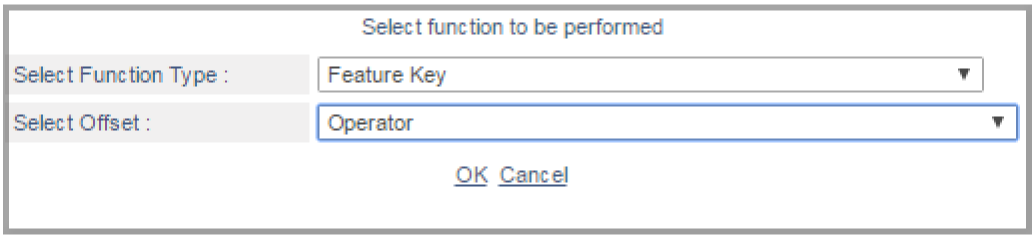


From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

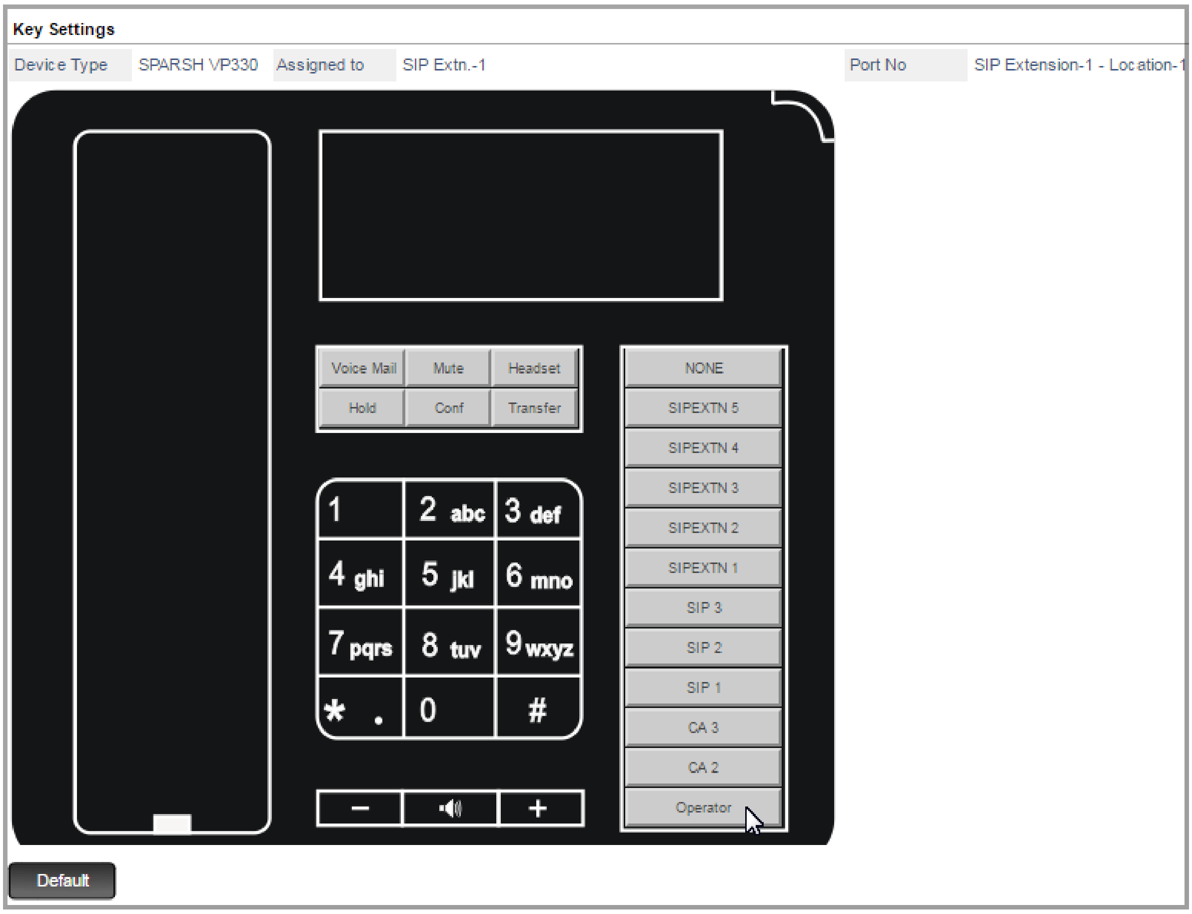
- Select **Operator** from the list of features in the **Select Offset** box.



- Click **OK**.



The *Operator* feature appears on the key label.



- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want to assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a SIP extension, select **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **1** or any other trunk number as Offset.

  Click **OK**, each time you select a Function Type and Offset in the dialog box.

  You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- *The phone will enter the Auto Configuration mode, when you assign/re-assign certain features in the key maps. To know more, refer to the SPARSH VP330 User Guide.*
- *Even if you assign keys for the following feature in the Key Templates, these features will not function:*

Function Type	Offset
Macro	
SA Command	
Special Keys	Digit Pause
	Digit A
	Digit B
	Digit C
	Digit D
	Enter
	Local Menu
Feature	Enter SE Programming Mode
	Enter SA Programming Mode
	Personal Directory Programming
	Abbreviated Dialing
	Emergency Conference
	Self Ring Test
	SA Command Prefix
	PMS - User Defined Fields
	Department Group Call Forward

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**[24].

- *If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.*
- *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.*

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

RTP Port

- Define **RTP Port**:
  - **RTP Listening Port:** This is the port on which the phone listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:
  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.

24. *SPARSH VP330 supports TLS Version V1.0 only. To configure the TLS version, refer "Advance Options" in Security Settings.*

OR

- **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP330 is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.
  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:
  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.
  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.
  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.
- Click **Submit** to save settings.
- If you have completed the configuration of the SPARSH VP330 Phone Settings at Location 1, follow the same steps as described above to configure the SPARSH VP330 Phone at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP330 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Language
- Call Progress Tone
- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- Emergency Numbers

The SIP Extension registered at Location 1, 2, 3, will also restart, if:

- The SE Password of ANANT UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP510

SPARSH VP510, the Premium IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. To know the list of features supported, refer to *"ANANT UCS Features Supported in Terminals"*.

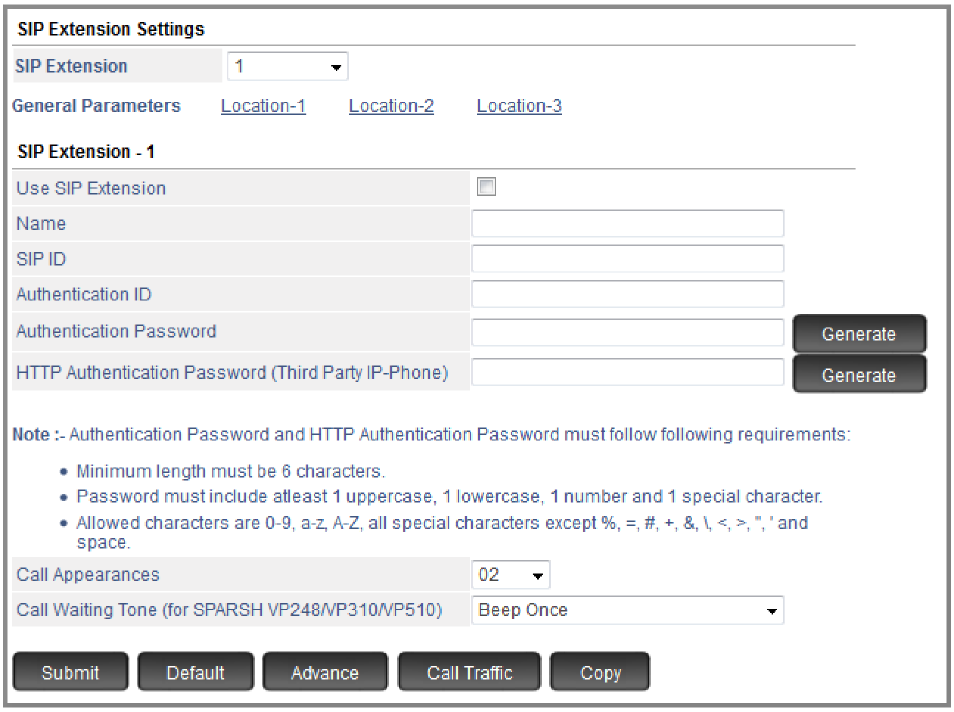
For instructions on how to use SPARSH VP510, refer to the *EON510_SPARSH VP510 User Guide.*

To be able to use SPARSH VP510[25] - Extended IP Phone, you must configure the following:

- SIP Extension General Parameters, see *"Configuring SIP Extension General Parameters"*.
- SIP Extension Settings, see *"Configuring SIP Extension Settings"*
- Extended Phone Settings, see *"Configuring Matrix Extended Phone Settings"*
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See *"Extension Voice Mail Settings"*.

Configuring SIP Extension Settings

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

- You may select the **SIP Extension** number you want to configure.

25. *ANANT UCS supports only IPv4 Addresses for registering SPARSH VP510.*

The parameters of the SIP Extension number you selected will appear on this page.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.
- In **Name**, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.
- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see *"Assigning Access Codes to a Range of Extensions"*.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:
  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See *"Black List IP Address - SIP Extensions"* for more details. This activity will be logged in the *"System Activity Log"* as well as *"Simple Network Management Protocol (SNMP)"*.

- In **Call Appearances**, define the maximum number[26] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- During an on-going conversation, if there is a second incoming call, the system plays beeps to indicate the second incoming call. You can set the frequency of the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** beeps as per your requirement. You can select from the following options:
  - Off
  - Beep Once
  - Beep until Answered

  Default: Beep Once

  However, when a ongoing call is being taped or recorded, the call waiting tone for any new incoming call will not be played.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

__Auto Sign-In parameters — Send Configuration Mail__ and __Mail Status,__ are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:

  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.

  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

26. *The calls that are routed through the system will depend on the number of transcoding channels available.*

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The *"SIP Hardware Template"* contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks. Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click **SIP Hardware Template**.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic *"SIP Hardware Template"* to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01. The *"Station Basic Feature Template"* has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (CoS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Template**.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

- Under **Configuration**, click **Station Advanced Feature Template**.
- Select the Template number, for example 02, and customize this template.
- Click **Submit** and return to the 'SIP Extension Settings' page.
- In the **Station Advanced Feature Template** field, select the number of the template you customized.
- Click **Submit** to save changes.

Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required. Default: 01

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer "Call Pick Up" for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

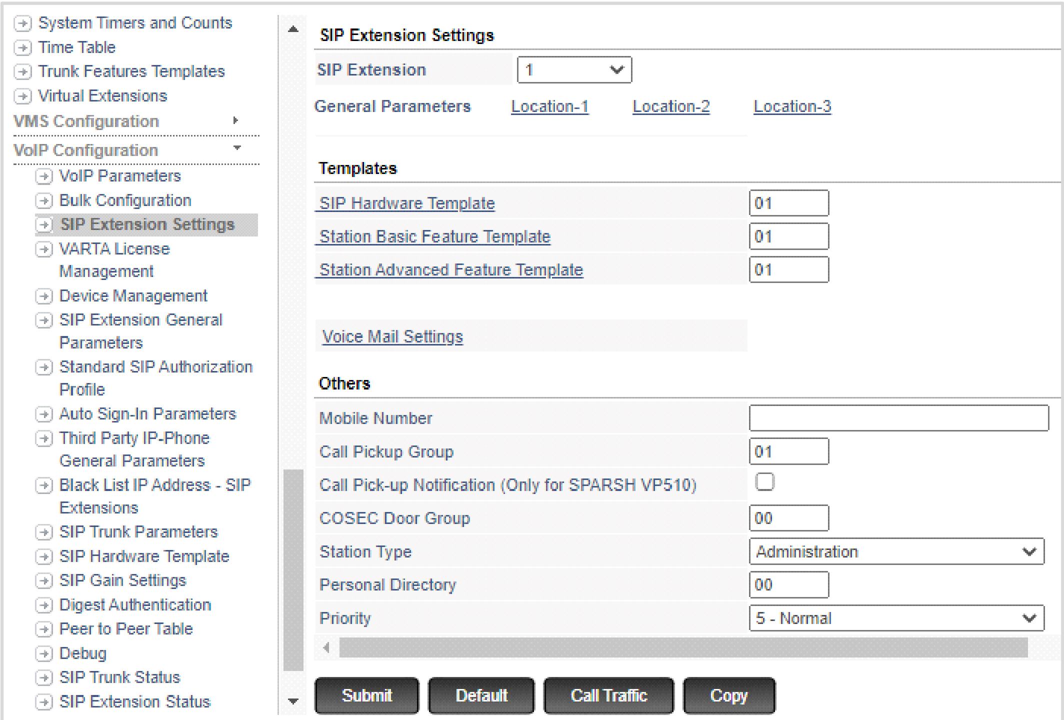
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (CoS)". Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.**

  If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on configuring the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal.

  Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended IP Phones/Soft Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones/Soft Clients —SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330, SPARSH VP210 Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client— at each location. In this case we assume that SPARSH VP510 is connected at Location 1, 2 and 3.

If you want to use more than one SPARSH VP510 Extended IP Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP248 at any of the locations, refer to *"Configuring Matrix Extended Phone Settings"*.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

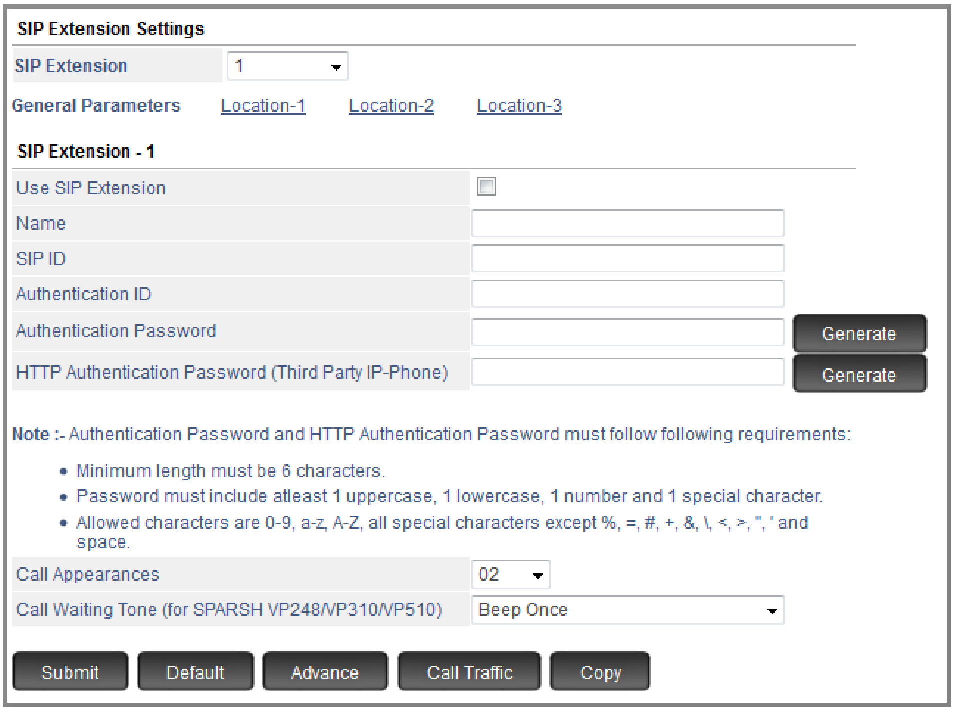
If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710".*

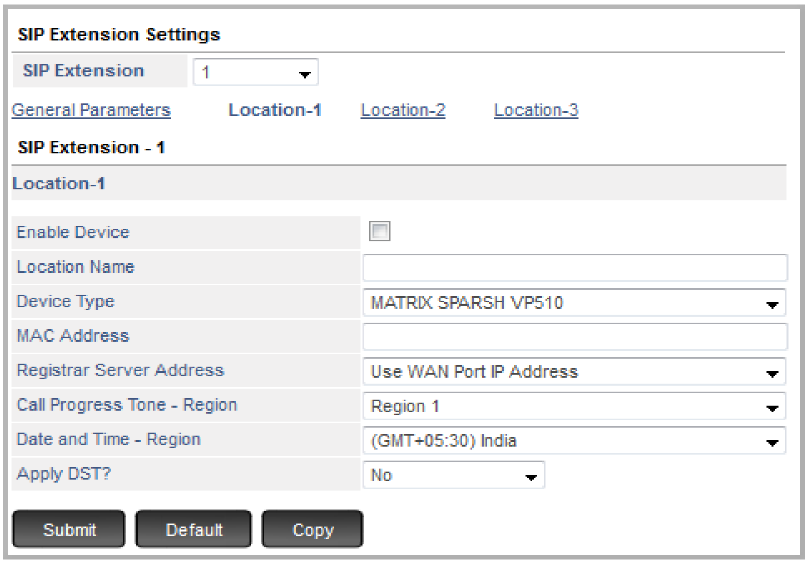
If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.
- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.
- Select **MATRIX SPARSH VP510** as the **Device Type** at this location.
- Enter the **MAC Address**[27] of the SPARSH VP510 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: Blank.

ANANT UCS validates the Extended Phone on the basis of the MAC Address, and provides configuration on validation.

As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the configured MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP510 with the SIP Registrar of ANANT UCS, according to your installation scenario:
  - If the SPARSH VP510 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.
  - If the SPARSH VP510 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.
  - If the SPARSH VP510 is connected in the Global Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see *"Configuring Network Parameters"*.
  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

    By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.
- To set the call progress tone generation standards of the country where the SPARSH VP510 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See *"Call Progress Tones"* to know more.
- To display the Date and Time of the country where the SPARSH VP510 is installed, select the **Date and Time - Region**. Default: (GMT+5:30) India.
- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of ANANT UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  ANANT UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to *"Daylight Saving Time (DST)"*. To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in *"Default Settings"*.

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27. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

When you select **Manual** as the DST option, the Real Time Clock of ANANT UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

- In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

- In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

  If you select **'Date-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Date**: Select the date on which DST begins (1-31).
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
  - **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

  Similarly, in the **DST End** configure the desired DST End Time details.

  If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

  - **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
  - **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
  - **Month**: Select the month when DST begins (January-December).
  - **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
  - **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

  Similarly, in **DST End** configure the DST End Time details.

  Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- Select the **Trunk CLIP Pattern** for the SPARSH VP510. This is the type of Calling Line Presentation on the phone for incoming calls from trunks. You can select any of these options:
  - **Name Only** (only the name of the caller will be displayed).
  - **Number Only** (only the number of the caller will be displayed).
  - **Number + Name** (both the name and the number of the caller will be displayed).

  Default: Number + Name.

- Select the **Display Language** for the SPARSH VP510. Default: English.

  ANANT UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP510. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- *SIP Extension users can change the language by accessing and navigating through the phone menu.*
- *The SA can change the Language by logging into the SA Jeeves.*

- Select a **Ringer Mode** for the phone from the four options:

  - Ring immediately (it rings immediately as a fresh calls lands on the phone).
  - Ring if idle (rings only if the phone is idle).
  - Ring after a delay (if the call is still not answered).
  - Silent.

  Default: Ring Immediate.

- If you selected *Ring after a delay* as Ringer Mode, set the **Ring Delay Timer (sec)**, if required, to the desired value.

  The Ring Delay Timer is the time in seconds the system waits on receiving a call before ringing on the phone. The range of this timer is 0 to 99 seconds. Default: 10 seconds.

- If you want to enable *Ringer Auto Acknowledge* mode, set the **Acknowledge Timer** (sec) to the desired value.

  The Ringer Auto Acknowledge mode determines when to stop the ring on the phone. There are two options for Ringer Auto Acknowledge:
  - Stop only when the call is answered.
  - Stop after a delay.

  To stop the ring on the phone after a delay, the Acknowledge Timer must be configured. The range of this timer is 00 to 99 seconds. Default: 00 seconds.

  To stop the ring only when the Call is answered or manually acknowledged, the Acknowledge Timer must be set to '00'. By default, Ring Auto Acknowledge is turned OFF.

- To assign the Ring Destination for the SPARSH VP510, select the desired destination for **Play Ring on**. You may choose
  - **Speakerphone:** The ring will be played on the Speakerphone.
  - **Headset:** The ring will be played on the Headset.

  Default: Speakerphone.

  When you select the Headset as the destination, make sure that you have selected the *Headset Connected?* check box and connected a Headset to the SPARSH VP510.

- Set the **Ringer Volume** to the desired level, from 0 to 7, according to your preference. Default: 4.

You can also set the Ringer Tune. For detailed instructions, refer to the EON510_SPARSH VP510 User Guide.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the handset of the SPARSH VP510, set the **Handset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of incoming speech (Receive Gain) on the handset of the SPARSH VP510, set the **Handset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Transmit Gain) on the headset of the SPARSH VP510, set the **Headset Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase/decrease the volume of outgoing speech (Receive Gain) on the headset of the SPARSH VP510, set the **Headset Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Transmit Gain of the Speakerphone MIC Volume, set **Speaker Transmit Volume Level** to the desired level, from 0 to 7. Default: 4.

- To change the Receive Gain of the Speakerphone MIC Volume, set **Speaker Receive Volume Level** to the desired level, from 0 to 7. Default: 4.

- To increase the volume of the incoming speech on the handset, select the **Handset High Gain Mode** check box. This is useful for individuals with hearing aids. Default: Disabled.

- To use a Headset with the SPARSH VP510, select the **Headset Connected?** check box**.** Default: Disabled.

  Make sure that you connect a Headset to the SPARSH VP510, if you enable this option.

- Select the **Auto Answer** check box to enable this feature on the SPARSH VP510. Default: Disabled.

  When you set the *"Auto Answer"* feature on the SPARSH VP510, the phone goes OFF-Hook automatically after a preset period of time, without the extension user having to pick up the handset or press the speaker or headset key. When you enable Auto Answer, you must configure the Auto Answer Timer.

- If you enabled Auto Answer on the phone, set the **Auto Answer Timer (sec)** to the desired value.

  This timer defines the time in seconds that the SPARSH VP510 should wait before going OFF-Hook to auto answer a call. The range of this timer is 1 to 9 seconds. Default: 1 second.

- Adjust the Backlight brightness of the phone's LCD display, by setting the **LCD Backlight Level** to the desired value, from 1 to 4. Default: 3.

- Set the **Back Light Off Timer (sec)** to the desired value, if required, from 000 to 999 seconds. Default: 10 seconds.

- Set the **LCD Contrast Level** to a level from 1 to 4 that is comfortable to you. Default: 3.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See *"Customizing Extended IP Phone Templates"* for more details.

  OR

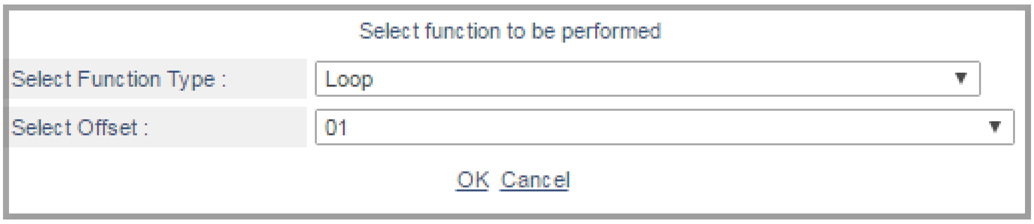
- You can personalize the key map of the SPARSH VP510 for this location. To do so,

- Select **Personalized** as the **Key Template** option.
- Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.

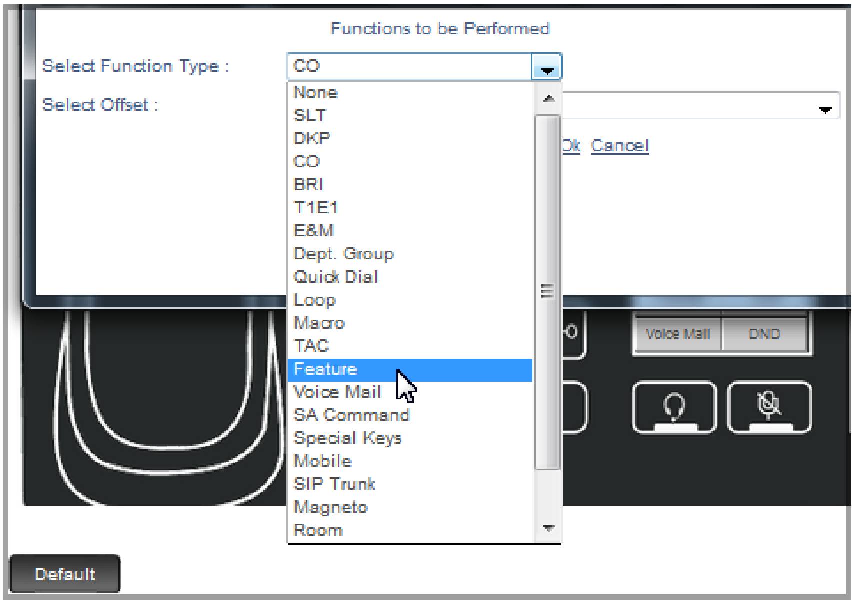
- Click the key you want to configure. For example, **CA 1**.

The **Functions to be Performed** by the key opens in a new window.



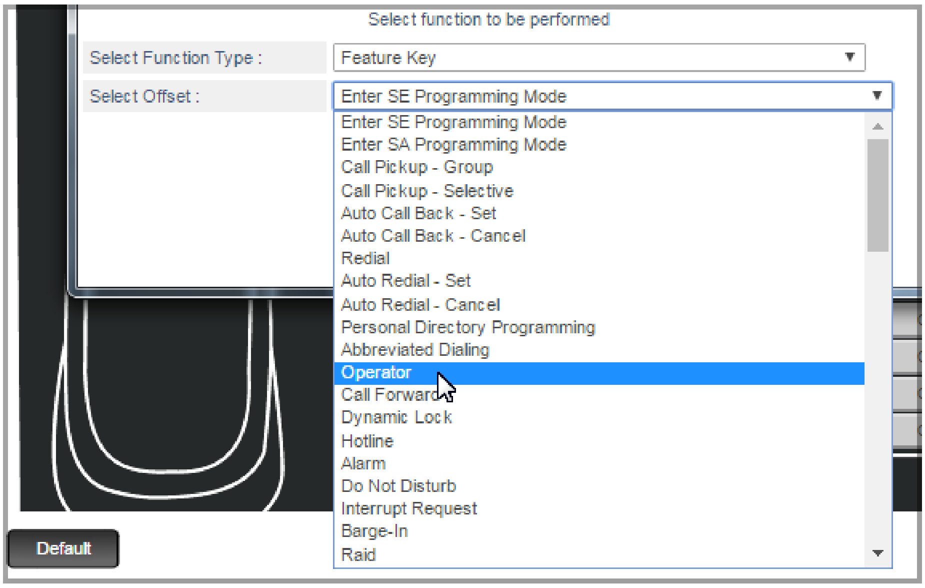
- In the **Select Function Type** list, select the function to be performed by the key. For example, you want to use the key to call the **Operator**.

  The Operator function is a Feature, so select the option **FEATURE** from the **Select Function Type** list box.

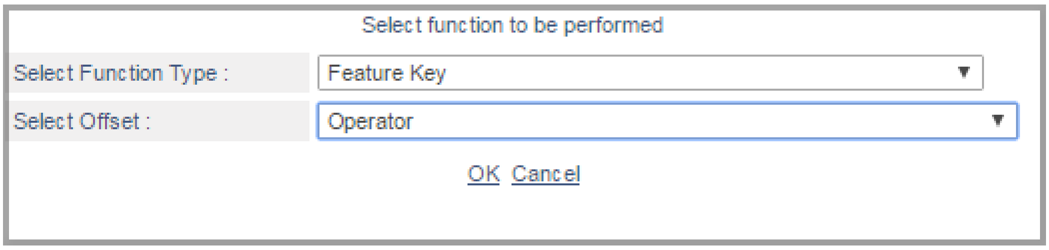


From the **Select Offset** drop down list, all the features that can be assigned to keys are listed.

- Select **Operator** from the list of features in the **Select Offset** box.

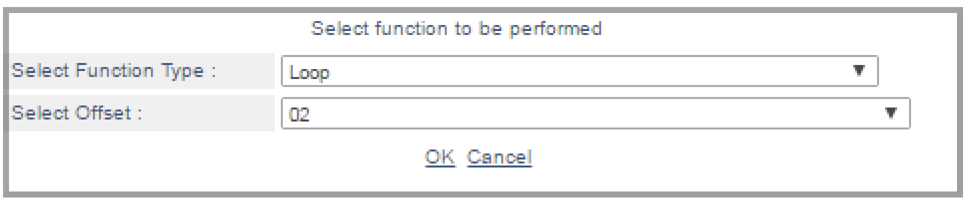


- Click **OK**.

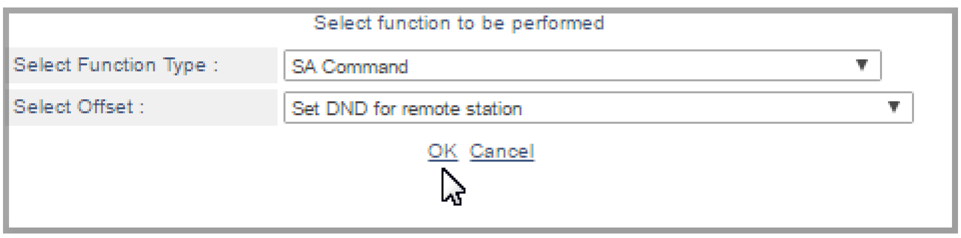


The *Operator* feature appears on the key label.

- To take a second example, if you want to assign **Remote DND** to the key currently assigned **CA 2** key, click the key.



- In the **Select Function Type** list box, select the option **SA Command**, as Remote DND is a System Administrator (SA) Command.



- In the **Select Offset** box, select the option **Set DND for remote station**.

- Click **OK**. The box closes. Remote DND feature will appear in abbreviated form as *R-DND* on the key label.

- Follow the same instructions to assign features to other DSS keys. Selecting the appropriate Function Type and the Offset for each feature/function.

  If you want assign a feature, select **FEATURE** as function type, and select the desired feature as Offset.

  If you want to use the key to call a SIP extension, select **SIP Extension** as Function Type and select the number of the extension as Offset.

  To assign direct access to a SIP Trunk, select **SIP** as Function Type and the desired trunk number from **01** or any other trunk number as Offset.

  Click **OK**, each time you select a Function Type and Offset in the dialog box.

  You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions to keys, close the window.

- If you assign/re-assign functions to the following keys, the Phone will restart:

- Speaker
- Headset
- Ringer Acknowledge
- Local Menu

- You can also connect a DSS Console (DSS532) with SPARSH VP510. For instructions:
  - to install the DSS532 with SPARSH VP510, see “Installing DSS532 with SPARSH VP510”.
  - to configure the DSS keys of the Console, see “Configuring DSS Console Keys connected to SPARSH VP510”.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

- *If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.*
- *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.*

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

If you select this check box, make sure you have selected SRTP Mode as Forced or Optional in the General Parameters under SIP Extension Settings.

RTP Port

- Define the RTP Port:
  - **RTP Listening Port:** This is the port on which the SPARSH VP510 listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:
  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP510 is connected behind a NAT router, configure **NAT Keep Alive**.
  - Select the **Enable NAT Keep Alive** check box to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.
  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:
  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.
  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.
  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP510, configure Debug parameters:
  - Select the **Enable Debug?** check box. Default: disabled.

    When Debug is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.
  - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

    The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).
  - You may select the **Debug Level** from the following options, by selecting the respective check box:
    - SIP
    - System
    - Hardware
    - Call
    - Network
    - VoPP

    You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network. For example, if the debug log of 'Call's is required, you can select this option, and disable all others.
- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP510 Settings at Location 1, follow the same steps as described above to configure the SPARSH VP510 at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP510 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will restart automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Call Progress Tone
- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- The SE Password of ANANT UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters in Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix SPARSH VP210

SPARSH VP210 is the proprietary Extended IP Phone by Matrix which is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price.To know the list of features supported, refer to *"ANANT UCS Features Supported in Terminals"*.

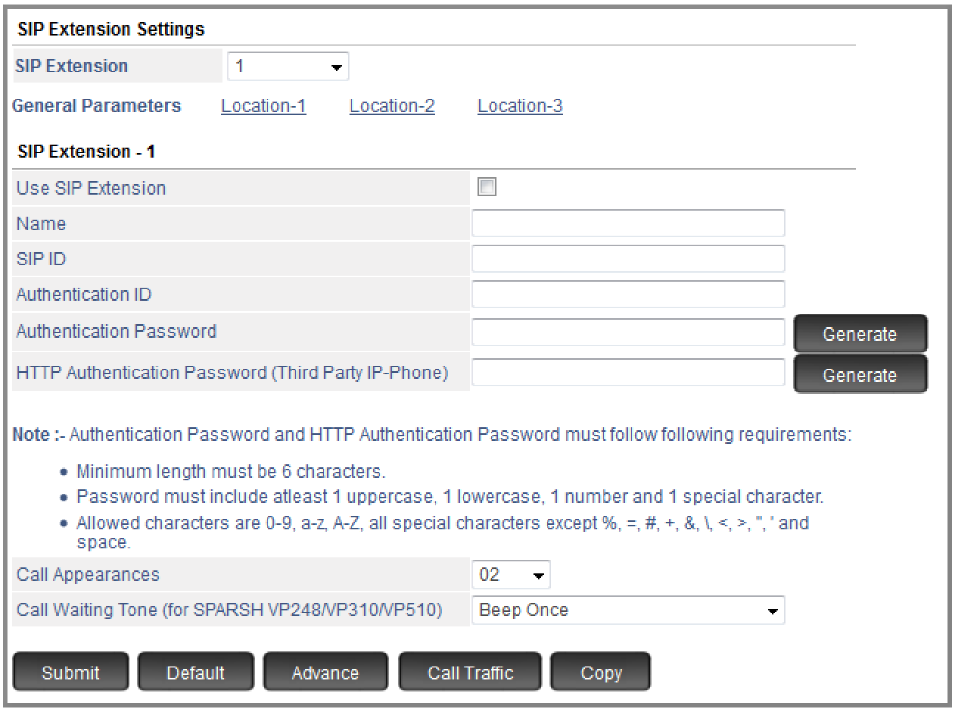
For detailed product information and operation instructions, refer to the *SPARSH VP210 (Extended) User Guide.*

To be able to use SPARSH VP210[28] as SIP Extensions, you must configure the following:

- SIP Extension General Parameters, see *"Configuring SIP Extension General Parameters"*.
- SIP Extension Settings, see *"Configuring SIP Extension Settings using Jeeves"*
- Extended IP Phone Settings, see *"Configuring Matrix Extended Phone Settings using Jeeves"*
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See *"Extension Voice Mail Settings".*

Configuring SIP Extension Settings using Jeeves

- Log into Jeeves as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

The parameters of the SIP Extension number you selected will appear on this page.

28. *ANANT UCS supports only IPv4 Addresses for registering SPARSH VP210.*

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one extension.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and “ (double quote) are allowed. Default: Blank.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When you enter the password manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) and digits 0 to 9 are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See *“Black List IP Address - SIP Extensions”* for more details. This activity will be logged in the *“System Activity Log”* as well as *“Simple Network Management Protocol (SNMP)”*.

- In **Call Appearances**, define the maximum number[29] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

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29. *For the calls that are routed through the CPU, the number of Vocoder channels that will be supported would be as per the license you purchase.*

Auto Sign-In parameters — Send Configuration Mail and Mail Status, *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

The ***Send Configuration Mail*** *button will appear only after you have enabled the SIP Extension and configured the SIP ID, Authentication ID and Password.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been programmed.

- For secure conversations over SIP, enable **SRTP Mode**. The ANANT UCS supports the following options:
  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.
  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.
  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: 01. The *"SIP Hardware Template"* contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and Fax-over-IP options and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,
  - Under **VoIP Configuration**, click the **SIP Hardware Template** link.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.

- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic *"SIP Hardware Template"* to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: The *"Station Basic Feature Template"* has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension (*"Class of Service (CoS)"*, *"Toll Control", "OG Trunk Bundle Group"* etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click the **Station Basic Feature Template** link.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic *"Station Basic Feature Template"* to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The *"Station Advanced Feature Template"* has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click the **Station Advanced Feature Template** link.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.

- Click **Submit** to save changes.

  Also see the topic *"Station Advanced Feature Template"* for instructions on customizing these templates and applying them on the extensions.

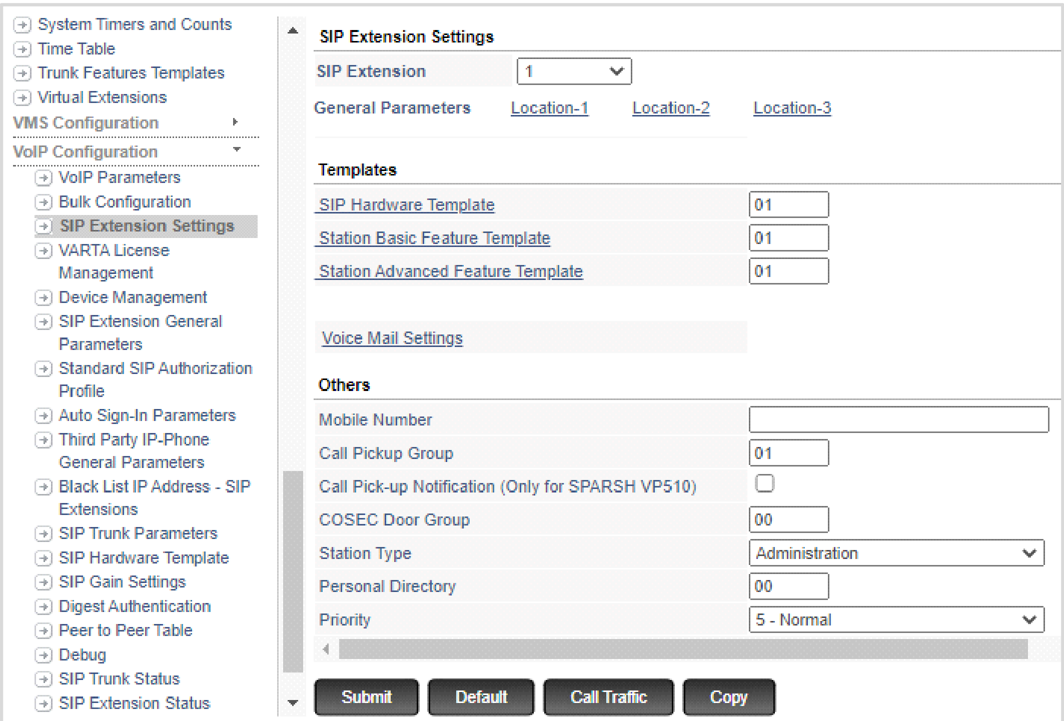
- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see *"Extension Voice Mail Settings"*.

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer *"Call Pick Up"* for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (CoS)". Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group. See *"COSEC Integration"* for more information.

- If this is an Operator extension and you want the system to play beeps during a conference to the participants, to indicate the presence or absence of the Operator, select the **Station Type** as **Assistant**.

  If you are using the system in the *Hotel Mode*, select the **Station Type** for the SIP Extension as **Administration/Assistant** or **Guest**. The system will consider the options Administrator and Assistant as same.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features *"Abbreviated Dialing"* and *"Dial By Name"*.

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be programmed by the SIP Extension users and by the System Engineer. Refer the topic *"Abbreviated Dialing"* for instructions on programming the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default; 5-Normal.

  Each extension of the ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description *"Priority"*.

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings using Jeeves

You can register three Matrix Extended Phones/Soft Clients at three different locations as a single SIP Extension.

You can connect the same or different types of Extended Phones/Soft Clients —SPARSH VP210, SPARSH VP248, SPARSH VP310, SPARSH VP510, SPARSH VP330,Extended SPARSH VP710, Matrix VARTA ADR100 Mobile UC Client, Matrix VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that SPARSH VP210 is connected at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to *"Configuring Matrix SPARSH VP248"* in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

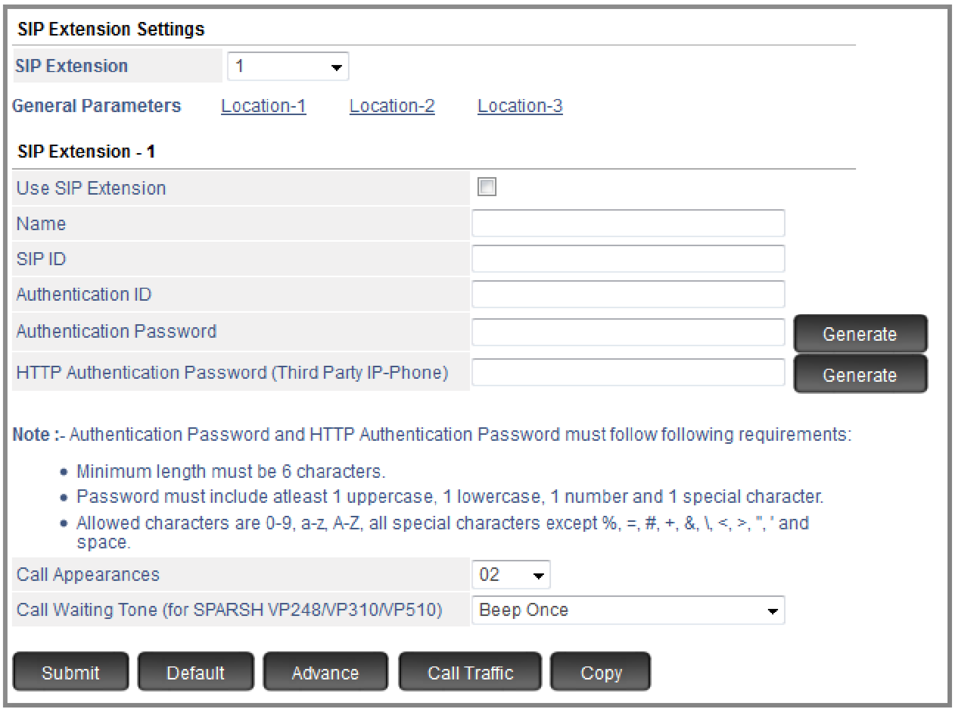
If you have connected Extended SPARSH VP710 at any of the locations, refer to *"Configuring Matrix Extended SPARSH VP710"*.

If you have registered Matrix VARTA Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.

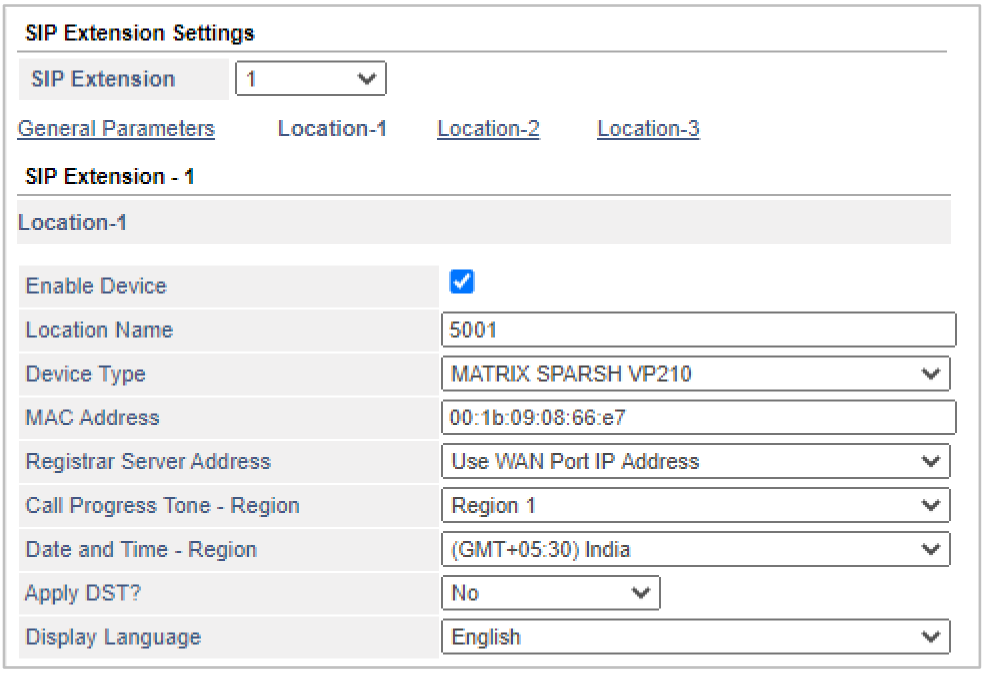
If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

If you want to use more than one SPARSH VP210 Phones as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP210** as the **Device Type** at this location.

- Enter the **MAC Address**[30] of the SPARSH VP210 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: blank.

  ANANT UCS validates the SPARSH VP210 on the basis of the MAC Address, and provides configuration on validation.

  As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the programmed MAC address.

- Select the appropriate **Registrar Server Address** to register the SPARSH VP210 with the SIP Registrar of ANANT UCS, according to your installation scenario:

  - If the SPARSH VP210 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the SPARSH VP210 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

30. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

- If the SPARSH VP210 is connected in the Global Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is programmed, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

  Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see *"Configuring Network Parameters"*.

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- To set the call progress tone generation standards of the country where the SPARSH VP210 is installed, select the **Call Progress Tone - Region**. Default: Region 1.

  See *"Call Progress Tones"* to know more.

- To display the Date and Time of the country where the SPARSH VP210 is installed, select the **Date and Time - Region**. Default: India.

- If you want to enable Daylight Saving Time (DST) on the phone, set **Apply DST?** to either **Manual** or **Scheduled** as per your requirement.

  When you select **Scheduled** as the DST option, the Real Time Clock of ANANT UCS is advanced and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  ANANT UCS supports 18 DST Types for Scheduled DST Adjustment. To know more, refer to "Daylight Saving Time (DST)". To know more about Scheduled DST assigned for the respective region, refer to *Time Zone* in "Default Settings".

  When you select **Manual** as the DST option, the Real Time Clock of ANANT UCS is advanced manually and set backward automatically according to the DST convention of the country/region where the IP Phone is installed.

  Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

  When DST option is set as 'Manual', you must set the DST Start and the DST End time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed. To do so,

  - In **Time Offset,** enter the time you wish to forward or backward the DST start time with.

  - In **DST Type**, select the desired option: **Date-Month Wise** OR **Day-Month Wise**.

    If you select '**Date-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

    - **Date**: Select the date on which DST begins (1-31).
    - **Month**: Select the month when DST begins (January-December).

- **Time (Hours)**: Select the time at which DST will begin to change.The time mode is of 24 hours ranging from 00 to 23 hours.
- **Time (Minutes):** Select the time at which DST will begin to change.The time mode is of 60 minutes ranging from 00 to 59 minutes.

Similarly, in the **DST End** configure the desired DST End Time details.

If you select **'Day-Month Wise'** in **DST Type**, you should now select the desired options in each of the following to specify the **DST Start** details.

- **Ordinal**: Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins.
- **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday- when DST begins.
- **Month**: Select the month when DST begins (January-December).
- **Time (Hours)**: Select the time at which DST must begin to change.The duration is of 24 hours ranging from 00 to 23.
- **Time (Minutes):** Select the time at which DST must begin to change.The duration is of 60 minutes ranging from 00 to 59.

Similarly, in **DST End** configure the DST End Time details.

Once the DST Ends, the time of the IP Phone is set back to the Standard time automatically.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- Select the **Display Language** for the SPARSH VP210. Default: English.

  ANANT UCS provides language support for English, French, German, Spanish, Portuguese, and Italian on the SPARSH VP210. When you select any of these languages, all the prompts and command strings will appear in the selected language.

SIP Extension users can change the language by accessing and navigating through the phone menu.

The SA can change the Language by logging into the SA Jeeves.

DSS Key Settings

- You can select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant, Guest or any other template you added. See "Customizing Extended IP Phone Templates" for more details.

  OR

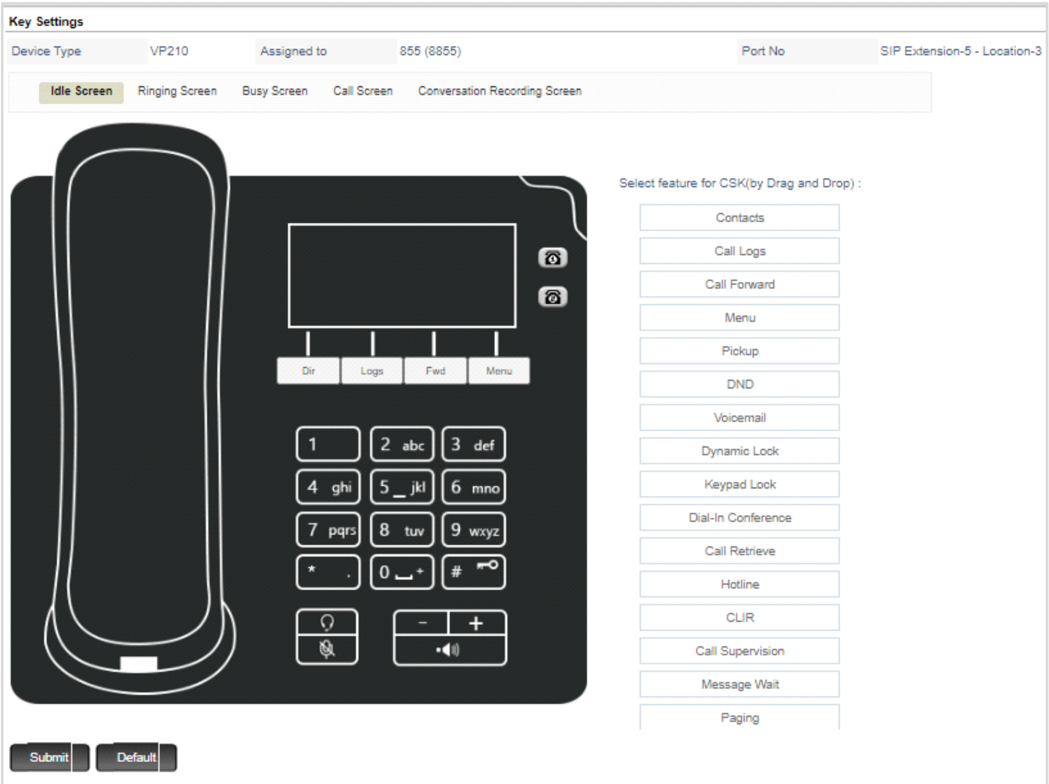
- You can personalize the key map of the SPARSH VP210 for this location. To do so,

  - Select **Personalized** as the **Key Template** option.

  - Click **Submit.**

- Click **Key Settings**.

- The key map of the Extended Phone opens in a new window on your screen.



- Click **Idle Screen**.
- Each Context key, 1 to 4 can be assigned features.
- The feature assignment cum priority list appears on the right. You can change the feature assignments/ priorities as per your preference.
- To set the priority, drag and drop the features in the order of your preference. This will have two implications — the Context Key will be assigned the desired feature as well it will set the priority.

- For example, if you wish to assign DND feature to Context Key 2, then drag and drop the DND feature at position 2 as below. Also make sure that the priority of Menu Key is kept as either of the four Context Keys.

- Click **Submit**.

- The key map will refresh and DND appears as Context Key 2.

Menu must be assigned to one of the first four Context Keys.

- Similarly, you can click **Ringing Screen, Busy Screen**, **Call Screen** or **Conversation Recording Screen** and can set the feature priorities as per your preference.

  To assign features/set feature priorities for other Context Keys, follow the same instructions. You can reinstate default key assignment any time, by clicking the **Default** button at the bottom of the window.

- When you complete assigning functions/priorities to the keys, close the window.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.
If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

RTP Port

- Define **RTP Port**:

  - **RTP Listening Port:** This is the port on which the phone listens for RTP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS**. Valid range is 00 to 63. Default: 26.
    OR
  - **RTP DiffServe/ToS**. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If the SPARSH VP330 is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

    The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

Debug

- To debug using Syslog Client supported by the SPARSH VP210, configure Debug parameters:

    - Select the **Enable Debug?** check box. Default: disabled.

      When the Debug flag is enabled, the phone will send the debug messages to the Syslog Server IP address. Debug report can be viewed on the Syslog Server or any other application which can capture the Syslog messages/debug sent by the phone.

    - Enter the IP Address and port of the remote Syslog Server and as **Syslog Server Address and Server Port**.

      The address of the Listening Port of the Syslog Server is from 1025-65535;514. Default: 514. Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

    - You may select the **Debug Level** from the following options, by selecting the respective check box:
        - Debug Level 1
        - Debug Level 2
        - Debug Level 3

      You may select any or all of these debug levels. The Syslog Client will send only the debug messages for the selected level to the remote server on the IP network.

- Click **Submit** to save settings.

- If you have completed the configuration of the SPARSH VP210 Phone Settings at Location 1, follow the same steps as described above to configure the SPARSH VP210 Phone at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP210 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:
- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address
- Enable Device
- Device Type
- Key Map in the Key Template assigned to phone
- Language
- Call Progress Tone

- Date and Time
- Apply DST?
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- Emergency Numbers

The SIP Extension registered at Location 1, 2, 3, will also restart, if:

- The SE Password of ANANT UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix Extended SPARSH VP710

Extended SPARSH VP710[31], the Smart Video IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. This IP Phone is an integration of SPARSH VP710, android based deskphone with VARTA ADR100 application. To know the list of features supported, refer to *"ANANT UCS Features Supported in Terminals"*.

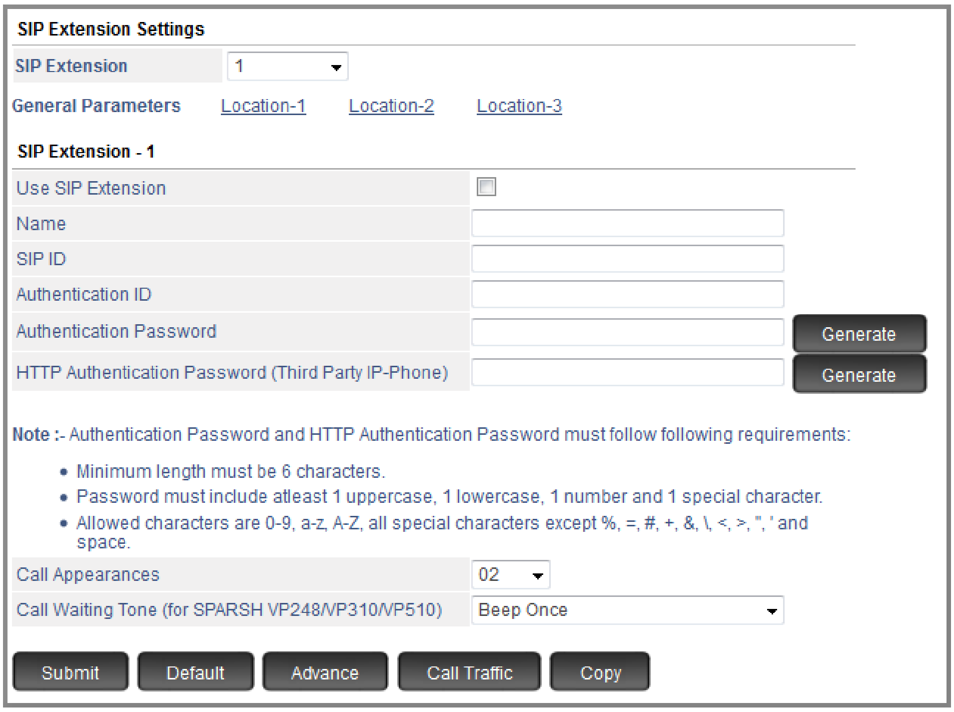
For instructions on how to use Extended SPARSH VP710, refer to the *EXTENDED SPARSH VP710 User Guide.*

To be able to use Extended SPARSH VP710[32], you must configure the following:

- SIP Extension General Parameters, see *"Configuring SIP Extension General Parameters"*.
- SIP Extension Settings, see *"Configuring SIP Extensions"*.
- Extended Phone Settings, see *"Configuring Matrix Extended Phone Settings"*.
- Voice Mail Settings, if you want to provide mailbox facility to the extension. See *"Extension Voice Mail Settings"*.

Configuring SIP Extension Settings

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 001 opens.

- You may select the **SIP Extension** number you want to configure.

31. *Check for availability.*
32. *ANANT UCS supports only IPv4 Addresses for registering Extended SPARSH VP710.*

The parameters of the SIP Extension number you select will appear on this page.

- Select the **Use SIP Extension** check box to enable the SIP extension. By default, it is disabled.
- In the **Name** field, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.
- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see *"Assigning Access Codes to a Range of Extensions"*.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.
- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See *"Black List IP Address - SIP Extensions"* for more details. This activity will be logged in the *"System Activity Log"* as well as *"Simple Network Management Protocol (SNMP)"*.

- In **Call Appearances**, define the maximum number[33] of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

***Auto Sign-In parameters — Send Configuration Mail** and **Mail Status,** are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:

  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.

  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.

    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The *"SIP Hardware Template"* contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks. Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click **SIP Hardware Template**.

33. *he calls that are routed through the system will depend on the number of transcoding channels available.*

- Select a Template number, for example 02.
- Customize Template number 02 and click **Submit**.
- Return to the **SIP Extension Settings** page.
- Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic *"SIP Hardware Template"* to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01. The *"Station Basic Feature Template"* has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different Station Basic Feature Templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension (*"Class of Service (CoS)"*, *"Toll Control"*, *"OG Trunk Bundle Group"*, etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Template**.
  - Select a Station Basic Feature Template number, for example 05.
  - Customize Station Basic Feature Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Station Basic Feature Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit** to save changes.

  Also, see the topic *"Station Basic Feature Template"* to know more about customizing the Station Basic Feature Templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The *"Station Advanced Feature Template"* has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click **Station Advanced Feature Template**.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit** to save changes.

  Also see the topic *"Station Advanced Feature Template"* for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see *“Extension Voice Mail Settings”*.

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required. Default: 01
  Call Pick Up allows the SIP Extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer *“Call Pick Up”* for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

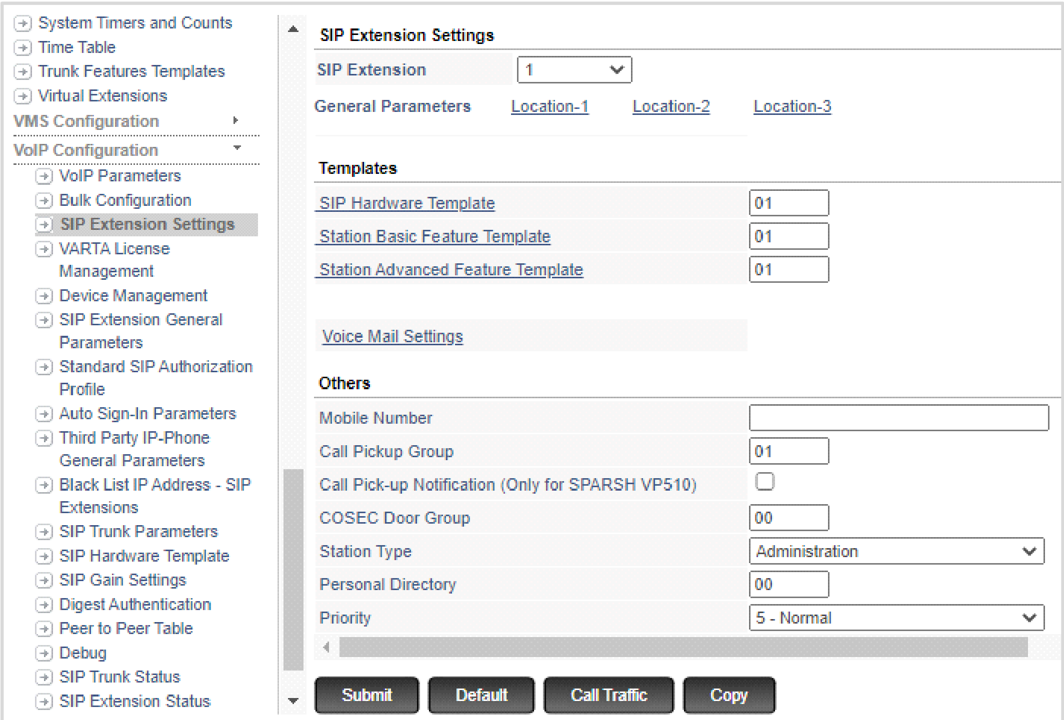
  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to “Call Pick Up” and “Class of Service (CoS)”. Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You must assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group. See *“COSEC Integration”* for more information.

  Click **Close** to close the window.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.** If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on configuring the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal.

  Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended IP Phones/Soft Clients at three different locations as a single SIP Extension. You can connect/register the same or different types of Extended Phones/Soft Clients —SPARSH VP248, SPARSH VP310, SPARSH VP330, SPARSH VP510, SPARSH VP210, Extended SPARSH VP710, VARTA ADR100 Mobile UC Client, VARTA AMP100 Mobile UC Client or VARTA WIN200 Desktop UC Client— at each location. In this case we assume that Extended SPARSH VP710 is connected at Location 1, 2 and 3.

If you want to use more than one Extended SPARSH VP710 IP Phone as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you have connected SPARSH VP248 at any of the locations, refer to *"Configuring Matrix SPARSH VP248"*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

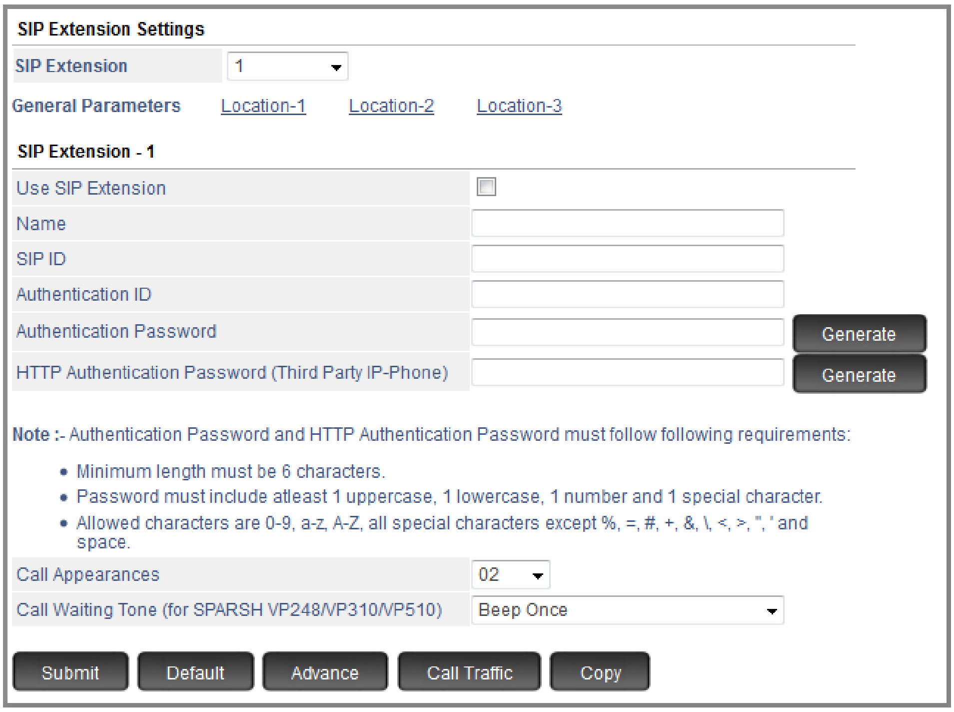
If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

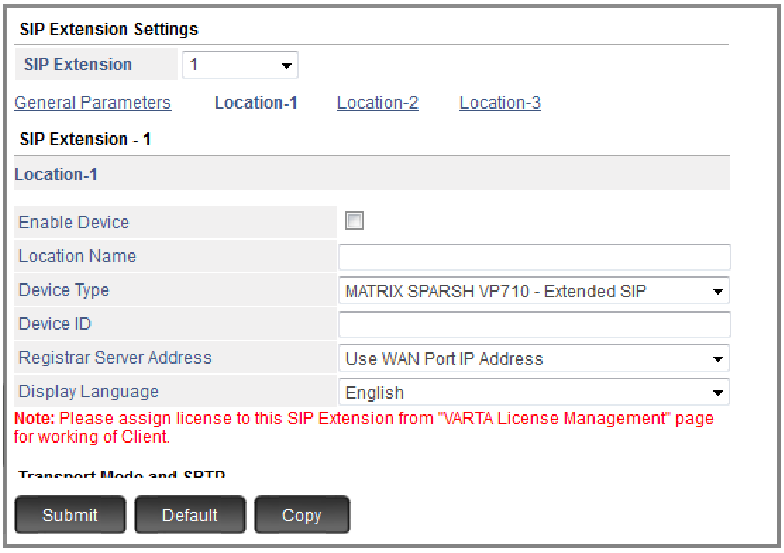
If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX SPARSH VP710 - Extended SIP** as the **Device Type** at this location.

- Enter the **MAC Address**[34] of the Extended SPARSH VP710 connected at this location in hexadecimal format: 00:1b:09:XX:XX:XX. Default: Blank.

  ANANT UCS validates the Extended Phone on the basis of the MAC Address and provides configuration on validation.

  As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the configured MAC address.

- Select the appropriate **Registrar Server Address** to register the Extended SPARSH VP710 with the SIP Registrar of ANANT UCS, according to your installation scenario:

  - If the Extended SPARSH VP710 is connected on the WAN network, select **Use WAN Port IP Address** as Registrar Server IP Address.

  - If the Extended SPARSH VP710 is connected on the LAN network, select **Use LAN Port IP Address** as Registrar Server IP Address.

  - If the Extended SPARSH VP710 is connected in the Global Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as Registrar Server IP Address.

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34. *MAC address is the address of the electronic hardware devices such as a computer, which is hard-coded into the device during manufacture and cannot be modified. No two devices can have similar MAC address and thus it uniquely identifies your phone. MAC address is assigned as per the IANA standard. The MAC Address of the phone will be used as source MAC address on all Ethernet frames.*

Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see *"Configuring Network Parameters"*.

- If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server IP Address.

  By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

- Select the **Language** for the Extended SPARSH VP710. Default: English.

  ANANT UCS provides language support for English, French, German, Spanish, Portuguese and Italian on the Extended SPARSH VP710. When you select any of these languages, all the prompts and command strings will appear in the selected language.

- *SIP Extension users can change the language by accessing and navigating through the phone menu.*
- *The SA can change the Language by logging into the SA Jeeves.*

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

- *If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.*
- *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.*

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

If you select this check box, make sure you have selected SRTP Mode as Forced or Optional in the General Parameters under SIP Extension Settings.

RTP Port

- Define the RTP Port:

  RTP Listening Port: This is the port on which the Extended SPARSH VP710 listens for SIP messages over UDP. This port is also used as the source port for sending RTP packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling as:

  - **SIP DiffServe/ToS.** Valid range is 00 to 63. Default: 26.
    OR
  - **Voice DiffServe/ToS**. Valid range is 00 to 63. Default: 46.
    OR
    Video DiffServe/ToS. Valid range is 00 to 63. Default: 46.

NAT Keep Alive

- If Extended SPARSH VP710 is connected behind a NAT router, configure **NAT Keep Alive**.

- Select the check box **Enable NAT Keep Alive** to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

- Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the phone should send Keep Alive message. Default: 120 seconds.

  The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after ending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

- Click **Submit** to save settings.

- If you have completed the configuration of the Extended SPARSH VP710 Settings at Location 1, follow the same steps as described above to configure the Extended SPARSH VP710 at Location 2 and Location 3.

  However, if you want to replicate the configuration of SPARSH VP710 Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to *“Copy Parameter Values”*.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Authentication ID
- Authentication Password
- Registrar Server IP Address
- MAC Address

- Enable Device
- Device Type
- Transport Mode and SRTP
- QoS
- RTP Ports
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- The SE Password of ANANT UCS is changed
- Specific parameters in VoIP Parameters are changed
- Specific parameters in Network Port parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix VARTA WIN200 UC Client

MATRIX VARTA WIN200, is a SIP (Session Initiation Protocol) based Unified Communication Desktop Client running on Windows OS, delivering full-array of the System features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise features of the System, UC Client provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using premium features like Presence subscription and notification, Corporate Voicemail access to enhance your overall mobile experience.

To use MATRIX VARTA WIN200 Desktop UC Client, make sure you have:

- Purchased and activated the VARTA Essential, VARTA Professional or VARTA Collaboration license. For more details, see *"Licenses Supported in ANANT UCS"* and *"License Management"*.

- Assigned the desired license to the SIP Extension. For more details, see *"VARTA License Management"*.

To know the list of featured supported, refer to *"ANANT UCS Features Supported in Terminals"*.

For detailed product information and operation instructions, refer to the *MATRIX VARTA WIN200 User Guide.*

Configuring MATRIX VARTA WIN200 using Jeeves

To be able to register and use the Desktop Client, you must configure the following:

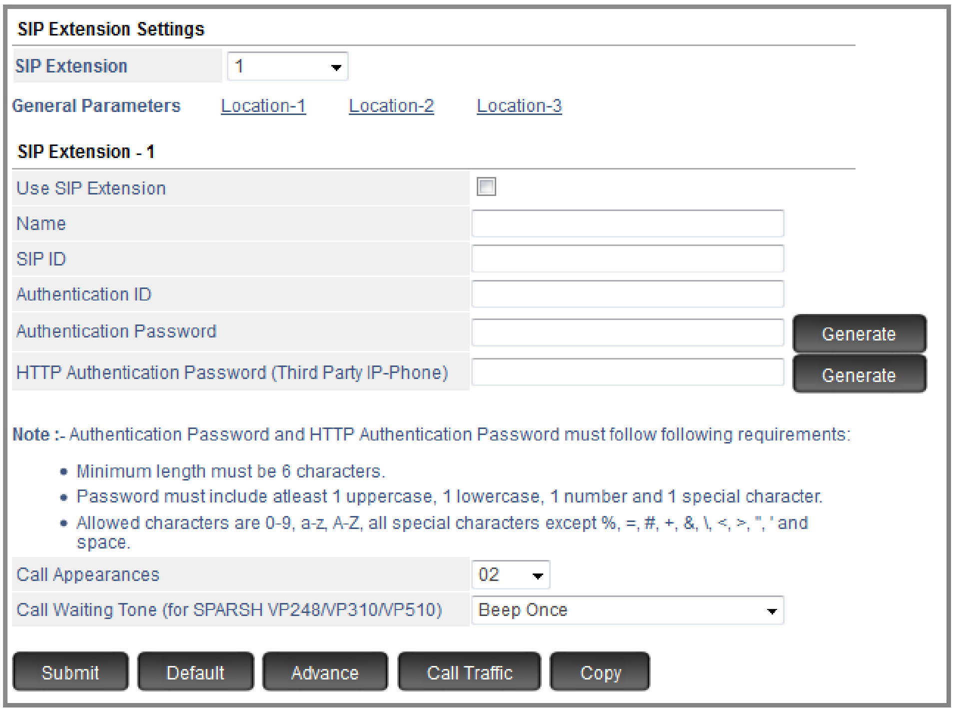
- SIP Extension General Parameters, see *"Configuring SIP Extension General Parameters"*.
- SIP Extension Settings, see *"Configuring SIP Extensions"*.
- Extended Phone Settings, see *"Configuring Matrix Extended Phone Settings"*.
- Voice Mail Settings, if you want to provide Voice Mail to the extension. See *"Extension Voice Mail Settings"*.

Configuring SIP Extension Settings

- Login as System Engineer.

- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

ANANT UCS supports IPv4 Addresses only for registering the Desktop UC Client.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.

- In **Name**, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see *"Assigning Access Codes to a Range of Extensions"*.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:
  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And &, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- In **Call Appearances**, define the maximum number of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail *and* ***Mail Status,*** *are applicable only for Mobile Clients — VARTA ADR100 and VARTA AMP100 applications.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - INVITE Request
  - SUBSCRIBE Request

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:

  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.

- **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.
- **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The "SIP Hardware Template" contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

    - Under **VoIP Configuration**, click **SIP Hardware Template**.
    - Select a Template number, for example 02.
    - Customize Template number 02 and click **Submit**.
    - Return to the **SIP Extension Settings** page.
    - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

  Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Station Basic Feature Template 01. The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (CoS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

    - Under **Configuration**, click **Station Basic Feature Template**.
    - Select a Template number, for example 05.
    - Customize Template number 05 and click **Submit**.
    - Return to the **SIP Extension Settings** page.
    - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.

- Click **Submit**.

  Also, see the topic “Station Basic Feature Template” to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The “Station Advanced Feature Template” has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click **Station Advanced Feature Template**.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit**.

  Also see the topic “Station Advanced Feature Template” for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by using the respective icon from Client GUI, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

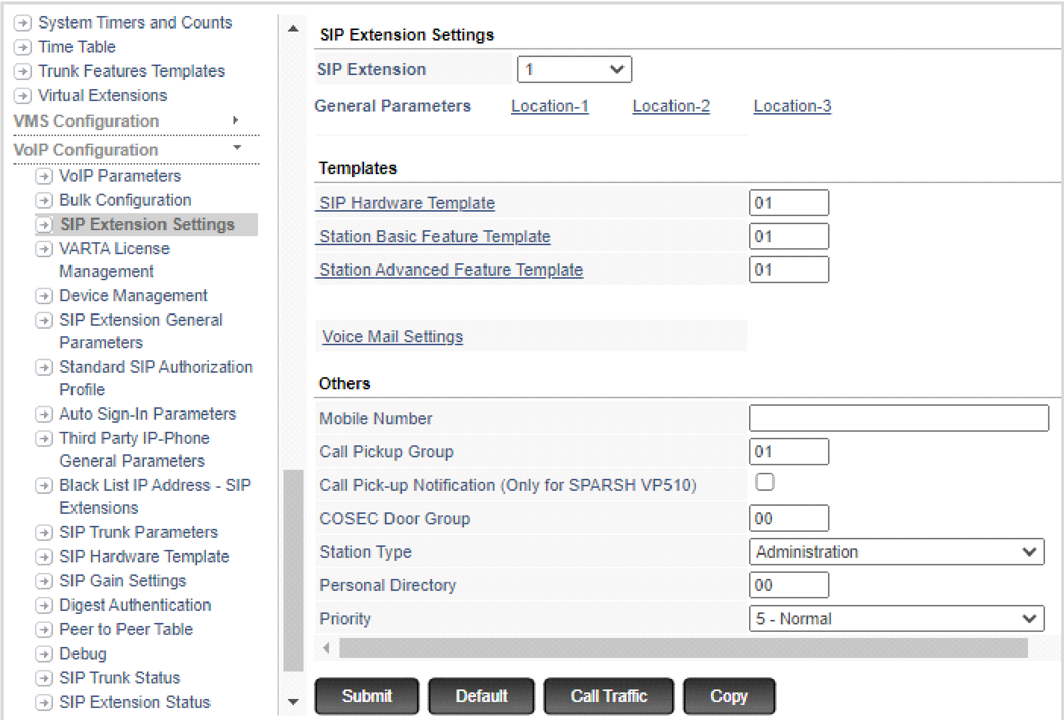
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to "Call Pick Up" and "Class of Service (CoS)". Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You can assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group '00' are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.**

  If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on configuring the Personal Directory.

If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal.

  Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended Phones/Soft Clients at three different locations as a single SIP Extension. You can connect the same or different types of Extended Phones/Soft Clients — SPARSH VP248, SPARSH VP330, SPARSH VP310, SPARSH VP510,SPARSH VP210, Matrix VARTA ADR100 Mobile UC Client, Matrix VARTA AMP100 Mobile UC Client or MATRIX VARTA WIN200 Desktop UC Client — at each location. In this case we assume that MATRIX VARTA WIN200 Desktop UC Client are registered at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to *"Configuring Matrix Extended Phone Settings"* in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

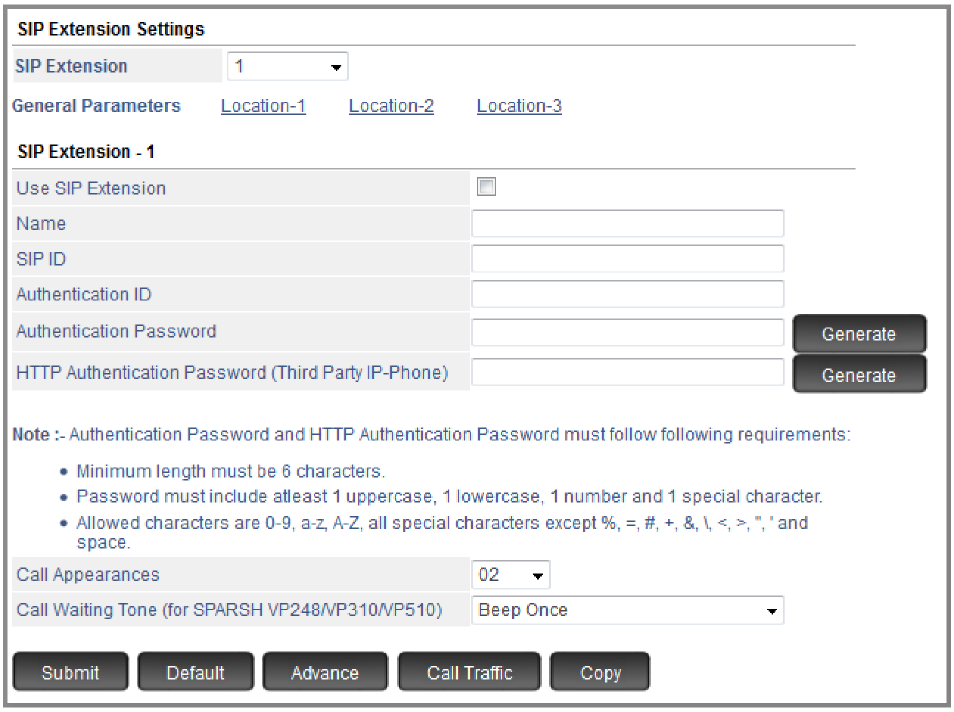
If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

If you have connected the Matrix Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710"*.

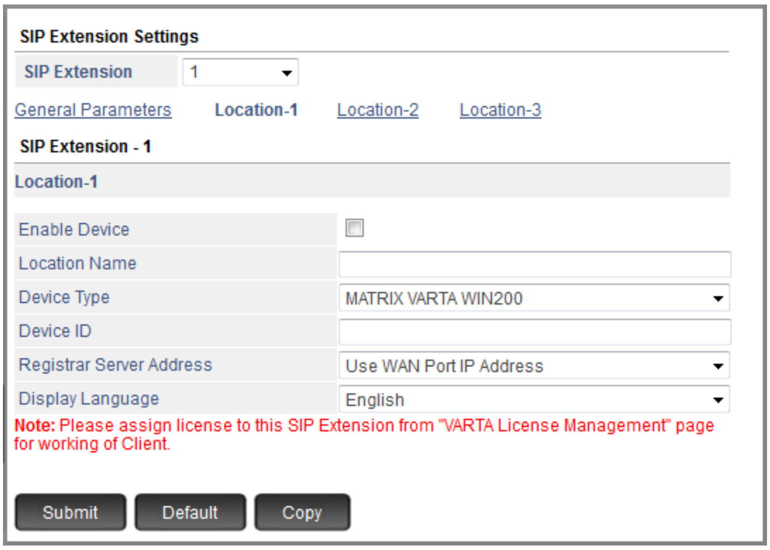
If you have registered Matrix VARTA ADR100 and VARTA AMP100 Mobile UC Clients in any of the locations, refer to *"Configuring Matrix VARTA ADR100/AMP100 UC Clients"*.
If you want to use more than one UC Clients as a SIP Extension, configure their settings as **Location 1**, **Location 2** and **Location 3**.

*If you want to use the IM functionality in the UC Client, you must configure it at **Location-1** only.*

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX VARTA WIN200** as the **Device Type** at this location. Make sure you assign the desired license to this SIP extension. For details, see "VARTA License Management".

- Enter the **Device ID** here. Default: blank.

ANANT UCS validates the desktop/PC on which you have installed the UC Client on the basis of the Device ID, and provides configuration on validation.

As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the configured Device ID.

- Select the appropriate **Registrar Server Address** to register the UC Client with the SIP Registrar of ANANT UCS, according to your installation scenario:
    - If you want the UC Client to register using the WAN network, select **Use WAN Port IP Address** as Registrar Server Address.
    - If you want the UC Client to register using the LAN network, select **Use LAN Port IP Address** as Registrar Server Address.
    - If the UC Client is registered in the Public (Global) Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as the Registrar Server Address.

      Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".
    - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Registrar Server Address.

      By default, Use WAN Port IP Address is selected as the Registrar Server IP Address.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

!

- *If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.*
- *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.*

- For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

  The UC Client supports RTP Relay. For detailed description, see "Configuring VoIP Parameters".

RTP Port

- Define the **RTP Port**. This is the port on which the client listens for RTP packets over UDP. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **SIP Quality of Service (QoS)** for SIP signaling:
    - **SIP DiffServe/ToS**: The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *SIP DiffServe/ToS* range is from 00-63, default: 26.

- **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. It also improves the voice quality.The Valid *Voice DiffServe/ToS* range is from 00-63, default: 46.

- **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. It also improves the video quality.The Valid *Video DiffServe/ToS* range is from 00-63, default: 46.

  Configure any decimal value as per your requirement from the table mentioned below:

Traffic Type	DSCP Value (dec)
Best Effort	0
Background	8
Excellent Effort	40
AudioVideo	40
Voice	56
Control	56

NAT Keep Alive

- If the UC Client is connected behind a NAT router, configure **NAT Keep Alive**.

  - Select the **Enable NAT Keep Alive** check box to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

  - Define as **Interval (sec)**, the time period, from 001 to 999 seconds, after which the client should send Keep Alive message. Default: 120 seconds. The time period you define should be less than the binding timer of the router.

Timers

- Set the following **Timers** to the desired value, where required:

  - **SIP INVITE Timer (sec):** This is the time in seconds that the client waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the client terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.

  - **SIP Provisional Timer (sec)**: This is the time in seconds that the client waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the client terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.

  - **General Request Timer (sec):** This is the time in seconds for which the client waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

- Click **Submit.**
- If you have completed the configuration of the UC Client at Location 1, follow the same steps as described above to configure the UC Client at Location 2 and Location 3.

  However, if you want to replicate the configuration of the UC Client same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Registrar Server Address
- MAC Address/IMEI/ESN Number
- Enable Device
- Device Type
- Transport Mode
- Enable SRTP
- QoS
- SIP/RTP Ports
- RTP Listening Port
- NAT Keep Alive
- SIP Timers
- Class of Service
- Trunk Access Code
- The SE Password of ANANT UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

Configuring Matrix VARTA ADR100/AMP100 UC Clients

Matrix offers the following Mobile UC Clients[35]:

- Matrix VARTA ADR100 for Android Smartphones/Tablets
- MATRIX VARTA AMP100 for iPhones

To use MATRIX VARTA UC Clients for Mobile, make sure you have:

- Purchased and activated the VARTA Essential, VARTA Professional or VARTA Collaboration license. For more details, see *"Licenses Supported in ANANT UCS"* and *"License Management"*.

- Assigned the desired license to the SIP Extension. For more details, see *"VARTA License Management"*.

To know the list of features supported, refer to *"ANANT UCS Features Supported in Terminals"*.

For VARTA Mobile Clients the system supports,

- automatic configuration and registration though Auto-Sign-In Configuration Mail. For this you must:

  - Configure the **Auto Sign-In Parameters**. For details, refer to *"Configuring Auto Sign-In Parameters"*.
  - Configure the **General Parameters** in **SIP Extensions Settings**. For details, refer to *"Configuring SIP Extension Settings"*.
  - Make sure you send the **Auto Sign-In Configuration Mail**. For details, refer *"Configuring SIP Extension Settings"*.

    You can also view the status of Auto Sign-In Email in *"Viewing SIP Extension Status"*.

- manual configuration and registration, follow the instructions in *"Configuring Mobile UC Clients using Jeeves"*.

MATRIX VARTA ADR100

MATRIX VARTA ADR100 is a proprietary SIP (Session Initiation Protocol) based UC Client Application running on Android Phones/Tablets, delivering full-array of Matrix ANANT UCS features to the user on-the-go along with an added advantage of UC features. Through tight integration with the enterprise mobility features of the ANANT UCS, VARTA ADR100 provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using premium features like Video Calling, IM, IM to SMS, Presence notification and corporate Voicemail System access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or traveling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhances the speed of communication and collaboration with office users and customers.

For detailed product information and operation instructions, refer to the *MATRIX VARTA ADR100 User Guide.*

35. *ANANT UCS supports only IPv4 Addresses for registering Mobile UC Clients.*

MATRIX VARTA AMP100

MATRIX VARTA AMP100 is a proprietary SIP (Session Initiation Protocol) based UC Client Application running on iPhones, delivering full-array of Matrix ANANT UCS features to the user on-the-go along with an added advantage of UC features. Through tight integration with the enterprise mobility features of the ANANT UCS, VARTA AMP100 provides advanced call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these, you can take the advantage of using premium features like Video Calling, IM, IM to SMS, Presence notification/subscription and corporate Voicemail access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or traveling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhances the speed of communication and collaboration with office users and customers.

For detailed product information and operation instructions, refer to the *MATRIX VARTA AMP100 User Guide.*

Configuring Mobile UC Clients using Jeeves

To be able to register and use the Mobile Handsets/Tablets as Matrix VARTA Mobile UC Clients, you must configure the following:

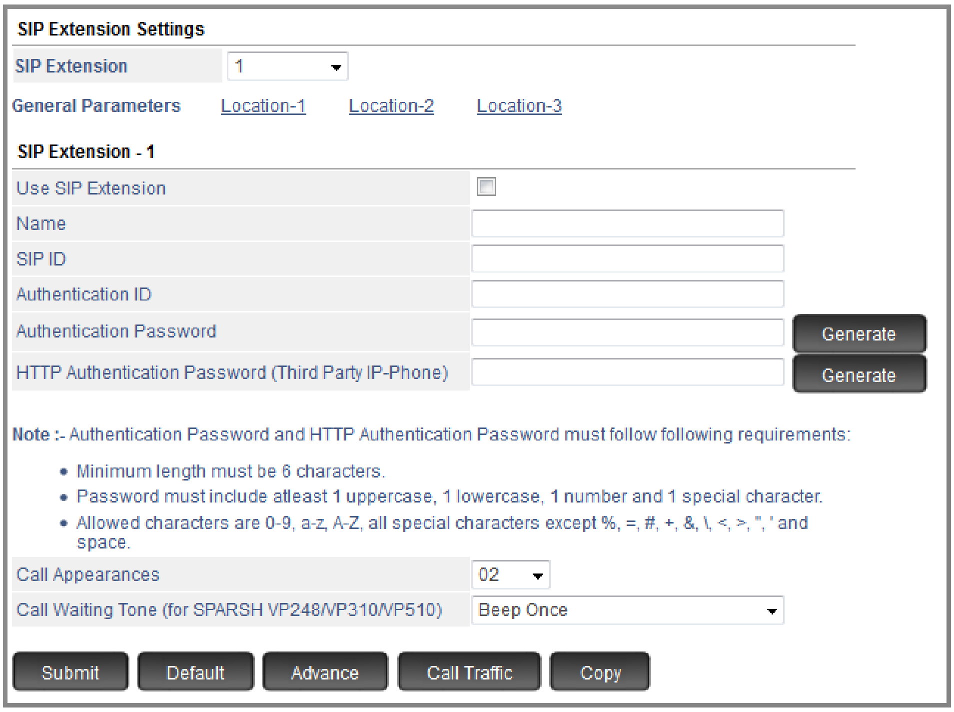
- SIP Extension General Parameters, see *“Configuring SIP Extension General Parameters”*.
- SIP Extension Settings, see *“Configuring SIP Extension Settings”*
- Extended Phone Settings, see *“Configuring Matrix Extended Phone Settings”*
- Voice Mail Settings, if you want to provide Voice Mail to the extension. See *“Extension Voice Mail Settings”*.

Configuring SIP Extension Settings

- Login as System Engineer.

- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

- You may select the **SIP Extension** number you want to configure.
- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.
- In **Name**, enter a name of SIP Extension, which may be the name of the person who will use the UC Client/ SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. The SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.

  You cannot assign the same SIP ID to more than one SIP extension.

  To assign SIP IDs according to your preference and requirement to a range of SIP Extensions, see “Assigning Access Codes to a Range of Extensions”.

  By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension.

However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- In **Call Appearances**, define the maximum number of simultaneous incoming calls that the SIP Extension user should be allowed to receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

- If you want the Mobile Clients to automatically configure and register with the Server, click the **Send** button adjacent to **Send Configuration Mail**.

  The Auto Sign-In Configuration Mail will be sent to the VARTA user on the **Email ID** configured above.

  The **Mail Status** will display either sent, failed or sending.

- *The parameters **Send Configuration Mail** and **Mail Status** will appear only after you have enabled/ configured (SIP Extension, SIP ID, Authentication ID and Password) and clicked the submit button.*

- *Make sure the Auto Sign-In parameters have been configured. For details, refer to "Auto Sign-In Parameters".*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - INVITE Request
  - SUBSCRIBE Request

By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- To allow the SIP Extension to monitor the status of another extension or Trunk, select the **Busy Lamp Field Subscription** check box. Default: disabled. See “Busy Lamp Field for Trunks” to know more.

When extension's state is changed from Ringing (early state as defined in BLF) to Mature (confirm state, as defined in BLF), because of implementation of ANANT UCS, it will send 'Terminate' state while moving from ringing to mature state. The interpretation of terminate message will vary from terminal to terminal.

- To allow the SIP Extension to view the status of other SIP-enabled Terminals, whether they are available or not, select the **Presence Subscription** check box. Default: disabled.

The SIP Extension, for which you have enabled Presence Subscription, will be able to view Presence of only those SIP Extensions which have PUBLISH enabled.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:
  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.
  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.
  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.

- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,

  - Under **VoIP Configuration**, click **SIP Hardware Template**.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Templates** to the SIP Extension. Default: Station Basic Feature Template 01. The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different Station Basic Feature Templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (CoS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this Station Basic Feature Template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Templates**.
  - Select a Template number, for example 05.
  - Customize Station Basic Feature Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Station Basic Feature Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit**.

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template**.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click **Station Advanced Feature Template**.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit**.

  Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

  Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by using the respective icon from Client GUI, without physically going to the ringing extension. It also allow incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

  For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

  Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

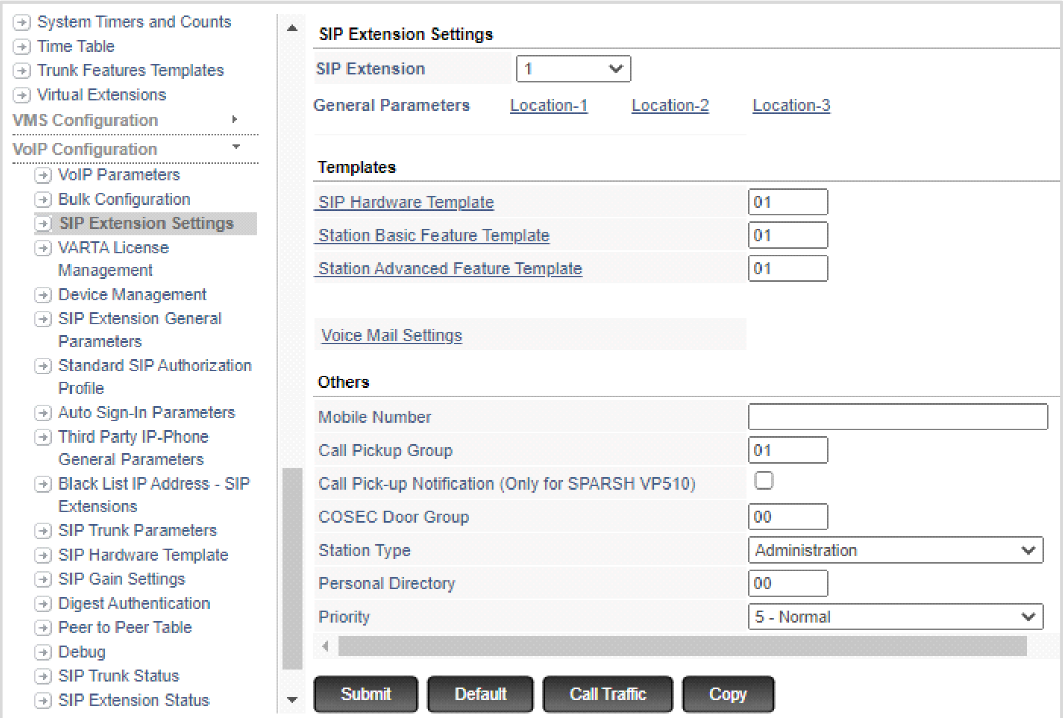
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to “Call Pick Up” and “Class of Service (CoS)”. Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You can assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.

- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type.**

  If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.

- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on configuring the Personal Directory.
  If Personal Directory is not to be assigned, enter 00 in this field.

- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal.

  Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.

- Click **Submit** to save your SIP Extension Settings.

Configuring Matrix Extended Phone Settings

You can register three Matrix Extended Phones at three different locations as a single SIP Extension. You can connect the same or different types of Extended Phones—SPARSH VP248, SPARSH VP330, SPARSH VP310, SPARSH VP510, SPARSH VP210, Extended SPARSH VP710 or Matrix VARTA UC Clients — at each location. In this case we assume that Matrix VARTA Mobile UC Clients are registered at Location 1, 2 and 3.

If you have connected SPARSH VP248 refer to *"Configuring Matrix Extended Phone Settings"* in *Configuring Matrix SPARSH VP248 as SIP Extensions*.

If you have connected SPARSH VP330 at any of the locations, refer to *"Configuring Matrix SPARSH VP330"*.

If you have connected SPARSH VP310 at any of the locations, refer to *"Configuring Matrix SPARSH VP310"*.

If you have connected SPARSH VP510 at any of the locations, refer to *"Configuring Matrix SPARSH VP510"*.

If you have connected SPARSH VP210 at any of the locations, refer to *"Configuring Matrix SPARSH VP210"*.

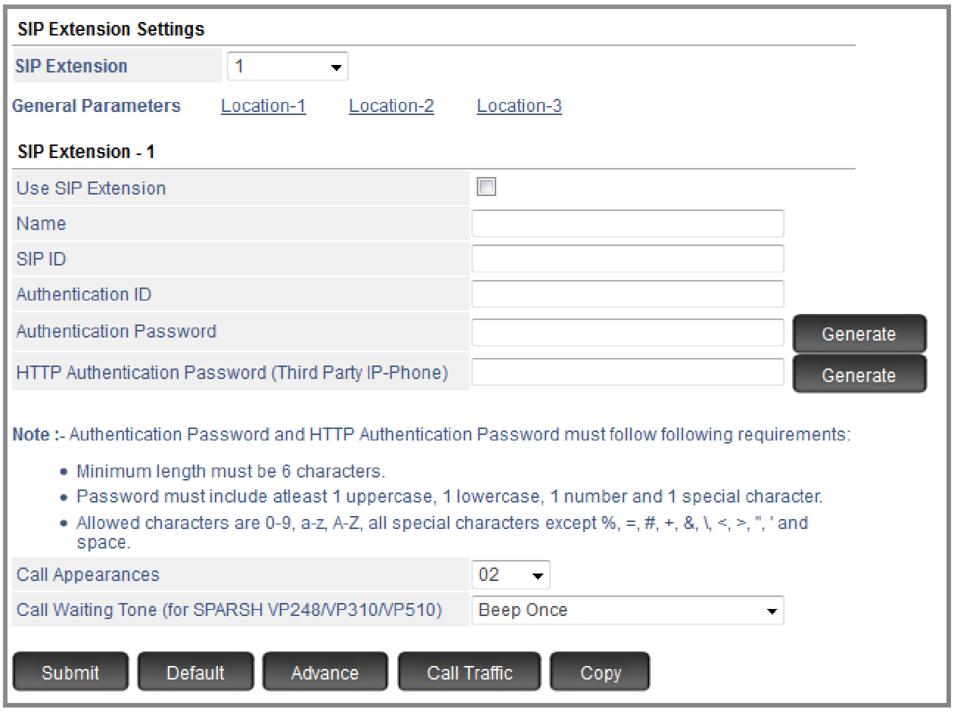
If you have connected the Extended SPARSH VP710 as SIP Extensions, for configuration instructions see *"Configuring Matrix Extended SPARSH VP710"*.

If you have registered MATRIX VARTA WIN200 Desktop UC Client in any of the locations, refer to *"Configuring Matrix VARTA WIN200 UC Client"*.

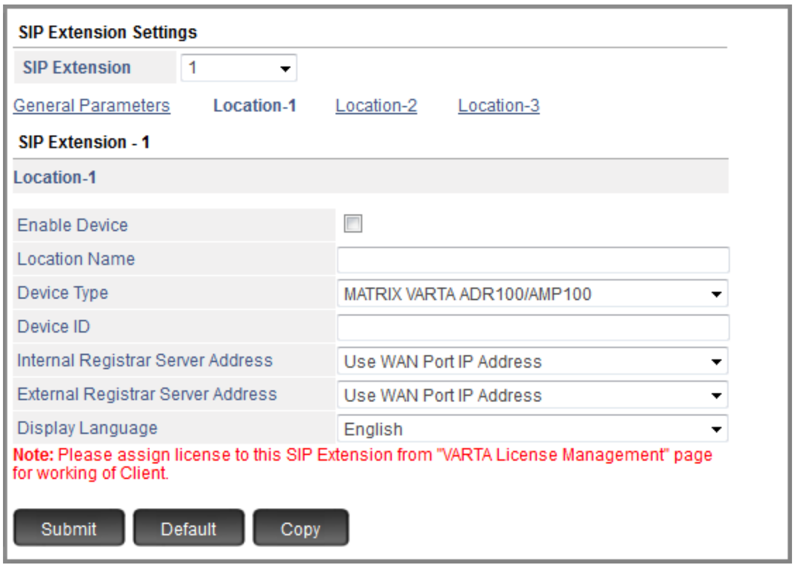
If you want to use more than one Matrix UC Clients as a SIP Extension, configure their settings at **Location 1**, **Location 2** and **Location 3**.

If you want to use the IM functionality in the MATRIX VARTA ADR100/AMP100, you must configure it at ***Location-1*** *only.*

- Click **Location 1**.



- The settings of the phone at **Location 1** appear.



- Select the **Enable Device** check box. Default: Disabled.

- Enter the **Location Name** for the phone to identify the phone. Location name may be the place where the phone is located (e.g.: Head office, branch, residence). The Location Name may consist of 18 characters (maximum). Default: Blank.

- Select **MATRIX VARTA ADR100/AMP100** (for the Android/iPhone Application) as the **Device Type** at this location. Make sure you assign the desired license to this SIP extension. For details, see "VARTA License Management".

- Enter the **MAC Address/IMEI[36]/ESN Number** of the phone/tablet on which you have installed the application.

  If you are using an iPhone, enter the **Device ID** here. Default: blank.

  ANANT UCS validates the phone/tablet on which you have installed the application on the basis of the IMEI/ESN Number or Device ID, and provides configuration on validation.

  As ANANT UCS allows registration of the SIP Extension from three different locations, it identifies the SIP Extension in each location by the configured IMEI/ESN Number/Device ID.

- Select the appropriate **Internal Registrar Server Address** to register the application with the SIP Registrar of ANANT UCS within a private network. Select the appropriate option as per your installation scenario:

  - If you want the application to register using the WAN network, select **Use WAN Port IP Address** as the Internal Registrar Server Address.

  - If you want the application to register using the LAN network, select **Use LAN Port IP Address** as Registrar Server Address.

  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as Internal Registrar Server Address.

    By default, Use WAN Port IP Address is selected as the Internal Registrar Server Address.

- Select the appropriate **External Registrar Server Address** to register the application with the SIP Registrar of ANANT UCS from a public network. Select the option according to your installation scenario:

  - If you want the application to register using the WAN network, select **Use WAN Port IP Address** as External Registrar Server Address.

  - If the application is connected in the Public Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Use Router/STUN's IP Address** as External Registrar Server Address.

    Make sure you configure either the **Router's Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see "Configuring Network Parameters".

  - If Dynamic DNS is configured in the Network Parameters, select **Use Dynamic DNS Host Name** as External Registrar Server Address.

    By default, Use WAN Port IP Address is selected as the External Registrar Server Address.

Transport Mode and SRTP

- Select the protocol to be used to transport the SIP messages. You can select the **Transport Mode** as **TCP** or **TLS**.

- *If you select TCP, make sure the SIP Over TCP check box is selected in VoIP Parameters.*
- *If you select TLS, make sure the SIP Over TLS check box is selected in VoIP Parameters.*

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36. *IMEI Number is the unique identification number of the GSM engine used in the Mobile handset.*

For secure conversations over SIP, select the **Enable SRTP?** check box. The SIP messages will be transported over SRTP only.

The application supports RTP Relay. For detailed description, see “Configuring VoIP Parameters”.

RTP Port

- Define **RTP Port**. This is the port on which the phone listens for RTP packets over UDP. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025-65278. Default: 8000.

Quality of Service

- Set the **Quality of Service (QoS)** for SIP signaling:
  - **SIP DiffServe/ToS:** The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *DiffServe* range is from 00-63, default: 26.
  - **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. It also improves the voice quality.The Valid *DiffServe* range is from 00-63, default: 46.
  - **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. It also improves the video quality.The Valid *DiffServe* range is from 00-63, default: 46.

  Configure any decimal value as per your requirement from the table mentioned below:

DSCP <=> IP Precedence Conversion Table			
DSCP Name	**DS Field Binary**	**Value Decimal**	**IP Precedence**
CS_0	000 000	0	0
CS_1	001 000	8	1
AF_{11}	001 010	10	1
AF_{12}	001 100	12	1
AF_{13}	001 110	14	1
CS_2	010 000	16	2
AF_{21}	010 010	18	2
AF_{22}	010 100	20	2
AF_{23}	010 110	22	2
CS_3	011 000	24	2
AF_{31}	011 010	26	3
AF_{32}	011 100	28	3
AF_{33}	011 110	30	3
CS_4	100 000	32	4

DSCP <=> IP Precedence Conversion Table			
DSCP Name	**DS Field Binary**	**Value Decimal**	**IP Precedence**
AF_{41}	100 010	34	4
AF_{42}	100 100	36	4
AF_{43}	100 110	38	4
CS_5	101 000	40	5
EF	101 110	46	5
CS_6	110 000	48	6
CS_7	111 000	56	7
CS Class Selector (RFC_{2474})			
AFxy Assure Forwarding (x=class, y=drop precedence) (RFC_{2597})			
EF Expedited Forwarding (RFC_{3246})			

Timers

- Set the following **Timers** to the desired value, where required:
  - **SIP INVITE Timer (sec):** This is the time in seconds that the phone waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the phone terminates the call process and gives an error tone to the user. The range of the SIP INVITE TIMER is 10-180 seconds. Default: 30 seconds.
  - **SIP Provisional Timer (sec)**: This is the time in seconds that the phone waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, the IP phone terminates the call process and gives error tone to the user. The range of SIP Provisional Timer is 10-180 seconds. Default: 60 seconds.
  - **General Request Timer (sec):** This is the time in seconds for which the phone waits for response of a transaction request. This timer starts on initiating a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the phone clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.
- Click **Submit.**
- If you have completed the configuration of the UC Client at Location 1, follow the same steps as described above to configure the application at Location 2 and Location 3.

  However, if you want to replicate the configuration of the UC Client same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to “Copy Parameter Values”.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

When you change any of the parameters listed below in the SIP Extension at Location 1, 2, 3, the phone will go in Auto Configuration mode automatically, if registered:

- Use SIP Extension
- SIP ID
- Name
- Authentication ID
- Authentication Password
- Internal Registrar Server Address
- External Registrar Server Address
- MAC Address/IMEI/ESN Number
- Enable Matrix Extended Phone Mode
- Extended Phone Type
- Transport Mode
- Enable SRTP
- QoS
- SIP/RTP Ports
- SIP Timers
- Class of Service
- Trunk Access Code
- The SE Password of ANANT UCS is changed
- Specific parameters in Network Port parameters are changed
- Specific parameters in VoIP Parameters are changed
- You restart the System
- Set the System to Default

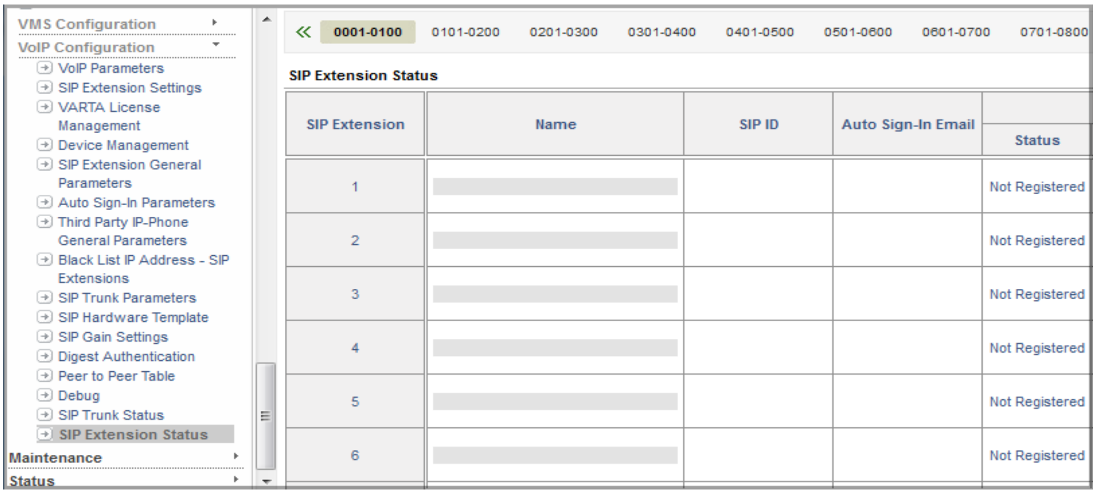
Viewing SIP Extension Status

You can view the Status of SIP Extension using Jeeves. To do this,

- Login as System Engineer.

- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Extension Status**.



- The SIP Extension Status page will open and display the following for each SIP Extension,
  - SIP Extension number
  - Name of the SIP extension
  - SIP ID assigned to the SIP Extension
  - Auto Sign-In Email; whether the mail is sent, failed or sending.
  - REGISTRATION status; whether the SIP Extension is registered or not.
  - Contact 1 (for *MATRIX VARTA AMP100/ADR100))*

    When the device is Registered - It will display the SIP ID, IP Address and Port and the Registration Expiry Timer. If there is a router in between it also displays the IP Address of the Network layer.

    When the device is in the Background - It will display Active and the time remaining for the expiry of the VARTA Client Inactivity Timer

    When Unregistered - The existing details will be cleared and it will be blank.

    To know more, refer “Apple Push Notification Service Support” or “Firebase Cloud Messaging (FCM) Support”.
  - Contact 2 - same as above.
  - Contact 3 - same as above.

*You can also view the SIP Extension Status from the **Status** link. To view, click the SIP Extension Status under Status.*

Configuring SIP Extensions using Bulk Configuration

In today's competitive world, time is an important asset for all the modern organizations. So, most of the organizations demand for a system that accomplish most of its functionality at a click of a button. As, we know, ANANT UCS supports 5000 Extensions and configuring these extensions one by one may be tedious and time consuming. So, to overcome this concern of the organizations, ANANT UCS supports Bulk configuration.

Bulk configuration allows you to configure a large number of SIP Extensions by simply uploading a CSV File in the Server. Once the CSV File is uploaded, the extensions listed in the file are configured in the server.

You can also configure extensions one-by-one, to know more, refer to "Configuring SIP Extension Settings as per the Extended Phone Type".

How to works

- Firstly, you need to download the CSV Generator Template from Jeeves. To know more, refer to "Download ing the CSV Generator Template".

- Secondly, customize the CSV Generator Template as per your requirement. To know how to customize the template, refer to "Preparing the CSV Generator Template".

- After you have prepared the Template, you need to generate the CSV File. To know more, refer to "Generating CSV File". After generating the file, you must export the file. To know how to export the file, refer to "Exporting the CSV File".

- Once the CSV file is exported, now you need to upload the file. To know how to upload the CSV File, refer to "Uploading the CSV File". The Extensions listed in the CSV File are configured in the Server.

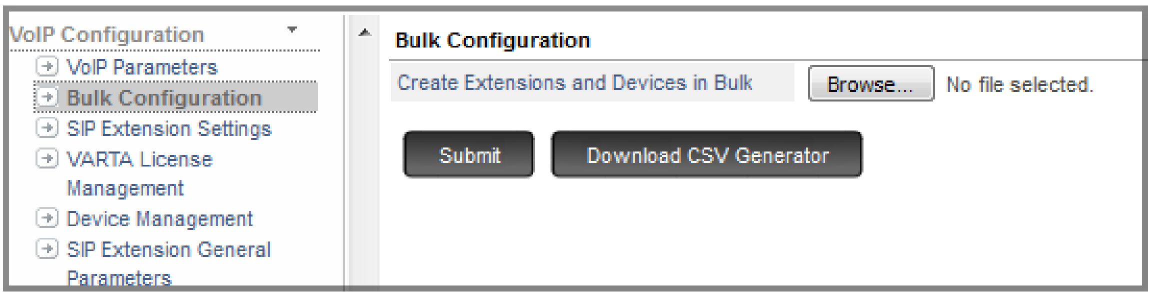
Before configuring the extensions using Bulk Configuration, make sure Microsoft Excel is installed on your PC.

How to configure

Downloading the CSV Generator Template

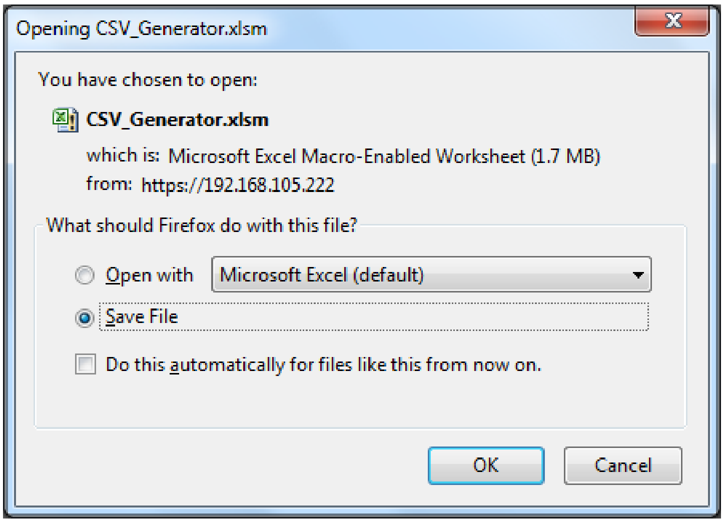
- Login as System Engineer.

- Under **VoIP Configuration**, click **Bulk configuration.**



- Click **Download CSV Generator** button to download the CSV Generator Template.

- The CSV_Generator.xlsm window opens.



- You can either open the CSV Generator Template or save the Template to a desired location on the local disk.

- *The above window display depends upon the browser you are using. Check the Download Settings of your browser and set the Download path accordingly.*
  OR
- *If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.*

Preparing the CSV Generator Template

After downloading the CSV Generator Template, you can now customize the template as per your requirement. The parameters listed in the CSV Generator Template are explained below in detail:

Parameters	Description
Start Extension Number	Enter the Extension Number for the first extension. For example: If you configure Start Extension Number as 5001, the system will start creating the extensions with 5001 as the first extension number followed by 5002, 5003.....for other extensions.
Total Extensions	Enter the total number of extensions you want to configure. For example: You want to configure 100 extensions, then enter 100 as the Total Extensions.

Parameters	Description
Start S/W Port No.	Enter the software port number from where you want to configure the first extension. For example: You want to configure the extensions from software port number 1, then enter 1 as the software port number.
Use SIP Extension	Select Yes to enable the SIP extensions. If you select No, the SIP extensions will not get registered with the server.
SIP Hardware Template	Enter the template number of the SIP Hardware Template you want to assign to the extensions. By default, the SIP Hardware Template number 01 is assigned. The SIP Hardware Template contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters. Check if the values in this template fulfills the requirements of the extensions. If the Template number 01 fulfills the feature requirements, retain the Template number 01. If a different set of SIP hardware features are to be allowed to the extensions, then prepare another template and enter that template number in this field. To know how to customize the template, refer to "SIP Hardware Template".
Station Basic Feature Template	Enter the template number of the Station Basic Feature Template you want to assign to the extensions. By default, Station Basic Feature Template number 01 is assigned. The Station Basic Feature Template has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. Check if the values in this template fulfills the requirements of the extensions. If the Template number 01 fulfills the feature requirements, retain the Template number 01. If a different set of features are to be allowed to the extensions, then prepare another template and enter that template number in this field. To know how to customize the template, refer to "Station Basic Feature Template".

Parameters	Description
Station Advanced Feature Template	Enter the template number of the Station Advanced Feature Template you want to assign to the extensions. By default, Station Advanced Feature Template number 01 is assigned. The Station Advanced Feature Template has a set of advanced features for the extensions such as such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service and related parameters. Check if the values in this template fulfills the requirements of the extensions. If the Template number 01 fulfills the feature requirements, retain the Template number 01. If a different set of features are to be allowed to the extensions, then prepare another template and enter that template number in this field. To know how to customize the template, refer to “Station Advanced Feature Template”.
Call Pickup Group	Enter the Call Pickup Group number you want to assign to the extensions. By default, Call Pickup Group number 1 is assigned. Call Pick Up allows the extension user to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allow incoming calls for the extension to be answered by the other extensions assigned the same Call Pick-Up group. For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group’. Refer “Call Pick Up” for instructions on how to create groups.
Station Type	Select the Station Type — *Administration/ Guest* you want to assign to the extensions.
Priority	Select the Priority level for the extensions from 1 to 9. By default, 5-Normal is selected as the Priority level. Each extension is assigned a Priority Level starting from 1,2, 3....to 9, with '1' being the lowest Priority and '9' being the highest Priority. To know more about the feature, refer to “Priority”.

You can connect/ register three Matrix Extended IP Phones / UC clients/ Standard SIP Phones at three different location as a single SIP extension. You can register the same or different types of Extended IP Phones/ UC clients/ Standard SIP Phones — at each location.

For Example: You can configure SPARSH VP330 at Location 1, 2 and 3. If you want to use more than one SPARSH VP330 Extended IP Phones as a SIP Extension, configure their settings at Location 1, Location 2 and Location 3.

For Auto Provisioning, you must configure the Standard SIP phones at Location1 only. To know more, refer to “Configuring Standard SIP Phones”.

Location 1	Description
Enable Device	Select Yes to enable the SIP extension.
Device Type	Select the desired Device Type for location 1.
Registrar Server Address	Select the appropriate Registrar Server Address to register the Extended IP Phones/ VARTA WIN200/ Standard SIP Phones with the SIP Registrar of ANANT UCS, according to your installation scenario. If extensions are connected on the WAN network, select ***Use WAN Port IP Address*** as the Registrar Server IP Address. If extensions are connected on the LAN network, select ***Use LAN Port IP Address*** as the Registrar Server IP Address. If extensions are connected in the Global Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select ***Use Router/STUN's IP Address*** as the Registrar Server IP Address. Make sure you configure either the Router's Public IP Address or Simple Traversal of UDPs through NATs (STUN) in Network Parameters. For details, see "Configuring Network Parameters". If Dynamic DNS is configured in the Network Parameters, select ***Use Dynamic DNS Host Name*** as the Registrar Server IP Address. By default, ***Use WAN Port IP Address*** is selected as the Registrar Server IP Address.
Internal Registrar Server Address	Select the appropriate Internal Registrar Server Address to register the UC clients (VARTA ADR100/ VARTA AMP100) with the SIP Registrar of ANANT UCS within a private network. Select the appropriate option as per your installation scenario: If you want the UC clients to register using the WAN network, select ***Use WAN Port IP Address*** as the Internal Registrar Server Address. If you want the UC clients to register using the LAN network, select ***Use LAN Port IP Address*** as the Internal Registrar Server Address. If Dynamic DNS is configured in the Network Parameters, select ***Use Dynamic DNS Host Name*** as the Internal Registrar Server Address. By default, ***Use WAN Port IP Address*** is selected as the Internal Registrar Server Address.

Location 1	Description
External Registrar Server Address	Select the appropriate External Registrar Server Address to register the UC clients (VARTA ADR100/ VARTA AMP100) with the SIP Registrar of ANANT UCS from a public network. Select the option according to your installation scenario: If you want the UC clients to register using the WAN network, select ***Use WAN Port IP Address*** as the External Registrar Server Address. If the UC clients is connected in the Public Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select ***Use Router/STUN's IP Address*** as External Registrar Server Address. Make sure you configure either the Router's Public IP Address or Simple Traversal of UDPs through NATs (STUN) in Network Parameters. For details, see "Configuring Network Parameters". If Dynamic DNS is configured in the Network Parameters, select ***Use Dynamic DNS Host Name*** as the External Registrar Server Address. By default, ***Use WAN Port IP Address*** is selected as the External Registrar Server Address.

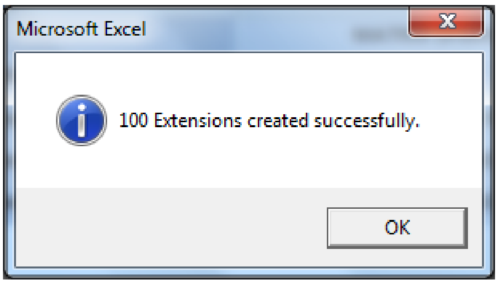
Similarly, you can configure the parameters listed in Location 2 and 3 as per your requirement.

In case, LAN Port is not defined in the system, and ***Use LAN Port IP Address*** *is selected as the* ***Registrar Server Address/ Internal Registrar Server Address*** *in the CSV Generator Template, then in this case, the system will automatically select* ***Use WAN Port IP Address*** *as the* ***Registrar Server Address*** *and will create the extensions.*

Generating CSV File

After you have configured the required parameters in the CSV Generator Template, you can now generate the CSV File. To do so,

- Click the **Generate csv** button present at the bottom of the CSV Generator Template.
- A pop window opens specifying the number of extensions that are created successfully.



- Click **OK**.

- The CSV File will be generated.

Save CSV File

Extension Number	S/W Port No.	Use SIP Extension	Name	Email ID	Mobile Number	Authentication Password	SIP Hardware Template	Station Basic Feature Template	Station Advanced Feature Template
5001	1	No				us!(VJ66	1	1	1
5002	2	No				d/Q26o[L	1	1	1
5003	3	No				vRC`79)n	1	1	1
5004	4	No				).gl‌Y6i0	1	1	1
5005	5	No				DQ/5`2rq	1	1	1
5006	6	No				tYOr8(-4	1	1	1
5007	7	No				(e)8G7pQ	1	1	1
5008	8	No				7-cLhM]5	1	1	1

You can now modify the generated CSV File by adding the following details for each and every extension in the file as per your requirement.

- **Name:** Enter the name for the extensions listed in the file. The name you enter here will be displayed as the Caller ID of the extensions on the remote user's phone, when the extension user makes calls.

  It can be the name of the person who will use the extension or the name of a Department. You can configure a name of a maximum of 18 alphanumeric characters.

- **Email ID:** Enter the email address for the extensions listed in the file. The Email ID you enter here will be used to send the auto-sign email in Mobile UC clients — VARTA ADR100 and VARTA AMP100 applications. Refer "Auto Sign-In Parameters", to know more. The Email ID is also used for various server features.

  You can configure an Email ID of a maximum of 64 characters.

- **Mobile Number:** Enter the mobile number for the extensions listed in the file, if required. The mobile number can be a maximum of 16 digits.

- **Location Name:** Enter the location for the extension so as to identify the extension. Location Name may be the place where the extension is located (e.g: 2nd Floor/ Desk number -8, Head office, Branch office, residence).

  The Location Name may consist of 18 characters (maximum).

- **MAC Address / Device ID:** Enter the MAC Address / Device ID of the extensions.

Similarly you can configure Location Name and MAC Address / Device ID in location 2 and location 3, if required.

You can also modify the CSV File by editing or deleting the values of the parameters as per your requirement.

- *Authentication ID is used by the system for user authentication of the SIP messages received from the extensions. This parameter is not listed in the CSV File and is automatically configured by the system in the SIP Extension Settings Page, when you upload the CSV File. The system will configure the* ***Authentication ID*** *same as the* ***Extension Number*** *configured in the CSV File for the extensions.*

  You can change the Authentication ID, if required. The Authentication ID must be unique for each SIP extension. It may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and " (double quote) are allowed.

- *Corresponding to the Authentication ID, the system generates a random* ***Authentication Password*** *and assign to all the extensions listed in th CSV File. You can edit the authentication password for the extensions, if required. Make sure you note down or copy the Authentication Password in a confidential file.*

  All the other parameters that are not listed in the CSV File but are present in the SIP Extension Settings Page will contain its default value. You can modify the values of these parameters, after the extensions are configured in the system. You can also use the copy button present at the bottom of the SIP Extension Settings Page to replicate the changes made in the configuration of an extension to all the other extensions configured in the system. To know more, refer to "Copy Parameter Values".

Exporting the CSV File

After modifying the CSV File, you can now export the file. To do so,

Save CSV File

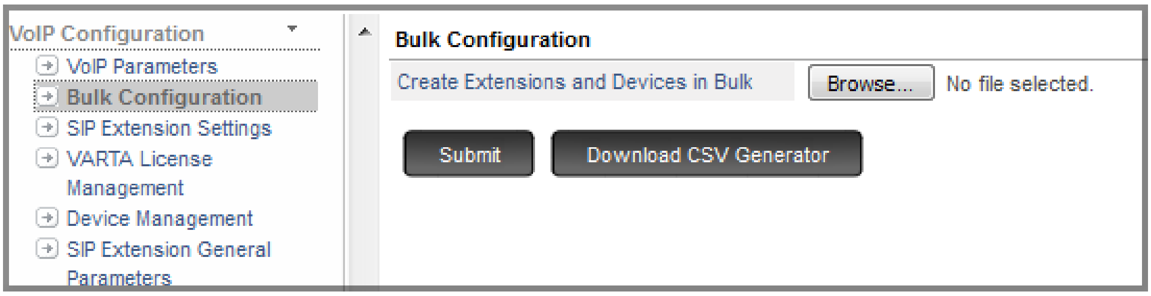
Extension Number	S/W Port No.	Use SIP Extension	Name	Email ID	Mobile Number	Authentication Password	SIP Hardware Template	Station Basic Feature Template	Station Advanced Feature Template
5001	1	No				us!(VJ66	1	1	1
5002	2	No				d/Q26o[L	1	1	1
5003	3	No				vRC^79)n	1	1	1
5004	4	No				).gIY6i0	1	1	1
5005	5	No				DQ/5^2rq	1	1	1
5006	6	No				tYOr8(-4	1	1	1
5007	7	No				(e)8G7pQ	1	1	1
5008	8	No				7-cLhM]5	1	1	1

- Click the **Save CSV File** button present at the top of the CSV File.
- The CSV File with the name format **CSV-Exported-File-DD-Mmm-YYYY HH-MM-SS**, where **DD-Mmm-YYYY signifies** the current date and **HH-MM-SS** signifies the current time, will be exported on your desktop.

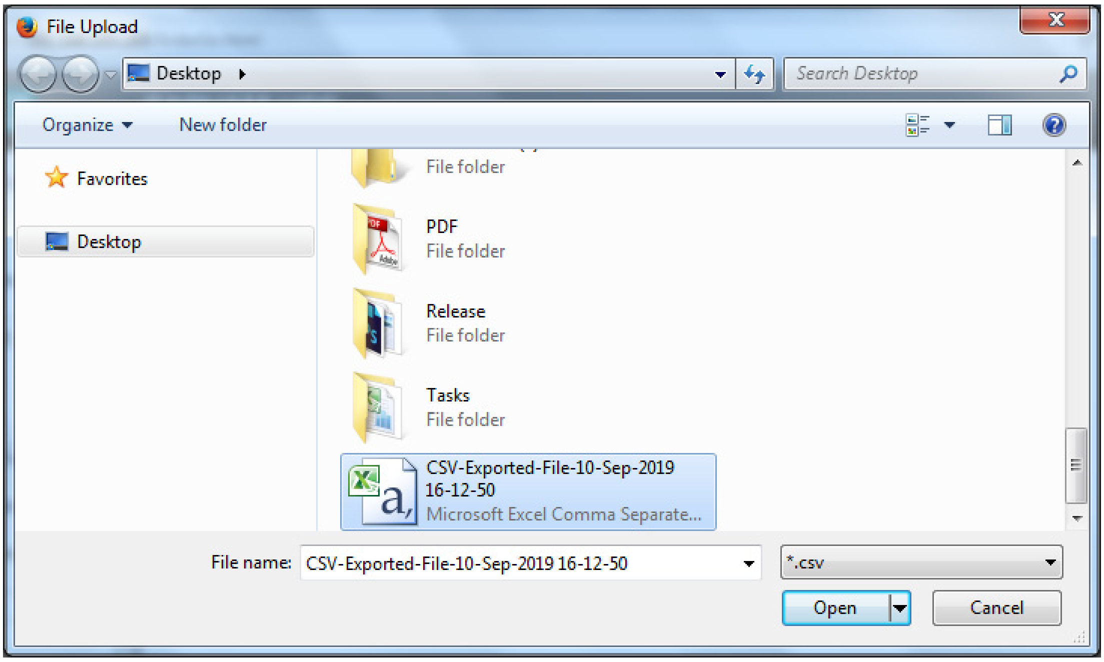
Uploading the CSV File

After you have exported the CSV File, you now have to upload the CSV File to create the extensions. To do so,

- Login as System Engineer.
- Under **VoIP Configuration**, click **Bulk configuration**.

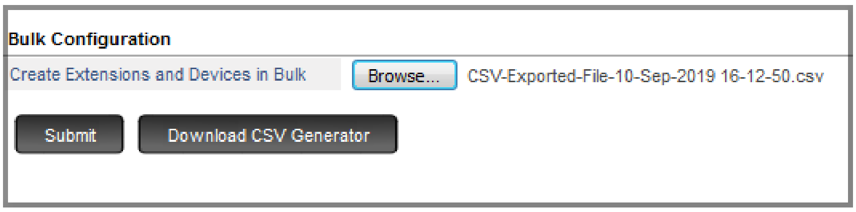


- Click the **Browse** button to select the CSV File from the local disk on the computer.



- After selecting the required CSV File, click **Open**.

  The system displays the name of the CSV File.



- Click **Submit** to upload the CSV File.

After you upload the CSV File, the following statuses will be displayed:

- **Successful:** The total number of extensions that are successful configured in the server.
- **Failed:** The total number of extensions that are not configured in the server. The list of extensions that are not configured are also displayed.

- *Consider extension 5001 is already configured at a particular location in the system, say location 1 of S/W Port No 1. Another extension 5002 with different set of configuration is listed at the same location, that is, location 1 of S/W Port No 1 in the CSV File, then in this case, when the CSV file is uploaded, the configuration of the extension 5001 will be overwritten with the configuration of the extension 5002.*
- *In case, you keep the Extension Number for any extension listed in the CSV File as blank, then, the system will not create this extension. Moreover, the system will consider it as an end of the list and any extension/s listed after this blank Extension Number will not be created.*
- *The total number of extensions the system creates will depend on the number of extensions configured in System Pre-requisites. For an instance, if you have configured 100 as the **SIP Extensions** in **System Pre-requisites** and upload a CSV File with a configuration of 500 extensions, then the system will create only the first 100 extensions.*
- *Make sure the SIP ID and MAC/ Device ID is unique for all the extensions configured in the system. In case, the SIP ID or MAC /Device ID listed in the CSV File conflicts with the SIP ID or MAC /Device ID of an extension already configured in the system, then in this case, the extension with the duplicate SIP ID or MAC/ Device ID will not be created.*
- *It may take about 5 minutes to create all the 5000 extensions in the system.*
- *The system will not create an extension listed in the CSV File, if the Extension Number configured for this extension in the CSV File matches with any of the access codes configured in the system. To know the access codes that are configured in the system, refer to “Access Codes”.*
- *Make sure you do not leave the mandatory parameters listed in the CSV Generator Template as blank, otherwise the system will not create these extensions. All the parameters listed in the CSV Generator Template are mandatory for creating the extensions.*

To configure rest of the parameters which are not mentioned in the CSV File, refer to “Configuring SIP Extension Settings as per the Extended Phone Type”.

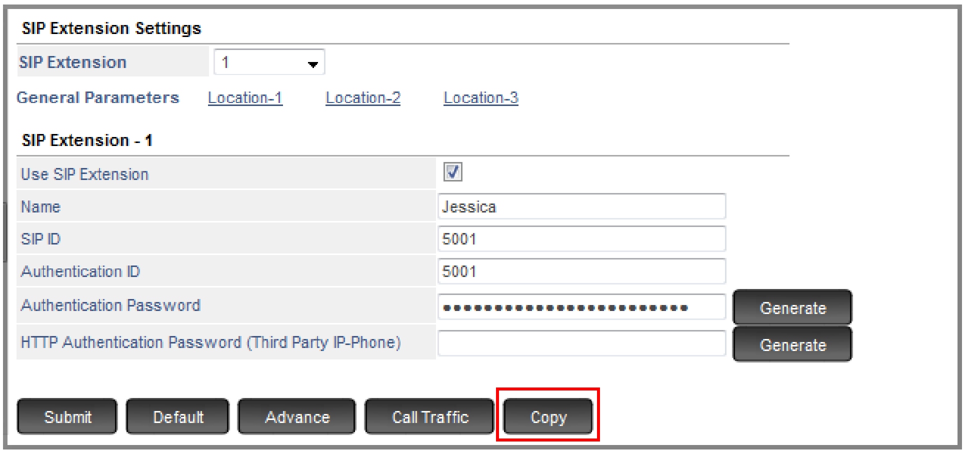
Copy Parameter Values

Once you have created the extensions, you can now configure the parameters which are not mentioned in the CSV File as per your requirement.

After you have configured the required parameters for an extension, you can now copy the configuration of this extension to all the other extensions by simply using the **Copy** button present at the bottom of the SIP Extension Settings page.

For an instance, you can configure the parameters like *DSS Key Settings, Quality of Service (QoS)* for an extension and copy the configuration of this extension to other extensions. To do so,

- Click **Copy**.



- The Copy page opens in another tab.

- Select the check boxes against the parameters you want to copy to other extensions. If you want to copy all the parameters to the other extensions, select the check box **Select All**.

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you only clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

- Click **Next**. The Select Extensions window opens.

- The number and/or name of the current extension is displayed in **Copy from**.
  The extensions to which you can copy the current extension settings are displayed on the left side box arranged sequentially in the increasing order.
  - **To select a extension from the List,**
    - Select the desired extension and click **>** button. The selected extension will appear on the right side box.
  - **To select multiple extensions from the List,**
    - Select the desired extensions and click **>** button. The selected extensions will appear on the right side box.
  - **To select all the extensions from the List,**
    - Click **>>** button. All the extensions will appear on the right side box.

  You can also de-select the selected extensions, if required.
  - **To de-select a extension from the List,**
    - Select the desired extension and click **<** button. The selected extension will appear on the left side box.
  - **To de-select multiple extensions from the List,**
    - Select the desired extensions and click **<** button. The selected extensions will appear on the left side box.
  - **To select all the extensions from the List,**
    - Click **<<** button. All the extensions will appear on the left side box.
- Click **Submit.** The settings of the current extension is copied to all the selected extensions.

Auto Sign-In Parameters

The system supports automatic configuration and registration of Mobile Clients — VARTA ADR100, VARTA AMP100 applications with the Server using Auto Sign-In.

Auto Sign-In enables Mobile Clients — VARTA ADR100, VARTA AMP100 applications — to configure and register with the server automatically at a click of a button.

For this you must:

- Configure the **Auto Sign-In Parameters**. For details, refer "Configuring Auto Sign-In Parameters"
- Configure the **General Parameters** in **SIP Extensions Settings**. For details, refer "Configuring SIP Extension Settings"
- Make sure you send the **Auto Sign-In Configuration Mail**. For details, refer "Configuring SIP Extension Settings"

You can also view the status of Auto Sign-In Email in "Viewing SIP Extension Status".

How it Works

After you configure the required Auto Sign-In parameters and have sent the Auto Sign-In Mail to the Mobile Client users, they need to follow the instructions given below:

- The Auto Sign-In mail has an attachment that contains the necessary configuration details. The Mobile Client user must open the attachment in the Auto Sign-In mail using the VARTA ADR100 or VARTA AMP100 application. For more information refer to the respective User Guides.

- The Server will receive the request and process it. The client will get configured and registered automatically at any free Location 1, 2 or 3 in SIP Extension Settings. If none of the Locations are free the request will not be served.

- You can check the Registration status in "Viewing SIP Extension Status".

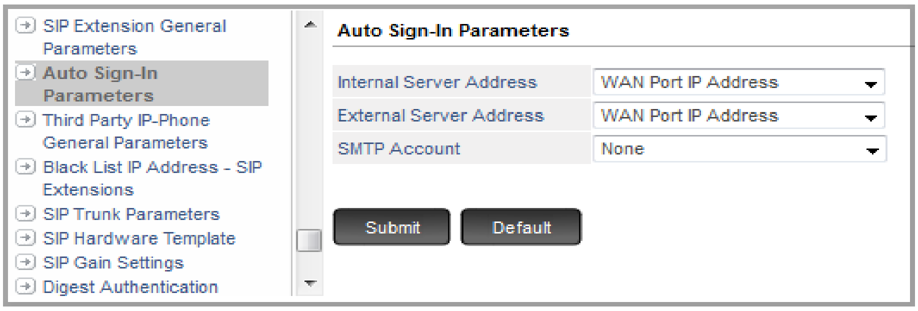
Configuring Auto Sign-In Parameters

The information you configure in Auto Sign-In Parameters will be sent in the mail to the Mobile Client user, when you click the **Send Configuration Mail** button.

- Login as System Engineer.

- Under **Configuration**, click **VoIP Configuration**.

- Click **Auto Sign-In Parameters**.



- Select the appropriate **Internal Server Address** to register the application with the SIP Registrar of ANANT UCS within a private network. Select the appropriate option as per your installation scenario:

  - If you want the application to register using the WAN network, select **WAN Port IP Address** as the Internal Server Address.

  - If you want the application to register using the LAN network, select **LAN Port IP Address** as Internal Server Address.

  - If Dynamic DNS is configured in the Network Parameters, select **Dynamic DNS Host Name** as Internal Server Address.

  - If you do not want to configure the Internal Server Address, select **Don’t Send**.

    By default, WAN Port IP Address is selected as the Internal Server Address.

- Select the appropriate **External Server Address** to register the application with the SIP Registrar of ANANT UCS from a public network. Select the option according to your installation scenario:

  - If you want the application to register using the WAN network, select **WAN Port IP Address** as External Server Address.

  - If the application is connected in the Public Network and ANANT UCS is located behind a Router, or behind a NAT Router and STUN is configured, select **Router/STUN's IP Address** as External Server Address.

    Make sure you configure either the **Router’s Public IP Address** or **Simple Traversal of UDPs through NATs (STUN)** in Network Parameters. For details, see “Configuring Network Parameters”.

  - If Dynamic DNS is configured in the Network Parameters, select **Dynamic DNS Host Name** as External Server Address.

  - If you do not want to configure the External Server Address, select **Don’t Send**. By default, WAN Port IP Address is selected as the External Server Address.

- Select the **SMTP Account**[37] through which you want the email to be sent.

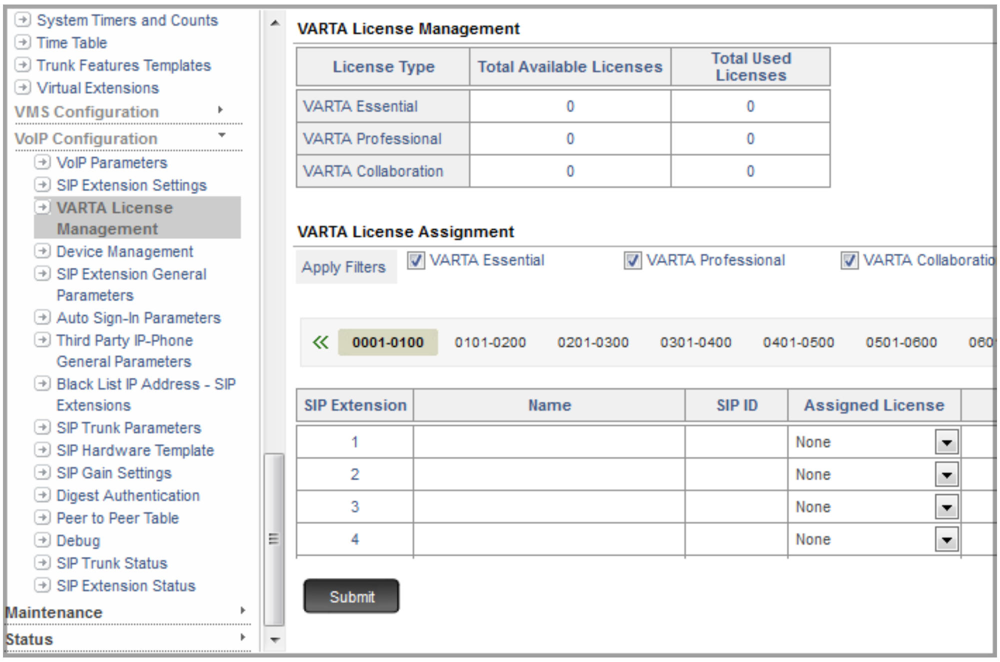
*If you select **Don’t Send** in both Internal as well as External Server Address, the server will send the Auto Sign-In mail but the VARTA Mobile Clients will not get registered.*

37. *Make sure that the SMTP settings are configured correctly.For more information, refer “SMTP Settings”.*

VARTA License Management

To view the VARTA License Status and to assign licenses to SIP Extensions,

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **VARTA License Management.**



VARTA License Management

The following information will be displayed for the license you activated (VARTA Essential/Professional/ Collaboration).

- **License Type:** This displays the name of the licenses — Essential, Professional or Collaboration.
- **Total Available Licenses:** This displays the total number of licenses activated.
- **Total Used Licenses:** This displays the total number of VARTA users registered as SIP extensions.

VARTA License Assignment

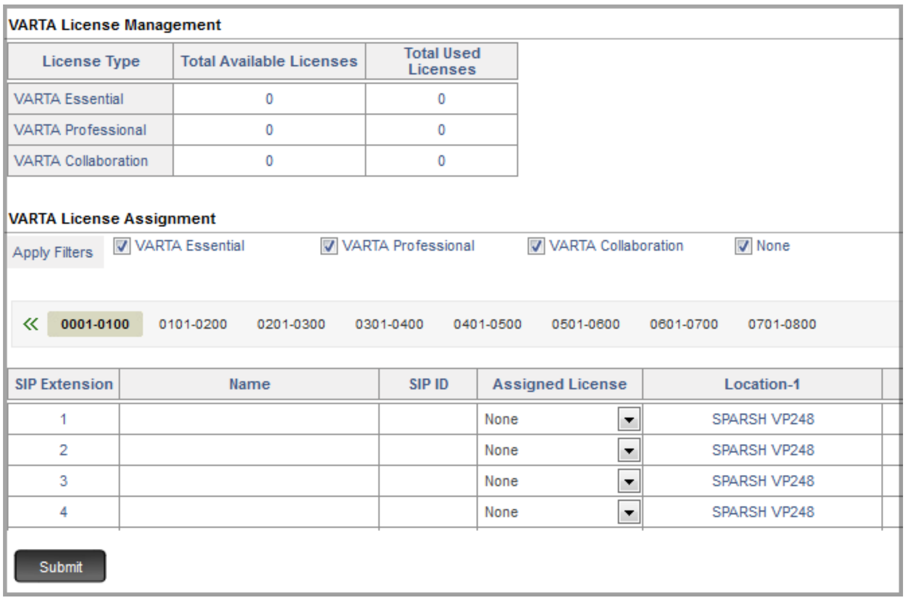
After registering and configuring MATRIX VARTA WIN200/VARTA ADR100/VARTA AMP100 as a SIP Extension, you must select the desired license in the **Assigned License** field below. You can also filter the SIP Extensions as per the license assigned.

- **Apply Filters:** By default you can view all the SIP Extensions, as all the filters are enabled.
    - Clear the **VARTA Essential** check box, if you do not want to view the SIP Extensions that are assigned this license.

- Clear the **VARTA Professional** check box, if you do not want to view the SIP Extensions that are assigned this license.

- Clear the **VARTA Collaboration** check box, if you do not want to view the SIP Extensions that are assigned this license.

- Clear the **None** check box, if you do not want to view the SIP Extensions that are not assigned any license.



- **SIP Extension:** This displays the SIP Extension Number with which you can register the VARTA UC Clients.

- **Name:** This displays the name assigned to the SIP Extension.

- **SIP ID:** This displays the SIP ID assigned to the SIP Extension.

- **Assigned License:** Select the license you wish to assign to the SIP Extension and click **Submit**.

- **Location 1, 2, 3:** This displays the Device Type selected on the SIP Extension Location 1, 2 and 3.

Configuring Standard SIP Phones

You can connect any of the following as SIP Extensions of ANANT UCS:

- Matrix SPARSH VP248
- Matrix SPARSH VP110
- Matrix SPARSH VP710
- Matrix SPARSH VP210
- Matrix SAPRSH VP510
- Any Standard SIP Phone
- Any SIP-enabled device including PC based Soft-phone
- Analog Terminal Adaptor

For detailed product information and operational instructions, refer to the product documentation supplied with the Standard SIP Phone/device.

ANANT UCS supports two separate methods of Provisioning the Standard SIP Phones. These two methods are:

- **Manual Provisioning:** In Manual Provisioning, the user must configure the required parameters of the Standard SIP Phone manually. So, it is not a simple plug-and-play solution for mass deployment, as it requires intervention of authorized technical personnel for phone configuration. To configure Standard SIP Phones using this method, see “Configuring Standard SIP Phones using Manual Provisioning”.

- **Auto Provisioning:** In Auto Provisioning, the Standard SIP Phone gets configured automatically by retrieving the required configuration file from ANANT UCS. The configuration file contains pre-configured values of necessary parameters required by the Standard SIP Phone. Thus, it eliminates the necessity of manually configuring the Standard SIP Phone parameters. When the Standard SIP Phone starts, it then gets configured automatically. Here, ANANT UCS acts as the Auto Provisioning Server by providing the configuration file to the Standard SIP Phones. Auto Provisioning enables mass deployment of Standard SIP Phones and provides a plug-and-play solution for them. To configure Standard SIP Phones using this method, see “Configuring Standard SIP Phones using Auto Provisioning”.

- *Auto Provisioning for Third Party SIP Phones is supported through SPARSH Port (using HTTP) and Secure SPARSH Port (using HTTPS Port).*

- *For Auto Provisioning, you must configure the Standard SIP phones at **Location1** only.*

  You will have to manually configure the Standard SIP phones at Location 2 and 3. After doing so, the Standard SIP Phone will register at the selected location and the location page will be disabled.To check the registration status, click the SIP Extension Status link.

To be able to use Standard SIP Phones/Devices as SIP Extensions, you must configure the following:

- SIP Extension General Parameters, see “Configuring SIP Extension General Parameters”.

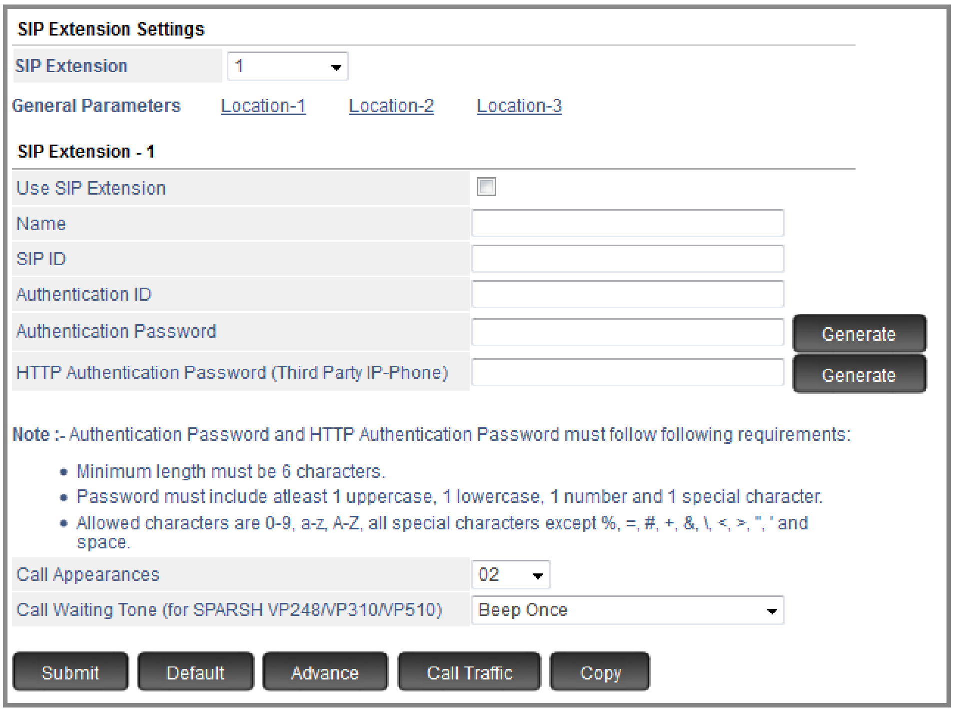
- SIP Extension Settings, see “Configuring SIP Extension Settings”.

- Standard SIP Authorization Profile, see “Standard SIP Authorization Profile”.

- Voice Mail Settings, if you want to provide mailbox to the extensions. See “Extension Voice Mail Settings”.

Configuring SIP Extension Settings

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings**.



The page of SIP Extension 1 opens.

- You may select the **SIP Extension** number you want to configure.

  The parameters of the SIP Extension number you selected will appear on this page.

ANANT UCS supports IPv4 and IPv6 Addresses for registering Standard SIP Phones.

- Select the **Use SIP Extension** check box to enable the SIP extension. Default: disabled.
- In **Name**, enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls.

  The name may consist of a maximum of 18 alphanumeric characters.

If no name is assigned to the SIP Extension, the system will display the name received in the INVITE message from the SIP Extension user when making outgoing calls.

- Enter the **SIP ID** for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the system can call a SIP Extension by dialing the SIP ID assigned to the SIP Extension. SIP ID of each

SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed. You cannot assign the same SIP ID to more than one extension.

By default, the SIP IDs are Blank.

The SIP ID will be set to default value (blank), when you restore the default settings of the system.

- In **Authentication ID**, enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank and it must be unique for each SIP extension. The number may be a string of maximum 6 digits. All ASCII characters except < > and " (double quote) are allowed. Default: Blank.

Make sure the User ID configured in "Digest Authentication" does not conflict with the Authentication ID configured above.

- In **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the Authentication Password in a confidential file.

To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See "Black List IP Address - SIP Extensions" for more details. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- In **HTTP Authentication Password** (Third Party IP-Phone), enter the password manually or click **Generate** to automatically generate a unique password which will be used to authenticate the Standard SIP Phone connected to the system. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password (Third Party IP-Phone) and provides configuration on validation.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, the password must:

  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **',** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

Make sure you note down or copy the HTTP Authentication Password in a confidential file.

To provide additional security, when the HTTP Authentication fails 5 times consecutively within 10 minutes due to wrong Authentication ID / Authentication Password, the system will block the IP Address for configuration of this Standard SIP Phone. However, you can register again after 10 minutes. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.

- In **Call Appearances**, define the maximum number of simultaneous calls that the SIP Extension should be allowed to make/receive. You can set up to 10 call appearances for a SIP Extension. Default: 2.

  When Call Appearance is set to 2, the SIP Extension can make/receive 2 calls at a time.

- Under **Auto Sign-In,** enter the **Email ID** of the extension user you wish to store. The Email ID can be a maximum of 64 characters. The Email ID is used for various server features.

Auto Sign-In parameters — Send Configuration Mail *and* ***Mail Status,*** *are applicable only for Mobile Clients — VARTA ADR100, VARTA AMP100 applications.*

- Under **Authentication**, enable Authentication of any or all of the following SIP Message Options by selecting the respective check boxes:
  - **INVITE Request**
  - **SUBSCRIBE Request**

  By default, the SIP Message Options INVITE and SUBSCRIBE are enabled.

Make sure that the Authentication ID for the SIP Extension has been configured.

- If you are going to register this SIP Extension with the same SIP ID at more than one location[38], you may enable the **Shared Call Appearance Subscription** check box on this SIP Extension. Default: Disabled.

  Shared Call Appearance provides notification on call states to all the Standard SIP Phones with the same SIP ID at different locations. To know more about this feature, see “Shared Call Appearance”.

- To provide voice mail facility to the SIP Extension, select the **Voice Mail Subscription** check box. Default: disabled.

- To allow the SIP Extension to monitor the status of another extension or Trunk, select the **Busy Lamp Field**[39] **Subscription** check box. Default: disabled. See “Busy Lamp Field for Trunks” to know more.

When extension's state is changed from Ringing (early state as defined in BLF) to Mature (confirm state, as defined in BLF), because of implementation of ANANT UCS, it will send 'Terminate' state while moving from ringing to mature state. The interpretation of terminate message will vary from terminal to terminal.

38. *ANANT UCS allows you to register SIP Extension with the same SIP ID at three different locations.*
39. *Busy Lamp Field (BLF), a typical feature supported by the system and Key Telephone Systems, is also supported on SIP Extensions.*

    In the system and Key Telephone Systems, this feature is typically used by the Operator to monitor the status of another extension, that is, whether it is available, ringing or busy. The status of the other extensions is indicated on the special function keys configured on the Operator's console. This helps the Operator decide whether to place the call, or transfer the call to that extension, or pick up the call ringing on that extension. With BLF Subscription enabled on the SIP Extension, the user can monitor the status of another extension or trunk.

- To allow the SIP Extension to view the status of other SIP-enabled Terminals, whether they are available or not, select the **Presence Subscription** check box. Default: disabled.

The SIP Extension, for which you have enabled Presence Subscription, will be able to view Presence of only those SIP Extensions which have PUBLISH enabled.

- To allow the SIP Extension to publish presence using the PUBLISH feature supported by the SIP Extension, select the **PUBLISH Enable** check box. Default: disabled.

  By default, **Authentication** for PUBLISH message is enabled. You may disable if you do not want to use Authentication.

You must configure the Authentication ID, if you have enabled both Publish and Authentication.

- For secure conversations over SIP, enable **SRTP Mode**. ANANT UCS supports the following options:
  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.
  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.
    - If you select this option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.
  - **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

    By default, SRTP Mode is Disabled.
- **Key Templates** are not applicable to Standard SIP Phones registered with ANANT UCS.
- Assign a **SIP Hardware Template** to the SIP Extension. Default: Template 01. The “SIP Hardware Template” contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template number 01 assigned to all SIP Extensions as well as to SIP Trunks.

  Check if the values in this template fulfill requirements of the SIP Extension. If Template 01 fulfills the feature requirements, retain Template 01.

  If a different set of SIP hardware features are to be allowed to this SIP Extensions, prepare another template and assign it to this extension. To do this,
  - Under **VoIP Configuration**, click **SIP Hardware Template**.
  - Select a Template number, for example 02.
  - Customize Template number 02 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 02 in the **SIP Hardware Template** field.

Also see the topic "SIP Hardware Template" to know more about customizing the templates and applying on the SIP Extensions.

- Assign a **Station Basic Feature Template** to the SIP Extension. Default: Template 01. The "Station Basic Feature Template" has a set of features like Time Table, Class of Service, Toll Control, Operator, Storage of Incoming and Outgoing Calls, Outgoing Trunk Bundle groups. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  If the default Station Basic Feature Template 01 fulfills the feature requirements of the SIP Extension ("Class of Service (CoS)", "Toll Control", "OG Trunk Bundle Group", etc.) retain this template, you may also customize this template. If you want to assign a different set of features to this SIP Extension, prepare a different Station Basic Feature Template and apply it to this extension. To do this,

  - Under **Configuration**, click **Station Basic Feature Template**.
  - Select a Template number, for example 05.
  - Customize Template number 05 and click **Submit**.
  - Return to the **SIP Extension Settings** page.
  - Select the number of the Template you customized, Template 05, in the **Station Basic Feature Template** field.
  - Click **Submit.**

  Also, see the topic "Station Basic Feature Template" to know more about customizing the templates and applying on extensions.

- Assign a **Station Advanced Feature Template** to the SIP Extension. Default: Template 01. The "Station Advanced Feature Template" has a set of advanced features for extensions such as Alarm Notification settings, Routing of Incoming Auto Attendant Calls, Call Duration Control, Floor Service, etc. There are 50 different templates to choose from. Each template can also be altered to suit your requirement and preferences.

  Check if the default template fulfills the feature requirements of the SIP Extension by clicking the **Station Advanced Feature Template** link.

  You may retain this template and customize it further, or customize another template if a different set of features are to be allowed to this SIP Extension. To customize/prepare another template,

  - Under **Configuration**, click **Station Advanced Feature Template**.
  - Select the Template number, for example 02, and customize this template.
  - Click **Submit** and return to the 'SIP Extension Settings' page.
  - In the **Station Advanced Feature Template** field, select the number of the template you customized.
  - Click **Submit**.

  Also see the topic "Station Advanced Feature Template" for instructions on customizing these templates and applying them on the extensions.

- In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see "Extension Voice Mail Settings".

The Voice Mail Settings link will be visible only if you have configured the respective SIP ID.

- Assign the SIP Extension to a **Call Pick-up Group**, if required.

Call Pick Up allows the SIP Extension to 'pick up' (answer) calls ringing on any other extension, by dialing a feature code, without physically going to the ringing extension. It also allows incoming calls for the SIP Extension to be answered by the other extensions assigned the same Call Pick-Up group.

For this to work, both the ringing extension and the extension picking up the call must be in the same 'Call Pick Up Group'. Refer “Call Pick Up” for instructions on how to create groups. You can create as many as 99 groups numbered from 01 to 99.

Enter the number of the Call Pick-Up Group you created for this SIP Extension in this field.

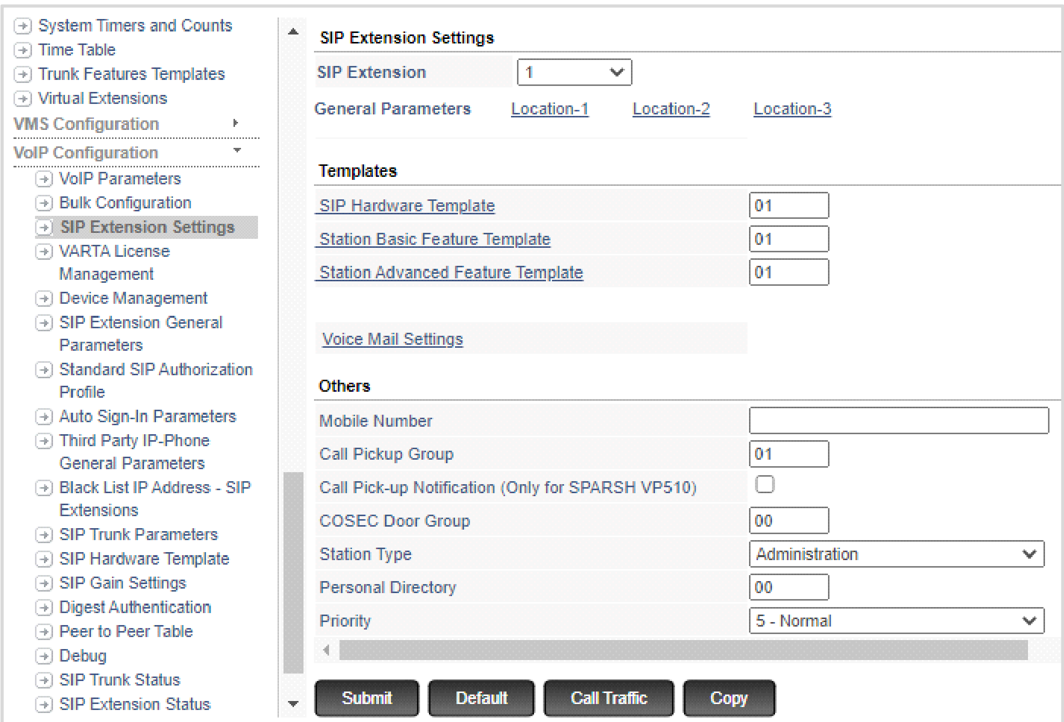
- Select the **Call Pick-up Notification (Only for SPARSH VP510)** check box, if you want the Call Pick-up Notification to be displayed on the phone LCD. The Call Pick-up Notification will be displayed for internal as well as external calls. The notification will be displayed on SPARSH VP510 only if the ringing extension is in the same Call Pick-up Group. The notification will be displayed only when SPARSH VP510 is in idle or dial state. The notification will display the name/number of the Caller along with the name/number of the Called. For example, if the name and number of the Caller is ABC, 2001 and the name and number of Called is XYZ, 1001, the LCD notification will be displayed as ABC -> XYZ. Along with the LCD display the phone back-light will also be lit.

  For this to work, make sure you have enabled Call Pick-up in COS as well as assigned a Call Pick-up Group to the extension. Refer to “Call Pick Up” and “Class of Service (CoS)”. Call Pick-up Notifications will be displayed for SIP Extensions and for calls landing through SIP Trunks. For details of the Notification, refer to the EON510_SPARSH VP510 V2 User Guide.

- You can assign the extension user to a **COSEC Door Group** for COSEC Integration. The users in the same group must be assigned the same group. You can create as many as 50 groups numbered from 00 to 50. Users who are assigned COSEC Door Group ‘00’ are not a part of any group.

Advanced Configuration Parameters

- If you want to provide other features like Personal Directory, Priority, or assign a Station Type to the SIP Extension, click the **Advanced** button at the bottom of the page.



- Enter the **Mobile Number** of the extension user you wish to store. The Number can be a maximum of 16 digits.
- If you are using the system in the *Enterprise Mode*, by default **Administrator** is selected as **Station Type**.

  If you are using the system in the *Hotel Mode*, by default **Guest** is selected as **Station Type**. You may select the **Station Type** as Administrator for the Hotel Administrative Staff. To know more refer to the *ANANT UCS Hospitality System Manual*.
- You may assign a **Personal Directory** number to the SIP Extension. Default: 00.

  A Personal Directory is a list of 25 frequently dialed numbers, each of which are stored by Index number (location code), Name and Trunk Access Codes ("Out Going Trunk Bundle Group Index"). The Personal Directory is necessary for using the features "Abbreviated Dialing" and "Dial By Name".

  When a Personal Directory is assigned to a SIP Extension, make sure you also configure this directory. The Personal Directory can be configured by the SIP Extension users and by the System Engineer. Refer the topic "Abbreviated Dialing" for instructions on configuring the Personal Directory.If Personal Directory is not to be assigned, enter 00 in this field.
- Select a **Priority** Level for the SIP Extension from 1 to 9. Default: 5-Normal. Each extension of ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension (phone) with higher priority calls an extension with lower priority, a triple ring is placed on the called extension. To know more, read the feature description "Priority".

  If this SIP extension is assigned to Operator, you may want to set a higher priority for this extension.
- Click **Submit**.

Configuring Standard SIP Phones using Manual Provisioning

To be able to use Standard SIP Phones/devices with ANANT UCS, you must configure the following:

- Configure the **SIP Extension Settings**. For details, see "Configuring SIP Extension Settings".[40]
- Configure specific parameters (for example, SIP ID/User Name, Authentication ID, Password, Server Address/Domain Name etc.) in the Standard SIP Phone/device which are required to register it with system. For more information, refer the product documentation supplied with the Standard SIP Phone/ device you want to use.

Configuring Standard SIP Phones using Auto Provisioning

ANANT UCS supports the following third party Standard SIP Phones for Auto Provisioning-

- Panasonic UTG200B
- Grandstream GXP110x
- Grandstream GXP2200
- Grandstream GXP21xx/116x/14xx
- Grandstream GXV3140/3175
- Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X

40. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected to the system.*

- Yealink SIP-T20P
- Yealink SIP-T3XG
- Cisco SPA50xG/51xG SIP Phone
- Cisco SPA525G SIP Phone
- Polycom IP Phone
- Snom IP Phone
- Htek 802
- Any Standard SIP Phone

Connecting the Standard SIP Phone

To be able to configure and register the Standard SIP Phone with ANANT UCS using Auto Provisioning,

- Connect and reboot the Standard SIP Phone.
- Make sure that **DHCP** is selected as the *Connection Type* in the Standard SIP Phone.
- For Auto Provisioning, you must configure the Standard SIP phones at Location1 only.
- The phone will automatically fetch the configuration file(s) from ANANT UCS and will get registered.

Using any third party DHCP Server

You can use any third party DHCP Server for assigning the 'Auto Provisioning Server Address' and 'Server Port' to the Standard SIP Phones. To do so,

Select **DHCP Option 66**, and configure the Server Address and Port of ANANT UCS in that third party DHCP Server in the following format - **http://IP Address:Port.**

If you are using Auto Provisioning and if the Standard SIP Phone has multiple SIP accounts, it is recommended to keep the first SIP account in default settings. After Auto Provisioning, the old configuration, if present for this account will be deleted automatically and it will get registered with ANANT UCS.

Configuring Panasonic Standard SIP Phones

You are recommended to complete the following steps before connecting **Panasonic UTG200B**:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings"[41]
- Configure the device specific settings applicable to your **Panasonic UTG200B** at Location1 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.

41. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

- Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box.	Panasonic UTG200B
Location Name		Configure the Location Name to identify the phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Panasonic UTG200B
Device Type		Select Device Type as Panasonic UTG200B.	Panasonic UTG200B
MAC Address		Enter the MAC Address of the Panasonic phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address and provides configuration on validation.	Panasonic UTG200B
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Panasonic UTG200B
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	Panasonic UTG200B
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Panasonic	Panasonic UTG200B
Display Language		Displays the Standard SIP Phone language and it is non-editable.	Panasonic UTG200B
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be a minimum of 6 characters to a maximum of 16 characters. Default: userpass	Panasonic UTG200B
Admin Password		Configure the Admin Password[c] of the Standard SIP Phone. It can be a minimum of 6 characters to a maximum of 16 characters. Default: adminpass	Panasonic UTG200B

Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30	Panasonic UTG200B
Daylight Saving Time	Enable DST	If DST is applicable, select the Enable DST check box. A list of DST parameters appear. Configure them as per your requirement. Default: Disabled.	Panasonic UTG200B
	DST Offset (min)	Configure the DST Offset in minutes. Valid Range: 0 to 720 minutes. Default: 60 minutes.	Panasonic UTG200B
	Start Day and Time of DST	To configure the time from when DST should be applied in the year, select the Month, Day, Week/Ordinal and Time (hh:mm) from the corresponding list boxes respectively.	Panasonic UTG200B
	End Day and Time of DST	To configure the time when DST should end, select the Month, Day, Week/Ordinal and Time (hh:mm) from the corresponding list boxes respectively.	Panasonic UTG200B
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Panasonic UTG200B

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Grandstream Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Grandstream** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings"[42].

42. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

- Configure the device specific settings applicable to your **Grandstream** at Location 1 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.
  - Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Device Type	Select Device Type as any of the desired Grandstream Standard SIP Phone you want to connect.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
MAC Address	Enter the MAC Address of the Grandstream phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Authenticate HTTP Provisioning request	Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Registrar Server Address	Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175

Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Grandstream	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Display Language	Select the desired Standard SIP Phone language. Default: English	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Send Phone Book	Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the ANANT UCS.	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Phone Book Download interval (min)	Configure the Phone Book Download interval in minutes. Valid Range: 0 - 720 minutes. Default: 60 minutes	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
User Password	Configure the User Password[b] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: user	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Admin Password	Configure the Admin Password[c] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: admin	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175
Time Zone	Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30 (Chennai, New Delhi, Mumbai)	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Date Display Format	Select the Date Display Format for the Standard SIP Phone. Default: yyyy-mm-dd	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Time Display Format	Select the Time Display Format for the Standard SIP Phone. Default: 24Hr	1. Grandstream GXP2200 2. Grandstream GXP21xx/116x/14xx 3. Grandstream GXV3140/3175
Primary NTP Server	Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	1. Grandstream GXP110x 2. Grandstream GXP2200 3. Grandstream GXP21xx/116x/14xx 4. Grandstream GXV3140/3175

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Yealink Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Yealink** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings"[43].
- Configure the device specific settings applicable to your **Yealink** at Location 1 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

43. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

Device Type		Select Device Type as any of the desired Yealink Standard SIP Phone you want to connect.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
MAC Address		Enter the MAC Address of the Yealink phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Yealink	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Send Phone Book		Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from ANANT UCS. Default: Enabled	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Web User Interface Language		Select the desired language in which the Web User Interface of the selected Standard SIP Phone variant should be displayed. Default: English	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Phone User Interface		Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

User Password		Configure the User Password[b] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Admin Password		Configure the Admin Password[c] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Daylight Saving Time Mode		Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
If DST Type = DST by Date	Start Date	Configure the time from when DST should be applied in the year by selecting the Month, Day and Hour from the corresponding list boxes respectively.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	End Date	Configure the time when DST should end by selecting the Month, Day and Hour from the corresponding list boxes respectively.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	Offset (minutes)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

If DST Type = DST by Week	DST Start Month	Select the Month from when DST should be applied.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	Start Hour of Day	Select the DST Start Hour of the Day.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Stop Month	Select the Month when DST should end.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Stop Day of Week	Select the Day of Week when DST should end.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
	End Hour of Day	Select the DST End Hour of the Day.	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default:	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default for T20P Phone: MM DD YY and for other phones WWW MMM DD	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	1. Yealink SIP-T19P/SIP-T21P/T22P/T26P/T28P/T4X 2. Yealink SIP-T20P 3. Yealink SIP-T3XG

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN’s IP Address as the Registrar Server Address. Make sure the Router’s Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Cisco Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Cisco** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see “Using any third party DHCP Server”.
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see “Configuring SIP Extension Settings”[44].
- Configure the device specific settings applicable to your **Cisco** at Location1on the SIP Extensions page. To do so,
    - Under **Configuration**, click **VoIP Configuration**.
    - Click **SIP Extension Settings** link.
    - Click **Location 1** and configure the following parameters.

Parameter		**Description/Action to be Taken**	**Applicable Standard SIP Phone Variant**
Enable Device		Select the Enable Device check box.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

44. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer’s documentation for more details.*

Device Type		Select Device Type as any of the desired Cisco Phone you want to connect.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
MAC Address		Enter the MAC Address of the Cisco phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of the profiles. Default: Cisco	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Send Phone Book		Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from ANANT UCS. Default: Enabled	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Phonebook Name		Configure the Name of the Phonebook. This name will be displayed on the Phone LCD. It can be of maximum 32 characters. Default: Corporate Directory	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: Blank	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

Admin Password		Configure the Admin Password[c] of the Cisco phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: Blank	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Phone User Interface Language		Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English-US	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Language File Download Path		Select the path from which the Standard SIP Phone must fetch the Language files. Default: None. Make sure you configure the desired Path (Server Address/es) in the "Third Party IP-Phone General Parameters".	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: GMT + 05:30	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Enable DST		If DST is applicable, select the Enable DST check box. A list of DST parameters appear. Configure them as per your requirement. Default: Disabled	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

Daylight Saving Time	DST Start Day of Week	Select the Day of Week from when DST should be applied.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	Start Hour of Day	Select the DST Start Hour of the Day.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Stop Month	Select the Month when DST should end.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Stop Day of Week	Select the Day of Week when DST should end.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
	End Hour of Day	Select the DST End Hour of the Day.	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: month/day	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24 Hr	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Offset Timer(min)		Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: 000	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	1. Cisco SPA50xG/51xG SIP Phone 2. Cisco SPA525G SIP Phone

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Polycom Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Polycom** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings"[45].
- Configure the device specific settings applicable to your **Polycom** at Location 1 on the SIP Extensions page. To do so,
    - Under **Configuration**, click **VoIP Configuration**.
    - Click **SIP Extension Settings**.
    - Click **Location 1** and configure the following parameters.

Parameter		**Description/Action to be Taken**	**Applicable Standard SIP Phone Variant**
Enable Device		Select the Enable Device check box.	Polycom IP Phone
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Polycom IP Phone
Device Type		Select Device Type as Polycom IP Phone.	Polycom IP Phone

45. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

MAC Address		Enter the MAC Address of the Polycom phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Polycom IP Phone
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Polycom IP Phone
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address	Polycom IP Phone
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of the profiles. Default: Polycom	Polycom IP Phone
Send Phone Book		Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the ANANT UCS. Default: Enabled	Polycom IP Phone
User Password		Configure the User Password[b] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: 123	Polycom IP Phone
Admin Password		Configure the Admin Password[c] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: 456	Polycom IP Phone
Phone User Interface Language		Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English Internal (en-in)	Polycom IP Phone

Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: (GMT + 05:30) Bombay, Calcutta, Madras, New Delhi	Polycom IP Phone
Enable DST		If DST is applicable, select the Enable DST check box. A list of DST parameters appear. Configure them as per your requirement. Default: Disabled	Polycom IP Phone
DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week.	Polycom IP Phone
If DST Type = DST by Date	Start Month	Select the Month from when DST should be applied.	Polycom IP Phone
	Start Day	Select the Day from when DST should be applied.	Polycom IP Phone
	Start Hour	Select the Hour from when DST should be applied.	Polycom IP Phone
	End Month	Select the Month when DST should end.	Polycom IP Phone
	End Day	Select the Day when DST should end.	Polycom IP Phone
	End Hour	Select the Hour when DST should end.	Polycom IP Phone
If DST Type = DST by Week	DST Start Month	Select the Month from when DST should be applied.	Polycom IP Phone
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	Polycom IP Phone
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	Polycom IP Phone
	Start Hour of Day	Select the DST Start Hour of the Day.	Polycom IP Phone
	DST Stop Month	Select the Month when DST should end.	Polycom IP Phone
	DST Stop Day of Week	Select the Day of Week when DST should end.	Polycom IP Phone
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	Polycom IP Phone
	End Hour of Day	Select the DST End Hour of the Day.	Polycom IP Phone
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: 1 Jan, Mon	Polycom IP Phone

Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24Hr	Polycom IP Phone
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Polycom IP Phone

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Snom Standard SIP Phones

You are recommended to complete the following steps before connecting any of the **Snom** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings"[46].
- Configure the device specific settings applicable to your **Snom** at Location 1 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.
  - Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box.	Snom IP Phone

46. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Snom IP Phone
Device Type	Select Device Type as Snom IP Phone.	Snom IP Phone
MAC Address	Enter the MAC Address of the Snom phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Snom IP Phone
Authenticate HTTP Provisioning request	Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Snom IP Phone
Registrar Server Address	Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	Snom IP Phone
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of the profiles. Default: Snom	Snom IP Phone
Send Phone Book	Select the Send Phone Book check box to enable downloading of the Phone Book (consisting of Extension and Global Directory Contacts) from the ANANT UCS. Default: Enabled	Snom IP Phone
Web User Interface Language	Select the desired language in which the Web User Interface of the selected Standard SIP Phone should be displayed. Default: English	Snom IP Phone
Web User Language File Download Path	Select the path from which the Standard SIP Phone must fetch the Language files. Default: None. Make sure you configure the desired Path (Server Address/es) in the "Third Party IP-Phone General Parameters".	Snom IP Phone
Phone User Interface Language	Select the desired language in which the Standard SIP Phone's User Interface should be displayed. Default: English	Snom IP Phone

Phone User Language File Download Path	Select the path from which the Standard SIP Phone must fetch the Language files. Default: None. Make sure you configure the desired Path (Server Address/es) in the "Third Party IP-Phone General Parameters".	Snom IP Phone
Time Zone	Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: 5.5 India (Calcutta)	Snom IP Phone
Date Display Format	Select the Date Display Format for the Standard SIP Phone. Default: mm/dd	Snom IP Phone
Time Display Format	Select the Time Display Format for the Standard SIP Phone. Default: 12Hr	Snom IP Phone
Call Progress Tone	Select the region to apply the Call Progress Tone prevailing there. Default: India.	Snom IP Phone
HTTP Server Login Username	Configure the HTTP Server Login User Name for web access. It can be a maximum of 16 characters. Default: Blank	Snom IP Phone
HTTP Server Login Password	Configure the HTTP Server Login Password[b] for web access. It can be a maximum of 16 characters. Default: Blank	Snom IP Phone
Admin Mode	Select Admin Mode check box to allow its access to the user. Default: Enabled	Snom IP Phone
Admin Mode Password	Configure the Admin Password[c] of the Standard SIP Phone. It can be a minimum of 1 character to a maximum of 16 characters. Default: 0000	Snom IP Phone
Primary NTP Server	Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Snom IP Phone

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address.

b. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

c. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Matrix SPARSH VP110 Standard SIP Phone

You are recommended to complete the following steps before connecting any of the **Matrix SPARSH VP110** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings"[47].
- Configure the device specific settings applicable to your **Matrix SPARSH VP110** at any one of the Location1 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box. Default: Disabled.	Matrix SPARSH VP110
Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Matrix SPARSH VP110
Device Type		Select Device Type as MATRIX SPARSH VP110.	Matrix SPARSH VP110
MAC Address		Enter the MAC Address of the SPARSH VP110 phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Matrix SPARSH VP110

47. *Some of the parameters may not be applicable depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Matrix SPARSH VP110
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	Matrix SPARSH VP110
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: SPARSH VP110.	Matrix SPARSH VP110
Send Key Settings		Select the Send Phone Book check box to apply the Key Settings[b] to the Phone. Default: Enabled	Matrix SPARSH VP110
Dial Plan[c]		Select the desired Dial Plan. Default: 1 The Phone will detect end of dialing as per the rules configured in the Dial Plan selected here.	Matrix SPARSH VP110
Transport Mode		Select the protocol to be used to transport the SIP messages. Default: TCP	Matrix SPARSH VP110
Enable SRTP?		Select the Enable SRTP? check box for secure conversations over SIP. Default: Disabled	Matrix SPARSH VP110
SIP DiffServe/ ToS		Enter the desired SIP DiffServe/ToS to set the Quality of Service (QoS) for SIP packets Default: 26	Matrix SPARSH VP110
RTP DiffServe/ ToS		Enter the desired RTP DiffServe/ToS to set the Quality of Service (QoS) for RTP packets. Default: 46	Matrix SPARSH VP110
SIP Port		Enter the port on which the phone will listen for SIP messages. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5060	Matrix SPARSH VP110

Min RTP Port		To define a range of RTP ports, configure the minimum local RTP port. Default: 11780	Matrix SPARSH VP110
Max RTP Port		To define a range of RTP ports, configure the maximum local RTP port. Default: 11800	Matrix SPARSH VP110
Allow HTTP		Select the Allow HTTP check box to enable HTTP Web access. Default: Enabled	Matrix SPARSH VP110
HTTP Port		Enter the port number on which the HTTP access is to be given. Default: 80	Matrix SPARSH VP110
Allow HTTPS		Select the Allow HTTPS check box to enable HTTPS Web access. Default: Enabled	Matrix SPARSH VP110
HTTPS Port		Enter the port number on which the HTTPS access is to be given. Default: 443	Matrix SPARSH VP110
Local Phone Book[d]		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the ANANT UCS. These will be stored in the phone's Local Phone Book. Default: Do not send	Matrix SPARSH VP110
Remote Phone Book		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the ANANT UCS. These will be stored in the phone's Remote Phone Book. Default: Send first Extension Numbers, remaining Global Directory Numbers	Matrix SPARSH VP110
Web User Interface Language		Select the desired language in which the Web User Interface should be displayed. Default: English	Matrix SPARSH VP110
Phone User Interface Language		Select the desired language in which the Phone's User Interface should be displayed. Default: English	Matrix SPARSH VP110
User Password		Configure the User Password[e] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	Matrix SPARSH VP110
Admin Password		Configure the Admin Password[f] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	Matrix SPARSH VP110
Ringer Device for Headset		When in Headset mode, select the ring destination for the SPARSH VP110. Default: Use Speaker	Matrix SPARSH VP110

Enable Distinctive Ring		Select the Enable Distinctive Ring check box, to set different ringing patterns to distinguish between different types of call events. The following types of call events are supported: • Internal Call • Trunk Call • Auto Call Back • Auto Redial • Alarm • Emergency • Operator Alarm • Message Wait • Priority • Emergency Conference Default: Disabled	Matrix SPARSH VP110
Call Progress Tone		Select the region to apply the Call Progress Tone prevailing there. Default: Custom.	Matrix SPARSH VP110
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	Matrix SPARSH VP110
Daylight Saving Time Mode		Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	Matrix SPARSH VP110
DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week. Default: DST by Date	Matrix SPARSH VP110
If DST Type = DST by Date	Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP110
	Start Day	Select the Day from when DST should be applied.	Matrix SPARSH VP110
	Start Hour	Select the Hour from when DST should be applied.	Matrix SPARSH VP110
	End Month	Select the Month when DST should end.	Matrix SPARSH VP110
	End Day	Select the Day when DST should end.	Matrix SPARSH VP110
	End Hour	Select the Hour when DST should end.	Matrix SPARSH VP110
	Offset Timer (min)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: Blank	Matrix SPARSH VP110

If DST Type = DST by Week	DST Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP110
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	Matrix SPARSH VP110
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	Matrix SPARSH VP110
	Start Hour of Day	Select the DST Start Hour of the Day.	Matrix SPARSH VP110
	DST Stop Month	Select the Month when DST should end.	Matrix SPARSH VP110
	DST Stop Day of Week	Select the Day of Week when DST should end.	Matrix SPARSH VP110
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	Matrix SPARSH VP110
	End Hour of Day	Select the DST End Hour of the Day.	Matrix SPARSH VP110
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: WWW MMM DD	Matrix SPARSH VP110
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24 Hr	Matrix SPARSH VP110
Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Matrix SPARSH VP110

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address.

b. To apply the desired Key Template:
   i) Select the **Send Key Settings** check box.
   ii) Click **Key Template** under **Configuration**.
   iii) Click the desired key template - Operator/ Executive/ Hotel Attendant/Guest of VP110. Assign features facilities to the keys in this template as per your requirement.
   For instructions, see “Customizing Extended IP Phone Templates”
   iv) Click **SIP Extension Settings** under **VoIP Configuration**. Select the SIP Extension Number on which SPARSH VP110 is registered.
   v) Under **General Parameters**, scroll to **Key Template** and select the template you configured as per your requirement.

The Personalized option of Key Template is not applicable for SPARSH VP110.

c. If you want to apply the rules of the Dial Plan configured in ANANT UCS, see “Dial Plan for SIP Extension”. You can also configure rules for the Dial Plan from each phone. To do so, refer to the *SPARSH VP110 User Guide*.

d. If an option other than "Do not send" is selected in Local Phone Book, it will overwrite all Local Phone Book's contacts of the phone.

e. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

f. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Htek 802 Standard SIP Phones

You are recommended to complete the following steps before connecting the **Htek 802** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see “Using any third party DHCP Server”.
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see “Configuring SIP Extension Settings”.
- Configure the device specific settings applicable to **Htek** at any one of the Location1 on the SIP Extensions page. To do so,
    - Under **Configuration**, click **VoIP Configuration**.
    - Click **SIP Extension Settings**.
    - Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box.	Htek 802 IP Phone
Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Htek 802 IP Phone
Device Type	Select Device Type as Htek 802.	Htek 802 IP Phone

MAC Address	Enter the MAC Address of the Htek 802 phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Htek 802 IP Phone
Authenticate HTTP Provisioning request	Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Htek 802 IP Phone
Registrar Server Address	Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	Htek 802 IP Phone
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of profiles. Default: Htek	Htek 802 IP Phone
Admin Password	Configure the Admin Password[b] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	Htek 802 IP Phone
Assign Voice Mail Key	Enable this check box to assign the first programmable key of the phone as Voice Mail.	Htek 802 IP Phone
SIP Port	Enter the port on which the phone will listen for SIP messages. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5060	Htek 802 IP Phone
Min RTP Port	To define a range of RTP ports, configure the minimum local RTP port. Default: 5004	Htek 802 IP Phone
Max RTP Port	To define a range of RTP ports, configure the maximum local RTP port. Default: 5014	Htek 802 IP Phone
Apply System Time Zone	Select this check box if you want to apply the System's Time Zone to the Standard SIP Phone.	Htek 802 IP Phone
Time Zone	Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	Htek 802 IP Phone

Primary NTP Server	Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Htek 802 IP Phone
Daylight Saving Time Mode	Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	Htek 802 IP Phone

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address

b. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Matrix SPARSH VP710 Standard SIP Phone

You are recommended to complete the following steps before connecting any of the **Matrix SPARSH VP710** Standard SIP Phones:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the **SIP Extension Settings** in ANANT UCS. For details, see "Configuring SIP Extension Settings".
- Configure the device specific settings applicable to your **Matrix SPARSH VP710** at Location1 on the SIP Extensions page. To do so,
  - Under **Configuration**, click **VoIP Configuration**.
  - Click **SIP Extension Settings**.
  - Click **Location 1** and configure the following parameters.

Parameter		Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device		Select the Enable Device check box. Default: Disabled.	Matrix SPARSH VP710

Location Name		Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Matrix SPARSH VP710
Device Type		Select Device Type as MATRIX SPARSH VP710.	Matrix SPARSH VP710
MAC Address		Enter the MAC Address of the SPARSH VP710 phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Matrix SPARSH VP710
Authenticate HTTP Provisioning request		Select the Authenticate HTTP Provisioning request check box to authenticate HTTP request for security reason. Default: Disabled. If you are using the Standard SIP Phone from the Public IP Address, we recommend you to enable this option. ANANT UCS validates the Standard SIP Phone on the basis of the SIP ID and HTTP Authentication Password and provides configuration on validation. Make sure, you have configured SIP ID as HTTP User Name and HTTP Authentication Password as HTTP Password in the Standard SIP Phone.	Matrix SPARSH VP710
Registrar Server Address		Select the appropriate Registrar Server Address according to your installation scenario[a] to register the Standard SIP Phone with the SIP Registrar of ANANT UCS. Default: Use WAN Port IP Address.	Matrix SPARSH VP710
Standard SIP Authorization Profile		Select the desired Standard SIP Authorization Profile from the list of profiles. Default: SPARSH VP710	Matrix SPARSH VP710
Dial Plan[b]		Select the desired Dial Plan. Default: 1 The Phone will detect end of dialing as per the rules configured in the Dial Plan selected here.	Matrix SPARSH VP710
Transport Mode		Select the protocol to be used to transport the SIP messages. Default: TCP	Matrix SPARSH VP710
Enable SRTP?		Select the Enable SRTP? check box for secure conversations over SIP. Default: Disabled	Matrix SPARSH VP710

DTMF Type		Select the appropriate DTMF Type to determine how the DTMF digits will be sent over the IP Network, when a DTMF digit is pressed.	Matrix SPARSH VP710
SIP DiffServe/ ToS		Enter the desired SIP DiffServe/ToS to set the Quality of Service (QoS) for SIP packets Default: 26	Matrix SPARSH VP710
RTP DiffServe/ ToS		Enter the desired RTP DiffServe/ToS to set the Quality of Service (QoS) for RTP packets. Default: 46	Matrix SPARSH VP710
SIP Port		Enter the port on which the phone will listen for SIP messages. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5060	Matrix SPARSH VP710
SIP TLS Port		Enter the port on which the phone will listen for SIP messages transported over TLS. This port is used as source port in SIP messages.This port is also used to send SIP messages to the remote peer. Default: 5061	Matrix SPARSH VP710
Min RTP Port		To define a range of RTP ports, configure the minimum local RTP port. Default: 11780	Matrix SPARSH VP710
Max RTP Port		To define a range of RTP ports, configure the maximum local RTP port. Default: 11800	Matrix SPARSH VP710
Allow HTTP		Select the Allow HTTP check box to enable HTTP Web access. Default: Enabled	Matrix SPARSH VP710
HTTP Port		Enter the port number on which the HTTP access is to be given. Default: 80	Matrix SPARSH VP710
Allow HTTPS		Select the Allow HTTPS check box to enable HTTPS Web access. Default: Enabled	Matrix SPARSH VP710
HTTPS Port		Enter the port number on which the HTTPS access is to be given. Default: 443	Matrix SPARSH VP710
Local Phone Book[C]		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the ANANT UCS. These will be stored in the phone's Local Phone Book. Default: Do not send	Matrix SPARSH VP710
Send Personal Directory		Enable this check box if you want to allow usage of Personal Directory. These contacts will be stored in the phone's Local phone Book.	Matrix SPARSH VP710

Remote Phone Book		Select the type of contacts (Extension or/and Global Directory Contacts) the phone must download from the ANANT UCS. These will be stored in the phone's Remote Phone Book. Default: Send first Extension Numbers, remaining Global Directory Numbers	Matrix SPARSH VP710
Web User Interface Language		Select the desired language in which the Web User Interface should be displayed. Default: English	Matrix SPARSH VP710
Phone User Interface Language		Select the desired language in which the Phone's User Interface should be displayed. Default: English	Matrix SPARSH VP710
User Password		Configure the User Password[d] of the Standard SIP Phone. It can be maximum of 16 characters long. Default: user	Matrix SPARSH VP710
Admin Password		Configure the Admin Password[e] of the Standard SIP Phone. It can be a maximum of up to 16 characters. Default: admin	Matrix SPARSH VP710
Ringer Device for Headset		When in Headset mode, select the ring destination for the SPARSH VP710. Default: Use Speaker	Matrix SPARSH VP710
Enable Distinctive Ring		Select the Enable Distinctive Ring check box, to set different ringing patterns to distinguish between different types of call events. The following types of call events are supported: • Internal Call • Trunk Call • Auto Call Back • Auto Redial • Alarm • Emergency • Operator Alarm • Message Wait • Priority • Emergency Conference Default: Disabled	Matrix SPARSH VP710
Call Progress Tone		Select the region to apply the Call Progress Tone prevailing there. Default: Custom.	Matrix SPARSH VP710
Time Zone		Select the Time Zone of the region/country where the Standard SIP Phone is installed. Default: + 05:30 India (Calcutta)	Matrix SPARSH VP710
Daylight Saving Time Mode		Select the Daylight Saving Time Mode that should be applied to the selected Standard SIP Phone. Default: Automatic.	Matrix SPARSH VP710

DST Type		Select the DST Type that should be applied to the Standard SIP Phone, either DST by Date or DST by Week. Default: DST by Date	Matrix SPARSH VP710
If DST Type = DST by Date	Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP710
	Start Day	Select the Day from when DST should be applied.	Matrix SPARSH VP710
	Start Hour	Select the Hour from when DST should be applied.	Matrix SPARSH VP710
	End Month	Select the Month when DST should end.	Matrix SPARSH VP710
	End Day	Select the Day when DST should end.	Matrix SPARSH VP710
	End Hour	Select the Hour when DST should end.	Matrix SPARSH VP710
	Offset (minutes)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: Blank	Matrix SPARSH VP710
If DST Type = DST by Week	Offset (minutes)	Configure the DST Offset timer value in minutes. Valid Range:-300 to +300. Default: Blank	Matrix SPARSH VP710
	DST Start Month	Select the Month from when DST should be applied.	Matrix SPARSH VP710
	DST Start Day of Week	Select the Day of Week from when DST should be applied.	Matrix SPARSH VP710
	DST Start Day of Week Last in Month	Select the DST Start Day of Week Last in Month.	Matrix SPARSH VP710
	Start Hour of Day	Select the DST Start Hour of the Day.	Matrix SPARSH VP710
	DST Stop Month	Select the Month when DST should end.	Matrix SPARSH VP710
	DST Stop Day of Week	Select the Day of Week when DST should end.	Matrix SPARSH VP710
	DST Stop Day of Week Last in Month	Select the DST Stop Day of Week Last in Month.	Matrix SPARSH VP710
	End Hour of Day	Select the DST End Hour of the Day.	Matrix SPARSH VP710
Date Display Format		Select the Date Display Format for the Standard SIP Phone. Default: WWW MMM DD	Matrix SPARSH VP710
Time Display Format		Select the Time Display Format for the Standard SIP Phone. Default: 24 Hr	Matrix SPARSH VP710

Primary NTP Server		Select the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phone. Default: None. Make sure you configure the desired NTP Server Address/es in the "Third Party IP-Phone General Parameters".	Matrix SPARSH VP710

a. If the Standard SIP phone is in the same network (LAN) as ANANT UCS, select Use LAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is connected to Internet over WAN, select Use WAN Port IP Address as the Registrar Server Address.

If the Standard SIP phone is in the Global Network and ANANT UCS is located behind a NAT Router and STUN is programmed, select Use Router/ STUN's IP Address as the Registrar Server Address. Make sure the Router's Public IP Address is configured in the Network Parameters.

If the Dynamic DNS is configured in the Network Parameters, select Use Dynamic DNS Host Name as the Registrar Server Address.

b. If you want to apply the rules of the Dial Plan configured in ANANT UCS, see "Dial Plan for SIP Extension". You can also configure rules for the Dial Plan from each phone. To do so, refer to the *SPARSH VP710 User Guide*.

c. If an option other than "Do not send" is selected in Local Phone Book, it will overwrite all Local Phone Book's contacts of the phone.

d. To avoid unauthorized access, we recommend you to change the User Password regularly. Make sure it is strong and is kept confidential.

e. To avoid unauthorized access, we recommend you to change the Admin Password regularly. Make sure it is strong and is kept confidential.

Configuring Any Standard SIP Phone

You are recommended to complete the following steps before connecting Any Standard SIP Phone:

- Decide the physical location of the Standard SIP Phone.
- Configure the settings in the third party DHCP Server. For instructions, see "Using any third party DHCP Server".
- Configure the SIP Extension Settings in ANANT UCS. For details, see "Configuring SIP Extension Settings"[48].
- Configure the device specific settings applicable to your Standard SIP Phone at any one of the Location 1, 2 or 3 on the SIP Extensions page. To do so,
    - Under **Configuration**, click **VoIP Configuration**.
    - Click **SIP Extension Settings**.

48. *Some of the features may not be supported depending on the Standard SIP Phone you have connected. Please refer the specific Standard SIP Phone manufacturer's documentation for more details.*

- Click **Location 1** and configure the following parameters.

Parameter	Description/Action to be Taken	Applicable Standard SIP Phone Variant
Enable Device	Select the Enable Device check box. Default: Disabled.	Any Standard SIP Phone
Location Name	Configure the Location Name to identify the Standard SIP Phone. The Location Name can be a maximum of 18 characters. Default: Blank.	Any Standard SIP Phone
Device Type	Select Device Type as Any Standard SIP Phone.	Any Standard SIP Phone
MAC Address	Enter the MAC Address of the Standard SIP phone to be connected at this location. Default: Blank. ANANT UCS validates the phone on the basis of the MAC Address, and provides configuration on validation.	Any Standard SIP Phone
Standard SIP Authorization Profile	Select the desired Standard SIP Authorization Profile from the list of profiles. Default: None.	Any Standard SIP Phone

*If you select the Device Type as **Any Standard SIP**, then you are recommended to configure the **Standard SIP Authorization Profile** to prevent any unauthorized access and misuse of the system.*

If you want to replicate the configuration of the SIP Phone Settings same as Location 1 to Location 2 and Location 3, you can use the **Copy** button present at the bottom of the page. To know more, refer to "Copy Parameter Values".

If you wish to Copy only the configurations from a location to all other locations where the same IP Phone is connected, make sure you clear the Device Type check box.

If you wish to copy the Device Type as well as all the configurations from a location to all the SIP Extensions, make sure all the check boxes are selected.

Standard SIP Authorization Profile

The Standard SIP Authorization Profile contains the list of default profiles of various Standard SIP Phones supported by the system.

Each Profile consists of details which you must configure for successful registration of the phone. Thus, the Standard SIP Authorization Profile ensures that only authorized phones are used as extensions of the system.

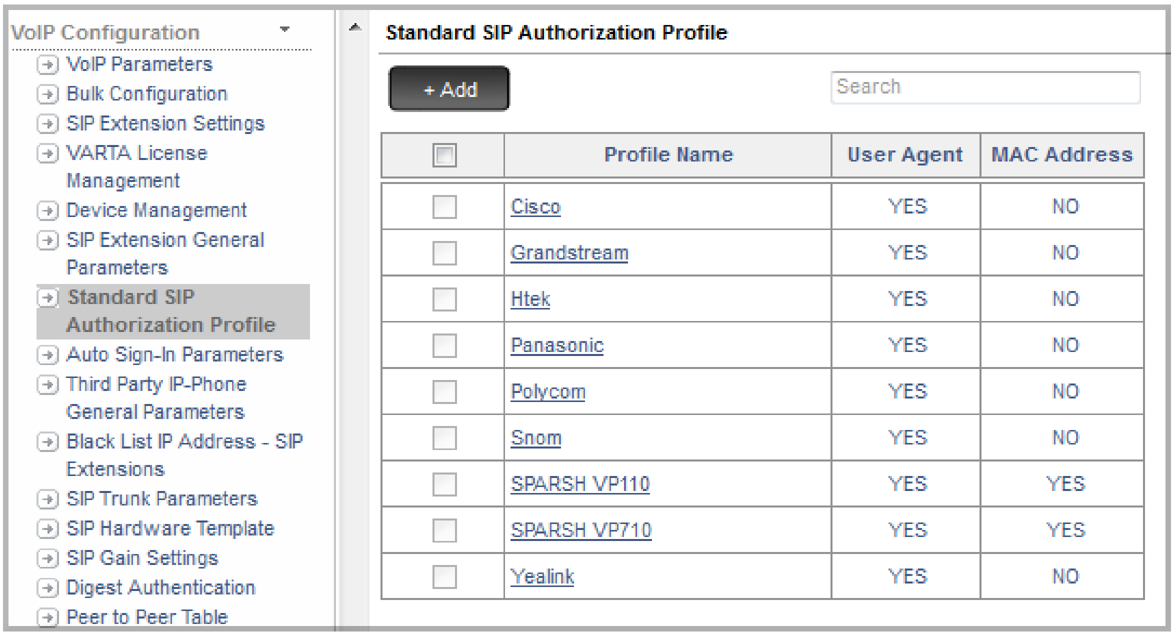
Using the Standard SIP Authorization Profile, you can register
- Matrix SPARSH VP110
- Matrix SPARSH VP710
- Third-party Standard SIP Phone
- Any Standard SIP Phone

When you configure the Device Type in Location - 1/2/3 of **SIP Extension Settings**, the default Standard SIP Authorization Profile is assigned to the phone. However, you may change the profile by selecting the desired option in **Standard SIP Authorization Profile** drop down list. To know more, refer "Configuring Standard SIP Phones".

Any changes in the assigned Standard SIP Authorization Profile may unregister the phones thereby, causing drop of ongoing calls.

Configuring Standard SIP Authorization Profile

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **Standard SIP Authorization Profile**.



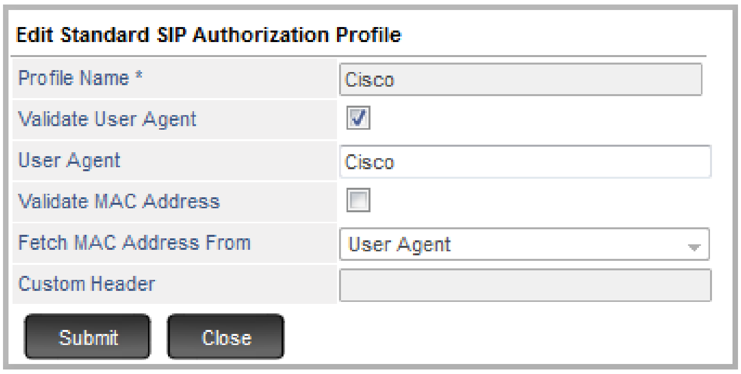
The list of default profiles supported by the system is displayed.

You can also add, edit, search or delete the Standard SIP Authorization Profiles according to your preference.

You cannot delete the default Standard SIP Authorization Profiles supported by the system.

Let us consider, that you want to register a Cisco Standard SIP Phone,

- Click **Cisco**.



The Standard SIP Authorization Profile details are displayed.

- **Profile Name** displays the name of the profile of the Standard SIP Phone.
- Select the **Validate User Agent** check box if you want the system to validate the User Agent received during phone registration request. Default: Enabled.
- In **User Agent**, enter the details which you want the system to match with the User Agent field received from the phone. This parameter is applicable only if you have enabled the **Validate User Agent** check box.
- Select the **Validate MAC Address** check box if you want the system to validate the MAC Address received during phone registration request.

*You are recommended to enable **Validate MAC Address** to prevent any unauthorized access and misuse of the system.*

- In **Fetch MAC Address From**, select the desired option - **User Agent** or **Custom Header**.

  If you select **User Agent**, the system will fetch the MAC Address from the User Agent field received during phone registration request.

  If you select **Custom Header**, the system will fetch the MAC Address from the configured Custom Header field during phone registration request.

  - In **Custom Header**, enter the Header name from where the MAC Address is to be fetched. For example, MAC. This parameter is applicable only if you have selected the option Custom Header in Fetch MAC Address From.

- Click **Submit**.

Similarly, you can configure the **Standard SIP Authorization Profile** of any Standard SIP Phone.

Customizing Profile

To add a new Standard SIP Authorization Profile,

- Click + Add and enter the details as per your requirement.
- Click **Submit**.

  The newly added Profile will get updated in the list.

To delete a Profile,

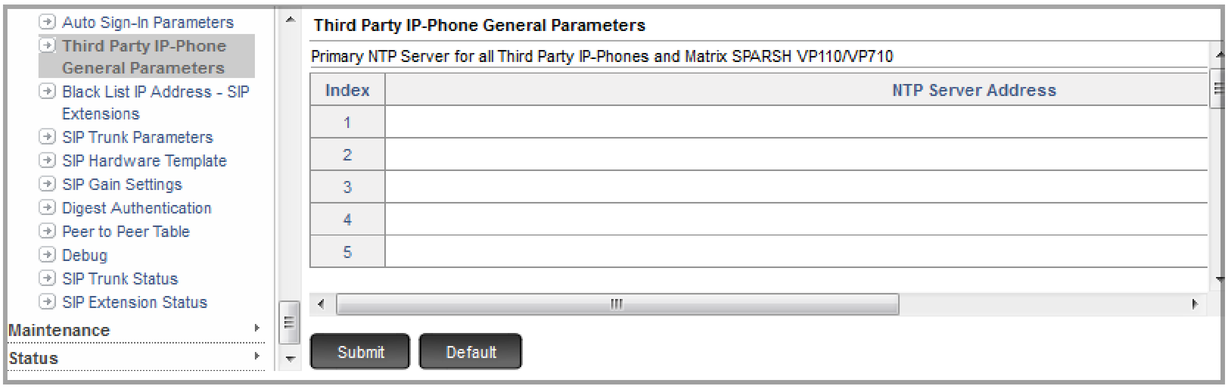
- Select the corresponding check box and click .

Third Party IP-Phone General Parameters

To configure the Third Party IP-Phone General Parameters,

- Under **Configuration**, click **VoIP Configuration**.
- Click **Third Party IP-Phone General Parameters**.

  The Third Party IP-Phone General Parameters page opens.



Primary NTP Server for all Third Party IP-Phones and Matrix SPARSH VP110/VP710

- Under **NTP Server Address**, configure the Primary NTP Server Address with which you want to synchronize the Date and Time of the Standard SIP Phones. Default: Blank.

  You may configure maximum 5 different Server Addresses.

Language File Download Path for Cisco IP-Phone

- Under **Path**, configure the path from which you want the Standard SIP Phone to fetch the Language files. Default: Blank.

  You may configure maximum 5 different Paths.

Phone User Language File Download Path for Snom IP-Phone

- Under **Path**, configure the path from which you want the Standard SIP Phone to fetch the Language files for the Phone user. Default: Blank.

  You may configure maximum 5 different Paths.

Web User Language File Download Path for Snom IP-Phone

- Under **Path**, configure the path from which you want the Standard SIP Phone to fetch the Language files for the Web user. Default: Blank.

  You may configure maximum 5 different Paths.

- Click **Submit**.

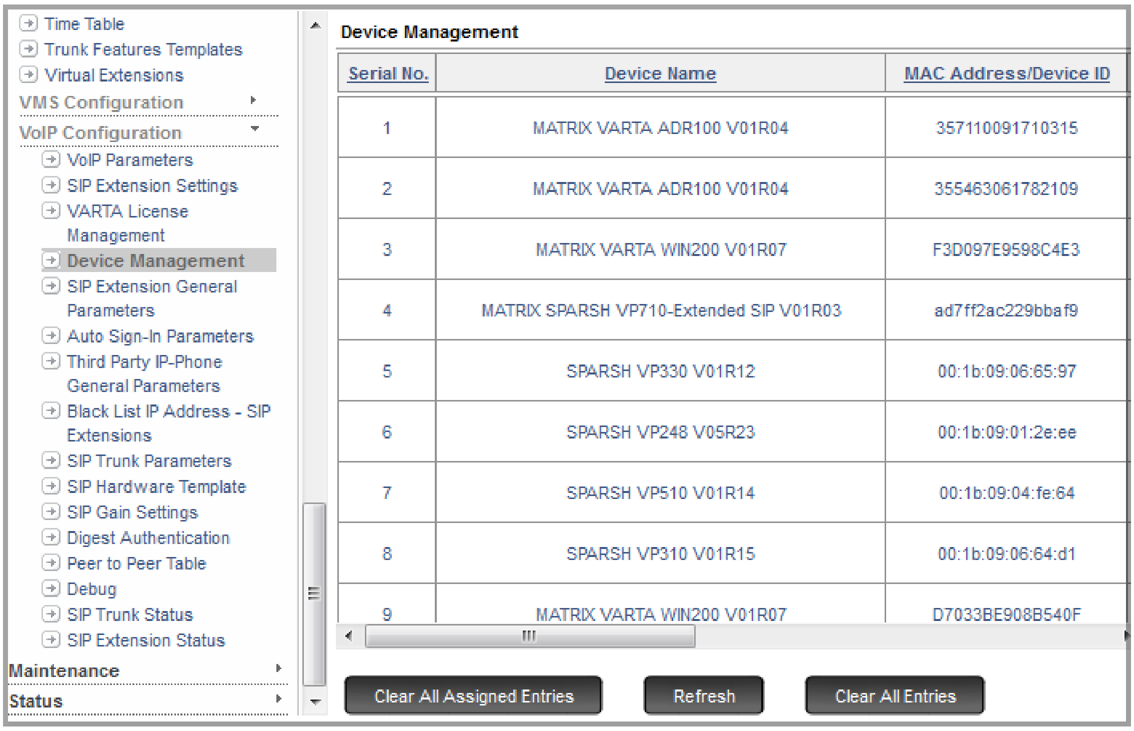
The Server Addresses/Path you configure on this page will appear in the Combo box of the respective parameter on the SIP Extension page.

Device Management

ANANT UCS supports Auto Detection and Auto Provisioning of third party IP Phones as well as Extended Clients. Device Management is a touch-free, plug and play feature and is an ideal solution for a large deployment of phones.

Once you connect the phones the details will be displayed on the Device Management page.To use this feature,

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **Device Management.**



- The following information will be displayed for each connected phone:
  - **Device Name:** It displays the name of the phone you connect/application you register.
  - **MAC Address / Device ID:** It displays the MAC or Device ID of the phone.
  - **IP Address:** It displays the IP Address of the phone.
  - **Last Seen:** It displays the date and time, when the system detects the connected phone.
  - **Assign:** If the phone is already configured, it displays the Name, Extension Number and Location Number. If you wish to edit the details, click on this detail link.

    If the device is not configured, it displays ‘Not Assigned’. If you wish to assign an extension to the phone, click on the **Not Assigned** link.

A new **Assign Extension** window opens. You can configure/edit the details:

- **Select SIP Extension**: Select the SIP Extension Number you want to assign to the phone.
- **Name**: Enter a name for the SIP Extension, which may be the name of the person who will use the SIP Extension or the name of a Department. The name you enter here will be displayed as the Caller ID of the SIP Extension on the remote user's phone, when the SIP Extension user makes calls. The name may consist of a maximum of 18 alphanumeric characters.
- **SIP ID:** Enter the SIP ID for the extension. The SIP ID is necessary for registering the SIP Extension with the Registrar of the system. It is the number with which you can call the SIP Extension. Any extension user of the ANANT UCS can call a SIP Extension by dialing the SIP ID assigned to the SIP extension. SIP ID of each SIP Extension must be a unique number string of a maximum of 6 digits. Any combination of digits from 0 to 9 and the characters * and # are allowed.
- **Authentication ID:** Enter the number which you want the system to use for user authentication of the SIP messages received from the SIP Extension. You cannot keep this field blank. The number may be a string of maximum 6 alphanumeric characters. All ASCII characters except < > and “ (double quote) are allowed.Default: Blank.

- *You must configure the Authentication ID, if any of the SIP Message Authentication Options, namely INVITE or SUBSCRIBE or PUBLISH, is enabled.*
- *Make sure the User ID configured in “Digest Authentication” does not conflict with the Authentication ID configured above.*

- In the **Authentication Password**, enter the password manually or click **Generate** to automatically generate a unique password. This password will be used by the system to authenticate the SIP messages received from the SIP Extension.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential. When the password is entered manually, it must:
  - be of minimum 6 characters and can be a maximum of 12 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
  - all ASCII characters (except Percentage **%**, Hash **#,** Equal to **=,** Plus **+**, And **&**, Backslash **\,** Less than **<,** Greater than **>,** Apostrophe **’,** Double Quote **"** and **Space**) are allowed.
    Default: Blank.

  To provide additional security, when the Authentication fails 10 times consecutively due to wrong Authentication ID / Authentication Password, the system will blacklist the IP Address and Port for registration of this SIP Extension. However, you can remove the IP Address from the Blacklist IP Address list. See “Black List IP Address - SIP Extensions” for more details. This activity will be logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

- **Select Location:** Select the location 1, 2, or 3 on which you want the extension to register. However, if you are configuring any third party IP Phone, then select location 1.
- **Location Name:** Enter the name that you want the system to display for the location you selected.
- **Registrar Server Address[49]:** Select the appropriate Registrar Server Address to register the device with the SIP Registrar of ANANT UCS, according to your installation scenario.

49. *When registering the VARTA clients, select the appropriate Internal / External Registrar Server Address according to your installation scenario.*

- Click **Submit**. The window closes and the details are displays on the Device Management page.

- **Reboot:** Click this button to reboot the phone remotely.

- *The phone will reboot only if it supports remote reboot.*
- *For Grandstream, Snom and Polycom IP Phones reboot is not supported.*

- Click **Clear All Assigned Entries**, to clear all the assignments.

Black List IP Address - SIP Extensions

Blacklist IP Address enables you to restrict unauthorized access to ANANT UCS.

To use this feature, you must enable the *Allow SIP Extensions Registration* check box and select the desired option for *Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts* in the "Security Settings".

ANANT UCS blacklists the IP Address from which an unauthorized attempt is made for registration. When any user attempts to register as a SIP Extension using false credentials— Authentication ID or Authentication Password and the authentication attempt fails for 10 times consecutively within 10 minutes, ANANT UCS blacklists the IP Address and port used for registration.

The blacklisted IP Address/es and ports are stored in the **Black List IP Address - SIP Extensions** table along with the date and time. This activity will also be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

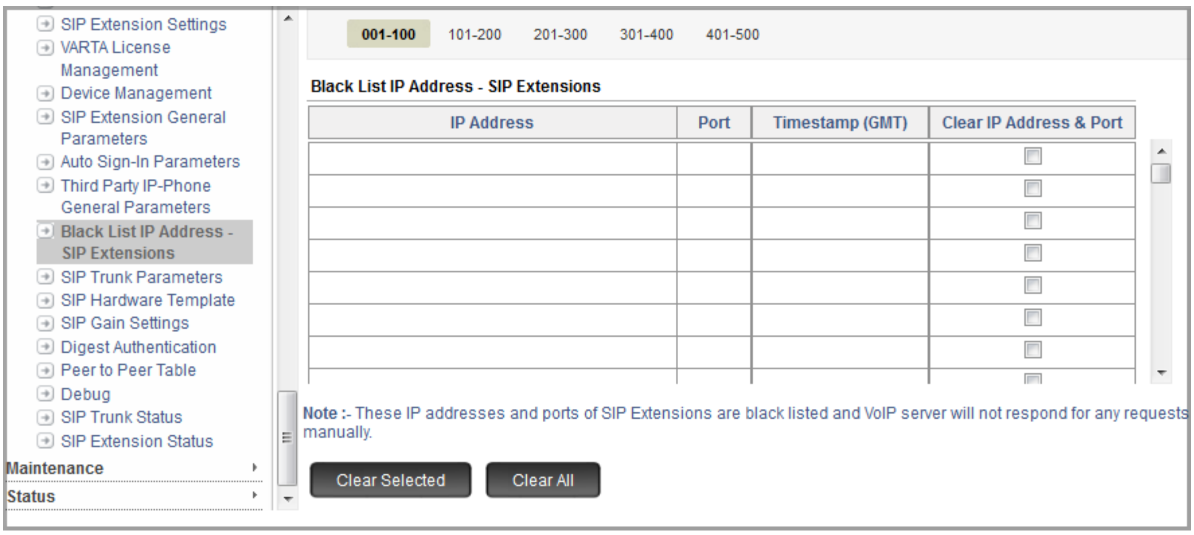
If you want to allow access of ANANT UCS to such black listed IP Address, you must remove it from the **Black List IP Address - SIP Extensions** table. If this IP Address is a Trusted IP Address, you can configure it in the Trusted IP Address/es table to avoid further black listing. For details, see "Security Settings".

Blacklist IP Address table stores upto 500 entries. When the buffer is full, ANANT UCS follows FIFO method to store further entries. You cannot edit any entries in this table. However, if required, you may remove a entry from this table.

To clear a entry from the **Black List IP Address - SIP Extensions** table,

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click the **Black List IP Address - SIP Extensions**.

  The Black List IP Address - SIP Extensions page opens.



- Select the **Clear IP Address & Port** check box of the entries that you want to remove from the Black List IP Address - SIP Extensions table.
- Click the **Clear Selected** button. The selected entries will be removed from the table.
- Click **Clear All** button to clear all the entries stored in the table.

DSS Keys Programming

The DSS (Direct Station Selection) Keys when personalized, provides you quick access to Stations, Trunks, Features/Functions of ANANT UCS. A few DSS Keys are provided on the phone itself. The number of DSS Keys provided on the phone depends on the type of phone. For details, “Extended IP Phone/VARTA UC Client - Operation”.

Matrix offers the DSS Consoles when attached to the phone, works as extensions to the phone. You can customize the DSS Console Keys as per your requirement by assigning the desired features/functions. For details, see “Direct Station Selection Console”.

To configure the DSS Keys on the phone, refer “Customizing Key Templates” and “Personalizing Key Maps”.

To configure the DSS Console Keys, refer “Programming DSS Console Keys”.

Key Templates

SPARSH IP Phones of Matrix, can be the extension of the Operators and Executives in an enterprise, and in hotels, it can be the extension of the Front Desk Attendants and Guests.

Each of these groups of users may require a different set of features on their phones. For example, when SPARSH is an Operator's extension, for efficient call management, more DSS keys may be required for Trunk Access, Call Appearances, Call Release, Direct Station Calling, than for accessing features.

When SPARSH is an Executive's extension, more DSS keys may be required for single-touch access to features, and fewer keys for Trunk Access and Direct Station Calling.

Similarly, when SPARSH is a Hotel Attendant's extension, keys are required for specific hotel functions such as Check-In/Check-Out, Changing Room Clean Status, Room Shift, etc. But, a different set of keys with special functions are required when SPARSH is provided as a guest extension, because guests are allowed only a limited number of features of ANANT UCS, such as calling the Front Desk/Floor Service, setting Do Not Disturb, Wake-up Calls, Call Forward, and checking Voice Mail.

Given the varying requirements of these groups of extension users, ANANT UCS provides programmable Templates of Key Maps for the Operator, Executive1, Executive2, Executive3, Hotel Attendant and Hotel Guest.

The default Key Maps in these Templates can be customized to match the requirement of the intended user group. The default Key Maps of Executive1, 2 and 3 are the same.

The customized Template is assigned to the SPARSH IP Phones. For example, you may customize the Key Template for the Operator and assign it to the Operator. Likewise, you may customize the Key Template for Executive and assign it to the Executive extensions.

The default Key Maps vary according to the models of the SPARSH IP Phone in use, as illustrated below.

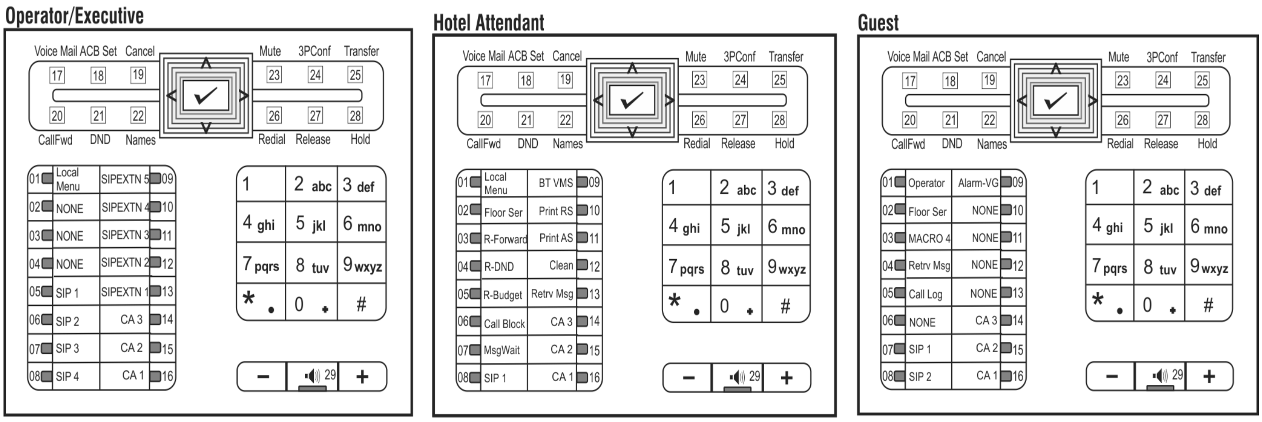
In case of Extended IP Phones, you can customize the Templates for the Operator, Executive1, Executive2, Executive3, Hotel Attendant and Hotel Guest as well as add new templates. You can add upto 64 templates. The new template you add, can be edited or remove as per your requirement. However you cannot remove the default Templates — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Hotel Guest. These can only be customized.

You can also Personalize the SPARSH IP Phone key maps, see “Personalizing Key Maps” for instructions.

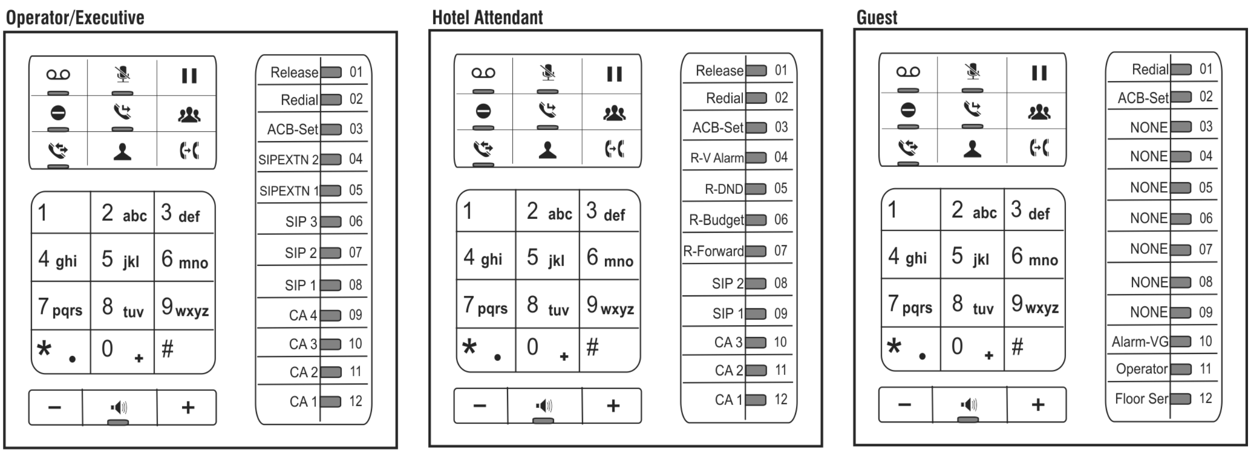
The keys in the Key Templates are numbered only for the purpose of locating the keys when programming. Key numbers do not appear on the key labels on the phone body.

Default Key Maps

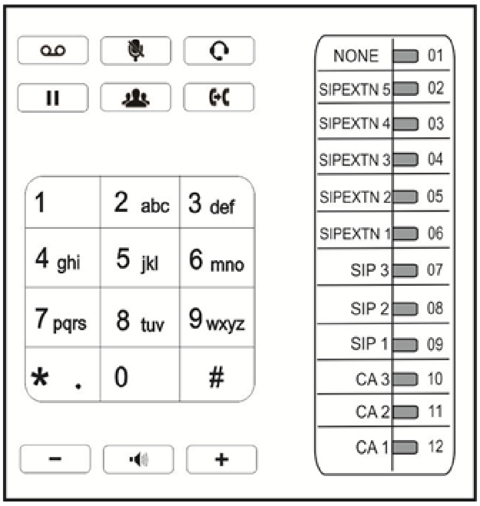
SPARSH VP248 Key Template (default)



SPARSH VP310 Key Template (default)

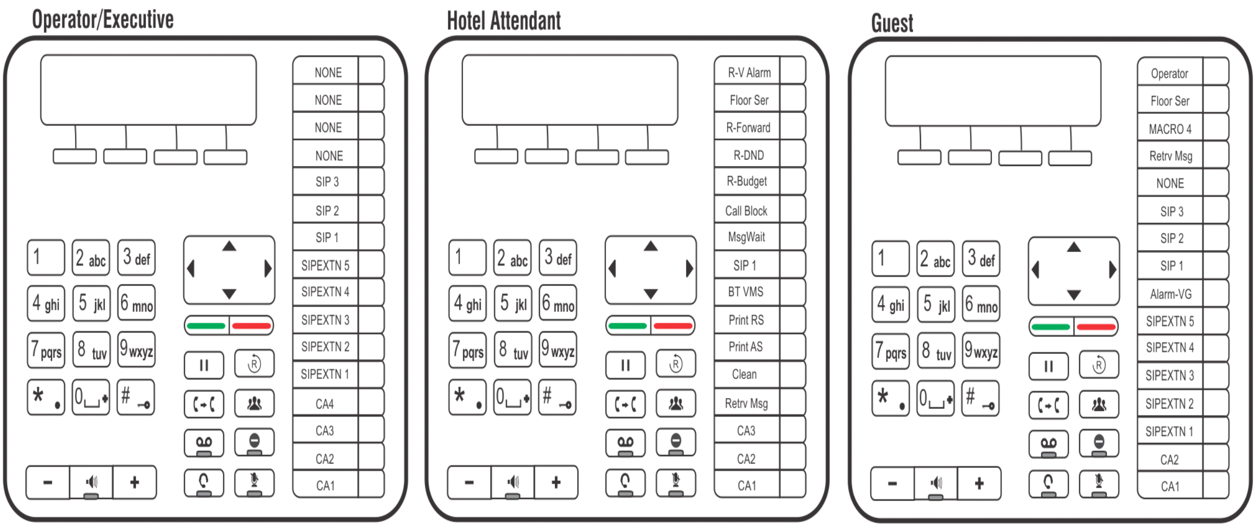


SPARSH VP330 Key Template (default)



The default Key Template is same for the Operator, Executive, Hotel Attendant and Guest.

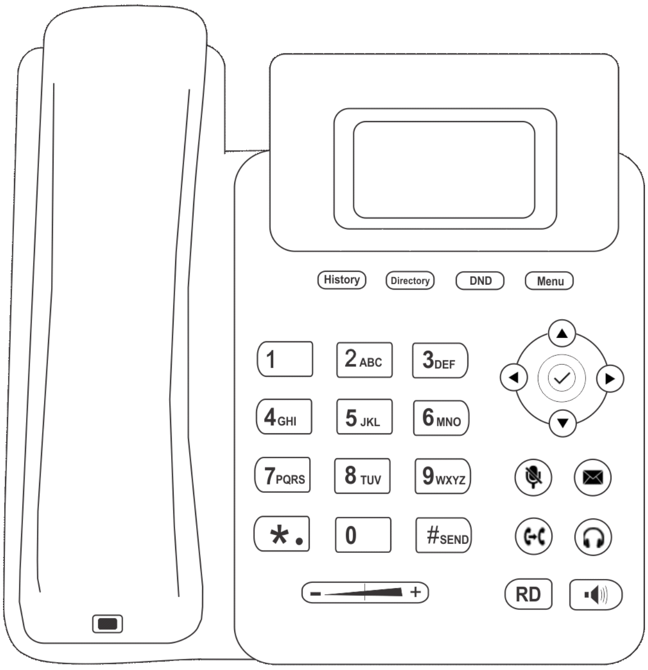
SPARSH VP510 Key Template (default)



SPARSH VP110

All the default templates are same for SPARSH VP110.

You can customize the following keys only — History, Directory, DND, Menu, Up, Down, Right, Left, OK, Mute and Transfer.

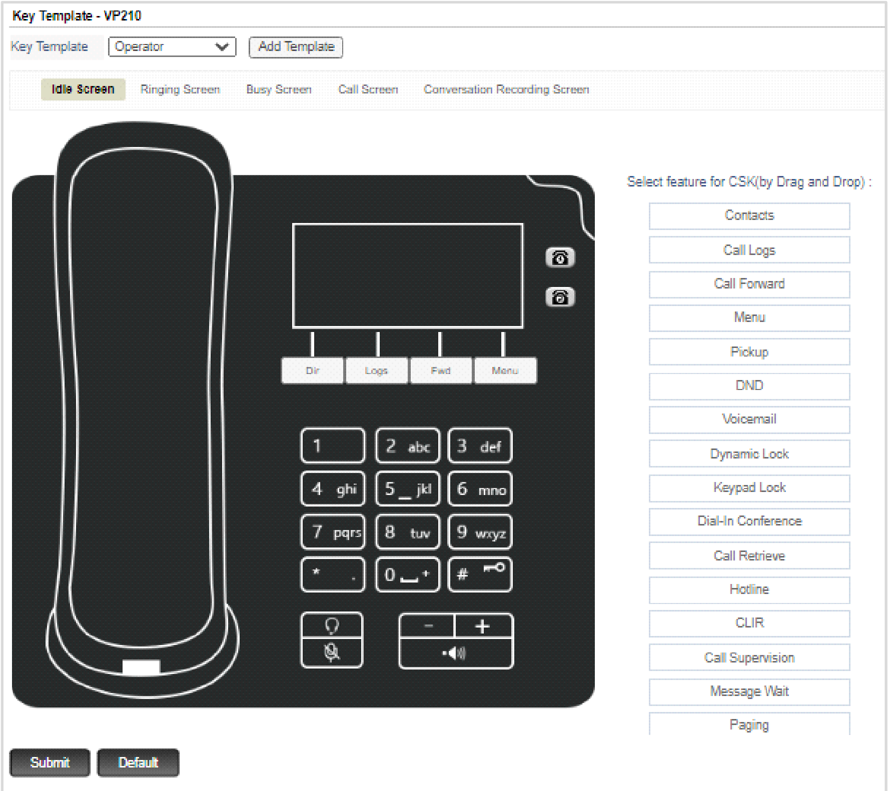


By using Key Templates you can prepare and assign common key maps to all or as many SPARSH IP Phones as you want, at one go.

ANANT UCS also offers the flexibility to personalize the Key Maps of each SPARSH IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 IP Phones, but you want to reassign some of the keys on two of these IP Phones, ANANT UCS allows you to selectively personalize the key maps of these two IP Phones.

SPARSH VP210 (Default)

All the default templates are same for SPARSH VP210.You can customize the Context keys only.



Customizing Key Templates

You can customize the Key Templates for:

- SPARSH VP248
- SPARSH VP330
- SPARSH VP310
- SPARSH VP510
- SPASH VP110
- SPARSH VP210

- *Before you begin programming the keys,*
  - *List the features/facilities that you want to change in each of the existing (default) Key Templates of Operator, Executive, Hotel Attendant and Guest.*
  - *Create customized Key Templates on sheets of paper.*
  - *For each template, decide the keys that will be reassigned the features you listed.*
  - *You may use the key templates printed above to decide the position of keys.*
  - *For each template that you customize, list down the SPARSH IP Phones along with their Software port numbers (SIP Extension numbers).*
  - *Similarly, list the SPARSH IP Phones which are to be assigned personalized Key Maps, along with their Software port numbers.*

Customizing Extended IP Phone Templates

You can:

- customize the existing key templates — Operator, Executive1, Executive2, Executive3,Hotel Attendant and Guests.
- add new key templates.

Customizing Existing Key Templates

To customize existing key templates for SPARSH VP248, SPARSH VP330, SPARSH VP310, SPARSH VP510 and SPARSH VP110 follow the instructions given below.

To customize existing key templates for SPARSH VP210, refer to “Customizing Existing Key Templates - SPARSH VP210”.

- Login as System Engineer.
- Under **Configuration**, click **Key Template**.
- There are separate links for each model of IP Phone. Click the desired phone model. The respective phone model is displayed.
- Click **Key Template** to select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Guests — you want to customize.

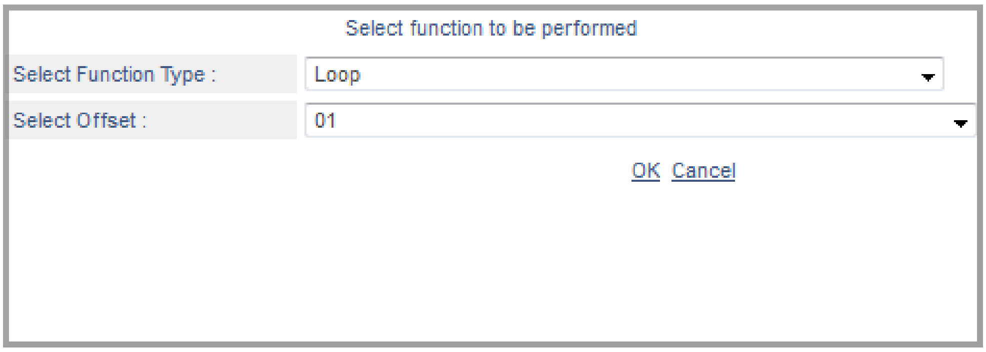
  Let us attempt to configure the sample Operator template for VP248.

  Click **VP248** to open the page. Select **Operator** as the **Key Template**.

For example, to customize the key template the keys that need to be reassigned features are as follows:

Existing function on the key	To be replaced by
CA 1	Hotline
CA 2	SIP Trunk 1 (SIP1)

- To assign Hotline, click the CA 1 key. A new window opens.

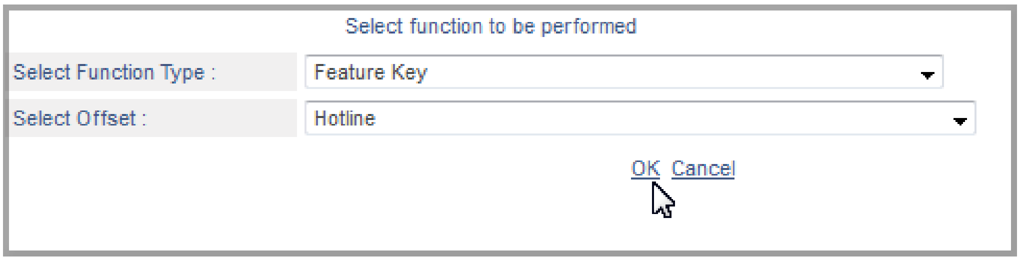


- As Hotline is a feature, in the **Select Function Type** list, click **Feature Key**.

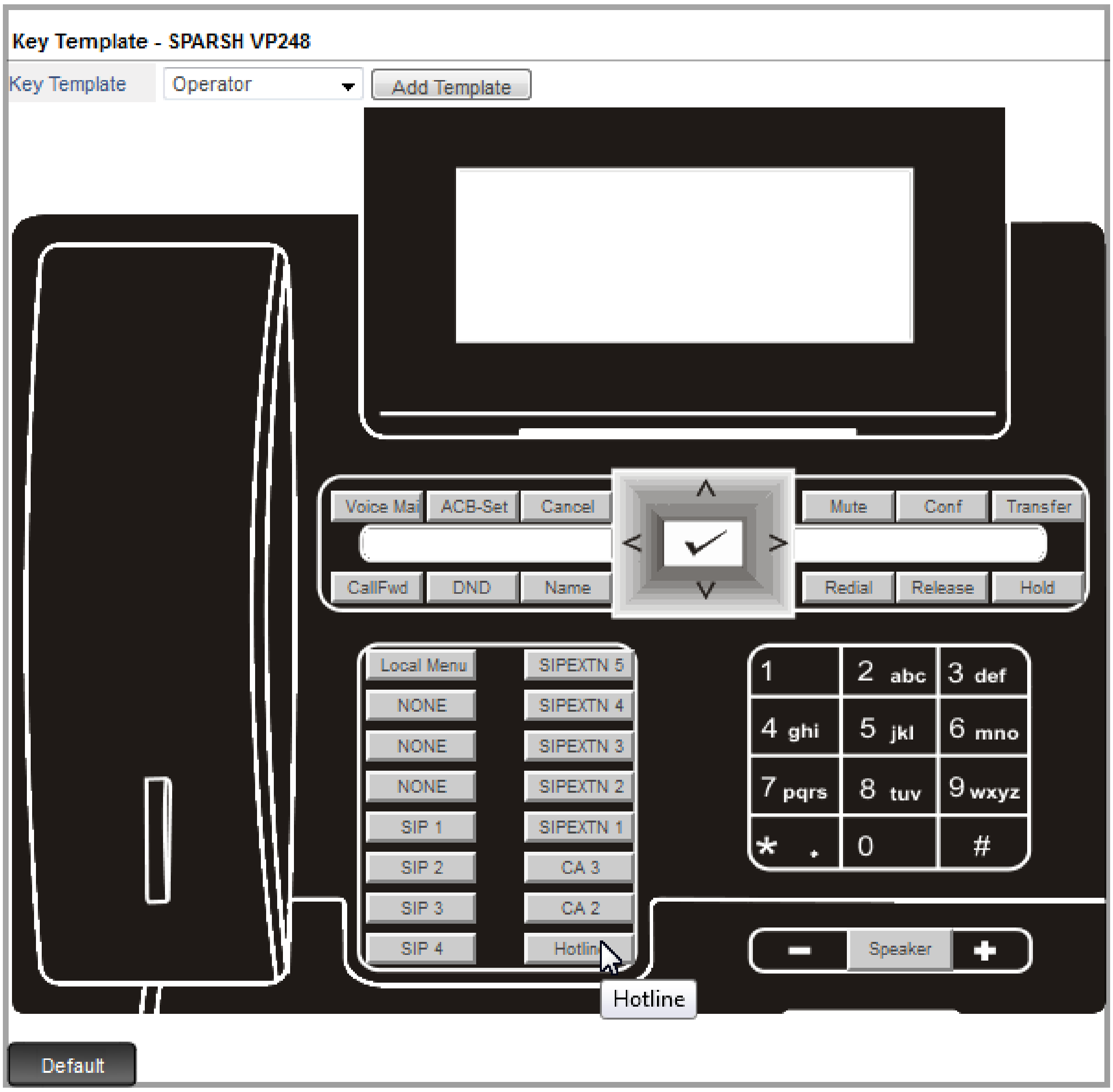
  All features that can be assigned to keys will appear in the **Select Offset** list.

- In the **Select Offset** list, click **Hotline**.

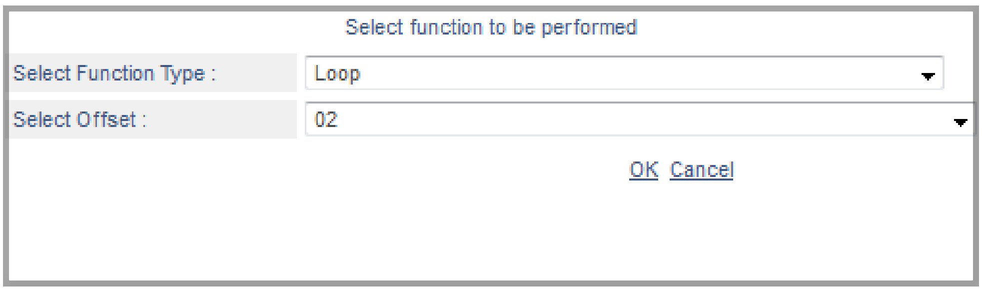
- Click on **OK** in the dialog box. The box will close.



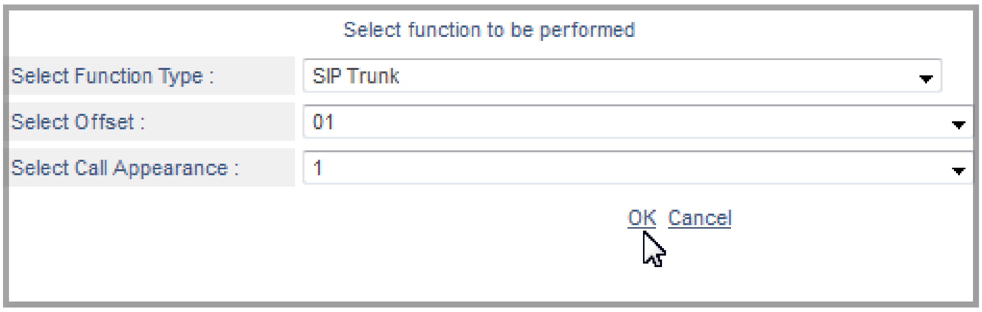
- The **Hotline** feature will appear on the key label.



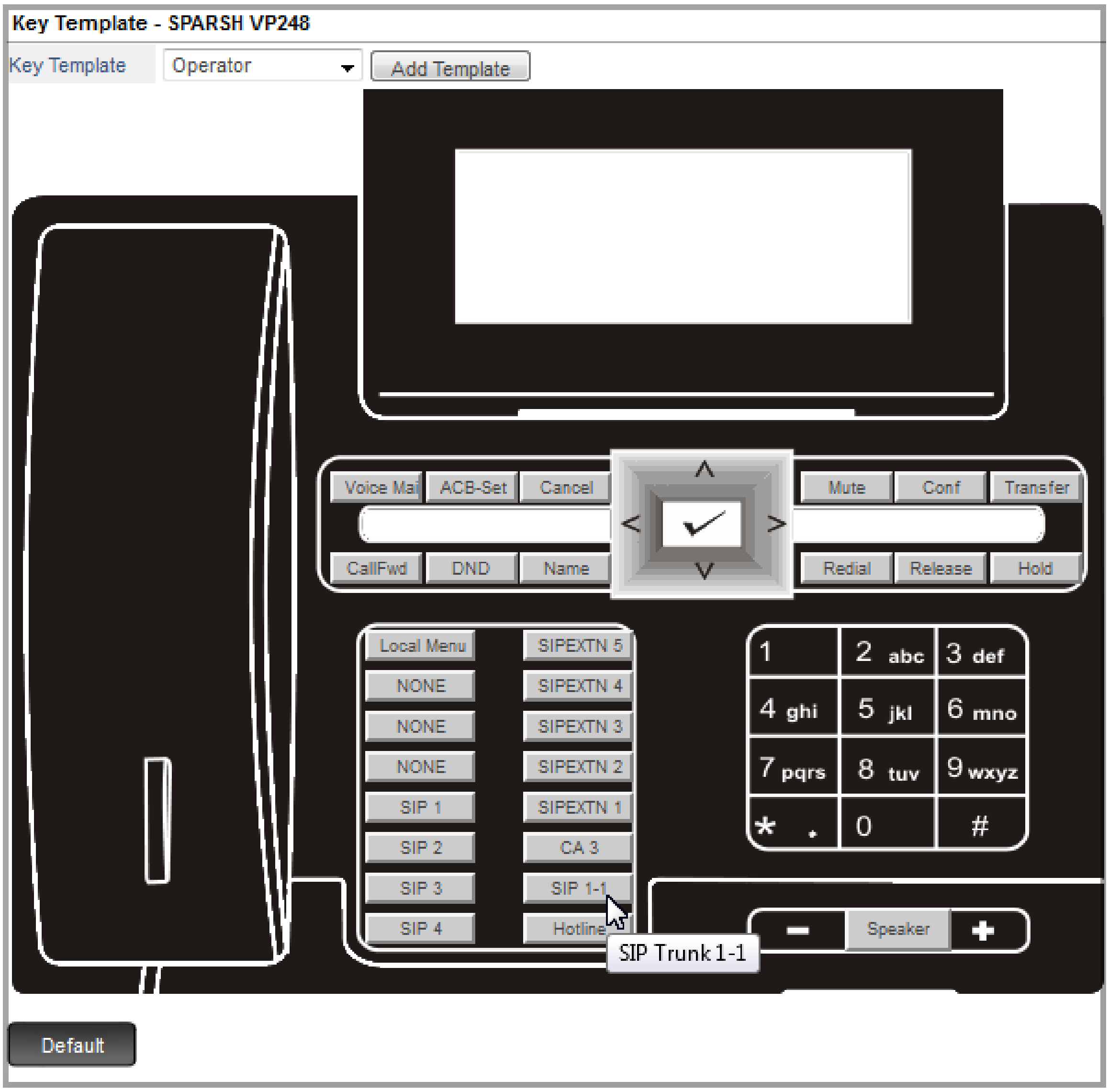
- To assign direct access to SIP Trunk 1, click CA 2 key.



Select **SIP Trunk** as function type and **01** as Offset.



SIP 1-1 appears as the key label.



When you have completed assigning functions to keys, click submit to save the settings.

Similarly, you can customize the other templates — Executive, Hotel Attendant, Guest.

To customize the templates for other phone models, follow the instructions as given above.

Customizing Existing Key Templates - SPARSH VP210

SPARSH VP210 has the provision to program the four Context Keys. These keys enable you to access the most frequently used functions/features at the press of a single button.

The screens — Idle Screen, Ringing Screen, Busy Screen, Call Screen, Conversation Recording Screen, all have different set of features that can be accessed. SPRASH VP210, enables you to customize these by allowing you to set the priorities of the features in each type of screen as per your preference. You can assign the features to the Context Keys depending on the state of the call.

- In the Idle Screen you can assign the desired feature/function to the Context Keys as well as set their priorities as per your requirement.

- In the other Screens you can only set the priorities of the features.

Refer to the details mentioned below for Default Key Assignment and the Feature Key Assignment/Feature Priority Assignment asper the different Call States:

Idle Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Dir
Context Key 2	Logs
Context Key 3	Fwd
Context Key 4	Menu

Feature Key Assignment/Feature Priority Assignment

Type of Screen	Feature Priority Selection List
Idle Screen	Contacts
	Call Logs
	Call Forward
	Menu
	Pickup
	DND
	Voicemail
	Dynamic Lock
	Keypad Lock
	Dial-In Conference
	Call Retrieve
	Hotline
	CLIR
	Call Supervision
	Message Wait
	Paging
	Meet Me Paging
	Room Monitoring
	Intercom
	Follow Me
	Walk-In
	PIN Dialing
	Department Group Call Forward
	Open a Door
	User Status

Ringing Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Transfer Complete
Context Key 2	Auto Call Back
Context Key 3	Message Wait Set
Context Key 4	Next

Feature Priority Assignment

	Feature Priority Selection List
Ringing Screen	Transfer Complete
	Auto Call Back
	Message Wait Set
	Forced Answer
	Release
	End Call

Busy Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Auto Call Back
Context Key 2	Interrupt Request
Context Key 3	Barge-In
Context Key 4	Next

Feature Priority Assignment

	Feature Priority Selection List
Busy Screen	Auto Call Back
	Interrupt Request
	Barge-In
	Forced Call Disconnection
	Message Wait Set
	Transfer Complete
	Release
	Trunk Reservation
	End Call

Call Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Mute
Context Key 2	Hold
Context Key 3	Transfer
Context Key 4	Next

Feature Priority Assignment

	Feature Priority Selection List
Call Screen	Mute
	Hold
	Transfer
	Conference
	Personal Call Park
	VMS Blind Transfer
	Global Hold
	General Call Park
	Call Chaining
	Conversation Recording
	Release
	Flashing on Trunk
	Account Code
	Open a Door or ACK
	New Call
	End Call

Conversation Recoding Screen

Default Key Assignment

Parameter	Default Key Assigned
Context Key 1	Mute
Context Key 2	Hold
Context Key 3	Transfer
Context Key 4	Next

Feature Priority Assignment

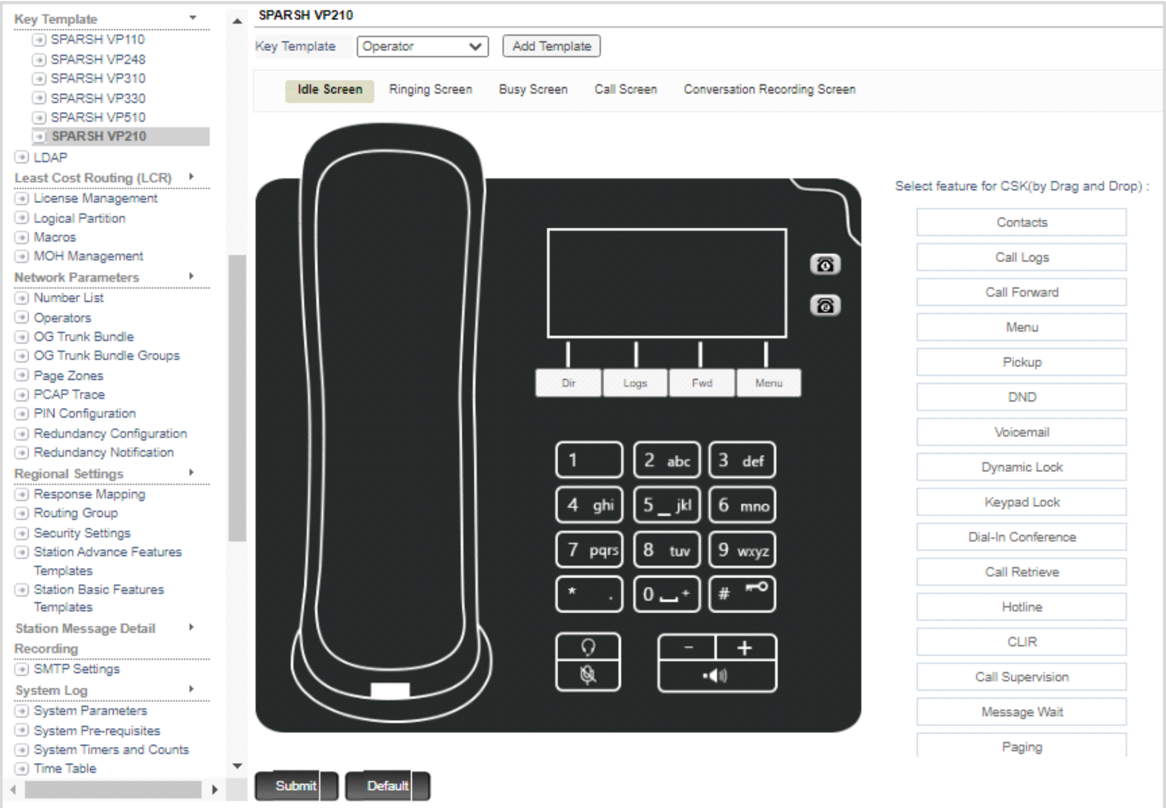
	Feature Priority Selection List
Conversation Recording Screen	Mute
	Hold
	Transfer
	Conference
	VMS Blind Transfer
	Global Hold
	Call Chaining
	Stop Recording
	Release
	End Call

To customize existing key templates for SPARSH VP210, follow the instructions given below.

- Log in as System Engineer.
- Under **Configuration**, click **Key Template** to open the page.
- There are separate links for each model of IP Phone. Click **VP210**.
- Click **Key Template** to select the desired key template — Operator, Executive1, Executive2, Executive3, Hotel Attendant and Guests — you want to customize.

  Let us attempt to configure the sample Operator template for VP210.

  Click **VP210** to open the page. Select **Operator** as the **Key Template**.



- Click **Idle Screen**.
- Each Context key, 1 to 4 can be assigned features.
- The feature assignment cum priority list appears on the right. You can change the feature assignments/ priorities as per your preference.
- To set the priority, drag and drop the features in the order of your preference. This will have two implications — the Context Key will be assigned the desired feature as well it will set the priority.
- Click **Submit**.
- The key map will refresh and the name of the Feature you selected (first four) will appear in abbreviated form as the key labels.

Menu must be assigned to one of the first four Context Keys.

- Similarly, you can click **Ringing Screen, Busy Screen**, **Call Screen** or **Conversation Recording Screen** and can set the feature priorities as per your preference.

  In these screens only priorities can be set and the 4th Context Key will always be assigned to **More >** feature. The first three Context keys will display the features assigned priorities 1, 2 and 3. To access other features press **More >**. The features will be displayed as per their set priorities.

Similarly, you can customize the other templates — Executive, Hotel Attendant, Guest.

You can also add new templates. To do so, refer “Add, Edit or Remove Key Templates”.

To customize the templates for other phone models, follow the instructions as given above.

Add, Edit or Remove Key Templates

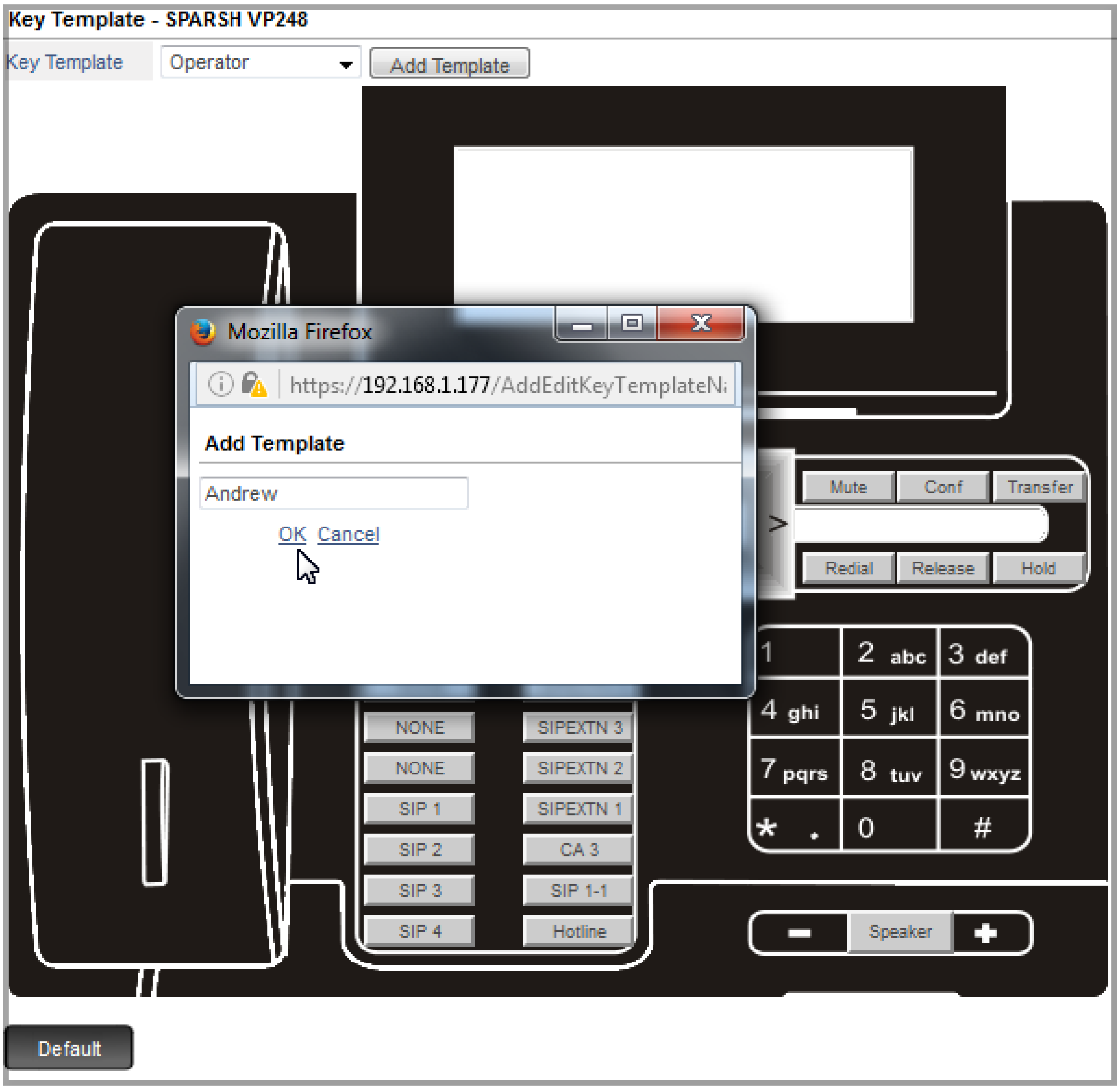
If you do not wish to customize any existing template, you can add new templates. By default, the Operator Template will be displayed as the phone key map. You can customize the key map as per your requirement or add a new template.

You can add upto 64 new templates for each phone model.

To add a template,

- Login as System Engineer.
- Under **Configuration**, click **Key Template**.
- There are separate links for each model of IP Phone. Click the desired phone model.The respective phone model is displayed.

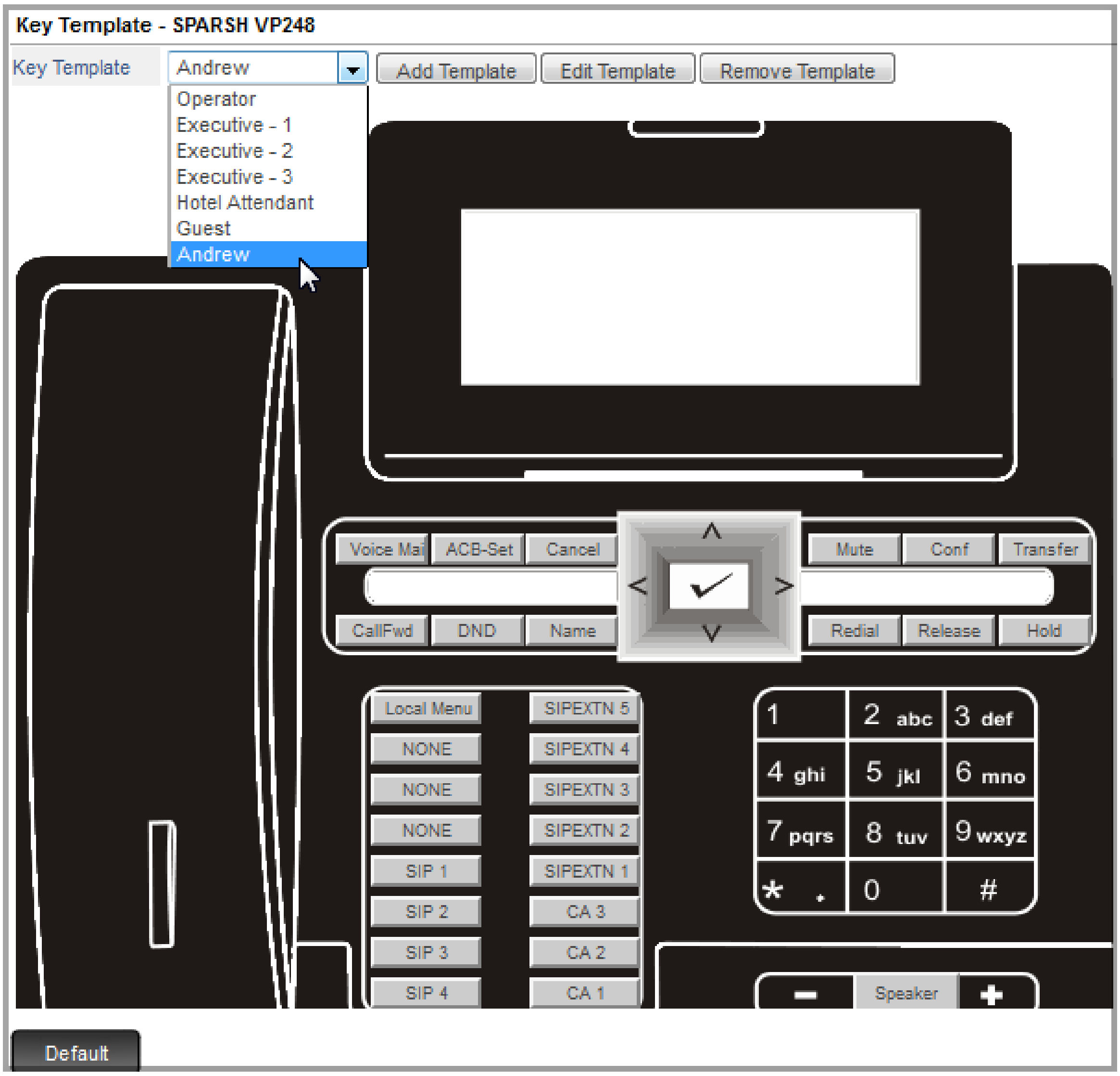
- Click **Add Template**. The Add Template window opens.



- Assign a name to the template. This template appears as one of the Key Template options.

- To customize the key template to match the requirement, click Andrew as the Key Template option.

Follow the instructions given under "Customizing Key Templates".



- When you have completed assigning functions to keys, follow the same steps if you wish to add other Key Templates.
- Assign the key templates you created to the respective SIP Extension location using Jeeves. See "DSS Key Settings" in "Configuring Matrix SPARSH VP330", "DSS Key Settings" in "Configuring Matrix SPARSH VP248", "DSS Key Settings" in "Configuring Matrix SPARSH VP510", "DSS Key Settings" in "Configuring Matrix SPARSH VP310" and "DSS Key Settings" in "Configuring Matrix SPARSH VP210" for instructions.
- To Edit the template name, click **Edit Template**.
- To delete the template, click **Remove Template**.

If you are customizing the key maps of SPARSH VP110, you can customize the following keys only — History, Directory, DND, Menu, Up, Down, Right, Left, OK, Mute, Transfer.

Personalizing Key Maps

You can personalize the Key Maps of individual SPARSH IP Phones, instead of using the Key Templates.

When you personalize the Key Map of an SPARSH IP phone, make sure 'Personalized' option as the *Key Template* in the *SIP Extension Settings - Location* of the phone.

The Personalized option is not applicable for SPARSH VP110.

For instructions on personalizing the Key Map of the Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP210” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”. The Personalized option is not applicable for SPARSH VP110.

Programming DSS Console Keys

Direct Station Selection (DSS) Consoles are devices that function as extensional buttons for SPARSH VP510 Phones, providing more buttons for single-touch Direct Station Calling and feature access.

You can attach four DSS532 Consoles to a single SPARSH VP510 to further increase the number of keys. Refer “Direct Station Selection Console” to know more.

For instructions to install the DSS532 with SPARSH VP510, see “Installing DSS532 with SPARSH VP510”.

Programming DSS Console Keys using Jeeves

You can configure the DSS Console either offline, that is before you have connected the DSS Console or after you have connected the Console.

To configure DSS Console keys, refer to “Configuring DSS Console Keys connected to SPARSH VP510”, if the console is connected to SPARSH VP510.

Configuring DSS Console Keys connected to SPARSH VP510

You can connect DSS532 to SPARSH VP510 for quick access to Extensions, Trunks, Features/Functions of ANANT or at the touch of a single key, making call operations easy.

You can configure the DSS Console keys offline, that is before connecting the console or after you have connected the Console. If you have connected the DSS Console the system will automatically detect the same.

- Login as System Engineer
- Under **VoIP Configuration**, click **SIP Extension Settings**.
- Select the desired SIP Extension Number, click the desired Location 1 or 2 or 3.
- On the Location Page make sure you have selected SPARSH VP510 as the Device Type. Scroll to **DSS Key Settings**.

- Click **Key Settings**. The **Key Settings** page opens.

Phone Details

- **Device Type** displays SPARSH VP510.
- **Assigned to** displays the Extension number assigned to the phone.
- **Port No** displays the software port number as well as the Location, for example: SIP Extension -1 - Location - 1.

- Click **ADD DSS**.

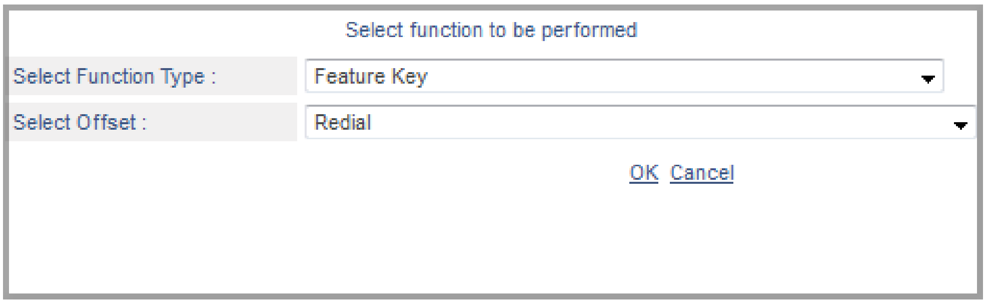
*If the DSS Consoles are connected the default Key maps will be displayed automatically. Click **ADD DSS** if you wish to add another DSS Console.*

- The default key map of DSS532 appears on your screen.

- By default **None** is assigned to all the DSS keys, you can now personalize the DSS Key map as per your requirement.

- Click the key on which you want to assign a feature/function.

- The options for the **Functions to be Performed** by the key will open in a new window.



- Select the desired **Function Type** to be assigned to the key and the desired **Offset** for the Function Type.
- Click **OK**.
- The function you selected will appear as the key label.
- Similarly, configure other keys.

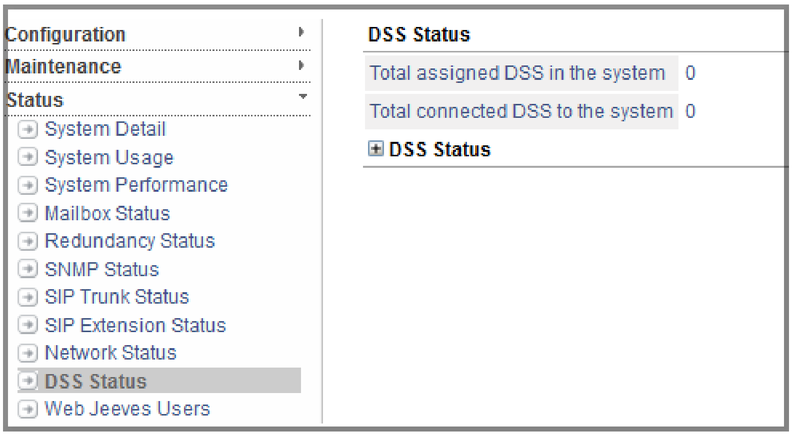
The following features cannot be assigned to DSS Keys:
- *Headset*
- *Speaker*
- *Ringer Acknowledge Key*
- *Local Menu*

- If you wish to clear all the key assignments, click the respective DSS Console check box and then click **Clear Key Assignment.**
- If you wish to interchange the key assignments of the DSS, click the check boxes of the respective DSS Consoles whose keys you wish to swap and then click **Interchange DSS Keys**.
- If you wish to remove a DSS Console, click **Remove DSS**.
- Click **ADD DSS** again to add another Console.
- To view the status of all the DSS Consoles, click "DSS Status".

DSS Status

To know the details of the DSS Consoles at a glance,

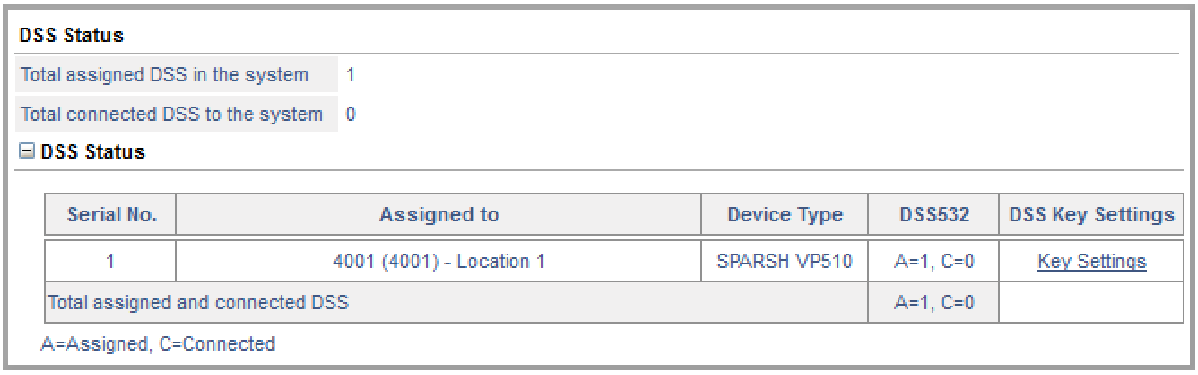
- Under **Status**, click **DSS Status**.



The following details are displayed:

- **Total assigned DSS in the system**
- **Total connected DSS to the system**

- Click **DSS Status** to expand.



Serial No.	Assigned to	Device Type	DSS532	DSS Key Settings
1	4001 (4001) - Location 1	SPARSH VP510	A=1, C=0	Key Settings
Total assigned and connected DSS			A=1, C=0	

The following information is displayed:

- **Serial Number**
- **Assigned to** displays the extension number to which the DSS Console is connected.
- **Device Type** displays SPARSH VP510, to which the DSS Console is connected.
- **DSS532** displays the status of the console, that is Assigned or Connected.
- **DSS Key Settings** provides the link to **Key Settings**. Click this link if you wish to configure the DSS Keys.

Configuring 'Operator'

Users understand the term 'Operator' as a person who handles multiple simultaneous calls and functions as the link between callers and called parties.

For the system however, an 'Operator' is a Routing Group; a group of extensions to which calls made by extensions by dialing '9' are to be landed. This also includes Auto Attendant calls on trunks during which the caller dials '9'.

Depending on the size of the Enterprise and the amount of call traffic to be managed, more than one Operator may be employed. Also, it is not uncommon to have different Operator extensions according to the time of the day. For instance, during working hours calls may be handled by the Receptionists or Front Desk Personnel, whereas during non-working hours, calls may be handled by the Security Personnel.

To meet this requirement, ANANT UCS offers configuration of up to 20 different Operators (Routing Group). However, at a time, only one Operator can be assigned to the extensions and trunks.

Each 'Operator' is assigned a Routing Group for the Time Zones - Working Hours, Break Hours and Non-Working Hours.

Each 'Operator' is assigned a Time Table, which defines the Working Hours, Break Hours and Non-working Hours for a week. The system follows this Time Table to assign a Routing Group as 'Operator' according to the current Time Zone.

Configuration of 'Operator' involves the following steps:

1. **Configuring Routing Groups as 'Operator':** A routing group may be made up of one or more than one extensions, depending on user requirement. If the user requires only one extension as 'Operator', include only one extension as member in the Routing Group for Operator. If the user requires five extensions as 'Operator', create a Routing Group of the five desired extensions to be used as 'Operator'.

   If the user requires Time-Zone based 'Operator', then prepare a different routing group for each Time Zone. If the user requires the same Operator for all Time Zones, use the same Routing Group number in all Time Zones.

2. **Configuring a Time Table for Operator:** This is applicable only when Operator extensions are different for different Time Zones.

   Define the Time Table to be followed for the Operator selection. The Time Table may be the same as the Time Tables assigned to trunks and extensions of ANANT UCS or may be configured to match with the timings of the persons who work as Operators. For example, if Operators in the Enterprise are working in shifts, the Time Table can be configured to match their timings.

3. **Assignment of 'Operator' to Extensions:** SIP Users and Virtual Extensions of ANANT UCS can be assigned to an 'Operator' in their Station Basic Feature Template.

   All extensions may be assigned to the same Operator, or different groups of extensions may be assigned to different Operators, so that call management is more efficient.

   Operator 1 is the default in the Station Basic Feature Templates. If you want to assign different extensions to different Operators, you must configure separate Station Basic Feature Templates with a different Operator for each extension group.

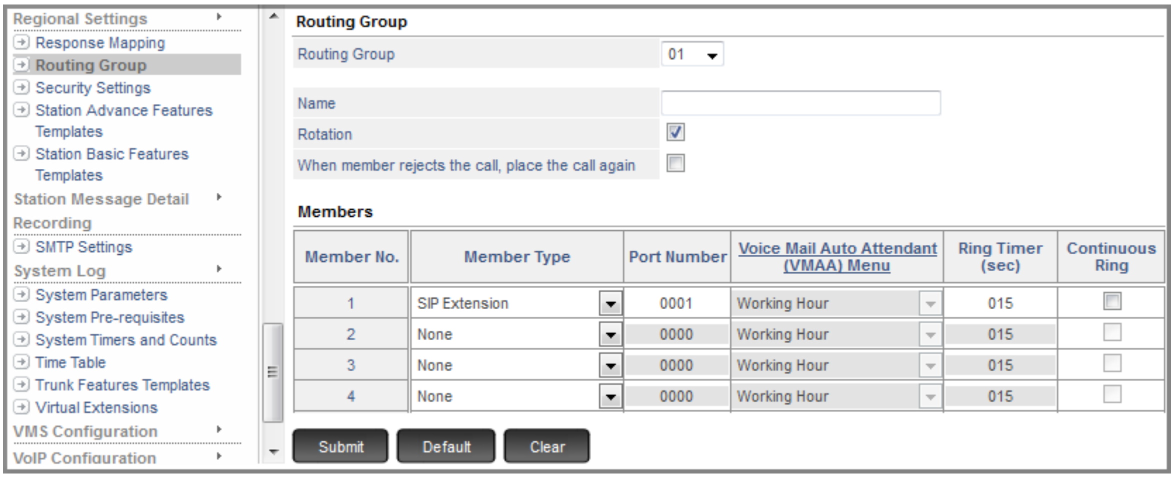
4. **Assignment of 'Operator' to Trunks:** Trunks are also assigned an 'Operator', so that when a caller dials '9' using Voice Mail Attendant, the call is routed to the Routing Group defined as Operator for the trunk for a particular Time Zone. For example, during working hours, a caller on trunk 01 dials '9', the call lands on 3001; when a caller on trunk 01 dials '9' during non-working hours the call lands on 3003 and when the caller dials '9' during break hours the call lands on 3002.

   Similarly, it is also possible to assign different Operators to different trunks.

Decide the number of Operators to be configured on the basis of the user's requirement.

Configuring Operator

- Login as System Engineer.
- Under **Configuration**, click **Routing Group**.



- Select a Routing Group Number from 1 to 96, you want to configure for Operator. By default, Routing Group 32 is assigned to Operator. You may configure this group, or select another one.
- For each Group you must configure the following:
  - Configure the **Name** you wish to assign to the Routing Group.
  - Select the **Rotation** check box to enable rotation of calls in the routing group which has multiple 'member' extensions. When enabled, each fresh call will land on the extension which is next to the one that received the last call. This ensures equal distribution of incoming calls to all the destinations within the routing group. The option is not relevant if the routing group has only one member extension.
  - If any member extension rejects an incoming call and the system again checks the routing group sequence, you can allow or restrict placing the same call on this extension. Select the **When member rejects the call, place the call** again check box**,** if you want the system to place the call again on the extension.
  - To configure **Members** in the Group,
    - Select the **Member Type**. You can select SIP Extension, Virtual Extensions, OTBG or the Voice Mail Auto Attendant.

Configure only as many extensions as you want in the routing group and set the remaining Member Types to 'None'.

For example: if you want to configure only one extension in the routing group, set the Member Type in the remaining columns (Member 02-Member 32) to 'None.'

- In **Port Number**, enter the software port number on which the SIP or Virtual Extension is connected.

  If you have selected OTBG then enter the OTBG number here.

- If you have selected the *Voice Mail Auto Attendant* as the Port Type, select the **Voice Mail Auto Attendant (VMAA) Menu** to assign to the respective routing group.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see "Voice Mail Auto-Attendant Menu".

- In **Ring Timer (sec)**, configure the time for which the extension, on which the call lands, should ring. By default, the ring timer is set to 015 seconds and can be changed.

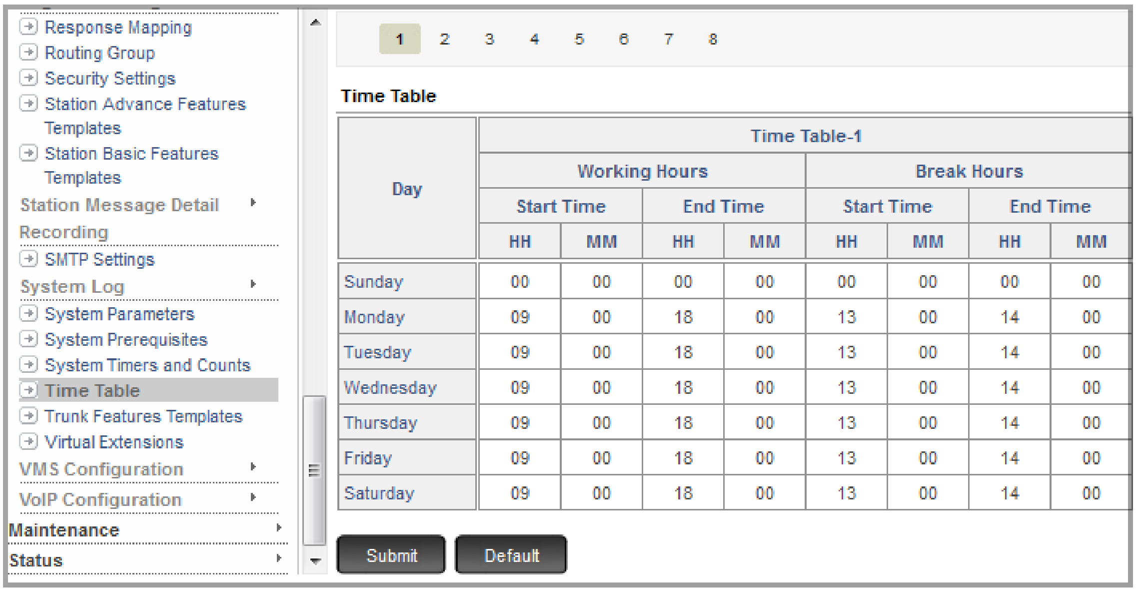
- Select the **Continuous Ring** check box, if you want an extension to ring continuously until the call is answered. The first extension will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This parameter is not relevant, if there is only one member extension in a routing group.

- Click **Submit**.

- Repeat the same steps to configure another Routing Group.

  If you have finished configuring Routing Groups for Operator, configure the Time Table for Operator.

- Under **Configuration**, click **Time Table**.



Day	Time Table-1							
	Working Hours				Break Hours			
	Start Time		End Time		Start Time		End Time	
	HH	MM	HH	MM	HH	MM	HH	MM
Sunday	00	00	00	00	00	00	00	00
Monday	09	00	18	00	13	00	14	00
Tuesday	09	00	18	00	13	00	14	00
Wednesday	09	00	18	00	13	00	14	00
Thursday	09	00	18	00	13	00	14	00
Friday	09	00	18	00	13	00	14	00
Saturday	09	00	18	00	13	00	14	00

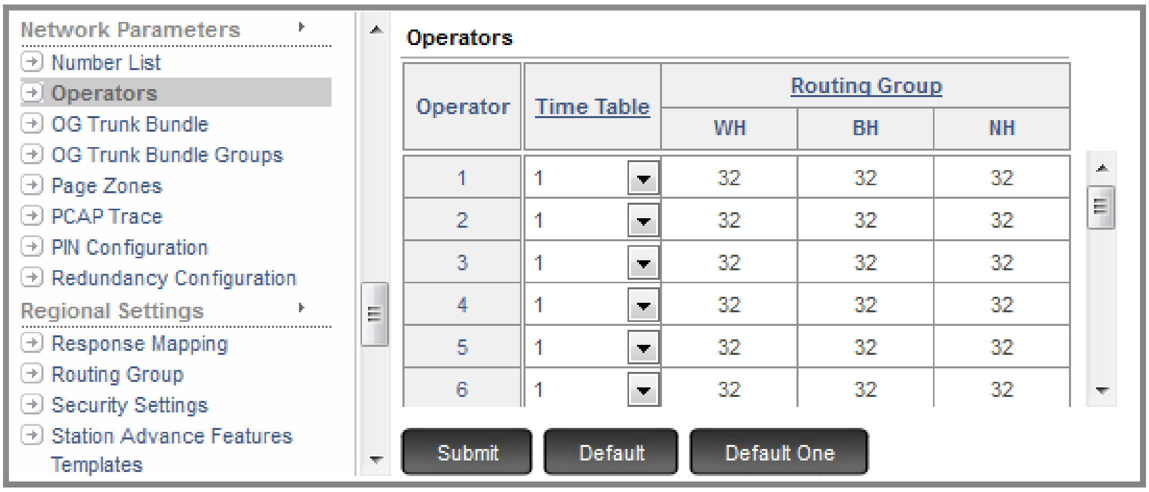
- By default, Time Table 1 is assigned to all Operators. If this Time Table meets your requirement, retain it. If not, select another Time Table. Customize it by defining the Working Hours, Break Hours and Non-Working Hours for the week.

- Click **Submit**.

- If you want to assign different Time Tables to different Operators, repeat the above steps to prepare the other Time Tables.

  If you have completed configuring the Time Table,

- Under **Configuration**, click **Operators**.



- Select the Operator number you want to configure. By default Operator 1 is assigned to all extensions and trunks.

- Select the number of the Time Table you prepared for the selected Operator. By default 1 is assigned.

- Enter the number of the Routing Group you prepared for the selected Operator for Working Hours, for Break Hours and for Non-Working hours. If the same Routing Group is to be kept for all Time Zones, enter the same number in fields of all three time zones.

- Click **Submit**.

- Repeat the above steps to configure another Operator.

  Now, you may assign the 'Operator' groups you have configured to the SIP extensions by configuring the number of the Operator (1-20) in the “Station Basic Feature Template” applied on these extensions.

  Similarly, you may assign the 'Operator' groups you have configured to the trunks in the “Trunk Feature Template” applied on these Trunks.

Extension Search

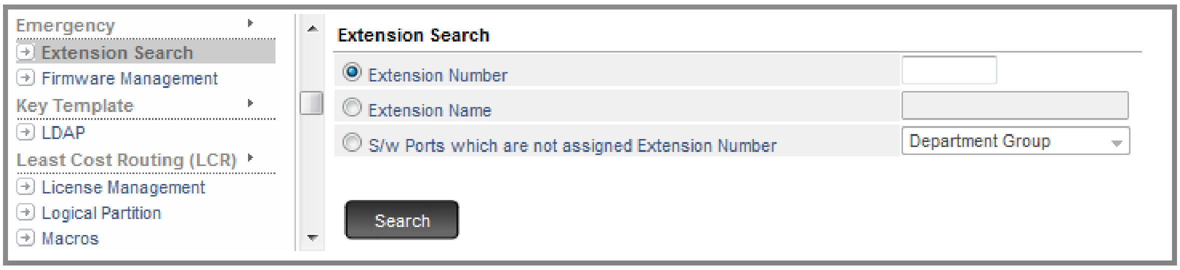
Using Extension Search, you can search for the extensions configured by you as well as find out the Software ports that are not assigned Extension Numbers.

You can search for any extension by:
- Extension Number
- Extension Name
- Software Ports not assigned Extension Numbers

- *The number of extensions that the system will search for depends on the total extensions configured in System Pre-requisites.*
- *If you are searching using Extension Name, make sure you enter the name as configured in the system.*

Configuring Extension Search

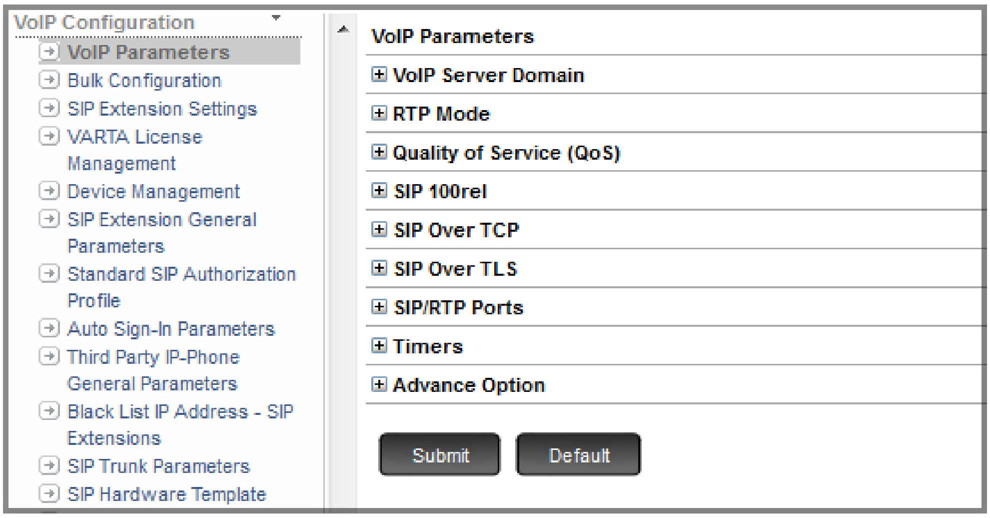
- Login as System Engineer.
- Under **Configuration**, click **Extension Search**.



- Select the desired **Extension Search** option from the following:
  - **Extension Number:** If you select this option, enter the *Number* of the extension you want to search.
  - **Extension Name:** If you select this option, enter the *Name* of the extension you want to search.
  - **Software Ports not assigned Extension Number:** If you select this option, select the type of extension— Department Group, SIP Extension, Virtual Extension — for which software ports are not assigned any extension numbers.
- Click **Search**. The search result appears on the screen.

Configuring VoIP Parameters

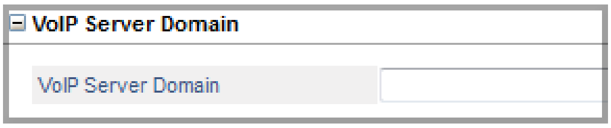
- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **VoIP Parameters**.



Changing any parameter on this page will result in call drop of all ongoing VoIP calls. All Open SIP Extensions registered will start functioning only after the next registration.

VoIP Server Domain

- Click **VoIP Server Domain** to expand.



- **VoIP Server Domain:** This parameter is of relevance only if you are configuring SIP Extensions.

  The system is capable of maintaining a domain for registering SIP clients (any SIP-enabled device) as SIP Extensions.

  Configure the Server Domain if you want SIP clients to register with the Registrar Server of the system using the domain handled by the system[50]. The Domain Name can be a maximum of 40 characters. Default: Blank.

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50. *SIP clients can be registered with the system either using the domain handled or using the WAN or LAN Port IP Address.*

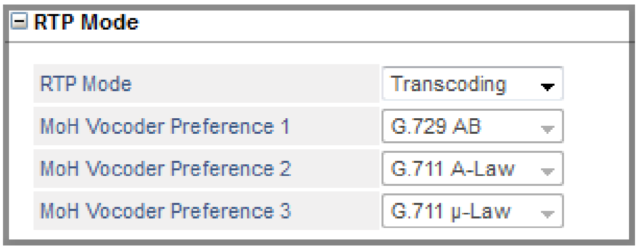
    If domain is configured, the system will listen for the SIP message which is redirected to the configured domain only. It will also listen for SIP messages on the WAN IP address and LAN IP address.

    But if domain is not configured, the system will listen for SIP messages only on the WAN IP Address and LAN IP address.

If you configure the Server Domain for registration of SIP clients, you must also map the Domain Name and the IP Address of the WAN Port to the DNS Server in the network.

RTP Mode

- Click **RTP Mode** to expand.



- **RTP Mode:** Select the desired RTP mode using which you want ANANT UCS to route speech in SIP to SIP calls.

  You can select from the following:

  - **Transcoding:** When this option is selected, RTP packets will be routed using transcoding[51] channels for SIP to SIP calls. SIP users (Extended SIP Clients and Standard SIP Clients) will be able to access all the features of ANANT UCS. This option uses two transcoding channels for SIP to SIP calls. Thus the maximum number of SIP to SIP calls per system is equal to the number of transcoding channels divide by 2.

  - **RTP Relay:** When this option is selected, RTP packets will be routed without using transcoding channels for SIP to SIP calls.The System will use transcoding channels for some features and Standard SIP Clients will be able to use limited features of ANANT UCS. Refer to "ANANT UCS Features supported with RTP/Direct RTP" for more details.

  - **Direct RTP:** When this option is selected, no transcoding channel will be used for SIP to SIP calls and RTP packets will be sent to and fro directly between SIP end points. If transfer of RTP packets is not possible between SIP end points, system will use RTP Relay as the fallback option. System will use transcoding channels for some features and Standard SIP Clients will be able to use limited features of ANANT UCS. Refer to "ANANT UCS Features supported with RTP/Direct RTP" for more details.

  - *Direct RTP is not supported on SIP Trunks and MATRIX VARTA ADR100/AMP100/WIN200, therefore the system will use RTP Relay.*
  - *If RTP Relay or Direct RTP is selected, maximum 1024 Audio calls or 102 Video calls are supported in ANANT UCS.*
  - *If you change the RTP Mode, the system will release all ongoing calls.*
  - *In RTP Mode Relay/DRTP, System will generate max 300 unsecured MoH streams (without SRTP) or max 100 SRTP MOH streams simultaneously.*
  - *Video calls will use RTP Relay even when Transcoding is set as the RTP Mode.*

- **MoH Vocoder Preference:** If RTP Mode is set as RTP Relay or Direct RTP, the system will play the MoH when any SIP end point is put on hold. The MoH is played as per the configured MoH Vocoder Preference.

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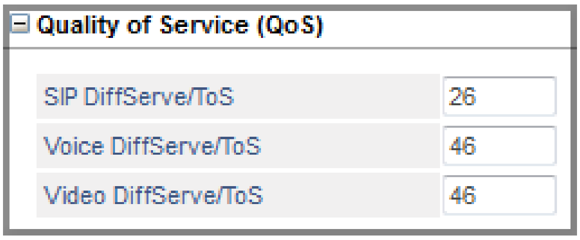
51. *ANANT UCS has in-built soft transcoding support.*

Select the Vocoders in the order of their preference, for **MoH Vocoder Preference 1**, **MoH Vocoder Preference 2** and **MoH Vocoder Preference 3**.

If required, you can customize the MoH to be played as per your requirement. For detailed instructions, see “MoH Management”.

Quality of Service (QoS)

Click **Quality of Service (QoS)** to expand.

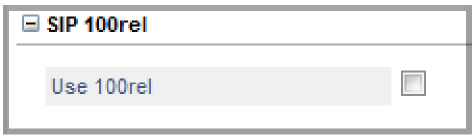


- **Quality of Service (QoS):** QoS refers to priority of IP packets on network layer. It can be configured for both signaling (SIP) and media (Voice and Video). Configure the following types of QoS:
  - **SIP DiffServe/ToS:** The system sends all the SIP signaling messages with this QoS setting. This field defines the priority bits for SIP messages. The Valid *DiffServe* range is from 00-63, default: 26
  - **Voice DiffServe/ToS:** The system sends all the Voice packets with this QoS setting. This field defines the priority bits for Voice packet. The Valid *DiffServe* range is from 00-63, default: 46
  - **Video DiffServe/ToS:** The system sends all the Video packets with this QoS setting. This field defines the priority bits for Video packet. The Valid *DiffServe* range is from 00-63, default: 46

QoS parameters are applicable for all packets (SIP/ Media) leaving both LAN and WAN port as well as TCP connection.

SIP 100rel

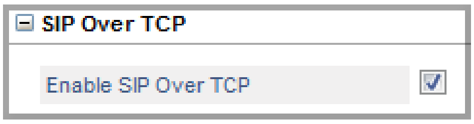
- Click **SIP 100rel** to expand.



- **Use 100rel:** Select this check box, if you want to support reliable transmission of (SIP) provisional responses and PRACK (Provisional Acknowledgment). Default: Disabled.

SIP Over TCP

- Click **SIP Over TCP** to expand.

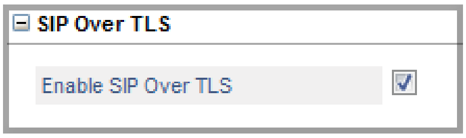


- **Enable SIP Over TCP:** ANANT UCS supports transporting of SIP messages over User Datagram Protocol (UDP), Transfer Control Protocol (TCP) as well as Transport Layer Security connection (TLS). Despite the advantages that SIP over TCP and SIP over TLS offer, it is more common to use UDP to transport SIP messages.
  By default, SIP Over TCP is enabled. To be able to send SIP messages over TCP, you must configure 'TCP' as the *'Default Transport for Outgoing Messages*' on the 'SIP Trunk Parameters' page.

  If you do not want to transport SIP messages using TCP, clear the SIP Over TCP check box.

SIP Over TLS

Click **SIP Over TLS** to expand.



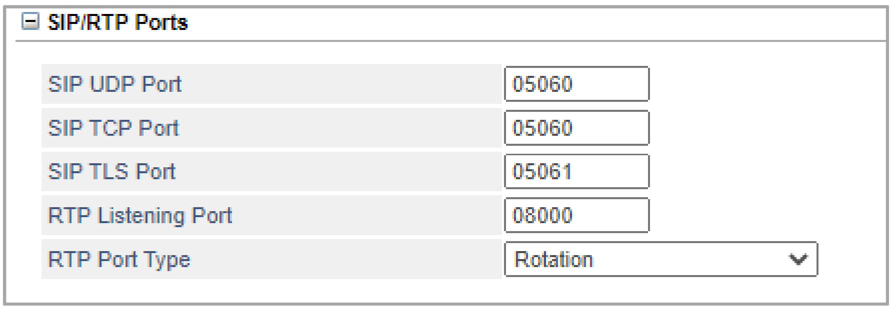
- **SIP Over TLS:** ANANT UCS supports transporting of SIP messages over Transport Layer Security. TLS offers secure SIP signaling.

  By default, SIP Over TLS is enabled. To be able to send SIP messages over TLS, you must configure 'TLS' as the *'Default Transport for Outgoing Messages'* on the 'SIP Trunk Parameters' page.

  If you do not want SIP messages to be transported using TLS, clear the SIP Over TLS check box.

SIP/RTP Ports

Click **SIP/RTP Ports** to expand.



- **SIP UDP Port:** This port defines the port on which ANANT UCS listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025 to 60000. Default: 05060.

- **SIP TCP Port:** This port defines the port on which ANANT UCS listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025 to 60000. The default SIP TCP Port is 05060.

- **SIP TLS Port:** This port defines the port on which ANANT UCS listens for SIP messages transported over TLS. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1025 to 60000. The default SIP TLS Port is 05061.

- **RTP Listening Port:** This port defines the port on which ANANT UCS listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1025 to 50992. The default RTP Listening Port is 08000.

- **RTP Port Type:** Select the desired Port Type — Rotation, First Free — from the drop-down.The default RTP Port Type is Rotation.

  If you select Rotation, for each new session the system checks and allocates the next available RPT port in a sequential manner. Make sure all the RTP Ports are forwarded.

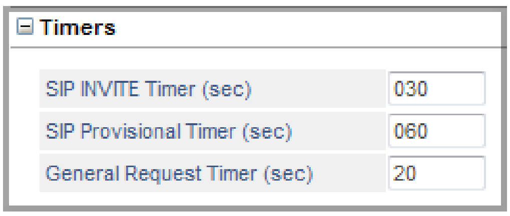
  If you select First Free, for each new session the system allocates the first available RTP Port regardless of its sequence.That is, the system will start checking from the starting RTP port for every new session and allocate the first free port available.In this case the audio and video ports need to forwarded separately. However, for a video call both audio and video ports will be consumed. For a video call maximum 400 ports can be forwarded.

  RTP Listening port = configured RTP Listening port
  Ending number of audio port = RTP Listening port + (number of audio calls * 4)
  Range of audio port = RTP Listening port to Ending number of audio port
  Starting number of video port = RTP Listening Port + 8191
  Ending number of video port = The total number of video port starting range + (number of video calls * 4)
  Range of Video port = Starting number of video port to Ending number of video port

  Let us understand this with the help of an example:
  Number of audio calls to be made 100 and number of video calls to be made 50
  RTP Listening port = 12000
  Ending number of audio port = 12000 + (100*4) = 12400
  Range of audio port = Range 12000 to 12400
  Starting number of video port = 12000 + 8191 = 20191
  Ending number of video port = 20191 + (50 * 4) = 20391
  Range of Video port = 20191 to 20391

Timers

Click **Timers** to expand.



- **SIP INVITE Timer (sec):** This is the time in seconds for which the system waits for a response from the called party after sending INVITE message. This timer starts after sending INVITE message to the called party and stops on receipt of the provisional response or the final response or when the user disconnects the call. On expiry of the timer, the call process is terminated by ANANT UCS and an error tone is played to the user. The range of the SIP Invite Timer is 010-180 seconds. Default: 30 seconds.

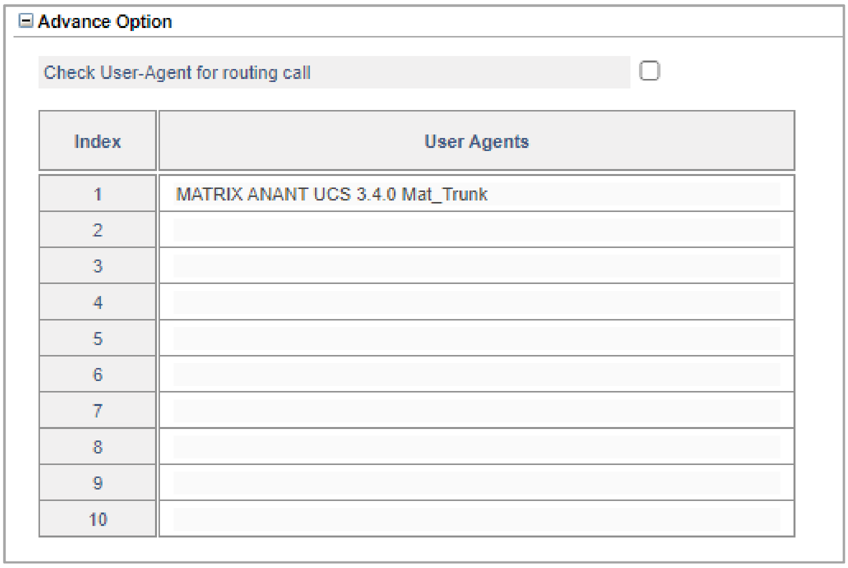
- **SIP Provisional Timer (sec):** This is the time in seconds for which the system waits for the final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On expiry of the timer, ANANT UCS terminates the call process and plays error tone to the user. The range of SIP provisional Timer is 010-180. Default: 60 seconds.

- **General Request Timer (sec):** The time in seconds for which the system waits for response for a transaction request. This timer starts on the initiation of a transaction. This timer stops on receipt of a response for the request. On expiry of the timer, the system clears the transaction. This timer is used for Registration request, etc. The range of the General Request Timer is 10-60 seconds. Default: 20 seconds.

The Timers will be applicable only after System Restart.

Advance Option

Click **Advance Option** to expand.



- **Check User-Agent for routing call:** Default: Disabled. Select the check box to enable. If enabled ANANT UCS will check if the call is originated from a trunk or an extension.

  In the User Agent table, under **User Agents**, configure the User Agent Strings of the other PBXs from which you want incoming call to land on the extensions. In this table the first index is reserved for Matrix Systems and is not editable. You can configure the User Agent Strings of the other PBXs from index 2 to 10.

  Now, let us consider a scenario:
  - We have two PBX's, either Matrix at both the ends or at one end Matrix and at the other any Third party PBX. Let us consider PBX A (Matrix PBX) and PBX B (Third Party PBX).
  - Both the PBX's have the same extension numbers configured, for example 2001.

- You have configured a P2P trunk between the two PBX's.

Now, when 2001 of PBX A receives an incoming call from 2001 of PBX B through the P2P trunk, then the system checks if **Check User-Agent for routing call** is enabled.

If enabled, then the system will check the User Agent in the INVITE request and compares the same with the entries in User Agent Table. Only is an exact match is found the system will place the call, else it will reject it.

CHAPTER 6

Configuring Trunks

ANANT UCS supports 99 SIP Trunks.

Templates for Configuring Trunk Lines

ANANT UCS offers the following Templates to make the configuration of Trunks easy.

- SIP Hardware Template (for SIP Trunks and SIP Extensions)
- Trunk Feature Template (for SIP Trunks)

Using these templates, you can configure all Trunks that are to be assigned the same set of hardware and software features at one go, instead of configuring each trunk individually.

SIP Hardware Template

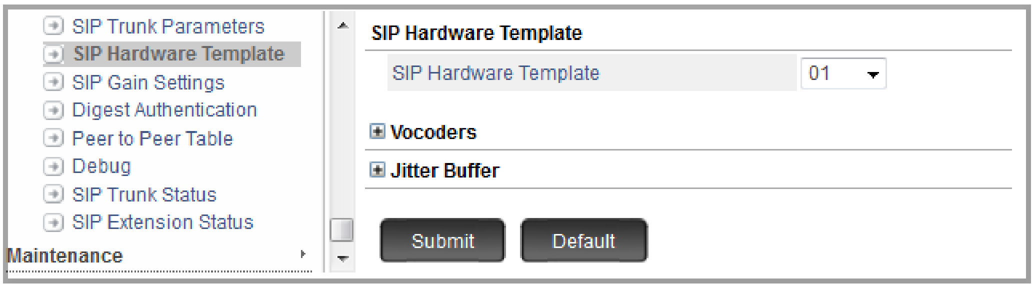
The SIP Hardware Template contains a set voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and other related parameters.

A SIP Hardware Template must be assigned to all SIP Trunks as well as SIP Extensions. Using the SIP Hardware Template, you can configure SIP Trunks and Extensions to have the same set of features at one go.

ANANT UCS offers 32 SIP Hardware Templates, which can be customized as per the requirement and applied on SIP Trunks and Extensions.

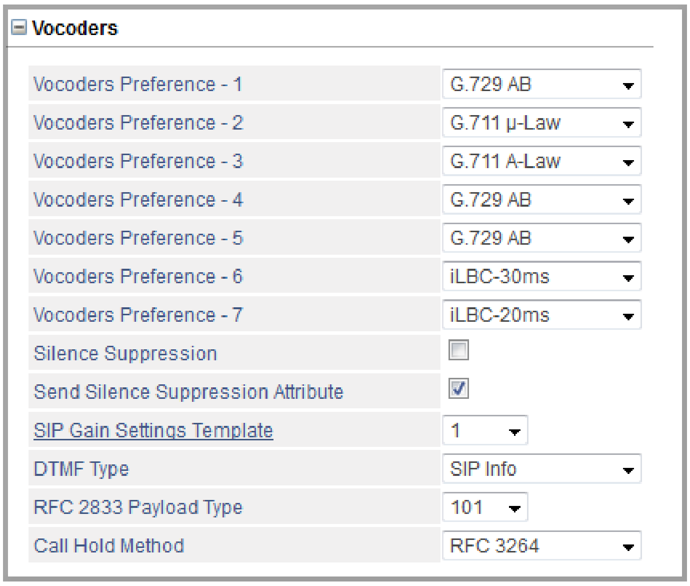
SIP Hardware Template Parameters

Each of the SIP Hardware Template parameters is described below in brief.



- **SIP Hardware Template Number:** Select the Template Number, from 1 to 32 that you wish to configure.

Vocoders



Vocoders are the various voice codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality.

- Click **Vocoders** to expand.

  Vocoders Preference 1 to 7: You can set 7 Vocoder options in the order of preference for a SIP trunk.

  The Vocoders supported by ANANT UCS in the order of preference, from 1st to 7th, by default are: G.729 AB, G. 711µ-Law, G. 711 A-Law, G.722, iLBC - 30 ms and iLBC - 20 ms.

- *If you do not want to select any Vocoder, you can select the option 'None' in the Template. However, if all Vocoder Preferences from 1 to 7 are set to 'None', incoming and outgoing calls will be blocked.*
- *G.723 is not applicable when the RTP Mode is set as Transcoding in "Configuring VoIP Parameters".*

- **G.723 Bit Rate (kbps)[52]:** You can select the Bit Rate for G.723 codec as 5.3 kbps or 6.3 kbps. When G.723 is negotiated, the selected Bit Rate will be applied only when sending the RTP packets. When receiving RTP packets from the remote end, both Bit Rates of G.723 will be accepted. The default G.723 Bit Rate is 6.3kpbs.

- **Silence Suppression:** If you select this check box, the system suppresses the 'Silence' packets, allowing only the Voice packets to pass through. ANANT UCS supports Silence Suppression for all Vocoders. By default, this option is disabled.

- **Send Silence Suppression Attribute:** Select this check box, if you want to include "silencesupp" media attribute in SDP body. If you do not want to include "silencesupp" media attribute in the SDP body, clear the check box. By default, it is enabled.

Silence Suppression Attribute is not dependent on the Silence Suppression option set.

- **SIP Gain Settings Template:** You can increase or decrease the level of Incoming Speech (Receive Gain) and Outgoing Speech (Transmit Gain) on the SIP Trunks/Extensions by changing the Rx Gain and Tx Gain

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52. *This parameter is visible only when the RTP Mode is set as RTP Relay or Direct RTP in "Configuring VoIP Parameters".*

to the desired level. By default, SIP Gain Template 1 is assigned to all the SIP Hardware Templates. If you want to assign a different Template, you must customize the SIP Gain Settings Template first and then assign the number of the SIP Gain Settings Template in this Template. To customize the SIP Gain Settings, see “Gain Settings”.

- **DTMF Type:** This parameter will determine how the DTMF digits will be sent over the IP Network, when a DTMF digit is pressed. ANANT UCS supports two DTMF options: i) RTP (RFC 2833), ii) SIP Info. You may select the appropriate option. By default, RTP (RFC 2833) is selected.

- **RFC2833 Payload Type:** If you have selected RTP (RFC 2833) as the DTMF Type, you must configure the value of RFC2833 Payload Type. The RTP packets will be tagged as DTMF as per the set value. The value of RFC2833 Payload Type can be set from 96 to 127. By default, 101 is selected.

- **Call Hold Method:** You can select RFC 3264 and RFC 2543, as per your requirement.

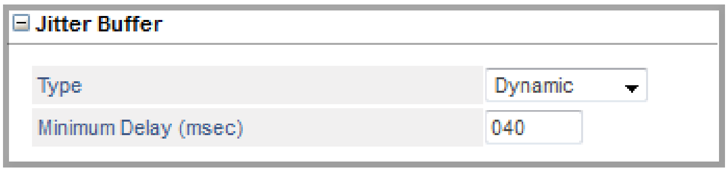
  If you select RFC 3264, the following information will be sent in the SDP:
  - Connection Information: IP Address as used in Contact
  - Media attribute (a): sendonly

  If you select RFC 2543, the following information will be sent in the SDP:
  - Connection Information: 0.0.0.0
  - Media attribute (a): sendonly

  By default RFC 3264 is selected.

- Click **Submit**.

Jitter Buffer



The speed at which voice packets travel through a network depends on the condition of the network. All voice packets may not come at the same speed. This variation in the delay in receiving packets, known as Jitter, affects voice quality. Jitter Buffer helps overcome the delay in receiving voice packets and improves voice quality. Jitter Buffer receives voice packets, stores them and sends it to the DSP to process it at evenly spaced intervals, thus improving voice quality.

- Click **Jitter Buffer** to expand.

- **Type:** ANANT UCS supports two types of Jitter Buffer: Static and Dynamic. Static Jitter Buffer's internal delay is static, whereas, the Dynamic Jitter Buffer's internal delay adapts itself to the jitter in the network. Select the type of Jitter Buffer - Static or Dynamic - you want to use. If you selected Static Jitter Buffer or Dynamic Jitter Buffer, you may set the 'Minimum Delay'. The value configured in the Minimum Delay determines the size of the Static/Dynamic Jitter buffer.

- **Minimum Delay (msec):** This parameter is to be configured for both Static and Dynamic Jitter Buffer. The Minimum Delay determines the size of the Jitter Buffer. When Jitter Buffer type is selected as Static, the Minimum Delay defines the size of the Static Jitter Buffer. The Static Jitter Buffer will store each received voice packets for the time you set and then it will send it to DSP for voice processing.

When Jitter Buffer type is Dynamic, the Minimum Delay specifies the minimum time for which the Dynamic Jitter Buffer will store the received voice packet before sending it to the DSP for voice processing. 'Minimum Delay' can be from 10 to 280 milliseconds. By default Minimum Delay is set to 40 milliseconds.

- Click **Submit**.

Customizing SIP Hardware Template

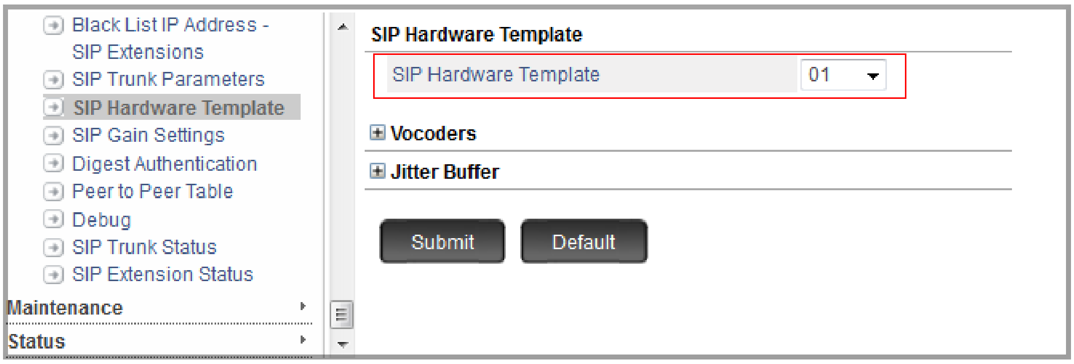
You can customize 32 SIP Hardware Templates. By default, SIP Hardware Template 01 is assigned to all SIP Trunks as well as SIP Extensions.

If the default SIP Hardware Template 01 fulfills the user requirements, retain Template 01. If you want to change the values of certain SIP Hardware Parameters, but apply the same parameter values to all SIP Trunks and Extensions, simply customize the desired parameters in Template 01.

If different hardware parameters are to be applied to different SIP Trunks and SIP Extensions, customize the different SIP Hardware Templates and apply them to the SIP Trunks and SIP Extensions accordingly.

Customizing SIP Hardware Template

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Hardware Template**.



- Select a Template number you wish to customize, for example Template 02.
- Change the values of the SIP Hardware Template parameters as desired.
- Click **Submit**.
- Now, apply this Template 02 on the SIP Trunks and SIP Extensions.

To apply the customized template on SIP Trunks,

- Under **VoIP Configuration**, click **SIP Trunk Parameters**.
- Select the desired **SIP Trunk No** to which this Template is to be assigned, for example SIP Trunk 02, 03 or 04.

- Click **Others** to expand. Enter the number of the Template you customized, 02, in **SIP Hardware Template** of each of these SIP Trunks.
- Click **Submit.**

To apply the customized template on SIP Extensions,

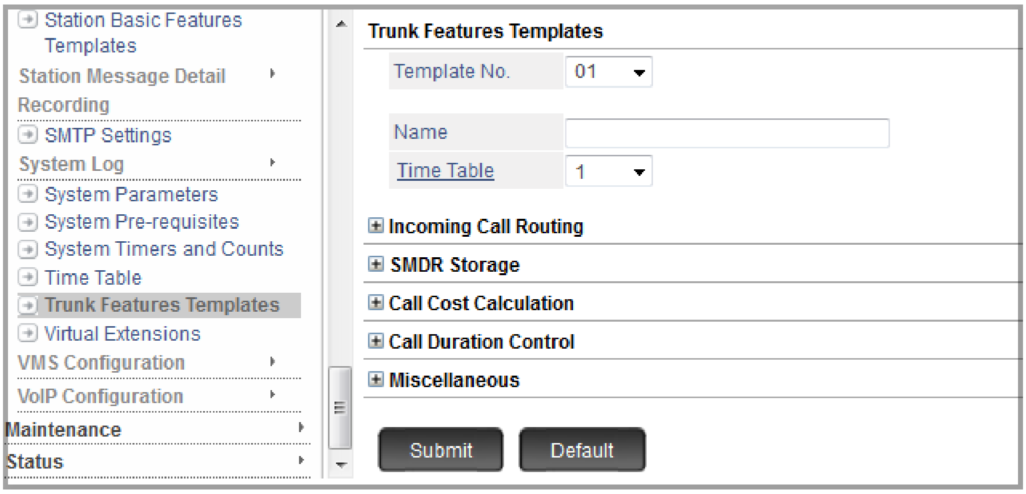
- Under **VoIP Configuration**, click **SIP Extension Settings**.
- Select the desired **SIP Extension** number to which the Template is to be assigned, for example SIP Extensions - 5,6,7.
- Enter the number of the Template you customized, 02, in **SIP Hardware Template** of each of these SIP Extensions.
- Click **Submit.**
- Repeat the same steps to customize another template and apply it on the SIP Trunks and Extensions.

Trunk Feature Template

The Trunk Feature Template is a set of general features that define the behavior of a Trunk port. ANANT UCS offers 50 such Templates. A Trunk Feature Template is assigned to all the SIP Trunks.

Trunk Feature Template Parameters

The Trunk Feature Template contains the following features:



- **Template No:** Select the Template Number, from 1 to 50 that you wish to configure.
- **Name:** Configure the Name you wish to assign to the Template you selected.
- **Time Table:** Select a Time Table for the Trunk port.

  A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week.

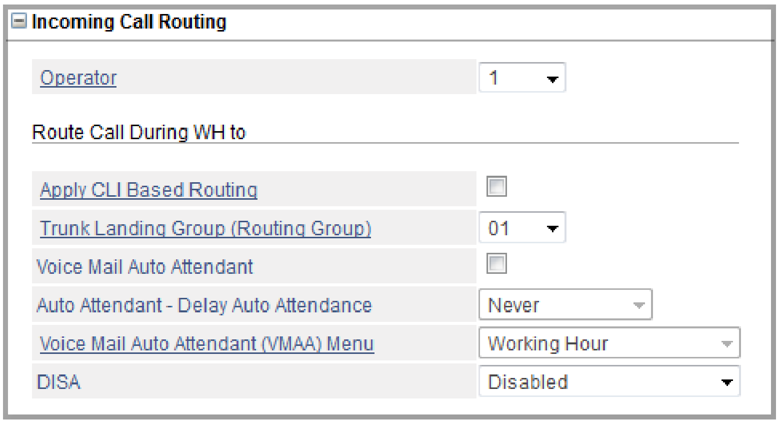
Certain features of the ANANT UCS like Operator, Voice Mail Auto Attendant, Trunk Landing Group, require the trunk to behave differently in each Time Zone. For example, it can be made to land on the Operator extension during working hours and on the extension of the dining area during Break (lunch) hours and on the extension of the Security Personnel during non-working hours.

So, a Time Table is assigned to trunks defining the Time Zones for the entire week, so that the system can execute the Time Zone-dependent features and facilities according to the Time Table.

There are 8 different Time Table templates to select from. By default, the Time Table 1 is assigned to all Trunk Feature Templates. All days of the week except Sunday are 'working hours 9:00-18:00' with break hours '13:00 -14:00 hrs'.

You may also customize the default Time Table 1 OR customize and assign a different Time Table to the Trunk Feature Template. Please refer the topic “Time Tables” for more details.

Incoming Call Routing



The system can route the incoming calls in different ways, such as CLI based Routing, routing calls through the Voice Mail Auto Attendant, etc. According to your requirement you can configure the parameters to route the call.

- Click **Incoming Call Routing** to expand.
- **Operator:** Define the Operator for the Trunks on which the template is applied. Operator is used to route the call when the caller dials '9' during an Auto Attendant call. This parameter is of significance only if Voice Mail Attendant/DISA is enabled on the trunk.

  The system supports multiple Operators. In each Time Zone any one of the 20 Operators can be selected.

  Trunks may be assigned to a single Operator, or different groups of Trunks may be assigned to different Operators, so that call management is more efficient. For instance certain Trunks may be assigned to Operator 1, while some may be assigned to Operator 2 and the rest to Operator 3.

  Operator 1 is default in the Trunk Feature Template. If you want to assign different trunks to different Operators, you must create a separate Trunk Feature Template with a different Operator for each trunk group.

  Refer the topic “Configuring 'Operator'” to know more.
- **Apply CLI Based Routing:** Select the check box to enable CLI Based Routing on the Trunk for each Time Zone: Working Hours (WH), Break Hours (BH) and Non-working Hours (NH). Default: disabled.

If you enable CLI Based Routing on the trunk for a Time Zone, make sure you also configure the CLI Based Routing Table. To know more, refer the feature description “CLI Based Routing”.

- **Trunk Landing Group (Routing Group):** This parameter allows you to configure the group of extensions on which incoming calls on the trunks (to which this template is assigned) are to be landed. This group of extensions is referred to as 'Trunk Landing Group' (TLG).

  To configure the TLG, you must first configure Routing Groups. Refer “Trunk Landing Group (TLG)” for instructions on configuring trunk landing groups. Also refer the topic “Routing Group”.

  There are as many as 96 Routing Groups which can be assigned as TLG. By default, Routing Group 01 is assigned as TLG for all Time Zones. If you have prepared a different TLG for each Time Zone, for example, Routing Group 02 for Working Hours and Break Hours, Routing Group 03 for Non-Working Hours, then enter the number of these Routing Groups in the TLG field.

- **Voice Mail Auto Attendant:** Select this check box if you want the calls to be answered by the Auto Attendant of the Voice Mail System. The Voice Mail System of ANANT UCS answers calls and processes them according to the Voice Mail Auto Attendant Menu assigned to the trunk.

  - **Auto Attendant - Delay Auto Attendance:** Set this time, if you want to enable “Delayed Auto Attendant” on the trunk.

    When you enable Delay Auto Attendance, ANANT UCS routes the incoming call on the trunk to the Trunk Landing Group assigned to this trunk. It waits for the duration of the Auto Attendant Delay Time for any of the extensions in the Trunk Landing Group to answer the call.

    If none of the extensions in the Trunk Landing Group answers the call before the expiry of the Auto Attendant Delay Timer, ANANT UCS hands over the call to the Voice Mail Auto Attendant.

    To enable Delay Auto Attendance, set the time to the desired value from the list. By default, it is set as Never, that is, it is disabled.

  - **Voice Mail Auto Attendant (VMAA) Menu:** If you have enabled the Voice Mail Auto Attendant, select the VMAA Menu to assign to the respective Trunk Feature Template.

    You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of the VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- **DISA:** This parameter is to be configured, if you want to enable “Direct Inward System Access (DISA)” on the trunk ports on which you will apply the template.

  DISA can be enabled or disabled for each Time Zone, namely Working Hours (WH), Break Hours (BH) and Non-Working Hours (NH).

  For each Time Zone, you may select the desired DISA option from the following:

  - **Disabled:** Select this option if you want to disable DISA.

  - **PIN Auth.-Multiple calls:** Select this option if you want to enable DISA with PIN Authentication and allow multiple calls during the DISA login session.

  - **CLI Auth.-Multiple calls:** Select this option if you want to enable DISA login with CLI Authentication and allow multiple calls to be made during the DISA login session.

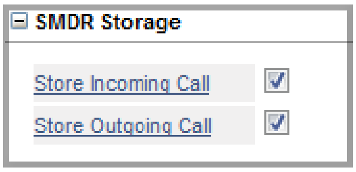
Caller numbers that do not match with the CLI Table will be routed as per the logic of the Trunk Feature Template.

- **CLI Auth.-One call:** Select this option if you want to enable DISA session with CLI Authentication, and allow only a single call to be made during the DISA login session. This form of DISA is used when ANANT UCS is installed in a Gateway application.

  By default, DISA is disabled for all Time Zones.

- Click **Submit**.

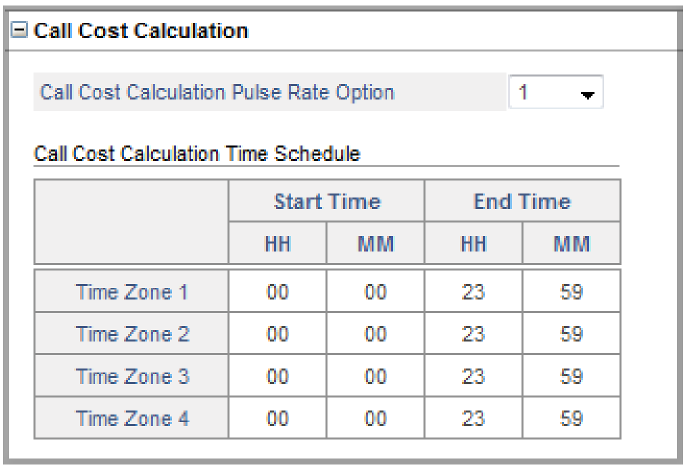
SMDR Storage



The Station Message Detail Recording (SMDR) feature of ANANT enables you to record the details of Incoming (IC) and Outgoing (OG) calls made from/on the trunks. To obtain SMDR as a report, you must enable SMDR Storage, and set filters. See “Station Message Detail Recording (SMDR)” to know more.

- Click **SMDR Storage** to expand.
- **Store Incoming Call:** By default this check box is selected. The system will store the details of incoming calls on the trunk. Please refer the topic “Station Message Detail Recording-Storage” for more details. Clear the check box to disable.
- **Store Outgoing Call:** By default this check box is selected. The system will store the details of outgoing calls from the trunk. Please refer the topic “Station Message Detail Recording-Storage” to know more. Clear the check box to disable.
- Click **Submit**.

Call Cost Calculation



ANANT UCS can calculate the cost in amount for the external calls made by the extension users. You need to configure the following parameters:

- Click **Call Cost Calculation** to expand.
- **Call Cost Calculation Pulse Rate Option:** This parameter is to be configured only if you want to apply the "Call Cost Calculation (CCC)" feature on the trunks on which the template is applied.

  You have four options for Pulse Rate Types. Select from Pulse Rate Type for Pulse Rate Option 1 to 4 which you want to apply on the trunks.
- **Call Cost Calculation Time Schedule:** This parameter is to be configured only if you want to apply the "Call Cost Calculation (CCC)" feature on the trunks on which the template is applied.

  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you must first define the Time Zone (time of the day) for which a particular Pulse Rate should be applied and the Time Schedule for each Time Zone.

  You can configure up to four different Time Zones - Time Zone 1, Time Zone 2, Time Zone 3 and Time Zone 4 with different Pulse rates in the CCC. For details refer "Configuring Pulse Rate Types".

  Now, configure the Call Cost Calculation Time Schedule, by specifying the Start Time and the End time (in 24 hours: minutes format) for each Time Zone.

  The default Time Schedule (starts and end time) for each Time Zone Index are as follows:

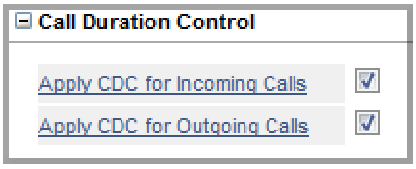
Time Zone Index	Start Time	End Time
Time Zone 1	00:00	23:59
Time Zone 2	00:00	23:59
Time Zone 3	00:00	23:59
Time Zone 4	00:00	23:59

  If your service provider offers the same Pulse Rate for the entire day,

  - configure only one Time Zone Index with the Pulse Rate, for instance, Time Zone 1, in the CCC-Normal Pulse Rate Table.
  - Now, set the Time Schedule for Time Zone, Time Zone 1, with the start and end time in Hours: Minutes format;
  - set the start and end time of the other Time Zone Index, Time Zone 2 to Time Zone 4, to 00:00 (hours: minutes).

  Similarly, if your service provider supports two different Pulse Rates in a day, set the Start and the End time for two Time Zones and set the other two to 00:00.
- Click **Submit**.

Call Duration Control

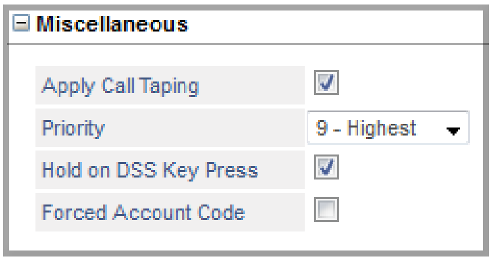


Call Duration Control (CDC) allows you to limit the duration of the conversations. CDC helps increase availability of trunks for making outgoing calls and for receiving incoming calls, which is important in high call traffic situations.

- Click **Call Duration Control** to expand.
- **Apply CDC for Incoming Calls:** By default this check box is selected (enabled). Clear the check box to disable.
- **Apply CDC for Outgoing Calls**: By default this check box is selected (enabled). Clear the check box to disable.
- Click **Submit.**

To know more, refer to "Call Duration Control (CDC)".

Miscellaneous



- Click **Miscellaneous** to expand.
- **Apply Call Taping:** By default, this check box is selected, Call Taping will be allowed. Clear the check box to disable. To know more refer "Call Taping".
- **Priority:** Select a Priority Level for the trunks on which the template will be applied.

  Each trunk of the ANANT UCS can be assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority.

  Whenever there are incoming calls on multiple trunks, the call on the trunk with higher priority will be answered by the Operator extension first. To know more, read the feature description "Priority".

  By default, the Priority of all trunks is set to '9-Highest'. Decide what Priority Level you will assign to the trunks and set the desired level for the trunk.
- **Hold on DSS Key Press:** By default, this check box is selected. This option defines the 'Hold' state of the external called party, when an extension user presses a DSS key to dial another port.

For example, the SIP extension user (on SIP-001) is in the middle of speech with an external party on a SIP Trunk -002.
If SIP Extension-001 user presses a DSS key to call another extension SIP Extension-003, two situations are possible, depending on whether the Hold on DSS Key Press check box is enabled or disabled:

- **When the Hold check box is enabled:** SIP Trunk-002 will be played music-on-hold. SIP Extension-001 user will hear Ring Back Tone and the call will be placed on SIP Extension-003.

- **When Hold check box is disabled:** SIP Trunk-002 will be disconnected. SIP Extension-001 will hear Ring Back Tone, and call will be placed on SIP Extension-003.

- **Forced Account Code:** Select this check box if you want the extension users to dial the Account Code whenever they grab a trunk to dial out a number. The system will allow extension users to dial out numbers only after they have dialed the Account Code or Name.

  By default, the check box is disabled. Refer the feature description for "Account Codes" to know more.

Account Codes feature must also be enabled in the Class of Service of extension users who wish to use this feature.

- Click **Submit**.

Customizing Trunk Feature Templates

By default, Trunk Feature Template 01 is assigned to all SIP Trunk. The default values of the parameters of this template are sufficient to meet the common requirements of most users.

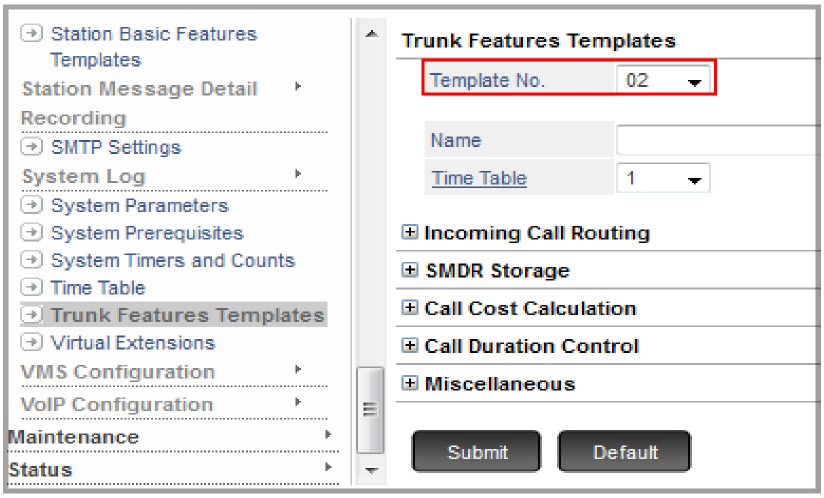
If the default values of the Template fulfill the necessary requirements, retain this template. If you want to change some of the feature settings and apply the template to the SIP trunks, you may simply customize this template.

However, if you want to assign different feature settings for different SIP trunks, you are recommended to prepare and apply separate Trunk Feature Templates for each SIP Trunk.

Customizing Trunk Feature Template

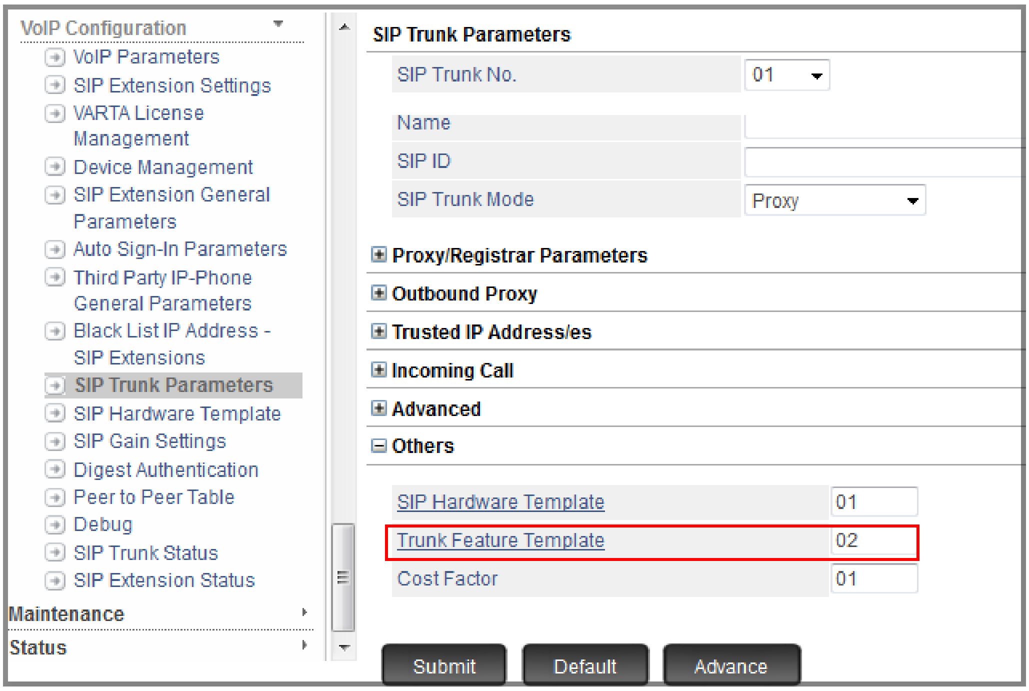
- Login as System Engineer.

- Under **Configuration**, click **Trunk Feature Template**.



- Select a Trunk Feature Template Number, for instance 02.
- Customize the Template by changing the values of the desired parameters.
- Click **Submit**.
- Apply the Trunk Feature Template you customized to the SIP Trunk.
- To apply the template on SIP Trunks,
  - Under **VoIP** Configuration, click **SIP Trunk Parameters**.
  - Select the desired SIP Trunk number to which this Template you customized is to be assigned, for instance SIP Trunk No.1.

- Click **Others** to expand. Enter the number of the Trunk Feature Template you customized, 02, in the **Trunk Feature Template**.



- Click **Submit**.

- Repeat the same steps to create another template and apply it on the desired SIP Trunk.

- Remember to click **Submit** to save your settings on each page.

Configuring SIP Trunks

SIP trunks may be Proxy or Non-Proxy. All SIP trunks are considered as Proxy trunks by default.

Regardless of whether a SIP Trunk is Proxy or Non-Proxy, to use the SIP trunk, it must be enabled. Calls can be initiated after suitable configuration of the SIP trunk number in the Outgoing Trunk Bundle Group.

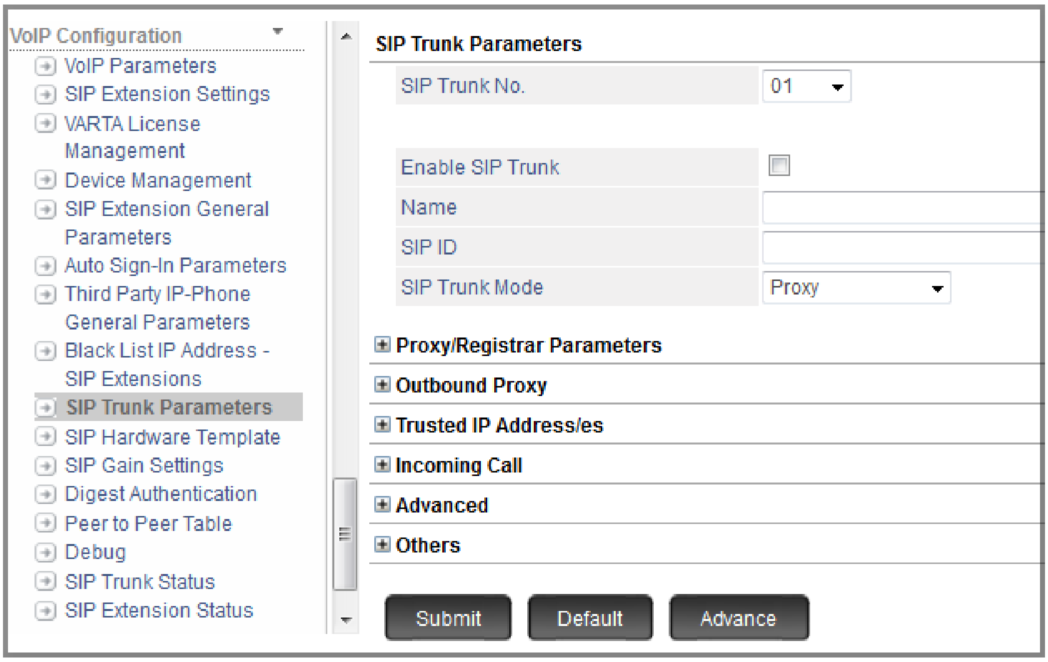
For Proxy SIP Trunks, you must configure the following parameters required for registration with the Proxy Server.

- Enable the SIP trunk.
- Configure the SIP ID, Registrar Server Address, Registrar Server Port, Authentication User ID, Authentication Password as provided by your ITSP.
- If your ITSP uses Outbound Proxy, enable the Outbound Proxy for the SIP trunk and also configure the Outbound Proxy Server Address and Port as provided by your ITSP.

The maximum number of SIP trunks supported by ANANT UCS is 99.

Configuring SIP Trunks

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Trunk Parameters.**



Configure the following SIP Trunk parameters for the SIP trunks:

- **SIP Trunk No.:** Select the **SIP Trunk No.**, from 1 to 99 that you wish to configure.

- **Enable SIP Trunk:** By default, this check box is disabled, that is, the SIP trunk is disabled, disallowing incoming and outgoing calls. You may keep it disabled if the SIP trunk is not functioning.

  To be able to make incoming and outgoing calls from the SIP trunk, select the **Enable SIP Trunk** check box.

- **Name:** You may assign a 'Name' to each SIP trunk to facilitate identification. Whenever there is an incoming call without CLI on this port, the Name you have configured will be displayed on the landing extension.

  The Name assigned to the SIP trunk may consist of a maximum of 18 characters. The Name of the port may be the name of the ITSP the SIP trunk is subscribed with (recommended).

- **SIP ID:** Enter the SIP User ID provided by the ITSP. SIP User ID is the ID that callers will use to call this SIP trunk.

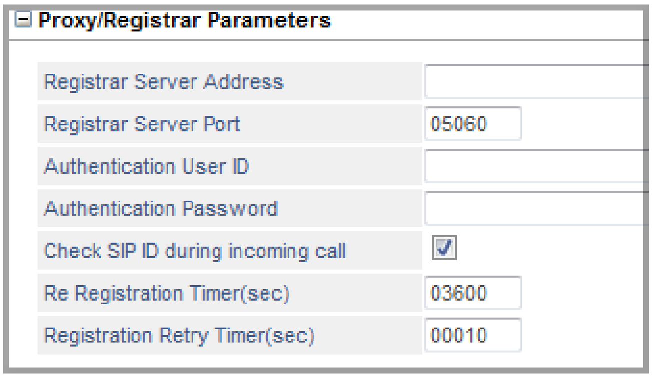
  The SIP User ID may be a number or text for remote parties to call on the SIP trunk. For example, if SIP URI provided by the ITSP is 12345@abc.com, enter 12345 in this field. SIP User ID may consist of a maximum of 40 characters. All ASCII characters are allowed.

- **SIP Trunk Mode**: You may select SIP Trunk Mode as **Proxy** or **Peer-to-Peer**, according to your requirement.

  If you are using the services of an Internet Telephony Service Provider (ITSP), select **Proxy** to register this SIP trunk with the ITSP.

  If you are not using this service, select **Peer-to-Peer.**

Proxy/Registrar Parameters



- Click **Proxy/Registrar Parameters** to expand.

- **Registrar Server Address:** Enter the Proxy/Registrar Server Address. Both IPv4 and IPv6 addresses are supported. The Server Address may be an IP Address or a Domain name, of maximum 40 characters. By default, it is Blank.

- **Registrar Server Port:** Enter the Registrar Server Listening Port. The valid range is from 1025 to 65535. By default, 5060 is set as the Listening Port of the Registrar Server.

- **Authentication User ID:** Enter the Authentication ID provided by the ITSP for this SIP trunk. The Authentication User ID may be a string of 40 characters (maximum), including ASCII characters. By default, it is Blank.

- **Authentication Password:** Enter the Authentication Password provided by the ITSP for this SIP trunk. The Authentication Password may be a string of 128 characters (maximum), including ASCII characters. By default, it is Blank.

- **Check SIP ID During Incoming Call:** By default, this check box is enabled. During Incoming Call Routing to select a SIP trunk, ANANT UCS compares the SIP ID received in the Request URI of the INVITE message with the SIP ID configured on the SIP Trunk.

  If you do not want ANANT UCS to check the SIP ID received in the Request URI of the INVITE message, disable the **Check SIP ID During Incoming Call** check box.

- **Send REGISTER Message:** With this parameter you can select whether or not the system should send REGISTER message from the SIP trunk. By default, this check box is enabled allowing REGISTER message to be sent from the SIP trunk.

- **Allow registration for any contact**: Enable this check box if you wish to register the SIP trunk with an ITSP. In this case even if you do not receive proper contact header from the ITSP, the SIP trunk will get registered.

- **Send OPTIONS as Heartbeat:** With this parameter you can select whether or not the system should send the OPTIONS message periodically to the Proxy Server to monitor its availability. Calls can be made and received only if the Proxy Server is alive.

  If the Proxy Server is unavailable, like no response is received, the status of the SIP Trunk will display “Heartbeat Failed” along with the Reason for Failure.

To view status of the Proxy Server, go to ***SIP Trunk Status****.*

  If you enable **Send OPTIONS as Heartbeat**, you must configure the Heartbeat Interval.

  - **Heartbeat Interval**: Define the Heartbeat Interval (Seconds), the time period, from 10 to 999 seconds, after which ANANT UCS should send the OPTIONS message to the Proxy Server. Default: 30 seconds.

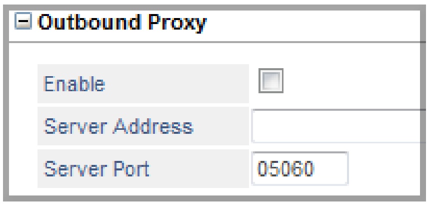
- **Re-registration Time (sec):** The Registrar Server deletes an entry of its client from its database on expiry of a fixed timer, which is set by the Registrar Server. ANANT UCS sends a registration request before this Timer expires to remain registered on the server.

  Enter the value of the Timer after which the system should send the registration request to maintain registration binding with the server. The valid range of this timer is from 00001- 65535. By default the Timer is set to 3600 seconds.

- **Registration Retry Time (sec):** This Timer stands for the period between retries for registration. If the registration attempt fails, ANANT UCS sends the registration request on the expiry of this Timer again. The system continues to send the registration request till it gets registered. The valid range of this timer is from 00001- 65535. By default the Timer is set to 00010 seconds.

- **Add 'rinstance' in Register:** 'rinstance' is any random value which can be used by the WAN Port to fetch its own contact binding, that is, to know the Registration Expiry Timer assigned by the server. By default, this check box is enabled.

Outbound Proxy



- Click **Outbound Proxy** to expand.

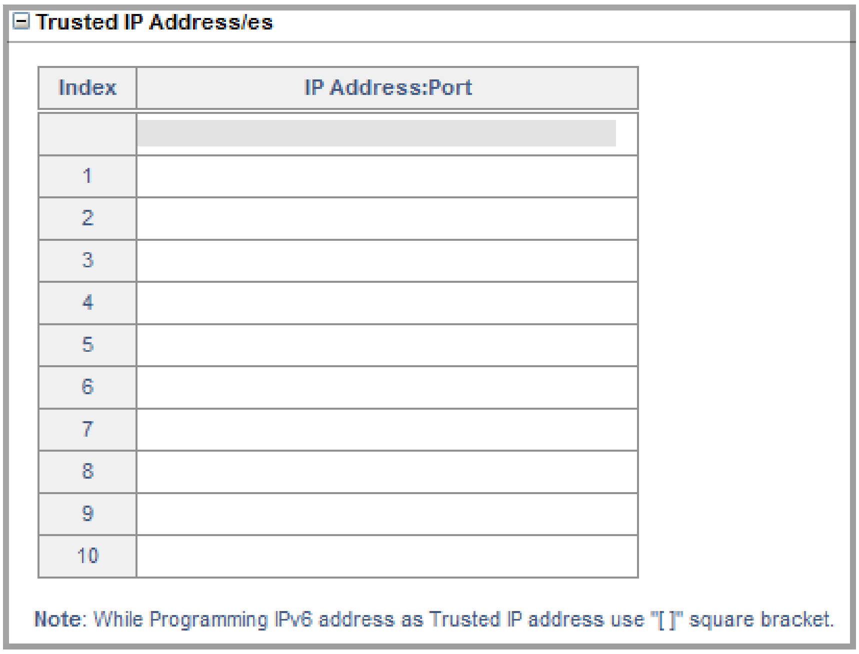
  These parameters are relevant only if the ITSP has a SIP outbound server to handle voice calls. If yes, configure the following parameters:

- **Enable:** Select this check box to enable Outbound Proxy. By default the check box is disabled.

- **Server Address:** Enter the Outbound Proxy Server's address. Both IPv4 and IPv6 addresses are supported. It may be an IP Address or Domain name. A maximum of 48 characters, including ASCII characters are allowed. By default it is Blank.

- **Server Port:** Enter the Outbound Proxy Server's Listening Port. The valid range for this is 1025-65535. By default the Server Port is 5060.

Trusted IP Address/es



- You must configure IP Address table to allow incoming calls from specific IP addresses on this SIP Trunk. Both IPv4 and IPv6 addresses are supported.

If you select *Peer to Peer* as the SIP Trunk Mode and you do not configure this table, incoming calls on this SIP Trunk will be rejected.

If you select *Proxy* as the SIP Trunk Mode and you do not configure this table, incoming calls will be allowed only from the Registrar Server Address or Outbound Proxy Address. All other calls on this SIP Trunk will be rejected.

If you have selected SIP Trunk mode as Peer to Peer, configure the following.

- **Allow from all IP Addresses:** Enable this check box, if you want to allow incoming calls on this SIP Trunk from all IP Addresses. Default: Disabled.
- **Apply Digest Authentication:** Enable this check box, if you want to allow incoming calls from callers only after the callers have authenticated themselves (with their User ID and Password). If the caller does not enter valid credentials in two attempts, the system will reject the call.

  If you have enabled *Allow from all IP Addresses* check box, Apply Digest Authentication check box will be enabled automatically. You must configure the Digest Authentication table.

  To configure the table, click the **Apply Digest Authentication** link. For detailed instructions, see "Digest Authentication".
- **Consider Peer to Peer Table for Trusted IP Address:** Enable this check box, if you also want to allow incoming calls from the domain names or IP Addresses configured in the Peer to Peer table. Default: Enabled. To know more about Peer to Peer table, see "Peer-to-Peer Calling".

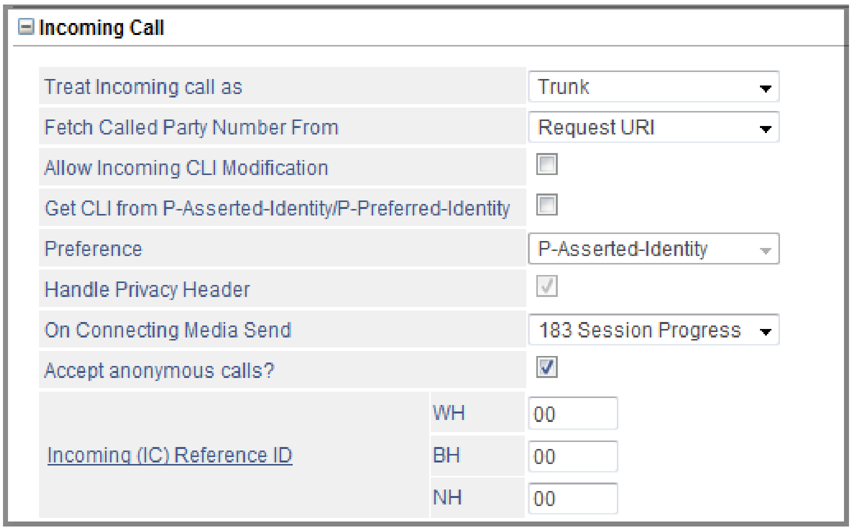
If you have selected SIP Trunk mode as Proxy or Peer to Peer, configure the Trusted IP Address table.

The first entry in the table will display the *Proxy/Registrar Server Address:Port* or *Outbound Proxy Address: Port* as configured for this SIP Trunk (applicable only for Proxy Mode). For the Index numbers 1 to 10,

- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls.

  Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.
- Click **Submit.**

Incoming Calls



- **Treat Incoming calls as:** If you select Peer-to-Peer as the SIP Trunk mode, you may configure the trunk to Treat Incoming call as Trunk or Station.

  If you select **Trunk**, the incoming calls will be routed as per the **Trunk Feature Template** assigned to the SIP Trunk.

  If you select **Station**, the system will route the incoming call as follows:

  - When only number is received in the “To:” field or the Request URI (configurable, default: Request URI) of the INVITE message, ANANT UCS will check the number in the Closed User Group Table. If a match is found in the CUG table the call will be routed as per the corresponding Outgoing Trunk Bundle Group.

  - If the CUG Table is not configured or if no match is found for the number received in the “To:” field of the INVITE message, the system will check if there is an extension number that matches with the number received in the “To:” field of the INVITE message. If a match is found the call is routed to the desired extension number.

  - When a Trunk Access Code and a number is received in the “To:” field of the INVITE message, the system will route the call as per the Outgoing Trunk Bundle Group assigned in the Template of the SIP Trunk.

    By default, Trunk is selected.

  - *If Station is selected as the option for Treat Incoming call as, the user will only be able to:*
    - *Dial Flexible Numbers*
    - *Dial Operator Code*
    - *Dial Trunk Access Code for making outgoing calls*
    - *Access the Global Directory*
    - *Make calls within the Closed User Group*

- **Fetch Called Party Number From**: ANANT UCS extensions may be assigned DDI numbers provided by the ITSP. When INVITE is received from ITSP, the ITSP may send the DDI number either in the “Request URI” of the INVITE message or in the “To:” field of the INVITE message.

By default, Request URI is selected. Ask your ITSP if you need to change this parameter.

- **Allow Incoming CLI Modification**[53]**:** To apply Incoming CLI Modification on the SIP trunk, select the **Allow Incoming CLI Modification** check box. By default, it is disabled.

  Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see “Incoming CLI Modification”.

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the check box disabled.
*For an incoming call on the SIP trunk, the Incoming CLI Modification will be applied only when both — **the Allow Incoming CLI Modification** check box and the **Enable Incoming CLI Modification** check box in System Parameters — are enabled.*

- **Get CLI from P-Asserted-Identity/P-Preferred-Identity:** It contains the caller ID information for the call on the Invite SIP Packet. The privacy header contains information on which parts of the caller ID are private. Select this check box to enable, if supported by your Service Provider. By default it is disabled.

- **Preference:** Select **P-Preferred-Identity** or **P-Asserted-Identity** as the **Preference**.

- **Handle Privacy Header:** By default Handle Privacy Header is enabled. If the system receives Privacy Header as:

  - **Privacy: ID**, anonymous will be displayed as the CLI to the called extension.

  - **Privacy: None**, the CLI received from the Service Provider will be displayed to the called extension.

    If No Privacy header is received the CLI received from the Service Provider will be displayed to the called extension.

  If you do not want the system to handle the Privacy Header, clear the check box.

- **On Connecting Media Send**: If DISA is enabled on the SIP Trunk, select the response you want the system to send on Connecting Media, that is **200 OK** or **183 Session Progress**.

- **Accept Anonymous Calls?:** The option is for accepting calls without CLI that land on the SIP trunk. By default the check box is disabled. You may select this check box to enable, that is allow calls without CLI on this SIP trunk port.

- **Incoming (IC) Reference ID:** Assign an Incoming Reference ID for the SIP trunk for Working Hours, Break Hours, Non-Working Hours. By default, 00 is assigned as Incoming Reference ID for all three time zones.

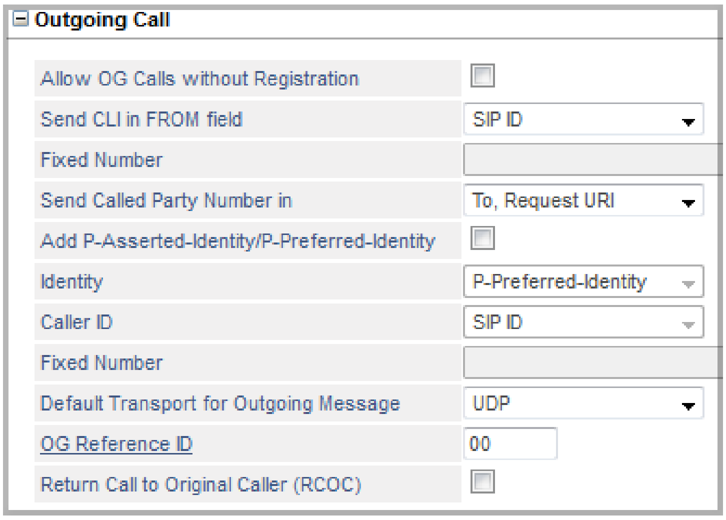
- **Station Basic Feature Template:** Configure this parameter if you have selected Station as the Treat Incoming call as option. Assign a “Station Basic Feature Template” to the SIP trunk. There are 50 different Station Basic Feature Templates to choose from. Each template can also be altered to suit your requirement and preferences. Default: Template 01 is assigned to all SIP trunks.

- **Station Advanced Feature Template:** Configure this parameter if you have selected Station as the Treat Incoming call as option. Assign a “Station Advanced Feature Template” to the SIP trunk. There are 50

53. To comply with the Indian Government Laws and Regulation, this parameter is not applicable for India Region.

different templates to choose from. Each template can also be altered to suit your requirement and preferences. Default: Template 01 is assigned to all SIP trunks.

Outgoing Call



- **Allow OG Calls without Registration:** This parameter is to be enabled to allow outgoing calls to be made from the SIP trunk, even when the SIP trunk is not registered. If this check box is disabled the system will not allow outgoing calls to be made if the status of the SIP trunk is 'not registered'. By default, this check box is disabled.

- **Send CLI in FROM field:** This parameter allows you to configure the CLI of the SIP Trunk to be sent to the remote party on outgoing calls made using the SIP trunk. You may select any of the following options as desired:

    - **CLIR:** Select this option if you do not want the CLI to be sent.

    - **SIP ID:** You may select this option if you want the SIP ID configured on the SIP Trunk to be sent as CLI.

    - **Calling Party Wise:** Select this option if you want to send the Calling Extension Number (the number of the extension making the outgoing call through the SIP trunk) as CLI.

      When reverse DDI is configured on the SIP Trunk, the DDI number of the calling extension will be sent, instead of its extension number.

      If the calling extension has disabled the parameter ‘Send DDI as CLI’ in its Station Advanced Feature Template, then its Pilot number configured in the Outgoing Reference Table will be sent as CLI.

      If calling extension has enabled CLIR, no CLI will be sent on the SIP Trunk.

    - **Fixed Number:** Select this option if you want a specific number to be sent as CLI. When you select this option, you must also define the number to be sent as CLI.

      You may select this option if you wish to send any of your trunk line numbers as CLI on the SIP Trunk so as to enable the called party to call back the calling party using this CLI.

Since it is not possible to call back a SIP ID, Fixed Number offers you a solution, using which you can send a trunk line number as CLI on the SIP Trunk. Using this CLI, the called party can call back the calling party.

If you select this option, you must configure the **Fixed Number**. The Fixed Number may consist of a maximum of 40 characters. Valid characters are from 0-9 and plus (+).

By default, SIP ID is set as the Send CLI in FROM field for all SIP Trunks.

- *When extension number of the calling extension is blank, and the 'Send CLI in FROM field' configured for the SIP Trunk is other than "SIP ID", then also SIP ID will be sent as CLI.*
- *When Emergency numbers are dialed using the SIP Trunk and even if CLIR is set as the 'Send CLI in FROM field' option, the system will send the number of the caller as CLI.*

- **Send Called Party Number In**: ANANT UCS provides you the option to Send Called Party Number in "To" or "Request URI" field. You may select — "To, Request URI", "To", "Request URI". Default: To, Request URI.

  - If you select *To*, then the called party number will be sent in "To" field, whereas SIP ID configured on the trunk will be sent in the "Request URI" field.

  - If you select *Request URI*, then the called party number will be sent in the "Request URI" field, while SIP ID configured on the trunk will be sent in the "To" field.

  - If you select *To, Request URI*, then the called party number will be sent in both the fields.

  If the SIP ID is not configured and you select the option — *To* or *Request URI*, then the called party number will be sent in both the fields.

  If the called party number is not available in any of the above cases then the remote server address will be sent in the selected field.

- **Add P-Asserted-Identity/P-Preferred-Identity:** Select Add P-Asserted-Identity/P-Preferred-Identity check box, if supported by your Service Provider.

- **Identity:** Select P-Preferred-Identity or P-Asserted-Identity as Identity.

- **Caller ID:** Select the desired Caller ID option:

  - **SIP ID:** If you want the SIP ID configured on the SIP Trunk to be displayed as the Caller ID.
  - **Calling Party Wise:** If you want the Calling Extension Number (the number of the extension making the outgoing call through the SIP trunk) to be displayed as the Caller ID.
  - **Fixed Number:** If you want a specific number to be sent as the Caller ID. When you select this option, you must also define the number to be sent as the Caller ID.

- **Fixed Number:** If you select Fixed Number as the Caller ID option, you must configure the desired number in Fixed Number. The Fixed Number may consist of a maximum of 40 characters. Valid characters are from 0-9 and plus (+)

- **Default Transport for Outgoing Message:** Configure this parameter for Proxy SIP Trunks only. ANANT UCS supports three options for transporting outgoing SIP messages:

  - **UDP:** Outgoing messages are transported using UDP. By default this option is selected.

- **TCP:** Outgoing messages are transported using TCP. If you select this option, you must enable 'SIP Over TCP' on the 'VoIP Parameters' page.

- **TLS:** Outgoing messages are transported using TLS. If you select this option, you must enable 'SIP Over TLS' on the 'VoIP Parameters' page.

  By default, UDP is selected as the Default Transport for Outgoing Message.

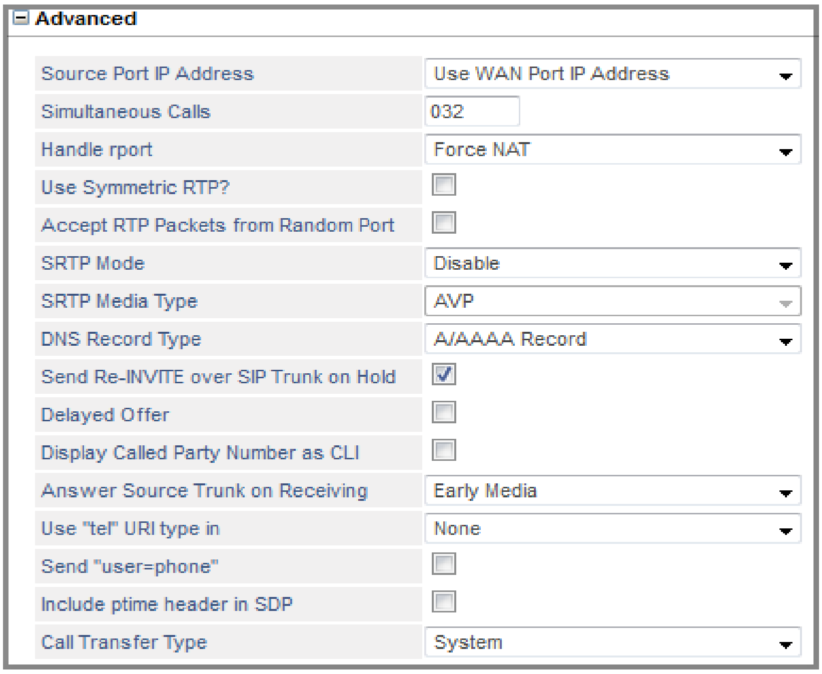
- *The Default Transport for Outgoing Message options are checked only if you have enabled SIP over TCP or SIP over TLS.*
- *If the SIP over TCP and SIP over TLS are disabled, all outgoing SIP messages will be transported over UDP only.*

- **Outgoing (OG) Reference ID:** Assign an Outgoing Reference ID for the SIP Trunk, from 00 to 99. By default, 00 is assigned as Outgoing Reference ID.

- **Return Call to Original Caller (RCOC):** Enable this check box if you want to apply the 'Return Call to Original Caller' on this SIP trunk.

  If this feature is enabled on the SIP trunk, the system will route calls returned by remote parties back to the extensions that originally made the call from this Trunk (the original callers' extensions). To know more, refer the feature description for "RCOC (Return Call to Original Caller)".

Advanced



- **Source Port IP Address:** Select the Source Port IP Address for the SIP trunk. You may select from any of the following options, as applicable to your installation scenario.

  - **Use WAN Port IP Address:** Select this option, if the WAN Port is connected directly to the public Internet.

- **Use IP Address fetched using STUN:** Select this option, if the WAN Port is located behind a NAT router other than Symmetric.

- **Use Router's Public IP Address:** Select this option, if the WAN Port is located behind a NAT router (any type).

- **Use LAN Port IP Address:** Select this option, if the LAN Port is connected within the LAN network.

- **Simultaneous Calls[54]:** Select the number of simultaneous calls you want to allow on this SIP Trunk. Default: 32.

  The number of simultaneous SIP calls depends on the number of simultaneous calls allowed by the ITSP with whom you have subscribed this SIP Trunk.

  If the ITSP supports less than 32 simultaneous calls on SIP Trunks, you must program this parameter accordingly.

- **Handle rport:** Select the desired option from:

  - **Force NAT:** By default, this option is selected. The system will not check Contact/Via header etc. while sending SIP messages and will follow Symmetric Signaling.
  - **RFC 3581:** Select this option, if you want the system to follow Standard RFC's while sending SIP messages.

  By default it is Force NAT.

- **Use Symmetric RTP?:** The use of Symmetric RTP makes it possible for a SIP device to send the RTP on the same connection on which it is listening for RTP. This is done only on peer to peer SIP trunks.

  Enable this Check box, if the WAN Port is located on a public IP and you want outgoing calls to the SIP Client located behind the NAT Router OR if you need to receive incoming calls from the SIP Client located behind the NAT router. By default, Use of Symmetric RTP is disabled.

- **Accept RTP Packets from Random Port:** By default, this is disabled, that is, the System will accept RTP packets from the negotiated RTP port only.

  Enable this flag, if you want the system to accept RTP packets from any random port.

Accept RTP Packets from Random Port parameter is applicable only if:
- *Use Symmetric RTP? is disabled.*
- *RTP Mode is set as Transcoding or RTP Relay. For details, see* *"Configuring VoIP Parameters".*

- **SRTP Mode:** ANANT UCS supports SRTP (Secure Real Time Protocol) for secure conversations over SIP. ANANT UCS supports the following options:

  - **Disable:** ANANT UCS uses normal RTP for transporting the speech packets.

  - **Optional:** ANANT UCS uses SRTP for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will use normal RTP for transporting the speech packets.

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54. *This parameter is only applicable if you have set the RTP Mode as Transcoding Mode in "Configuring VoIP Parameters".*

- **Forced:** ANANT UCS uses only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, ANANT UCS will reject incoming calls from and drop outgoing calls made to such users.

  By default, SRTP Mode is Disabled.

- **SRTP Media Type**: If you select **Optional** as the **SRTP Mode** option, you must configure the **SRTP Media Type**. You may select **AVP** or **SAVP**. By default, AVP is selected as the SRTP Media Type.

- **DNS Record Type:** Set this parameter as provided to you by your Service Provider. You may select any of the following options:

  - **A/AAAA Record:** Select this option, if you want ANANT UCS to send the query to the DNS Server to fetch the IP Address of the Target Server on which further SIP messages are to be sent.
    A Record maps to an IPv4 Address of the Target Server, while AAAA (quad-A) Record maps to an IPv6 Address.

  - **SRV Record:** Select this option, if you want ANANT UCS to send the SRV query to the DNS Server to fetch the Destination Port. The system will also make a DNS A query to fetch the IP Address of the Target Server on which further SIP messages are to be sent.

  - **NAPTR/SRV Record:** Select this option, if you want ANANT UCS to send queries to the DNS Server to fetch the Transport, Port and IP Address of the Target Server on which further SIP messages are to be sent.
    By default, A Record is set as the DNS Record Type.

This parameter will be checked when:
- *SIP Trunk Mode is configured as Proxy.*
- *Registrar Server Address is configured as Domain and Outbound Proxy Address is blank* ***OR*** *Registrar Server Address is configured as Domain/IP Address and Outbound Proxy Address is configured as Domain.*

- **Send Re-INVITE over SIP Trunk on Hold:** With this parameter you can select whether or not the system should send Re-INVITE message from the SIP Trunk to the Remote Peer, when an external call over the SIP Trunk is put on hold by the extension user. Set this parameter as per the requirement of the Remote Peer. The Remote Peer can be a Proxy Server or a SIP Device.

  By default, Send Re-Invite over SIP Trunk on Hold is enabled. For more information see “Call Hold”.

- **Delayed Offer:** Select this check box to enable, if you want the SIP Trunk to generate INVITE without SDP. By default it is disabled.

- **Display Called Party Number as CLI:** If you want ANANT UCS to display the called party number received in the INVITE message as the CLI, select the Display Called Party Number as CLI check box. By default, Display Called Party Number as CLI option is disabled.

  This parameter is useful when a single SIP Trunk having DDI Numbers and Operator are shared by more than one organization. If you enable this option, make sure:
  - you configure the names and corresponding numbers of the organizations sharing the SIP Trunk in the Global Directory of ANANT UCS.
  - the Operator has an Extended IP Phone or a Mobile UC Client.

With this option enabled the Operator will be able to handle calls more efficiently. When there is an incoming call, ANANT UCS matches the number with the numbers in the Global Directory. If a match is found ANANT UCS displays the company name configured for that entry to the Operator, that is, the CLI will display the called party number and name.

After the Operator answers the call, the CLI will change and display the calling party number and name (if configured in the Global Directory).

If you keep this option disabled, the calling party number and name will be displayed as the CLI, both during an incoming call and after the call is answered by the Operator.

You can configure the Display Called Party Number as CLI option only from Jeeves.

- **Answer Source Trunk on Receiving**: When a call received on any trunk of ANANT UCS is routed through the SIP Trunk, select the response after which the call on the source trunk must be answered. You can select **Early Media** or **200 OK**.

- **Use “tel” URI type in:** If your ITSP requires “tel” in the URI so that it can handle/ route calls to/from global numbers, select the desired option from any of the following:
  - TO, Request URI
  - FROM
  - TO, Request URI and FROM

  The system will send 'tel' URI in the selection headers and Request URI according to the selection you make. This will be used while making an Outgoing call.
  By default **None** is selected.

- **Send "user=phone":** Select this check box, if you want ANANT UCS to add user=phone in the Request URI/FROM/TO header of the INVITE message. Default: Disabled.

  ANANT UCS will send user=phone, only if the SIP ID is numeric.

- **Include ptime header in SDP:** Enable this check box if you want to add ptime header in SDP offer and answer.

- **Call Transfer Type:** Select the **Call Transfer Type as** supported by your ITSP. System will send the Call Transfer request to the transferee only if calls are routed through the same SIP Trunk. You may select:

  - **System:** In this option, ANANT UCS will handle the Call Transfer locally.
  - **Network:** In this option, the Call Transfer is handled by the Network.

  By default it is System.

Call Redirection

When an incoming call is diverted/forwarded to an external number using the same SIP Trunk, configure the following parameters as required by your ITSP:

- **Call Redirection Type:** Select the desired Call Redirection Type. You may select **Generate new Call** or **Send 302**. By default it is Generate new Call.

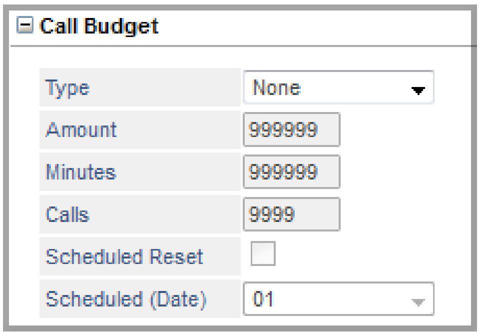
- **Sending of 181 responses upon Redirection:** If you want the system to send 181 response to the source upon redirection, enable the Sending of 181 responses upon Redirection check box. By default it is disabled.

- **Include "Diversion" while diverting the call**: If you have selected Generate new Call as the Call Redirection Type, you may select the **Include "Diversion" while diverting the call** check box. The system includes incoming call request information in Diversion header, along with the reason of redirection. By default it is disabled.

- **Include "History-Info" while diverting the call:** If you have selected Generate new Call as the Call Redirection Type, you may select the **Include "History-Info" while diverting the call** check box. The system includes the incoming call request information as well as the new call information in the INVITE/ REINVITE requests generated on the target. By default it is disabled.

- **CLI while Diversion/Redirection:** If you have selected Generate new Call as the Call Redirection Type, select the desired option in **CLI while Diversion/Redirection**. You may select **Self** or **Received**. By default it is Self.
  If you select **Self**, the system will send SIP Trunks' Identity and if you select **Received** the system will send the received CLI to the target.

Call Budget



Call Budget: By default Call Budget is enabled, if you wish to change the default settings or disable it, configure the following parameters:

- **Type:** Select the type of Call Budget on Trunk—Amount or Minutes or Calls—to be applied on this SIP trunk. By default, Minutes is selected as the Call Budget type.To disable select Type as None.

- **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 99999.

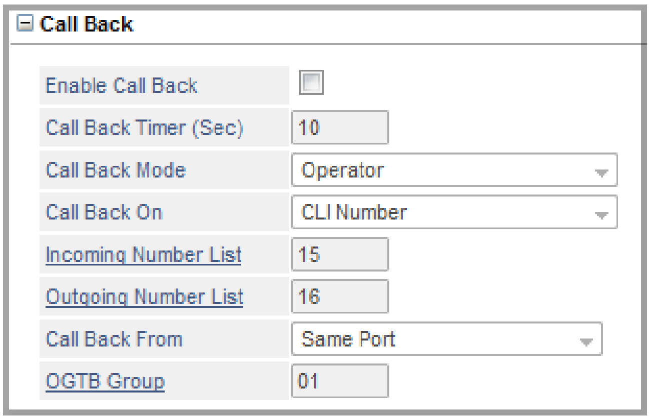
- **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.

- **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.

- **Scheduled Reset:** Select this check box to enable, if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.

- **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.

Call Back



This parameter is related to the “Call Back on Trunk Port’ feature. If you want to enable the 'Call Back on Trunk Port' feature on this SIP trunk, configure the following parameters:

- **Enable Call Back:** Select this check box to activate the Call Back on Trunk Port feature. By default, this check box is disabled (clear) on all SIP trunks. By default, the Check box is disabled.

- **Call Back Timer:** This is the duration for which the system waits for the caller to disconnect before applying the Call Back. The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds for all SIP Trunks.

- **Call Back Mode:** Select from the following options how a ‘Call Back’ call answered by the remote party should be routed:
  - PIN Authentication - Multiple Calls
  - CLI Authentication - Multiple Calls
  - CLI Authentication - Single Call
  - Operator

  By default, Operator is selected as the Call Back Mode.

- **Call Back on:** This parameter allows you to select if the call back should be made to the same number that was received or to a different number. If you want the call back to be made to the same number select the ‘CLI number”. If you want the call back to be made to a different number, select ‘Alternate Number’.

  By default, CLI number is selected for Call Back.

- **Incoming Number List:** Configure the number strings that are eligible for Call Back in this List. By default, Number List 15 is assigned to Call Back Incoming Number List.

  Number List 15 is also assigned to all SIP trunks. If you want the same numbers strings to be configured commonly for all SIP trunks, retain this list.

  If you want a different set of number strings to be configured for this SIP Trunk, select a different Number List, and assign it to the SIP trunk port.

  You may configure the Incoming Number List either from the ‘Number List’ page or by clicking the ‘Incoming Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Outgoing Number List:** Configure the number strings that are to be called back in this List. For each number string you configured in the ‘Incoming Number List’, you must configure in the corresponding index in the Outgoing Number List a number to which the call back is to be made. For example, for the number string programmed at Index 1 in the Incoming Number List, a corresponding number string at the same Index, Index 1, should be configured in the ‘Outgoing Number List’.

  By default, Number List 16 is assigned to Outgoing Number List.The same Number List 16 is also assigned to all SIP trunks.

  You may configure the default number list, or a different number list and assign it to this SIP Trunk port.

  You may configure the Outgoing Number List either from the ‘Number List’ page or by clicking the ‘Outgoing Number List’ link to reach the Number List page.

  Refer the topic “Number Lists” to know more, and for configuration instructions.

- **Call Back from:** This parameter determines the trunk port to be used to make the call back. The call back can be made using the same port or an Outgoing Trunk Bundle Group (OGTBG).

  Select ‘Same port’ if you want the call back to be made using the same port on which the missed call is received. If you select OGTBG, the call back will be made using the OGTBG, which you have defined.

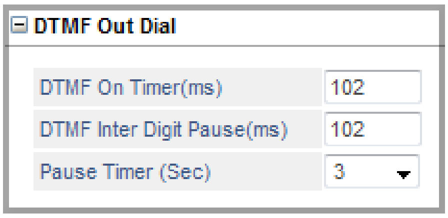
  By default, Same port is selected.

- **OGTB Group:** If you selected OGTBG for making the call back in the previous parameter, you must define the OGTBG that must be used in this parameter.

  By default, OGTBG 01 is assigned to all SIP trunks.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTBG that you define here for Call Back.

DTMF Out Dial



- **DTMF ON Timer (ms):** This is the time for which the DTMF digit will remain ON, while being out dialed by ANANT UCS. This parameter finds its application in the feature "Multi-Stage Dialing". The range of this timer is from 051 to 255 milliseconds. By default, ON Time is defined as 102 msec.

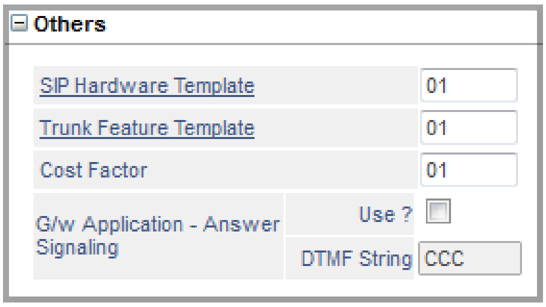
- **DTMF Inter Digit Pause (ms):** This is the time for which ANANT UCS will wait before dialing the successive digits.

- **Pause Timer (sec):** This Timer is required for inserting delay while digits of a number string are out dialed from the SIP trunk. The Pause Timer will be applicable when the letter 'P' is configured in the DTMF number string which is to be out dialed as DTMF digits on the SIP trunk. The range of this timer is from 1 to 9 seconds. By default the Timer is set to 3 seconds.

  For example, if 'PPP3' is to be out dialed and Pause timer is configured as 3 seconds, ANANT UCS will out dial the digit 3 after 9 seconds, after a delay of individual P (3+3+3 =9). The range of this Timer is from 1 to 9.

  This parameter is used for the "Multi-Stage Dialing" feature.

Others



- **SIP Hardware Template:** Assign a "SIP Hardware Template" to the SIP Trunk. The SIP Hardware Template contains voice quality related features such as Voice Codec selection, Tx and Rx Gains, Echo Cancellation, Jitter Buffer and related parameters.

  There are 32 different templates to choose from. Each template can also be altered to suit your requirement and preferences. By default, Template 01 is assigned to all SIP Trunks. Template number 01 is also assigned to all SIP Extensions.

  First, check if the values in Template 01 fulfill the feature requirements of the SIP Trunks. Retain this template, if it fulfills the feature requirements of all SIP Trunks and if the same features are to be allowed to all SIP Trunks.

If different sets of SIP hardware features are to be allowed to different SIP Trunks, then prepare separate SIP Hardware Templates and apply them on the SIP Trunks. To do this,

- Under **VoIP Configuration**, click **SIP Hardware Template**.
- Select a Template number, for example 03.
- Customize Template number 03 and click **Submit**.
- Now go back to the SIP Trunks page.
- Enter the number of the Template you customized, Template 03 in the SIP Hardware Template field of the SIP Trunk on which you want to apply this template.
- Click Submit.

- Repeat the same steps to customize and assign a different SIP Hardware Template to another SIP Trunk.

- Also, refer the topic “SIP Hardware Template” to know more about customizing the templates and applying on the SIP Trunks.

- **Trunk Feature Template:** A Trunk Feature Template is a set of features like Time Table, Operator, Auto Attendant, DISA, Trunk Landing Group, SMDR Storage, etc., that defines the behavior of a Trunk. Apply a Trunk Feature Template to the SIP trunk. By default, Trunk Feature Template 01 is applied on all SIP trunks. Refer the topic “Trunk Feature Template” to know more.

  Click the 'Trunk Feature Template’ link to open the page. Check if the default Template 01 fulfills your requirement for the SIP trunk.

  If the default Template 01 does not fulfill your requirement, prepare another Trunk Feature Template[55], and enter the newly prepared Template number for the SIP trunk.
- **Cost Factor:** This parameter is of relevance only if 'Least Cost Routing' feature is applied on the SIP trunk.

  Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.

  Assign a Cost Factor to the SIP trunk, for instance, 02 and configure Least Cost Routing Table accordingly.

  For example, if you want to route all outgoing calls starting with number '9' through the SIP trunk 01 only,

  - You must first assign a Cost Factor (01-99) to SIP trunk 01, for example, 02.
  - Click the 'Least Cost Routing - Number Based' link to open the page.
  - Enter '9' in the 'Number' column, Cost Factor '02' as Preference 1, 2, 3 and 4.
  - Click Submit.

    All outgoing calls assigned Cost Factor trunk 02 will be made from SIP trunk 01.

- **Gateway Application - Answer Signaling:** This parameter is to be configured if the SIP trunk is being used in a gateway application as a source port (from where calls originate). The calls originated on the SIP trunk (source port) are routed using another SIP Trunk (terminating port).

  - **Use?:** Enable this check box if you want the SIP trunk to be used in a Gateway Application.

  - **DTMF String:** Program the DTMF digits to be sent to signal call maturity to the source port.

55. *The default template is applied on all SIP trunks supported by ANANT UCS. Changes to the default template will be applied on all trunks also. So, you are advised to prepare a new template and apply it to the desired trunks.*

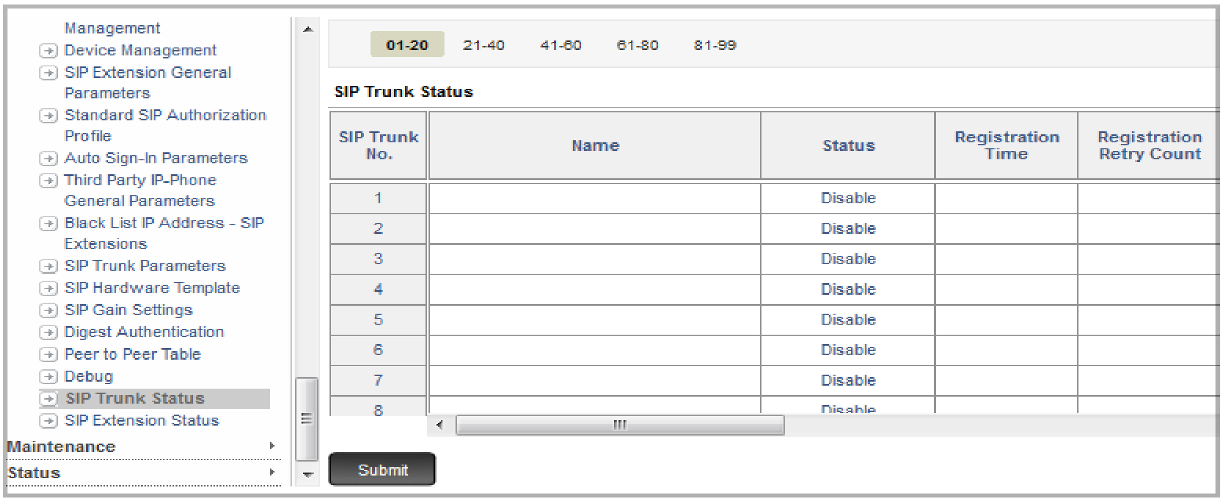
- Click **Submit.**

Viewing SIP Trunk Status

You can also view the settings of the SIP Trunk Parameters.

To do this,

- Under **VoIP Configuration**, click **SIP Trunk Status**.



- For each SIP trunk (number), the following settings will be displayed:
  - SIP Trunk Number
  - Name
  - Status
  - Registration Time
  - Registration Retry Count
  - Reason for Failure
  - Call Budget Type
  - Allotted Amount/Minutes/Calls
  - Consumed Amount/Minutes/Calls
  - Scheduled Reset
  - Budget Reset Scheduled (Date)
  - Reset Consumed Budget (this is not a status indicator. It is for resetting the Consumed Call Budget manually)

*You can also view the SIP Trunk Status from the **Status** link. To view, click the SIP Trunk Status link under Status.*

Configuring LCR

Least Cost Routing (also referred to as Automatic Route Selection) is an expense control feature of ANANT UCS.

Least Cost Routing (LCR) is useful when there are different trunk lines for making outgoing calls, and the service providers of these trunks offer different tariffs for calls made to certain locations or numbers or during a particular time of the day.

When a call is made from an extension of ANANT UCS, LCR recognizes where the call is going. Depending upon how the LCR is configured, the system routes the call through the assigned trunks.

The system can be configured to select the most cost effective trunk for the time of the day when the call is made from the extension or to select the most cost effective trunk for the destination number dialed from the extension or to select the most cost effective trunk considering both time of the day and destination number.

Accordingly, ANANT UCS supports four types of LCR which can be configured, namely:

1. **Time-based LCR:** This type of LCR may be used when you have trunk lines of more than one service provider, and each offers a different tariff according to the time of the day.

   For example, Service Provider 1 offers a lower tariff for calls made between 9 am to 8 pm, while Service Provider 2 offers a lower tariff for calls made between 8 pm to 9 am.

   When Time-based LCR is configured, the system uses the Online-dialing logic, whereby digits dialed by the user are directly passed on to the trunk.

2. **Number based LCR:** This type of LCR may be used when you have trunk lines of more than one service provider, and each offers different tariffs according to the area or distance, or phone numbers dialed. For instance, Service Provider 1 provides cheaper calling rates for calls made from City A to City B, than Service Provider 2 and Service Provider 3.

3. **Time and Number based LCR:** This type of LCR is a combination of number and time based LCR, that is, the service providers offer different tariffs according to the time of the day as well as area/distance.

   For example, Service Provider 1 offers lower rates for calls made from City A to City B during peak hours 9 am to 8 pm, as compared to Service Provider 2, whereas Service Provider 2 offers cheaper rates for calls made from City A to City B during off peak hours (8 pm to 9 am).

   When Time+Number-based LCR is configured, the system uses Store and Forward dialing logic, whereby digits dialed by the user are first stored at a memory location in the system, and then dialed out on the assigned trunk.

4. **Service Provider-based LCR:** This type of LCR may be used when the same Service Providers offer different rates for calls made to numbers within their own network and for calls made to numbers of another Service Provider's network. For example, Service Provider 1 offers lower rates to call a Service Provider1 number in City A and in City B, than for calling numbers of Service Provider 2 in the same cities.

   This type of LCR may also be used when the same Service Providers apply different charges for different subscriber services provided by them. For example, Service Provider 1 offers both Voice Mail as well as Call Recording services and applies different charges for providing both these services.

When Service Provider-based LCR is configured, whenever a number is dialed out, the system ignores the area code, checks the number in the 'Service Provider-based LCR table', and routes the call according to the trunk configured for that number.

ANANT UCS also supports LCR based on Carrier Pre-Selection. This type of LCR is useful where there exist different service providers for local and long distance calls. Refer the topic "Least Cost Routing-Carrier Pre-Selection" to know more.

Cost Factor

For LCR to work, all trunks that are allotted to extensions for making outgoing calls, must first be assigned a Cost Factor.

Cost Factor is a number assigned to each trunk for identification. This number also serves as a preference number for the trunk. The Cost Factor can be from 1 to 99. Trunks having the same preference must be assigned the same Cost Factor. Different trunk types can also be assigned the same Cost Factor. These trunks are used for routing calls.

By default all trunks are assigned Cost Factor number 01.

After assigning Cost Factor to Trunks, you must configure the Type of LCR to be used on Trunks in the Outgoing Trunk Bundle Group (OGTBG) allotted to the extensions for making calls.

Assigning Cost Factor to Trunks

- On a sheet of paper, make a list of all the trunks assigned to extensions for making outgoing calls.
- Make a table of the trunk types and assign a cost factor to each trunk type, as shown below.

Trunk Number	Service Provider	Cost Factor
SIP Trunk-01	BSNL	01
SIP Trunk-02	BSNL	02
SIP Trunk-03	You Telecom	03
SIP Trunk-04	BSNL	04
SIP Trunk-05	Airtel	05
SIP Trunk-06	Jio	06
SIP Trunk-07	BSNL	07
SIP Trunk-08	Tata Docomo	08

Configure the Cost Factor number you assigned to the Trunk in the SIP trunk parameters. For instance, assign Cost Factor 01 to SIP Trunk-01 and Cost Factor 02 to SIP Trunk-02.and so on.

Configuring Time-based LCR

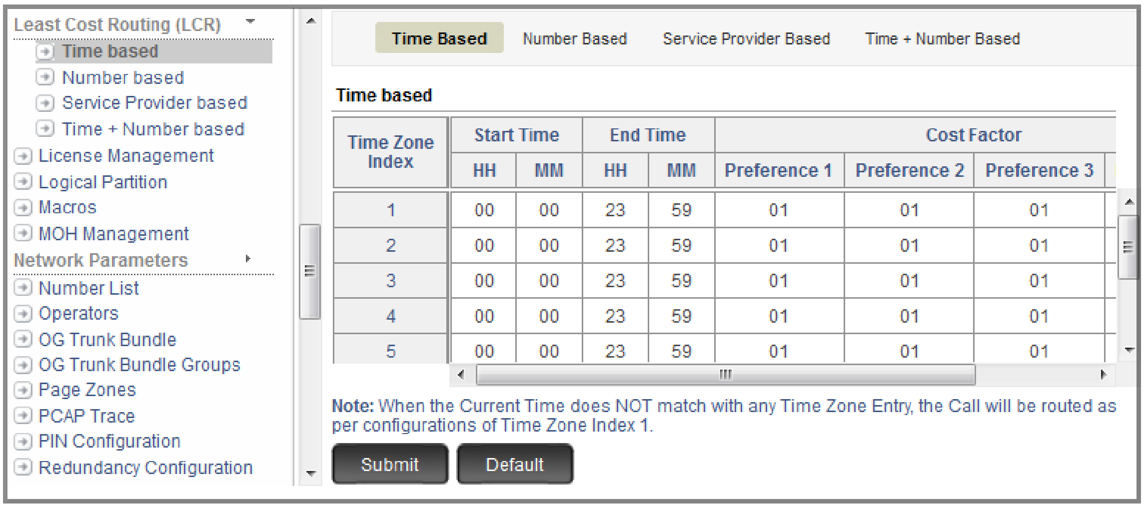
- You can configure Time-based LCR for as many as 8 different Time Zones.
- On a sheet of paper, make a table for Time-based LCR.
- Define the Time Zone, that is, the start and end time, when the LCR should be applied for the outgoing calls. The Time Zone you define is stored at an Index number from 1 to 8.
- For each Time Zone that you define, select the Trunk as your first preference, that is, Preference 1. Select the trunk of your second, third and fourth preference. When the trunk you selected as first preference is busy, the system will route the call through the next trunk you have set that is free.
- Refer to the table you prepared for assigning Cost Factor to trunks.
- For example, you want calls made during 9 am to 8 pm to be routed through BSNL SIP trunks (SIP-01 and -02). If these trunks are busy, you want the system to route calls through the Reliance SIP trunks (SIP-03). When this line is busy, you want the system to attempt to route calls through the Tata SIP Trunk (SIP-04).
- You want calls made between 8 pm to 9 am to be routed through BSNL SIP trunk 01 only.
- At Time Zone Index 1, define the Time Zone start and end time in 24 Hours:Minutes format, enter the Cost Factor you assigned to SIP-01 (01) and SIP-02 (02) as Preference 1 and Preference 2 respectively. Enter the cost factor you assigned to SIP-03 (07) and SIP-04 (08) as Preference 3 and Preference 4 respectively.

Time Zone Index	**Time Zone**		**Cost Factor**			
	Start Time (HH:MM)	End Time (HH:MM)	Preference 1	Preference 2	Preference 3	Preference 4
1	09:00	20:00	01	02	07	08
2	20:01	08:59	01	01	01	01
3						
4						
5						
6						
7						
8						

- Similarly, at Time Zone Index 2, define the Time Zone in 24 Hours: Minutes format. Enter the Cost factor you assigned to SIP-01, that is, 01 as Preference 1, 2, 3, and 4. When calls are made during this time period, they will be routed through SIP-01 only.
- If you have finished defining Time Zones and the preferred trunks for the time zones, configure the Time-based LCR using Jeeves.

Configuring Time-based LCR

- Login as System Engineer.
- Under **Configuration**, click **Least Cost Routing (LCR)**.
- Click **Time-based.**



- Enter the values of the Time-based LCR you prepared on the sheet of paper in the appropriate fields.
- Click **Submit.**

Configuring Number-based LCR

- You can configure Number-based LCR for as many as 99 different Numbers, which are stored against Index numbers from 01 to 99.
- On a sheet of paper, make a table for Number-based LCR.
- Enter each of the number strings at an Index number from 01 to 99. A Number string may be a complete telephone number, a truncated phone number or an area code. The number can be a maximum of 64 characters (Digits + Wildcards). Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. See “Wildcard Characters” to know the various number patterns you can use.
- For each number string you enter, select a Trunk as your first preference, that is, Preference 1. Select the trunk of your second, third and fourth preference. When the trunk you selected as first preference is busy, the system will route the call through the next trunk you have set that is free.
- Refer to the table you prepared for assigning Cost Factor to trunks.

  For example, you want all numbers starting with the number '9', which is prefixed with a '0' when making long distance calls, so enter '9' and '09' as the number strings. For '9' as well as '09', select the SIP trunks through which the calls should be made in order of preference.

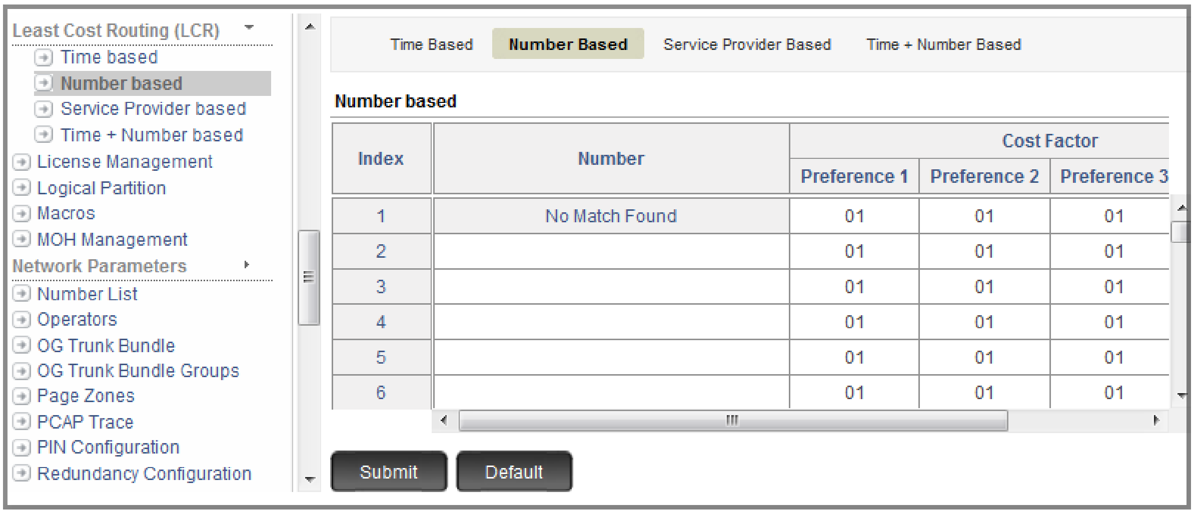
Similarly, all local numbers start with 2, so enter this number in the number string column, and select the SIP trunks in the order of preference. As in this example, you have only two SIP trunks, so you may keep the same two trunks as your preference.

Index	Number	Cost Factor			
		Preference 1	Preference 2	Preference 3	Preference 4
1	9	04	03	05	06
2	09	04	06	05	03
3	2	01	02	01	02
4					
5					
:					
:					
99					

- If you have finished entering the number strings, and selecting the preferred trunks for the numbers, configure the Number-based LCR.

Configuring Number-based LCR

- Login as System Engineer.
- Under **Configuration**, click **Least Cost Routing (LCR)**.
- Click **Number-based.**



- Enter the values of the Number-based LCR you prepared on the sheet of paper in the appropriate fields.
- In Number you can enter may be a complete telephone number, a truncated phone number or an area code. You may enter upto 64 characters (Digits + "Wildcard Characters") in this field. Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

- Click **Submit**.

Wildcard Characters

ANANT UCS supports following characters.

Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.
+	+ (plus) can be configured as a first character of the Destination Number string in the SIP Trunk only.
[-]	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
[,]	Comma within a bracket is used as a separator between the groups of numbers.
[^]	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
T (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

Refer the following table to understand how you can configure the Numbers.

Numbers	Description
1XX	Allows you to dial any number in a range from 100 to 199.
[2-5]XX	Allows you to dial any 3 digit number in a range from 200-599.
[2,3,8]XX	Allows you to dial any 3 digit number in the range from 200-299, 300-399, 800-899.
[2-9]XXXXXX	Allows you to dial any 7 digit number in the range from 2000000-9999999.
23[^2]1	Allows you to dial a 4 digit number: 2301, 2311, 2331, 2341, 2351, 2361, 2371, 2381, 2391.
2630[500-550]	Allows you to dial a 7 digit number in the range from 2630500-2630550.
[^6-7]X	Allows you to dial a 2 digit number in the range from 00 to 99 except the numbers from 60 to 79.
1234	Allows you to dial 1234 number only.
011T	Allows you to dial any number starting with 011. The number must be of minimum 3 digits and maximum digits must be as configured for the port.

Configuring Time and Number-based LCR

- This is a combination of the Time Zone-based and Number-based LCR. You may use this feature if your service providers offer lower call rates for calls made to certain numbers during a certain time of the day.
- On a sheet of paper, make a table for Time and Number-based LCR.
- Define the Time Zones when the service providers offer lower tariff. You can define up to 8 time zones.
- For each Time Zone you define, specify the Number strings on which lower tariff is applied during that Time Zone. The number can be a maximum of 64 characters (Digits + Wildcards). Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. See “Wildcard Characters” to know the various number patterns you can use.
- For each Number string you enter for a particular time zone, assign Cost Factor. Select a trunk as your first preference. Select trunks of your second, third and fourth preference. Refer to the table you prepared for assigning Cost Factor.
- You can enter up to 99 different number strings, which are stored at Index numbers from 01 to 99. The Number strings may be complete telephone numbers, truncated phone numbers or area codes.

  When the trunk you selected as first preference is busy, the system will route the call through the next trunk you set as preference if it is free.

  For example, service provider of SIP-01 and SIP-02 (assigned Cost Factor 01 and 02) offers the lowest rate for calls made to Area Code 022 between 8 am to 12 pm, followed by service providers of SIP -03 (assigned cost factor 04) and SIP-04 (assigned cost factor 03).
- Define Time Zone 1 Start and End time as 08:00 to 12:00 hours.
- Enter area code 022 as Number string at Number Index 1.
- Assign Cost Factor preference for the number string in this sequence: 01, 02, 04, 03

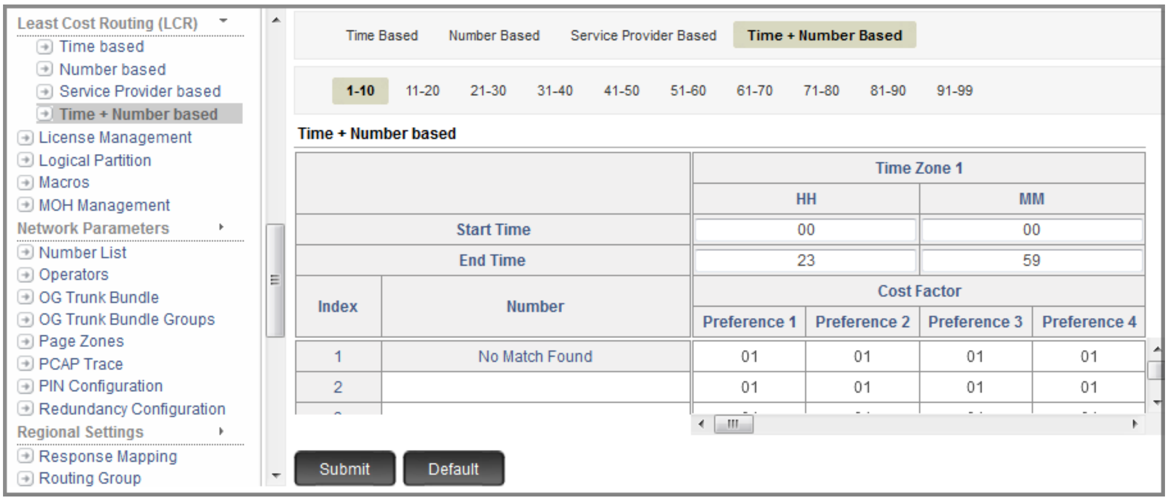
		Time Zone1				Time Zone2				Time Zone3				:	Time Zone 8
		HH		MM		HH		MM		HH		MM			
Start Time		08		00		12		00		09		00			
End Time		12		00		18		00		20		00			
		Cost Factor				Cost Factor				Cost Factor					
Index	Number	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 3	Preference 4		
1	022	01	02	04	03	04	03	02	01	03	05	06	04		
2	011	01	02	04	05										
3	080	01	02	05	06					03	05	06	04		
:															

		Time Zone1				Time Zone2				Time Zone3				:	Time Zone 8
		HH		MM		HH		MM		HH		MM			
Start Time		08		00		12		00		09		00			
End Time		12		00		18		00		20		00			
		Cost Factor				Cost Factor				Cost Factor					
Index	Number	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 3	Preference 4	Preference 1	Preference 2	Preference 3	Preference 4		
99															

- If you have finished defining the time zones, entering the number strings, and selecting the preferred trunks for the number strings, configure the Number and Time-based LCR.

Configuring Time+Number-based LCR

- Login as System Engineer.
- Under **Configuration**, click **Least Cost Routing (LCR)**.
- Click **Time + Number based.**



- Enter the values of the Time+Number-based LCR you prepared on the sheet of paper in the appropriate fields.
- Click **Submit**.

Configuring Service Provider-based LCR

- In Service Provider-based LCR, whenever a number is dialed out, the system ignores the area code, and starts checking the numbers in the 'Service Provider-based LCR table' and routes the call according to the trunk configured for that number. For this, you must configure the two parameters Area Code and Ignore Digit Count in the Area Code Table.

- On a sheet of paper make a table for Service Provider-based LCR.

Index No.	Number	Area Code	Ignore Digit Count	Cost Factor			
				Preference 1	Preference 2	Preference 3	Preference 4
01	3	080	3	08	07	01	02
02	6	022	3	07	01	02	08
:	:		:				
99	2	03852	5	01	02	01	02

- As you can see, the Service Provider-based LCR Table is similar to the Number-based LCR table.

- You can configure as many as 99 different numbers which are stored against Index numbers from 01 to 99.

- The number strings may be the complete telephone number, a truncated phone number or the first digit of the phone number. The number can be a maximum of 64 characters (Digits + Wildcards). Valid characters: 0 to 9, *, #, X, T, Comma [,], Hyphen [-], Caret [^]. See "Wildcard Characters" to know the various number patterns you can use.

- For each number string that you enter against an Index number, you must also specify the Area Code and the Ignore Digit Count.

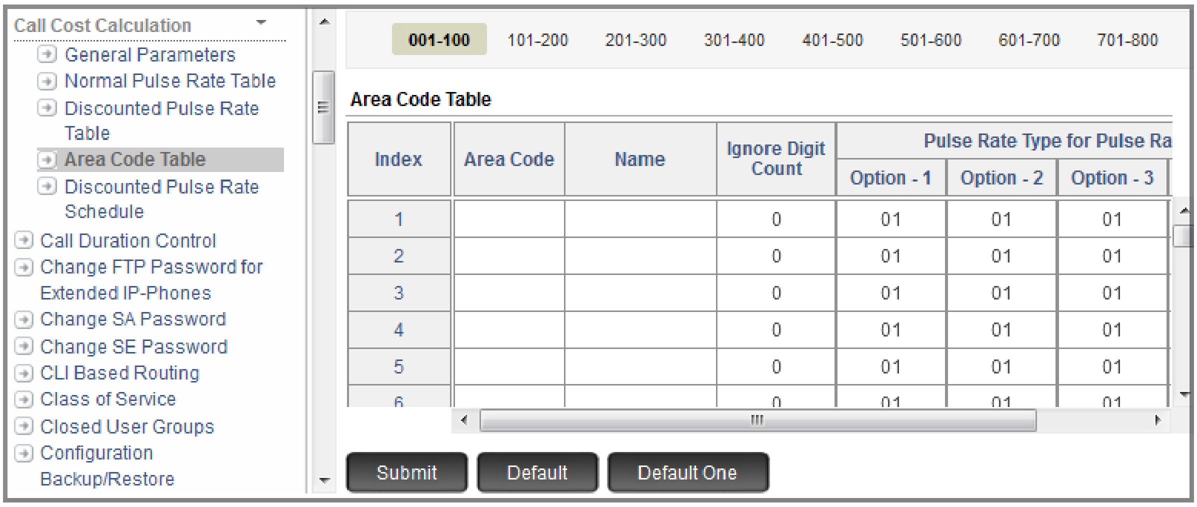
- The Ignore Digit Count is the number of digits in the area code that the system should ignore before checking the Service Provider-based LCR table. For each area code that you enter, the corresponding Ignore Digit Count will be the number of digits in the area code. For example, the area code for the number starting with '3' is 080, which consists of 3 digits. So, the Ignore Digit Count for the number/area code 080 will be 3.

- For each number string and area code that you enter, assign the Trunk of the service provider that you prefer as your first, second, third and forth preference for dialing that number/area code. Refer the table you prepared for assigning Cost Factor to trunks.

- If you have finished entering the number strings, their corresponding area codes and the Ignore Digit Count, and the preferred trunks, configure Service Provider-based LCR using Jeeves.

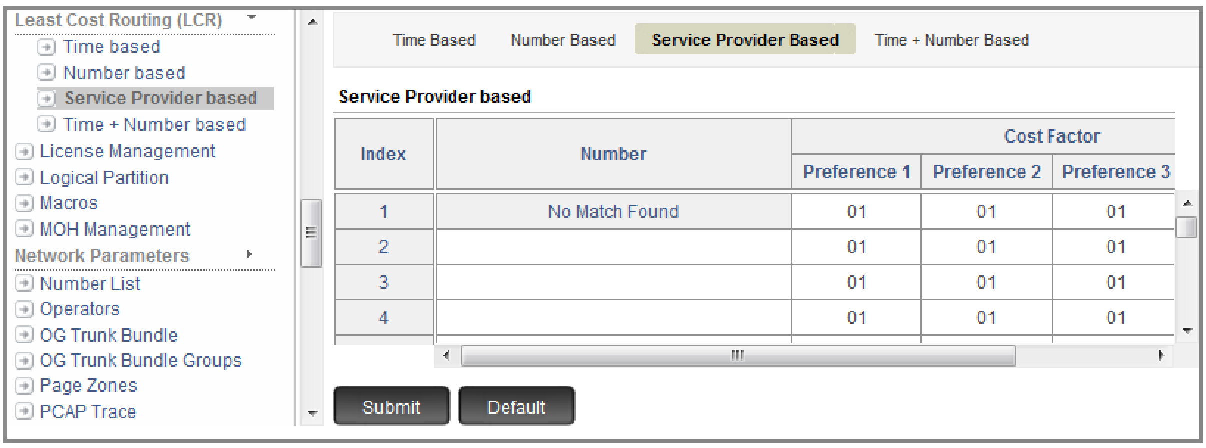
Configuring Service Provider-based LCR

- Login as System Engineer.
- To configure Area Code and Ignore Digit Count, Under **Configuration**, click **Call Cost Calculation**.
- Click **Area Code Table**.



Each area code is stored at an index number.

- Enter the Area Codes and the corresponding Ignore Digit Counts from the sheet you prepared for Service Provider based-LCR. You may also enter the respective name for each area code, if desired.
- Click **Submit.**
- Now, click the **Least Cost Routing (LCR)**.
- Click **Service Provider based.**



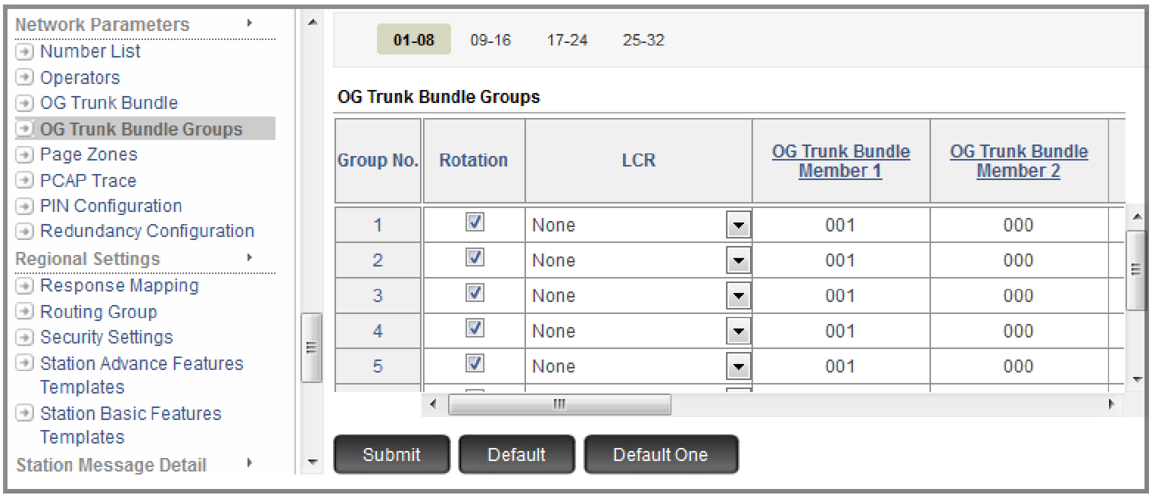
- Enter the values of the Service Provider-based LCR you prepared on the sheet of paper in the appropriate fields.
- Click **Submit.**

Configuring LCR Type on Trunks

After assigning Cost Factor to Trunks and configuring the LCR Type - Time-based, Number-based, Time and Number-based, Service Provider-based - you must now apply the desired LCR Type on the Outgoing Trunk Bundle Group (OGTBG) allotted to the extensions.

Configuring LCR Type in OGTB

- Login as System Engineer.
- Under **Configuration**, click **Outgoing Trunk Bundle Group**.



- For each OGTBG number assigned to extensions, select the desired LCR Type: Time-based, Number-based, Time+Number based, Service Provider-based (Cost Factor).
- Click **Submit.**

You can find the OGTBG number assigned to each extension from the Station Basic Feature Template assigned to the extension.

Configuring Emergency Number Dialing

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 Users license. For details, refer "Licenses Supported in ANANT UCS".

ANANT UCS supports dialing of Emergency Numbers immediately without any blocking.

The system will disregard features such as Toll Control, Call Budget, Automatic Number Translation, Call Duration Control on the extensions when dialing out Emergency Numbers. To know more, refer the topic "Emergency Dialing".

For Emergency Number Dialing, you need to configure the Emergency Number Table. In this table, each Emergency Number is to be assigned an Outgoing Trunk Bundle Groups (OGTB) through which the Emergency Number is to be routed.

The system loads the default Emergency Numbers and Outgoing Trunk Bundle Group (OGTBG) in the Emergency Number Table as per the "Configuring Region" you selected for the system. The Emergency Numbers loaded by default in the Emergency Number Table are non-editable, but you can re-assign the default OGTB, as per your requirement.

To use the feature "Emergency Calls (911) - Reporting to PSAP", make sure you assign only SIP Trunks in the outgoing trunk bundle group for dialing the number 911.

The default Emergency table as per the Region you select is given below:

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
Australia	000	106	112							
OGTB Group Number	32	30	31	1	1	1	1	1	1	1
Bangladesh	999									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Belgium	101	100	112							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Bhutan	110	112	113							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Canada	911									
OGTB Group Number	32	1	1	1	1	1	1	1	1	1

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
China	110	120	119							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Germany	110	112								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
India	100	101	108	112						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Indonesia	110	118	119	113	112					
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Italy	112									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Jordan	191	199								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Kazakhstan	03									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Kenya	999									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Kuwait	777									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Malaysia	999	112								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
Maldives	102	108								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Mauritius	999	115	144							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Mexico	911									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Namibia	911									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Nepal	100									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
New Zealand	111	112	911	08						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Oman	9999									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Pakistan	15	115	16	911	112					
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Philippine s	117	911	112							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Poland	997	999	998	112						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
Russia	02	03	01	112						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Singapore	999	995	112	911						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
South Africa	10111	10177	1022							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Spain	091	061	080	085	112					
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Sri Lanka	119	110	111							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Sudan	110	112	113							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Sweden	112									
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Taiwan	110	119								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Thailand	191	1669	199							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
Turkey	155	112	110							
OGTB Group Number	1	1	1	1	1	1	1	1	1	1

Feature	Emergency Number 1	Emergency Number 2	Emergency Number 3	Emergency Number 4	Emergency Number 5	Emergency Number 6	Emergency Number 7	Emergency Number 8	Emergency Number 9	Emergency Number 10
UAE	999	998	997	112						
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
UK	999	112								
OGTB Group Number	1	1	1	1	1	1	1	1	1	1
USA	911									
OGTB Group Number	32	1	1	1	1	1	1	1	1	1

For example, if you selected USA as Region, the default Emergency Number Table would look like this:

Default Emergency Number Table for USA

Index	Emergency Number	OG Trunk Bundle Group
01	911	32
02		01
03		01
04		01
05		01
06		01
07		01
08		01
09		01
10		01

If you selected Australia as Region, the default Emergency Number Table would look like this:

Default Emergency Number Table for Australia

Index	Emergency Numbers	OG Trunk Bundle Group
01	000	32
02	106	30
03	112	31
04		01
05		01

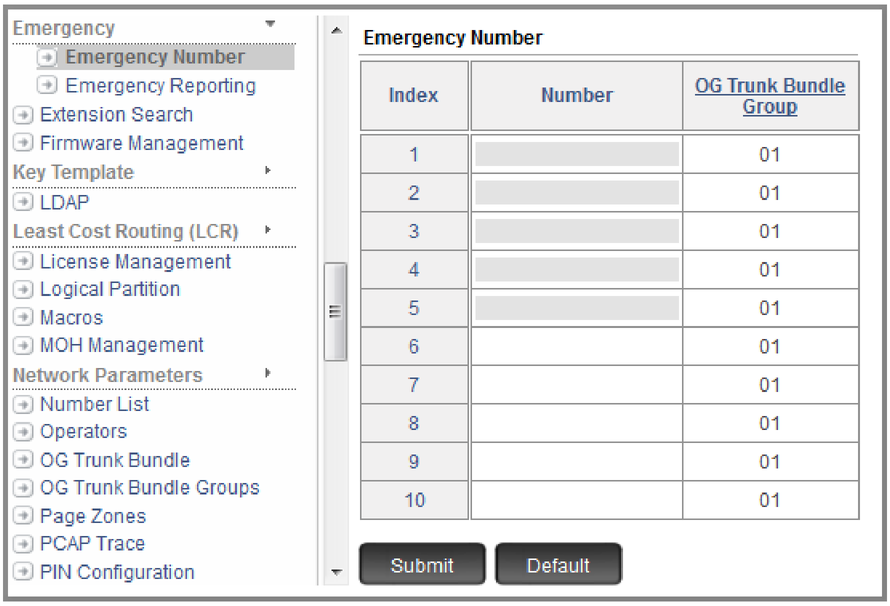
Index	Emergency Numbers	OG Trunk Bundle Group
06		01
07		01
08		01
09		01
10		01

Each Number is stored at an index from 01 to 10. The Emergency Number fields from index 01 to 5 are non-editable, but you can select a different OGTBG for each of these default Emergency Numbers.

You can add Emergency Numbers at index 06 to 10 in the table, and select the OGTBG as required.

Configuring Emergency Number Table

- Login as System Engineer.
- After you have selected the **Region.**
- Under **Configuration**, click **Emergency.**
- Click **Emergency Number.**



Click the **Default** button, the system will upload the default Emergency Numbers as per the Region you have selected.

*If you upgrade the firmware make sure you click **Default**. The system will upload the default Emergency Numbers as per the Region.*

The first five entries, at Index 01 to 05 on this table, are un-editable. These fields will be populated with the default Emergency Numbers of your country (which you selected as *Region*).

If the Emergency Numbers loaded by default are applicable for your region/country, all you need to do is re-assign, if required, the **Outgoing Trunk Bundle Group** assigned by default to each number.

If the Emergency Numbers loaded by default are not applicable for your region/country, you may add the Emergency Numbers and their OGTBG at index 06 to 10 in this table.

Make sure that the trunks configured in the OGTBG for each Emergency Number belong to the correct network and the ports through which the calls are to be routed are not disabled. For example, '112' is the default Emergency Number for the SIP Trunk. So, make sure that the SIP Trunk to be used for dialing this number is included in the OGTBG you assign to this number in the Table.

- Click **Submit**.

CHAPTER 7

Configuring Voice Mail System

Before you begin configuration of the VMS related parameters, consider the following points:

- You have purchased and activated the VMS Channels License. To know more, refer to the topic *"VMS Channels"* for more information.
- The number of VMS channels will be considered as the maximum channels available for reservation for Voice Mail Auto-Attendant parameter.
    - The channel reserved for Voice Mail Auto-Attendant configuration will not be changed if the number of channels configured in *Channel Reserved for Voice Mail Auto Attendant* parameter is increased.
    - The channels reserved for Voice Mail Auto-Attendant configuration will be changed to the number of VMS channels if the number of VMS channels available are less than the number of channels configured in *Channel Reserved for Voice Mail Auto Attendant* parameter. To know more, refer "Configuring VMS General Parameters".
- Decide which extension users are to be provided voice mail. Make a list of these extensions by their (software) port number, access codes. Configure the VMS parameters for these extensions. See "Extension Voice Mail Settings".
- If you intend to use the VMS Auto Attendant on trunks,
    - make a list of the SIP trunks.
    - configure welcome and greeting messages. You may record custom welcome messages that meet your requirements. Refer to "Voice Mail Auto-Attendant Menu".
    - configure Voice Mail Auto Attendant (VMAA) Menu for the time zones in the Trunk Feature Template(s) of the trunks on which you want to use the VMS Auto Attendant. Refer to "Trunk Feature Template".
    - assign the desired Voice Mail Auto Attendant (VMAA) Menu to the trunks in their Trunk Feature Template.
- The prompts used to route the call using the Voice Mail Auto Attendant can be customized as per your requirement. Refer to "Prompts Management".

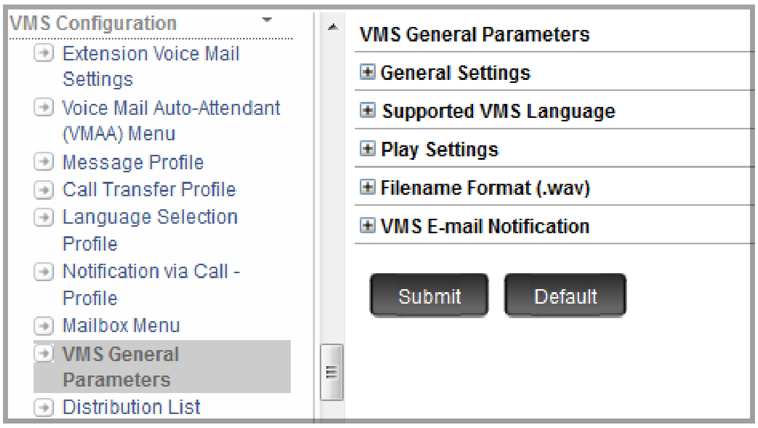
Configuring VMS General Parameters

VMS General Parameters allows you to customize VMS settings as per your requirement. You can:

- configure the VMS General Settings
- set the default language
- configure the Play Settings
- define the Filename format
- customize Voice Mail Notification Messages

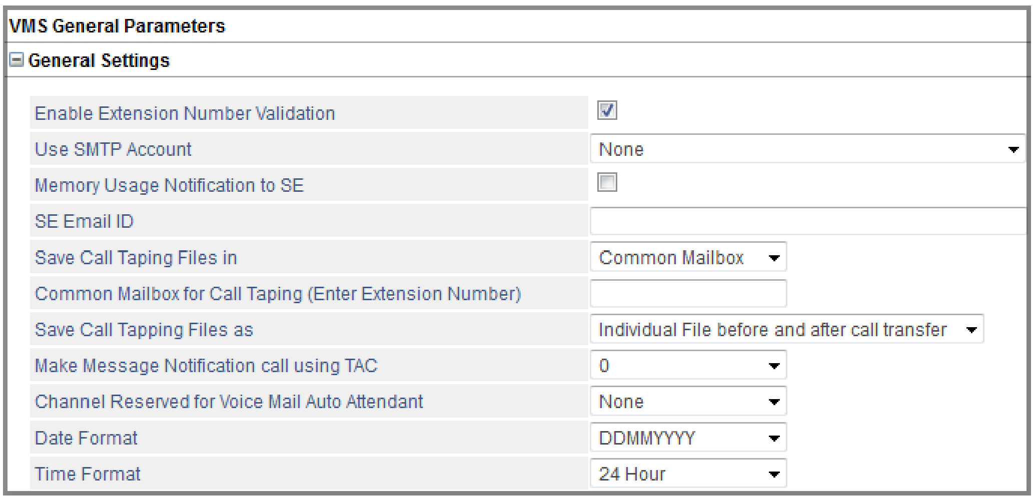
Configuring VMS General Parameters

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **VMS General Parameters**.



- Check the default values of the following parameters, and change them, if required, to the desired values.

General Settings



- **Enable Extension Number Validation:** The VMS Auto Attendant allows callers to directly reach the desired party in an organization, by giving them the option of dialing the extension number.
  When the Extension Number Validation check box is enabled, the VMS compares the extension number dialed by the external caller with the extension numbers configured in the system. If no match is found, the VMS responds with a message "*Invalid Number*".

  By default, Extension Number Validation is enabled when ANANT UCS is operating in the Enterprise mode. This flag is disabled when ANANT UCS is operating in the Hospitality mode.

- **Use SMTP Account:** Select the SMTP Account you wish to use for VMS Notifications.

  The SMTP Account you configure will be used for all VMS Email Notifications — Message Notifications and Memory Usage Notifications.

  You may add a new SMTP Account. To do so,
  - Select *Add New* option in Use SMTP Account.
  - Click **Settings** to configure the parameters of the New SMTP Account you created. For more information, see "SMTP Settings".

If you select None as the option, the system will not send any VMS Notifications — Memory Usage Notification to SE, VMS E-Mail Notification, Message Wait Notification via E-Mail — even if you have enabled and configured the respective notification.

  By default, None is selected.

- **Memory Usage Notification to SE[56]:** The VMS allows all the memory related notifications — VMS memory usage and mailbox memory usage — to be sent to the System Engineer via email. Select this check box to enable, if you want memory related notifications to be sent via email to the System Engineer. By default, it is disabled.

  You may customize the Notification Messages as per your requirement. For details, see "VMS E-Mail Notification".

  You must also specify the email address to which the notifications are to be sent in *SE Email ID*.

- **SE Email ID:** Enter the email address on which the notifications should be sent to the System Engineer. The System Engineer must have access to this email ID. The email ID may consist of a maximum of 64 characters. By default, it is Blank.

- **Save Call Taping Files in:** When Call Taping feature is enabled on your extension, you can save the Call Taping files either in the Common Mailbox or in your Personal Mailbox. By default, Call Taping files are saved in the Common Mailbox.

- **Common Mailbox for Call Taping (Enter Extension Number):** If you choose to save Call Taping files in Common Mailbox, you must specify the Access Code of the SIP Extension, Department Group or General Mailbox, whose mailbox you want to assign for Call Taping.

  If Call Taping files are saved in Common Mailbox, only the extension users who have access to the Common Mailbox will be able to retrieve and listen to the recorded conversations.

56. *This parameter will not function if you have selected None as the* ***Use SMTP Account*** *option. You may configure this parameter for future use.*

- **Save Call Tapping Files as:** If you select Common Mailbox, then select the type of file you want the system to generate for saving the taped conversation. You may select — *Individual File before and after call transfe*r or *Single File before and after call transfer* as per your requirement.

  If you select *Individual File before and after call transfe*r, the system will generate two separate files for saving the taped conversation, that is, one file containing the conversation taped before the call is transferred and another file containing the conversation taped after the call is transferred. However, if you select *Single File before and after call transfer,* the system will generate only one single file for saving the conversation taped before and after the call is transferred.

- **Make Message Notification calls using TAC:** Select the Trunk Access Code to be used by the system to make outgoing notification calls to external numbers.

- **Channel Reserved for Voice Mail Auto Attendant:** Select the number of channels of the VMS that you wish to reserve for the Voice Mail Auto Attendant. These channels will be used to answer incoming calls landing on Voice Mail Auto Attendant enabled Trunks only. These channels even if free will not be available to extension users to access their Mailbox.

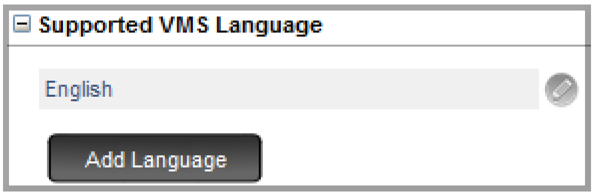
- **Date Format:** Select the Date Format — DD-MM-YYYY or MM-DD-YYYY. The Date format selected here will be applicable for all the VMS features, VMS E-Mail Notifications, etc. By default, DD-MM-YYYY is selected.

*This Date format is not applicable for **Message Profile**. The “Date Playback Format” will be applicable there.*

- **Time Format:** Select the Time Format — 24 Hour or 12 Hour. The Time format selected here will be applicable for all the VMS features, VMS E-Mail Notifications, etc. By default, 24 Hour is selected.

Supported VMS Language

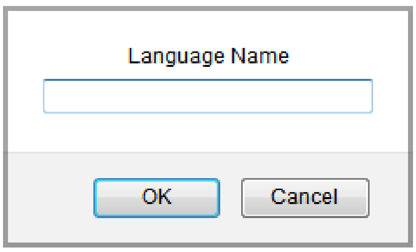
You may add a new language, edit or delete the existing languages.



To add a new language,

- Click the **Add Language** button.

  A window will pop up, asking you the language name.



- Enter the language name in the field. The allowed characters are A to Z, a to z, 0 to 9, - and _.

- Click OK.

To Edit a Language Name,

- Click .
- A window will pop up. You can edit the language name here.
- Click OK.

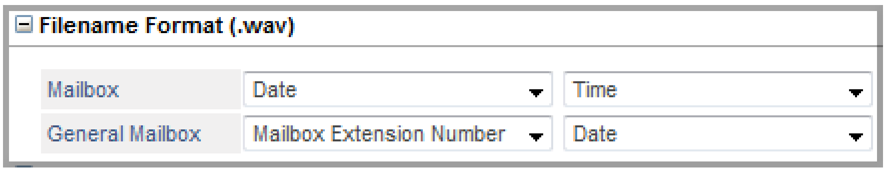
To delete a language, click .

If the language you wish to delete is already configured for any other parameter, the system will set English as the default language for that parameter.

Play Settings

- **Play '#' as**: Select the desired option — Hash or Pound. The system will play the selected option when '#' is pressed.
- **Play '*' as**: Select the desired option — Star or Asterisk. The system will play the selected option when '*' is pressed.
- **For all Options Menu, play digit**: Select the desired option — after each Option, before each Option. The system will play the digit as per the selected option for all Menu options.
- **For all Options Menu, for digit dialing play**: Select the desired option — Press or Dial. The system will play the selected option for digit dialing.

Filename Format (.wav)



You can define the Filename format for Mailbox (Personal) and General Mailbox by setting the sequence of the various parameters — None, Date, Time, Message Type, Calling Number, Called Number and Mailbox Extension Number[57]— as per requirement.

57. *The **Mailbox Extension Number** parameter is only applicable for General Mailbox Filename Format.*

While selecting the Filename format, make sure:

- *None is not selected as the first two options.*
- *Date and Time are included in the filename.*
- *The same option (other than None) is not selected twice.*

The table given below defines the parameters. You may sequence it as per your requirement.

Option	Meaning
None	The system will skip this option and move to the next option configured.
Date	Select this option if you want the system to add the Date[a] in the filename.
Time	Select this option if you want the system to add the Time[b] in the filename.
Message Type	Select this option if you want the system to add the Message Type[c] for the message stored.
Calling Number	Select this option if you want the system to add the calling number. If calling number is Blank, the system will add 'No Number'.
Called Number	Select this option if you want the system to add the called number. If Called number is not available, system will ignore this option and select the next option. This option is generally required when you wish to use the feature Call Tapping.
Mailbox Extension Number	This option is applicable only for General Mailbox. Select this option if you want the system to add the mailbox extension number from which the message is transferred to the General Mailbox.

a. The Date Format here will be as per the Date Format you have selected in General Settings.

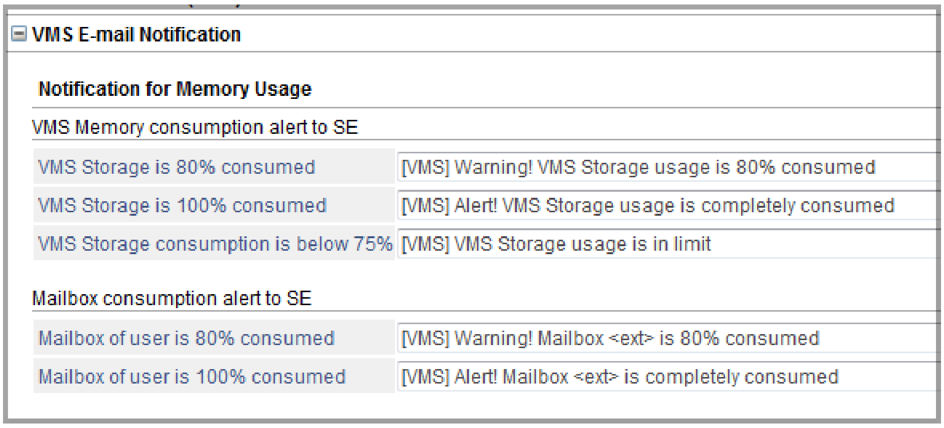
b. The Time Format here will be as per the Time Format you have selected in General Settings.

c. The Message Types supported by the system are: Call forward (CF), Call Taping (CT), Conversation Recording (CR), Broadcast Message (BM), Transfer to Mailbox (TM), Leave Message (LM), Send Message (SM), Call forward with LCS – No reply, all (LS), Redirect Message (RM), Message forward (MF).

The default sequential Filename Format is as given below:

- For Mailbox (Personal): *Date-Time-Message Type-Calling Number-Called Number.wav*
- For General Mailbox: *Mailbox Extension Number-Date-Time-Message Type-Calling Number-Called Number.wav*

VMS E-Mail Notification



Make sure you have configured the user Email ID for sending the notifications to user. For details, see “Message Wait Notification via E-Mail” in “Extension Voice Mail Settings”.

To know about the Message Wait Notification feature, see “Email Based Notification”.

The table below displays the default events for which notifications will be sent to the SE. You may customize it as per your requirement.

Event	Message	Description
Notification for Memory Usage		
VMS Memory consumption alert to SE		
VMS Storage is 80% consumed	[VMS] Warning! VMS Storage usage is 80% consumed	This email will be sent to the SE, when 80% of the VMS Storage has been consumed.
VMS Storage is 100% consumed	[VMS] Alert! VMS Storage usage is completely consumed	This email will be sent to the SE, when 100% of the VMS Storage has been consumed.
VMS Storage consumption is below 75%	[VMS] VMS USB memory usage is in limit	This email will be sent to SE, when VMS Storage consumption is below 70%.
Mailbox consumption alert to SE		
Mailbox of user is 80% consumed	[VMS] Warning! Mailbox <ext> is 80% consumed	This email will be sent to the SE, when 80% of the mailbox memory has been consumed, along with the Extension Number (if configured).
Mailbox of user is 100% consumed	[VMS] Alert! Mailbox <ext> is completely consumed	This email will be sent to the SE, when 100% of the mailbox memory has been consumed, along with the Extension Number (if configured).
Mailbox consumption alert to User		
Mailbox is 80% consumed	[VMS] Warning! Mailbox <ext> is 80% consumed	This email will be sent to the User, when 80% of the personal mailbox memory has been consumed, along with the Extension Number (if configured).

Event	Message	Description
Mailbox is 100% consumed	[VMS] Alert! Mailbox <ext> is completely consumed, please delete old messages to allow storing of new messages	This email will be sent to the User, when 100% of the personal mailbox memory has been consumed, along with the Extension Number (if configured).
General Mailbox is 80% consumed	[VMS] Warning! General Mailbox <ext> is 80% consumed	This email will be sent to the User, when 80% of the general mailbox memory has been consumed, along with the Extension Number (if configured).
General Mailbox is 100% consumed	[VMS] Alert! General Mailbox <ext> is completely consumed, please delete old messages to allow storing of new messages	This email will be sent to the User, when 100% of the general mailbox memory has been consumed, along with the Extension Number (if configured).
Notification for New Message in Mailbox		
Normal Message	[VMS] <msg_type> Message received from <cli>	This email will be sent to user's personal mailbox, when a message is received along with the Message Type and the Caller Number[a].
Conversation Recording Message	[VMS] Message received for conversation with <num1>	This email will be sent to user's personal mailbox, when the conversation between the extension user and the caller was recorded.
Call Tapping Message	[VMS] Message received for call recording between <num1> and <num2>	This email will be sent to user's personal mailbox, when the conversation between two Numbers was recorded.
Notification for New Message in General Mailbox		
New Message	[VMS] <msg_type> Message received for <ext> from <cli>	This mail will be sent to the SE Email ID configured in the General Settings, when a message with the message type[b] is received for an Extension Number (if configured) from the Caller Number.
Conversation Recording Message	[VMS] Message received for <ext> and conversation was with <num1>	This mail will be sent to the SE Email ID configured in the General Settings, when a message is received for an Extension Number (if configured) and conversation was with the Number.

a. If Caller Number is unavailable, Caller Name will be displayed. If both are unavailable, 'Unknown' will be displayed. The CLI Number can be an External Caller Number or an Internal Caller Number.

b. Message Type may be Normal or Urgent.

General Mailbox Settings

A General Mailbox is a common mailbox in the VMS, with which more than one extension users are associated. When the personal mailbox of any extension user is full, all new messages are diverted to the General Mailbox.

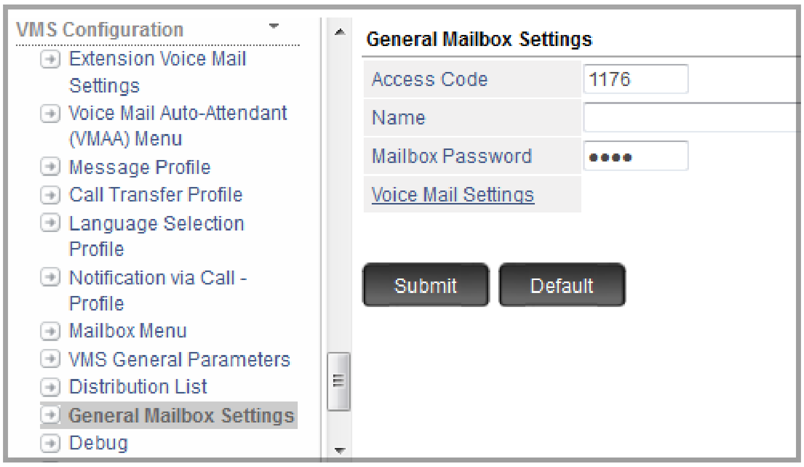
To access the General Mailbox, the extension users must have the feature General Mailbox enabled in their CoS.

The extension users can listen to the messages in the General Mailbox, by dialing the General Mailbox access code (configurable; default: 1176).

You can change the default settings of General Mailbox parameters using Jeeves.

Configuring General Mailbox Parameters

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **General Mailbox Settings**.



- Check the default values of the following parameters, and change them, if required, to the desired values.
    - **Access Code:** By default, 1176 is the access code for the General Mailbox. Extension users must dial this number, if they want to access the General Mailbox.

      If required, you may assign a different access code to the General Mailbox. The Access Code you assign may consist of a maximum of 16 digits. Digits 0-9, # and * are allowed.
    - **Name:** You can assign a Name to the General Mailbox. The name you assign can be a maximum of 18 characters. The Name must not have space as it's first character. <, >, :, ", /, \, |, ? and * characters are not allowed.
    - **Mailbox Password:** If you have selected the **Ask Password to access Mailbox** check box in *Extension Voice Mail Settings*, extensions users can access the General Mailbox by dialing the default Password-1111.

To avoid unauthorized access, we recommend you to change the default password. The password you assign may consist of a maximum of 4 digits. Valid Range: 0000 to 9999. Make sure the new password is strong and is provided to the extension users who need to access the General Mailbox only.

- **Voicemail Settings**: Click on this link to configure the Voicemail Settings for the General Mailbox. The Extension Voicemail Settings window opens.

  By default, the General Mailbox with access code-1176 will open. For information regarding the configuration, see "Extension Voice Mail Settings".

- Click **Submit** to save the settings.

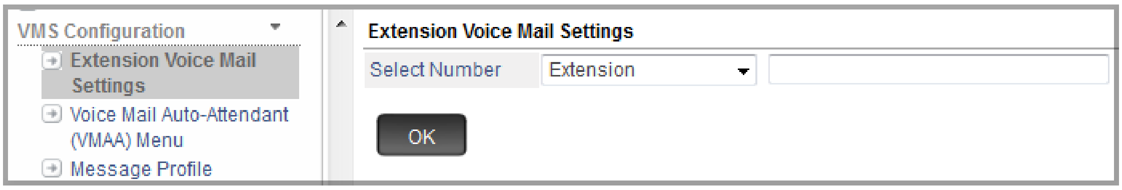
For more information and instructions on how to access the General Mailbox, see "Accessing the General Mailbox".

Extension Voice Mail Settings

Extension Voice Mail Settings allows you to configure the various VMS parameters assigned to — an Extension, a Department Group, an Operator or a General Mailbox.

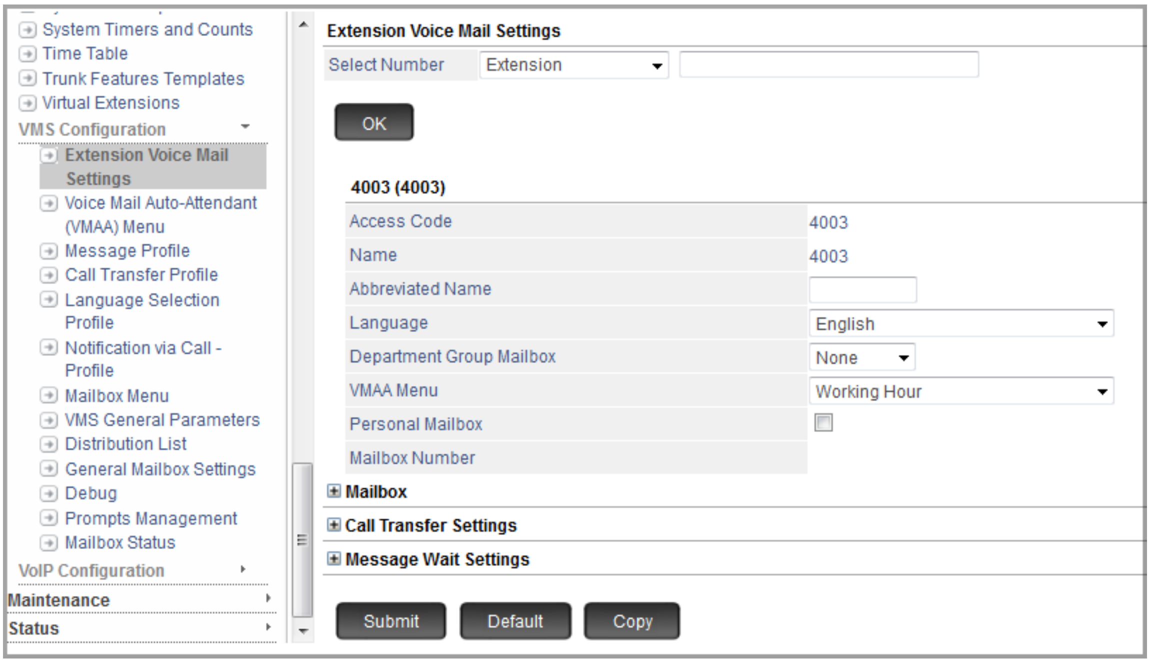
Configuring Extension Voice Mail Settings

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Extension Voice Mail Settings**.



- **Select Number:** You may select — Extension, Department Group, Operator or General Mailbox.
    - For **Extension**, enter the Extension Number/Name and then select the same from the drop down list.
    - For **Department Group**, select the desired Department Group Number from the drop down list.
    - For **Operator**, select the desired Operator from the drop down list.
- Click **OK**.

You may now configure the respective VMS parameters.



- **Access Code:** The Access Code of the respective Extension, Department Group or General Mailbox is displayed as a status.

- **Name:** The Name assigned to the respective Extension, Department Group or General Mailbox is displayed as a status.

- **Abbreviated Name:** You may configure the abbreviated name for the respective Extension or Department Group. It must be a minimum of 3 alphabetic characters and a maximum of 8 alphabetic characters.

  Abbreviated Name is applicable when you use *Dial by Name* feature to transfer the call to any extension using VMAA Menu. For details, refer “Dial By Name”.

- **Language:** Select the Language which you want the system to use when accessing the Personal Mailbox. The languages displayed as options in the drop down list is as per the *Supported VMS Language* you configure in VMS General Settings.

All the prompts related to Personal Mailbox will be played in the language you select here.

- **Department Group Mailbox:** Select the Department Group Mailbox Number which you want to assign to the respective Extension. You may select None or the Department Group Mailbox Number. The system will allow the access to the department group mailbox you select here.

  If you select None, the department group mailbox will not be accessible by the Extension even if the Extension is a member of the Department Group.

- **VMAA Menu:** Select the VMAA Menu Number which you want to assign to the respective Extension or Department Group. For more information, see “Voice Mail Auto-Attendant Menu”.

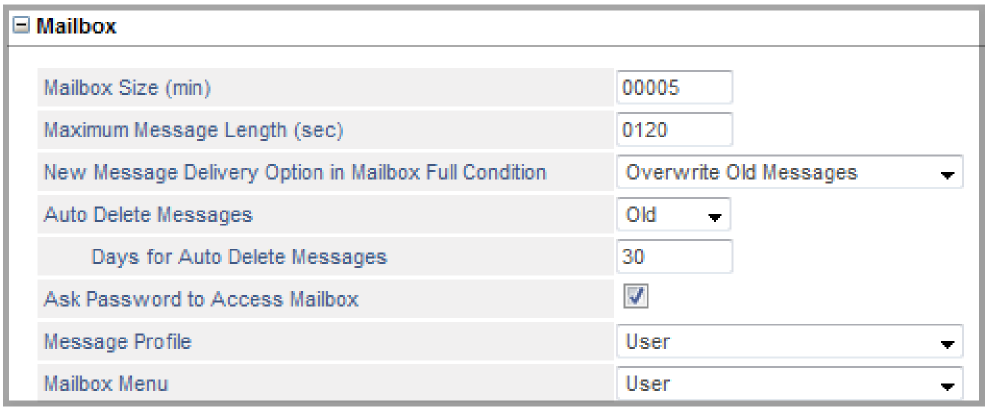
- **Personal Mailbox:** Keep the check box enabled if you want to assign the Personal Mailbox to the respective Extension, Department Group or General Mailbox. It allows you to access your Personal Mailbox.

  For Extension and Department Group:
  If you disable the check box, the Personal Mailbox will not be assigned and the system will not allow you to access your Personal Mailbox.

  For General Mailbox:
  If you disable the check box, the General Mailbox will not be assigned and the system will not allow you to access the General Mailbox.

- **Mailbox Number:** The Mailbox Number is displayed as a status when the Personal Mailbox is assigned to the respective Extension, Department Group or General Mailbox, if you have enabled the *Personal Mailbox* check box. If Personal Mailbox is not assigned, the Mailbox Number will be displayed as Blank.

Mailbox[58]



- In **Mailbox Size (min)**, configure the maximum allowed size for message storage. You may configure from 1 to 60000 minutes.

- In **Maximum Message Length (sec)**, configure the maximum time for which a message can be recorded by the caller. You may configure from 1 to 3600 seconds.

In case, the Maximum Message Length is more than the Mailbox Size, the Mailbox size will be considered for recording.

- In **New Message Delivery Option in Mailbox Full Condition**, select an option for delivering the New Message when your Personal Mailbox is full.

  You can select from any of the options described below:

  - **Do not offer to leave message**, if you do not want the system to allow any new message to be delivered when the Personal Mailbox is full.

  - **Deliver to General Mailbox**, if you want the system to deliver the new message to the General Mailbox when Personal Mailbox is full.

58. *Not applicable when you select **Operator** as the "Select Number" option.*

- **Overwrite Old Messages**, if you want the system to delete the old messages and allow the new message to be stored when the Personal Mailbox is full.

  In case, the recorded message size is greater than the old message that is to be overwritten then the recorded message will not be delivered. A prompt will be played for the same. Note: The old message will be deleted.

  The prompt will not be played in case of multiple recipients.

If the mailbox has no old messages, the recorded message will not be stored.

- **Overwrite New Messages**, if you want the system to delete the new messages to allow the new message to be stored when the Personal Mailbox is full.

  The number of messages that will be deleted would be as per the *Maximum Message Length* allowed.

If the mailbox has no new messages, the recorded message will not be stored.

- **Overwrite All (Old + New)**, if you want the system to delete the old and new messages and allow the new message to be stored when the Personal Mailbox is full.

- In **Auto Delete Message**, select an option to delete the messages automatically by the system.You may select None, Old or All.

  - Select **None** if you do not want the system to delete the message automatically from the Personal Mailbox.

  - Select **Old** if you want the system to delete the old read messages automatically after the number of days you configure in *Days for Auto Delete Messages.*

  - Select **All** if you want the system to delete all messages — read or unread — automatically after the number of days you configure in *Days for Auto Delete Messages.*

    - In **Days for Auto Delete Messages**, configure the number of days after which you want the system to automatically delete the messages.

- In **Ask Password to Access Mailbox**, By default the access to the mailbox is password protected. The “User Password” is required to access the mailbox. Whenever the mailbox owner accesses the mailbox, the VMS will ask for the (user) password.

  If you want to remove password protection, clear this check box.

Since a Mailbox can be accessed using the default User Password, 1111, extension users who are assigned a mailbox are recommended to change their User Password. To avoid unauthorized access, we recommend extension users to change the password regularly. Make sure it is strong and is kept confidential.

- In **Message Profile**, select the Message Profile you want to assign. For more information, see “Message Profile”.

- In **Mailbox Menu**, select the Mailbox Menu you want to assign. For more information, see “Mailbox Menu”.

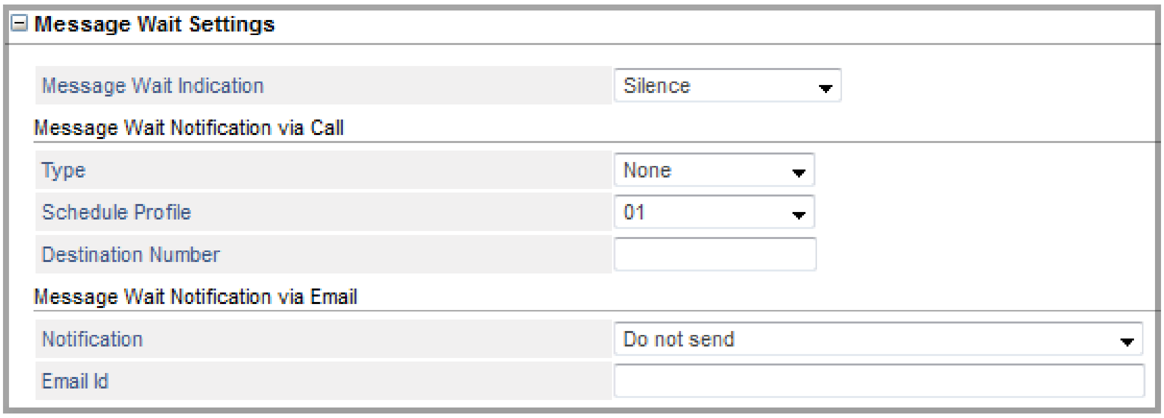
Call Transfer Settings[59]

- Under **Working Hour - WH**, select the Call Transfer Profile you want system to use for transferring the calls during Working Hours. The drop down list will include the profiles you configure in the "Call Transfer Profile".

- Under **Break Hour - BH**, select the Call Transfer Profile you want system to use for transferring the calls during Break Hours. The drop down list will include the profiles you configure in the "Call Transfer Profile".

- Under **Non-Working Hour - NH**, select the Call Transfer Profile you want system to use for transferring the calls during Non-Working Hours. The drop down list will include the profiles you configure in the "Call Transfer Profile".

Message Wait Settings[60]



- In **Message Wait Indication**, select the type of indication to be given to the extension user for new messages in the mailbox and message wait set by another extension user. This is only applicable when you select Extension as the "Select Number" option.

  Select the desired option per your requirement:

  - **Stuttered Dial Tone:** When the extension user goes OFF-Hook, s/he will hear a stuttered dial tone.

  - **Silence:** No indication will be provided to the extension user for Message Wait Notification, when s/he goes OFF-Hook.

---

59. *Not applicable when you select **General Mailbox** as the "Select Number" option.*
60. *Not applicable when you select **Operator** as the "Select Number" option.*

Default: Silence. Refer the feature description “Message Wait” to know more.

Message Wait Notification via Call[61]

The message wait notification will be sent to a number (destination number). This number can be an internal or an external number.

- In **Type**, you may select — Immediate, Scheduled or None.
  - Select **Immediate**, if you want the notifications to be sent as soon as a new message arrives in the mailbox of the extension user.
  - Select **Scheduled**, if you want the notification to be sent at fixed time schedules.
  - Select **None**, if you do not want to set message wait notification via call.

    Default: None.
- In **Schedule Profile**, select the **Notification via Call - Profile** number according to which you want the system to send the notifications. The Notification via Call Profile determines how notification calls are to be made to the destination numbers. To know more, see “Message Wait Notification via Call”.
- In **Destination Number**, configure the number that you want the system to use for sending the notification via calls.

  The destination number can be an internal or an external number. The destination number can be a maximum of 16 digits. Valid digits are 0 to 9, # and *.

  When the notification call is answered, the VMS informs the callee about the new message and allows the callee to access it.

  Refer the feature description “Message Wait Notification via Call” to know more.

Message Wait Notification via E-Mail[62]

The message wait notification will be sent to the e-mail address of the extension user.

- In **Notification**, you may select — Do not send, Without Attachment, With Attachment or With Attachment and mark voicemail as read. Default: Do not send.
  - Select **Do not send**, if you do not want the system to send email notification to the user for new message even if the email ID is configured. In this case, the system will send the Mailbox Memory Usage notification.
  - Select **Without Attachment**, if you want the system to send email notification to the user for new message if the email ID is configured.
  - Select **With Attachment**, if you want the system to send email notification to the user for new message along with the message as attachment. Make sure the email ID is configured. The attachment will be

61. *Applicable only when you select **Extension** as the “Select Number” option.*
62. *Not applicable when you select **Operator** as the “Select Number” option.*

sent only if the message size is less than or equal to 20 MB, else the email notification will be sent without the attachment.

- Select **With Attachment and mark voicemail as read**, if you want the system to send email notification to the user for new message along with the message as attachment and also mark the voicemail as read. Make sure the email ID is configured. The attachment will be sent only if the message size is less than or equal to 20 MB, else the email notification will be sent without the attachment.

- **E-mail Address:** Enter the email ID of the extension user to which the notification is to be sent. Maximum allowable length of the email ID is 64 characters. Default: blank.

*Extension users will receive notifications only for the mailbox memory utilization, if you configure the **E-mail Address** and select **Do not sent** as the **Notification** option.*

Refer the feature description “Email Based Notification” to know more.

- Click **Submit** to save Extension Voice Mail settings.

- Click **Copy** if you wish to copy the respective Extension’s Voice Mail settings to other extensions.

Copy Voice Mail Settings to window will open.

You can copy the Voice Mail Settings to a single Extension, multiple extensions or all extensions.

For single Extension, select **Extension Number** and enter the Extension Number.

For multiple extensions, select **Extension Numbers from** and enter the Extension Number range.

For all extensions, select **All Extensions**.

The Voice Mail Settings — Abbreviated Name, Message Wait Notification via Call parameters and Message Wait Notification via Email parameters will not be copied.

Voice Mail Auto-Attendant Menu

Voice Mail Auto-Attendant Menu is applicable when a trunk call is directly routed to the VMS.

Each trunk can be assigned a VMAA Menu. For details, see “Trunk Feature Template”.

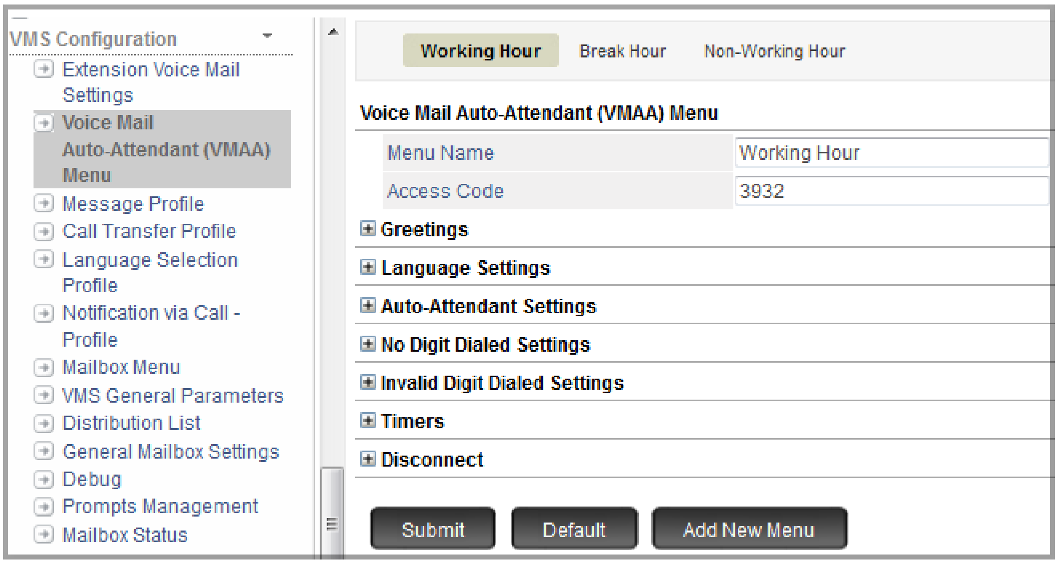
If VMAA Menu assigned to any action/trunk is deleted, the system will consider the first VMAA Menu for that action/ trunk by default.

VMS supports a maximum of 128 VMAA Menu.

By default, three VMAA Menus — Working Hour, Break Hour and Non-Working Hour— are provided to you. These three VMAA menus cannot be deleted but you may edit their settings as per your requirement.

How to Configure

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Voice Mail Auto-Attendant Menu**.



You may add a new menu, edit the default menus or delete a VMAA menu.

To add a new VMAA Menu,

- Click **Add New Menu**. A new *VMAA Menu xxx* will be created. You may now configure this as per your requirement.

To delete a menu, click **Delete**.

The default VMAA Menus cannot be deleted.

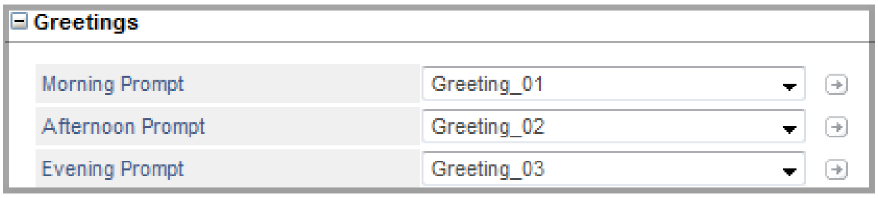
To edit a VMAA Menu,

- Click on the **VMAA Menu** tab you wish to edit.
- In **Menu Name**, configure the name of the VMAA Menu. By default, it is *VMAA Menu xxx* where xxx is the VMAA Menu Number from 001 to 128.
- In **Access Code**, configure the Access Code you wish to assign to the respective VMAA Menu. The caller can transfer the call to a VMAA Menu by dialing the respective access code. Access Code can be a maximum of 6 digits.

Make sure, the Access Codes assigned to the VMAA Menus do not conflict with the existing access codes. System will not save the configured VMAA access code if the same code is already assigned.

Greetings

Greetings allows you to select the prompts to greet the caller.



- In **Morning Prompt**, select the prompt which you wish to play to the caller as the Morning Greeting.

  Select **Do Not Play**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Morning Prompt.

- In **Afternoon Prompt**, select the prompt which you wish to play to the caller as the Afternoon Greeting.

  Select **Do Not Play**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Afternoon Prompt.

- In **Evening Prompt**, select the prompt which you wish to play to the caller as the Evening Greeting.

  Select **Do Not Play**, if you do not wish to play any prompt.

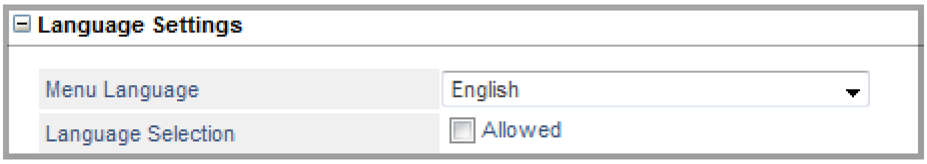
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

Once the files are uploaded these appear as options for Evening Prompt.

Language Settings

Language Settings allows you to set a menu language for the respective VMAA Menu and also gives the caller a choice to select the language.



- In **Menu Language**, select the language which you wish to set as the default language for the respective VMAA Menu. For all the calls routed to the VMAA Menu, the prompts and greetings will be played as per the Menu Language set.

- Select the **Language Selection** check box if you wish to allow the caller to choose a language.

  - In **Language Selection Profile**, select the language profile you want to assign to the respective VMAA Menu.

    - Click **Settings** to configure the parameters of the selected profile. For more information, see "Language Selection Profile".

  From here, the caller can select a different language other than the Menu Language configured. All the VMS prompts henceforth will be played in the language selected here.

Auto-Attendant Settings

Auto-Attendant Settings allows you to customize the Auto-Attendant Actions.

- In **Auto Attendant Prompt**, select the prompt which you wish to play to the caller according to the Auto-Attendant Actions.

  Select **Do Not Play**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Auto Attendant Prompt.

To add more Auto-Attendant Prompt options, click . You can add a maximum of 3 Auto Attendant Prompts in each VMAA Menu.

After playing all Auto-Attendant prompts, the system will wait for the input from the caller as per the timers configured. For details, see “Timers”.

- Select the **Extension Number Dialing** check box to allow the caller to dial the Extension Number while the Auto-Attendant prompts are being played.

  - Click **Settings** to configure the No Match Found Settings.

    - In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

      Select **None**, if you do not wish to play any prompt.

      You may add a new Prompt. To do so,

      - Click **Settings** . The **Prompts Management** page opens.

        You may add a new prompt from here. For further details, see “Prompts Management”.

      Once the files are uploaded these will appear as the options for No Match Found Prompt.

    - Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

    - In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

      - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

        Select **None**, if you do not wish to play any prompt.

        - Click **Settings** . The **Prompts Management** page opens.

          You may add a new prompt from here. For further details, see “Prompts Management”.

        Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

    - In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

      - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile. For more information, see “Call Transfer Profile”.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

- Select the **Confirm Name** check box, if you want the system to play the prompt asking the caller to enter the name of the called party, after dialing the extension number. To know more, refer to “Dial by Extension Number”.

- In **Confirm Name Prompt**, select the prompt which you wish to play to the caller.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Confirm Name Prompt.

  The system will place the call only when the entered name matches with the **Abbreviated Name** in Extension Voice Mail Settings. To know more, refer to “Extension Voice Mail Settings”.

- In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. Make sure you have configured the parameters for this profile. By default, *As configured for Dialed Extension* is selected.

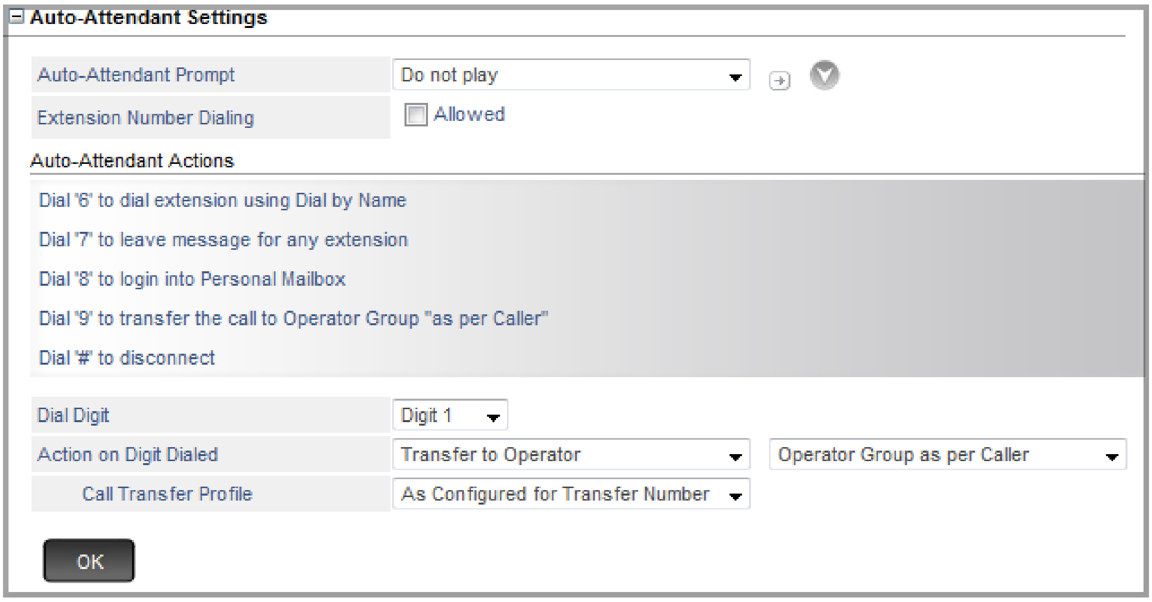
  - Select **As configured for Dialed Extension** if you want the VMS Auto Attendant to use the Call Transfer Profile which is assigned to the dialed extension.
  - Select **Wait for Ring** if you want the VMS Auto Attendant to wait for the extension to start ringing and then transfer the call.
  - Select **Blind** if you want the VMS Auto Attendant to transfer the call on the extension without checking whether it is busy or free.
  - Select **Attended** if you want the VMS Auto Attendant to transfer the call when the extension answers (goes OFF-Hook).

    Click **Settings** to configure the parameters of the selected profile. For more information, see “Call Transfer Profile”.

When the Call Transfer Profiles are added, or any existing profile is edited, these changes will be reflected in the options displayed.

Auto-Attendant Actions

The default Auto-Attendant actions taken by the system, when there is an incoming trunk call on the VMS are displayed here



You may add a new action, edit or delete the default actions.

To delete an Auto-Attendant action, mouse over on the respective action and click .

To add/edit an Auto-Attendant action,

- Click on the **Add Action** button to add a new action or click to edit an action.
- In **Dial Digit**, select the digit to be dialed for the respective action.
- In **Action on Digit Dialed**, you can select any one of the following:
    - Transfer to Operator
    - Transfer to Extension
    - Transfer to Department Group
    - Go to VMAA Menu
    - Play Information
    - Dial Extension Number by Name
    - Leave Voice Mail
    - Personal Mailbox Access
    - Change Language
    - Repeat Prompt
    - Go to Previous Menu
    - Disconnect

    - If you select **Transfer to Operator** or **Transfer to Extension** or **Transfer to Department Group**, select the respective Operator number or desired extension number or Department Group number.
        - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.

- If you select **Play Information**, click **Settings** to configure the parameters. For details, see "Play Information".

- If you select **Dial Extension Number by Name**, the system will allow caller to dial any extension number using name. Click **Settings** to configure the parameters. For details, see "Dial by Name".

- If you select **Leave Voice Mail**, the system will allow caller to leave the message directly to the extension number or department group. Click **Settings** to configure the parameters. For details, see "Leave Voice Mail".

- If you select **Personal Mailbox Access**, the system will provide the mailbox access to any user from remote location. Click **Settings** to configure the parameters. For details, see "Personal Mailbox Access".

- If you select **Change Language**, select the Language which you want the system to use for the VMAA Menu instead of the previously selected Language.

  Click **Settings** to configure the parameters of the selected language profile. For more information, see "Language Selection Profile".

  All the VMS prompts henceforth will be played in the language selected here.

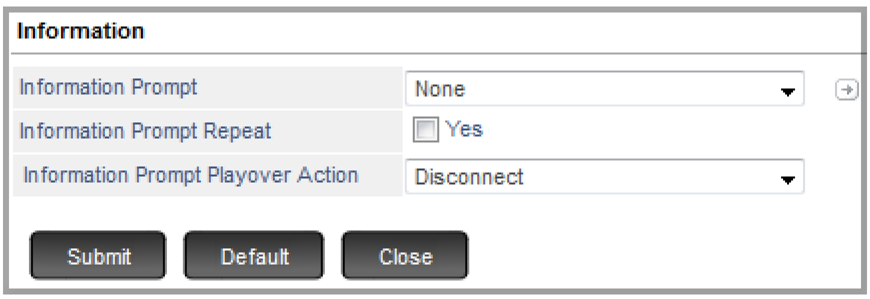
- If you select **Repeat prompt**, the system will repeat all the Auto-Attendant prompts configured in sequence.

- If you select **Go to Previous Menu**, the system will provide an option to the caller to go back to the previous VMAA Menu.

- If you select **Disconnect**, the system will disconnect the call.

- Select the **Ignore Digit Dialed during Prompt** check box, if you want the system to ignore the digits dialed by the caller while the prompts are being played.

Play Information



- In **Information Prompt**, select the prompt which has the information to be played to the caller. The information may be about the company or about the different products etc.

  Select **None**, if you do not wish to play any information prompt to the caller.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see "Prompts Management".

Once the files are uploaded these appear as options for Information Prompt.

- Select the **Information Prompt Repeat** check box to allow the Information prompt to be repeated to the caller.
- In **Information Prompt Repeat Count**, select the number of times you wish to play information prompt to the caller. The prompt will be played repeatedly till the Repeat Count expires.
- In **Repeat Count Expiry Prompt**, select the prompt you wish to play when the Information Prompt Repeat Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

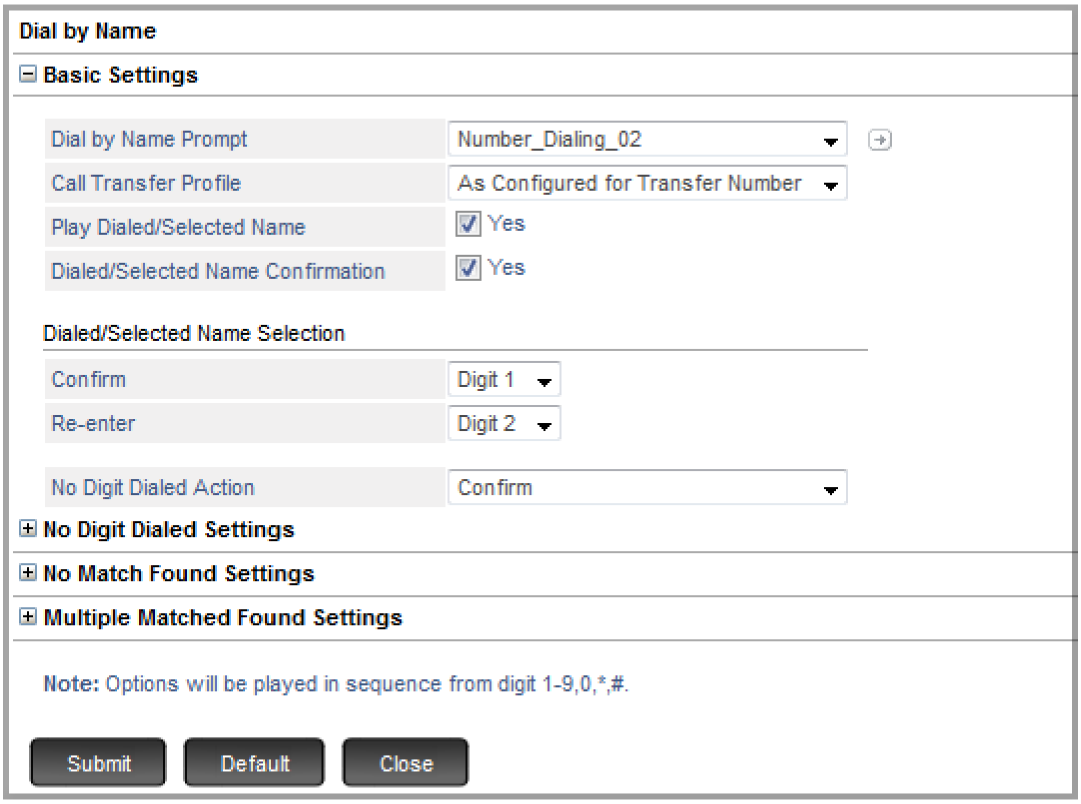
  Once the files are uploaded these appear as options for Repeat Count Expiry Prompt.

- In **Information Prompt Playover Action**, select — Transfer to Operator, Transfer to Department Group, Transfer to Extension, go to VMAA Menu or Disconnect.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.
- Click **Close** to close the window.

Dial by Name



Basic Settings

- In **Dial by Name Prompt**, select the prompt which you wish to play to the caller allowing him/her to dial the extension number using name.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Dial by Name Prompt.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile. For more information, see “Call Transfer Profile”.

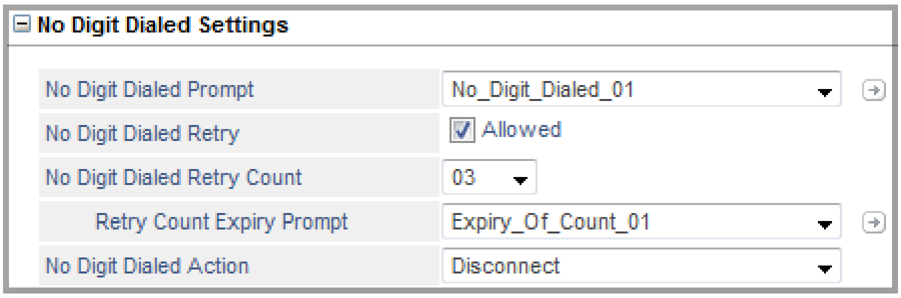
- Select the **Play Dialed/Selected Name** check box, if you want the system to play the name dialed by the caller and check the option **Dialed/Selected Name Confirmation**.

  If you disable the check box, the system will transfer the call directly to the extension number dialed by the caller as per the Call Transfer Profile configured.

- Select the **Dialed/Selected Name Confirmation** check box, if you want the system to confirm with the caller before transferring the call to the extension name dialed.
    - In **Confirm**, assign the Digit (0-9, *, # or None) that you want the system to play to the caller for confirming the Name dialed before transferring the call. If you assign *None*, the option will not be played.
    - In **Re-enter**, assign the Digit (0-9, *, # or None) that you want the system to play to prompt the caller to dial the extension name again.
    - In **No Digit Dialed Action**, select the option — Confirm or Re-enter. This is applicable when the caller has not dialed any digit for Selected Name Confirmation. The system will function as per the option you select.

No Digit Dialed Settings

No Digit Dialed Settings are applicable when caller has not dialed any digit and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt when caller has not dialed any digit.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompt the caller to dial the digit again.
    - In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the digit. The prompt will be played repeatedly till the Retry Count expires.
    - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

      Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

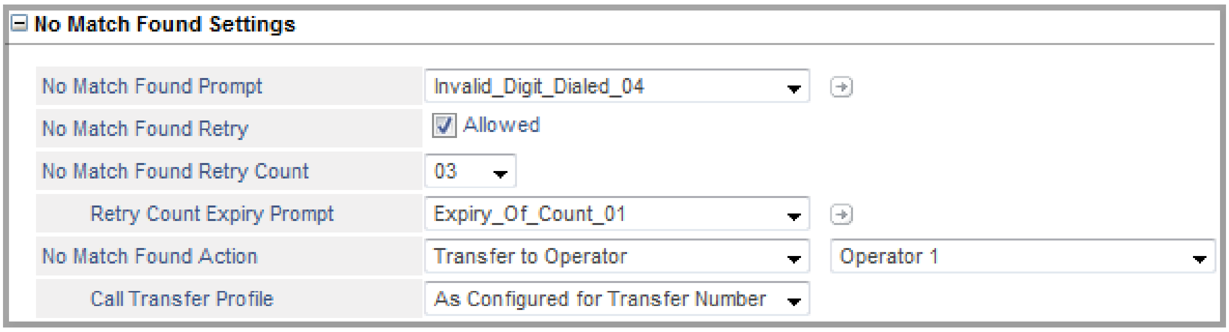
  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.

No Match Found Settings



- In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension name.

  Select **None**, if you do not wish to play any prompt.
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Match Found Prompt.
- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Name again.
- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.
- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

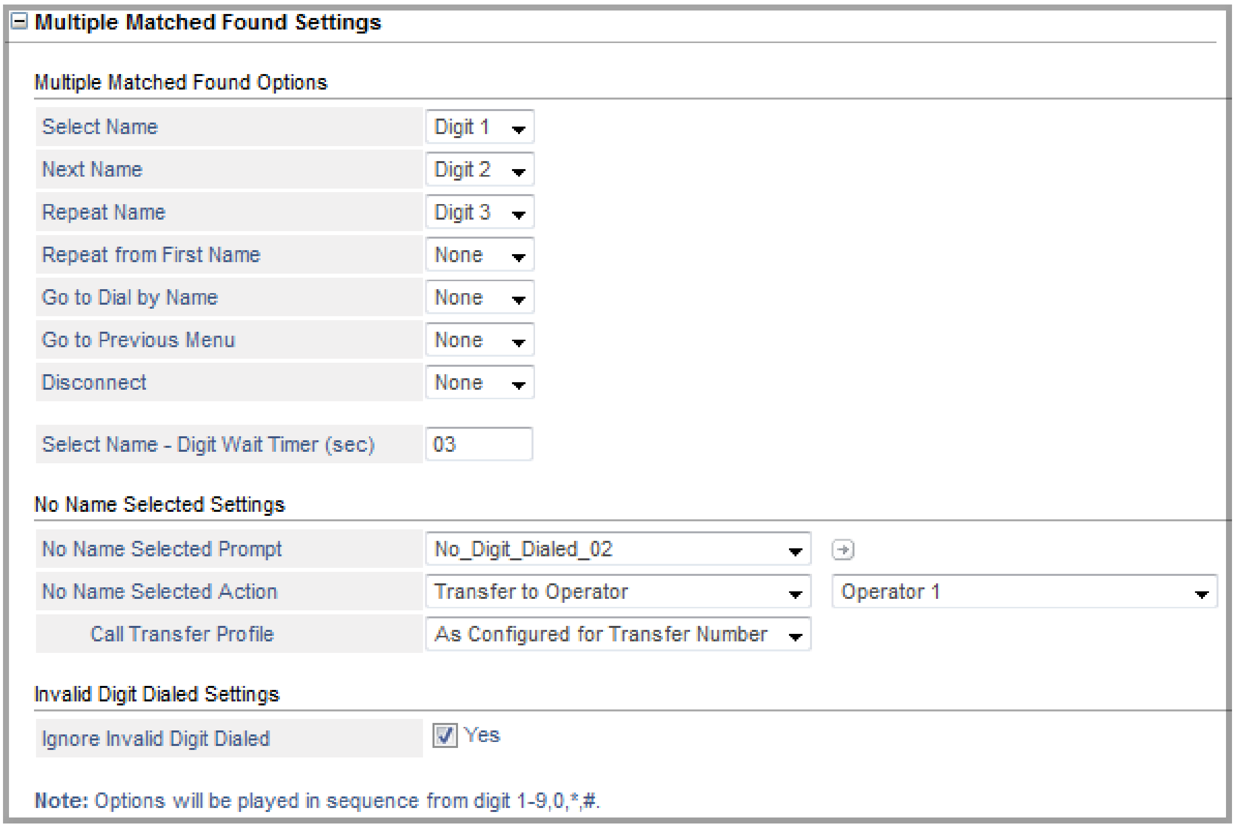
Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
    - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

      In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected

      Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".
    - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
    - If you select **Disconnect**, the system will disconnect the call.

Multiple Match Found Settings



Multiple Match Found Options

You can assign the Digits (0-9, *, # or None) to each of the options — Select Name, Next Name, Repeat Name, Repeat from First Name, Go to Dial by Name, Go to Previous Menu or Disconnect.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. These options will be played for each of the Names included in the Multiple Match Found list.

Brief description of each option is explained below:

- **Select Name**: The system will play a Name from the multiple match found list. The caller may select the this name for if he wishes to transfer the call to this Extension.
- **Next Name**: The system will play the next name from the multiple match found list.
- **Repeat Name**: The system will re-play the last name played from the multiple match found list.
- **Repeat from First Name**: The system will play the multiple match found list of names from the first matched name.
- **Go to Dial by Name**: The system will clear the multiple match found list and prompt caller to dial the Extension by Name again.
- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Previous Menu.
- **Disconnect**: The system will disconnect the call after playing the Disconnect prompt.

In **Select Name - Digit Wait Timer (sec)**, enter the time for which you want the system to wait to play the next name while playing the multiple match found list. Default: 03 seconds

No Name Selected Settings

- In **No Name Selected Prompt**, select the prompt you wish to play when no name is selected by the caller.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Name Selected Prompt.

- In **No Name Selected Action**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Previous Menu or Disconnect.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Go to Previous Menu**, the system will go back to the previous menu.
- If you select **Disconnect**, the system will disconnect the call.

Invalid Digit Dialed Settings

- Clear the **Ignore Invalid Digit Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.

  - In **Invalid Digit Dialed Prompt**, select the prompt you wish to play when the caller has dialed an invalid digit for option selection.

    Select **None**, if you do not wish to play any prompt when an invalid digit is dialed.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Invalid Digit Dialed Prompt.

- In **Invalid Digit Dialed Action**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Previous menu or Disconnect.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Go to Previous Menu**, the system will go back to the previous menu.
  - If you select **Disconnect**, the system will disconnect the call.

- Click **Close** to close the window.

Leave Voice Mail

- In **Leave Voice Mail to**, select an option where you want to leave the voice mail message.
  - If you select **Extension Number** or **Department Group**, the system will prompt the caller to leave the message for the Extension Number or Department Group configured.
  - If you select **Extension number dialed by caller**, the system will prompt the caller to enter the number to leave the message for the Extension Number or Department Group.

    Click **Settings** to configure the *Leave Voice Mail - Dialed Extension Number* parameters.

    - In **Leave Voice Mail Prompt**, select the prompt you wish to play to prompt the caller to leave the message.

      Select **None**, if you do not wish to play any prompt.

      - Click **Settings**. The **Prompts Management** page opens.

        You may add a new prompt from here. For further details, see “Prompts Management”.

      Once the files are uploaded these appear as options for Leave Voice Mail Prompt.
    - For **No Match Found Settings**, refer “No Match Found Settings”.
    - For **No Digit Dialed Settings**, refer “No Digit Dialed Settings”.
- Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply the Leave Voice Mail Settings configured on this page. Configure the “Message Leave Settings” and “No Mailbox Action”.

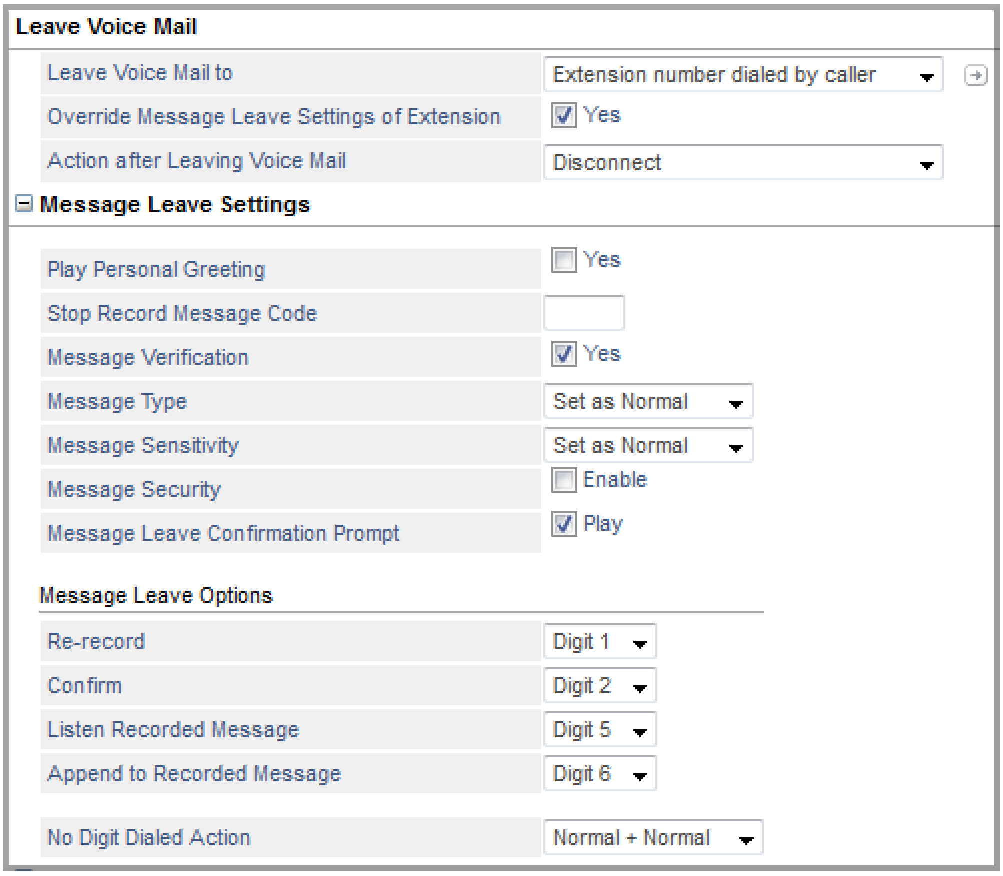
  Clear the check box, if you want the system to apply the Leave Voice Mail settings configured for the extension.
- In **Action after Leaving Voice Mail**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile. For more information, see "Call Transfer Profile".

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

Message Leave Settings

Message Leave Settings is applicable if you have enabled *Override Message Leave Settings of Extension* parameter. It allows you to configure the settings for the messages you receive from the caller i.e the message left for you by the caller.



- Select the **Play Personal Greeting** check box to allow the personal greetings to be played to the caller.

- *If the personal greeting is not recorded or is unavailable, no prompt will be played.*
- *Personal Greetings will not be applicable if your mailbox is defined as the destination for Call Tapping or Conversation Recording messages.*

- Select the **Play Conditional Greeting** check box to allow the conditional greetings to be played to the caller.

- *If the conditional greeting is not recorded or is unavailable, no prompt will be played.*
- *Conditional Greeting will be applicable only if your mailbox is defined as the destination for Call Transfer Unsuccessful.*

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the caller to dial to stop the recording of the message. The system will play this to the caller along with the prompt before he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code.

- Select the **Message Verification** check box to allow the recorded message to be verified by the caller before storing it in the Personal Mailbox or re-record it. You must configure the "Message Leave Options".

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

- In **Message Type**, select the option according to which you want the system to store the recorded messages. You may select — Set as Normal, Set as Urgent or Ask caller.
  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.
  - Select **Set as Urgent**, if you want the system to store the recorded message in the mailbox as Urgent.
  - Select **Ask Caller**, if you want the system to ask the caller to select the message type before storing the recorded message in the mailbox.
- In **Message Sensitivity**, select the option according to which you want the system to prioritize the messages as per their importance. You may select — Set as Normal, Set as Private or Ask caller.
  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.
  - Select **Set as Private**, if you want the system to store the recorded message in the mailbox as Private. The message received is considered as confidential and forwarding the message is restricted.
  - Select **Ask Caller**, if you want the system to ask the caller to select the message sensitivity before storing the recorded message in the mailbox.
- Select the **Message Security** check box, if you want the system to restrict the forwarding and downloading of the messages.

Message Security has higher priority than Message Type and Message Sensitivity set for the message.

  If you disable the check box, the system will not set any security for the message.

- Select the **Message Leave Confirmation Prompt** check box if you want the system to play the Message Leave Confirmation prompt to the caller after leaving the message. The Message Leave Confirmation prompt plays the Message Type, Message Sensitivity and Message Security that you configured.

  If you disable the check box, the system will not play the Message Leave Confirmation prompt to the caller after leaving the message.

Message Leave Options

You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the caller record a new message again. This is applicable only if you have enabled the Message Verification check box.
- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.
- **Urgent**: The system will save the recorded message along with the Message Type as *Urgent*. This is applicable only if you have selected the Ask Caller as the Message Type option.
- **Private(Confidential)**: The system will save the recorded message along with the Message Sensitivity as *Private*. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.
- **Listen Recorded Message**: The system will play the recorded message to the caller and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.
- **Append to Recorded Message**: The system will let the caller record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

  - *System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the caller to append the recorded message*
  - *The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.*

- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  If you have configured *Ask Caller* option in Message Type and/or Message Sensitivity, you can select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

No Mailbox Action

No Mailbox Action is applicable if you have enabled *Overide Message Leave Settings of Extension* parameter.

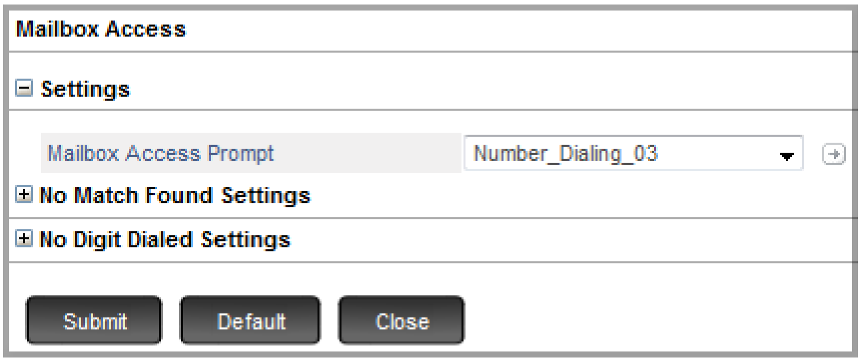
- In **No Mailbox Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile. For more information, see “Call Transfer Profile”.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

- Click **Close** to close the window.

Personal Mailbox Access



Settings

- In **Mailbox Access Prompt**, select the prompt you wish to play when the caller has selected *Personal Mailbox Access* as the Action on Digit Dialed option.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Mailbox Access Prompt.

No Match Found Settings

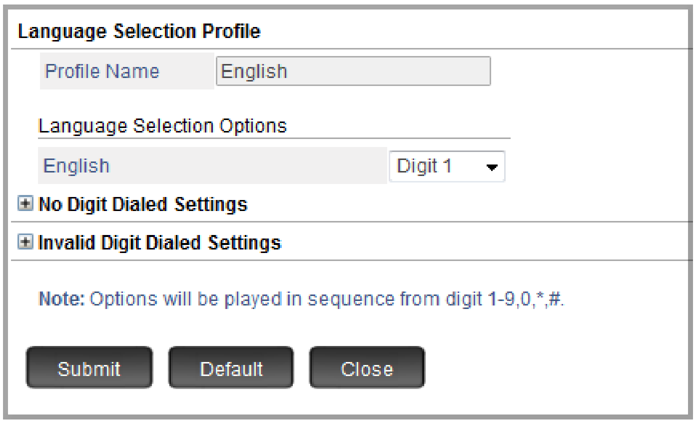
- For details, refer “No Match Found Settings”.

No Digit Dialed Settings

- For details, refer “No Digit Dialed Settings”.

- Click **Close** to close the window.

Change Language



You may add a Language Profile as per your requirement. To know more, refer to "Language Selection Profile".

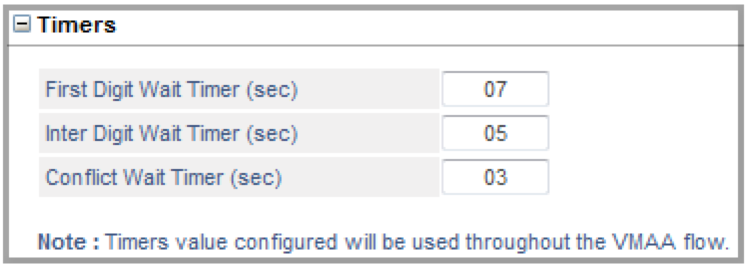
No Digit Dialed Settings

- For details, refer "No Digit Dialed Settings".

Invalid Digit Dialed Settings

- For details, refer "Invalid Digit Dialed Settings".

Timers

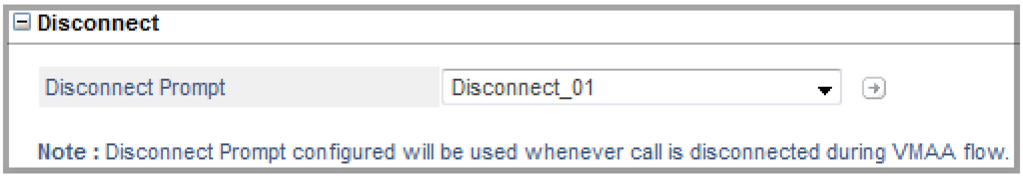


- In **First Digit Wait Timer**, enter the time for which you want the system to wait for the caller to dial the first digit for menu option selection or for entering the Extension or Department Group number. On the expiry of the First Digit Wait Timer, the system will apply "No Digit Dialed Settings". Default: 07 seconds

- In **Inter Digit Wait Timer**, enter the time for which you want the system to wait for the caller to dial the subsequent digit after the first digit is dialed. On the expiry of the Inter Digit Wait Timer, the system will apply "No Match Found Settings". Default: 05 seconds

- In **Conflict Wait Timer**, enter the time for which you want the system to wait for the extension user to dial the next digit to resolve conflicting access codes dialed by the extension user. Default: 03 seconds

Disconnect



- In **Disconnect Prompt**, select the prompt you wish to play when the caller has selected *Disconnect* as the Action on Digit Dialed option.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Disconnect Prompt.

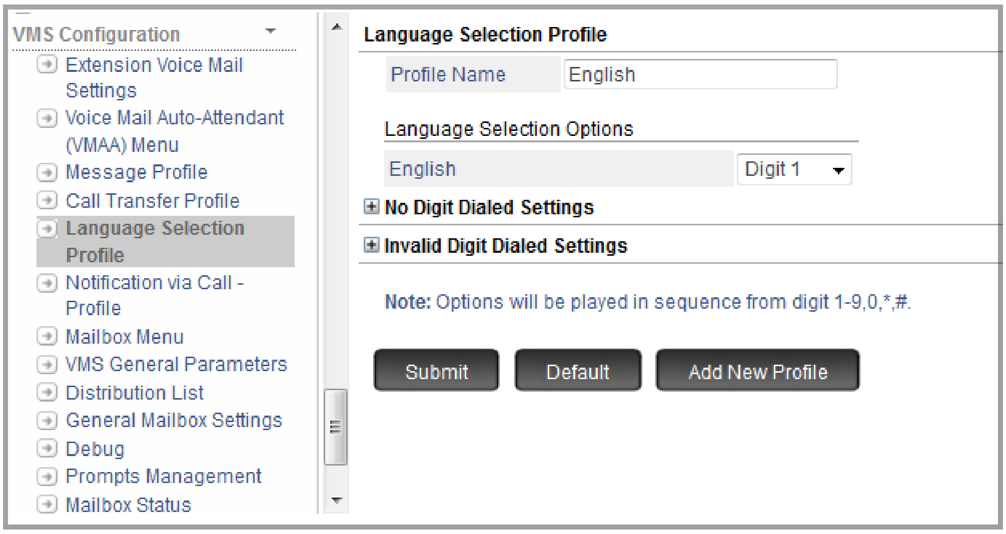
Language Selection Profile

Language Selection Profile is a menu which allows you to select the language. The languages you configure in the "Supported VMS Language" will be displayed under Language selection options. You may add a maximum of 8 profiles.

If Language Selection Profile assigned to any parameter is deleted, the system will consider the first Language Selection Profile.

How to Configure

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click the **Language Selection Profile**.



You may add a new profile, edit the default profile or delete a Language Profile.

To add a new Language Profile,

- Click **Add New Profile**. A new *Language Profile x* will be created. You may now configure this as per your requirement.

To delete a profile, click **Delete**.

The default Language Profile cannot be deleted.

To edit a Language Profile,

- In **Profile Name**, you may assign a name to the Language profile. By default, it is *Language Selection x,* where x is the Language Profile Number from 1 to 8.

Language Selection Options

The languages that are added in the system are displayed here. For details, see “Configuring VMS General Parameters”.

You can assign the Digits(0-9, *, # or None) to each of the Languages.

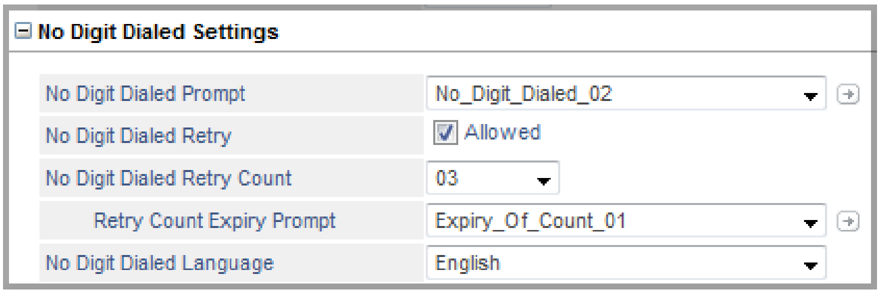
The digit you configure will be played as the digit to be dialed by the caller to select the respective language. The options will be played in the sequence — 1-9, 0, *, # — as per the Digits configured. The language for which the Digit selected is None will not be played.

The option you configure for a language will be played in the respective language.

For Example, you selected the Digit 5 to be played for Italian language. The option will be played along with the digit to dial (as configured) in Italian Language.

No Digit Dialed Settings

No Digit Dialed Settings are applicable when caller has not dialed any digit to select the desired Language and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt you wish to play when caller has not dialed any digit for option selection.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to allow the Language selection options to be played again if no digit has been dialed.

  - In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for language selection. The language selection options will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Language**, select — Use Language configured in Menu, Disconnect, or the language configured as Supported VMS Language.

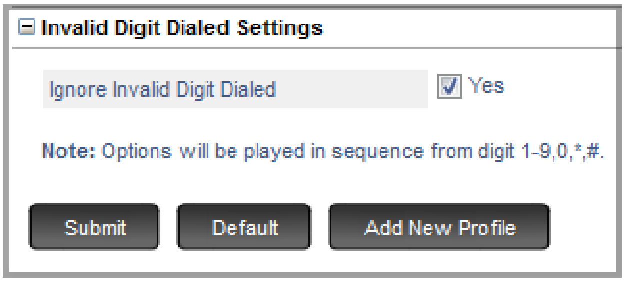
  - Select **Use Language configured in Menu**, if you want the system to use the Language you configured as Menu Language in VMAA Menu.

  - Select **Disconnect**, if you want the system to disconnect the call.

  If you select a language from the Supported VMS Languages you configured, the system will use the Language you select. See “Supported VMS Language”.

Invalid Digit Dialed Settings

Invalid Digit Dialed Settings are applicable when caller has dialed an invalid digit — a digit which is not configured as the Language Selection option.



- Clear the **Ignore Invalid Digit Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.

  - In **Invalid Digit Dialed Prompt**, select the prompt you wish to play when the caller has dialed an invalid digit for option selection.

    Select **None**, if you do not wish to play any prompt when an invalid digit is dialed.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Invalid Digit Dialed Prompt.

  - Select the **Invalid Digit Dialed Retry** check box to allow the Language selection options to be played again if an invalid digit has been dialed.

    - In **Invalid Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for language selection. The language selection options will be played repeatedly till the Retry Count expires.

    - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

      Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.
  You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **Invalid Digit Dialed Language**, select — Use Language configured in Menu, Disconnect, or the language configured as Supported VMS Language.
  - Select **Use Language configured in Menu**, if you want the system to use the Language you configured as Menu Language in VMAA Menu.
  - Select **Disconnect**, if you want the system to disconnect the call.
  - Select a language from the Supported VMS Languages you configured, if you want the system to use the Language you select. See "Supported VMS Language".

Message Profile

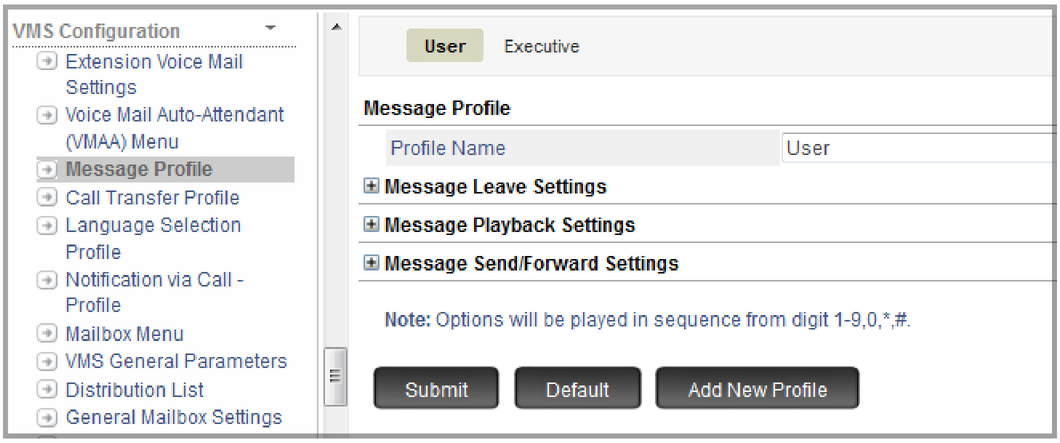
Message Profile allows you to customize the Message Leave Settings, Message Playback Settings and Message Sent/Forward Settings.

VMS supports a maximum of 24 Message Profiles.

By default, two Message Profiles — User and Executive — are provided to you. These two profiles cannot be deleted but you may edit their settings as per your requirement.

How to Configure

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Message Profile**.



You may add a new profile, edit the default profile or delete a message profile.

To add a new Message Profile,

- Click **Add New Profile.** A new *Message Profile xx* will be created. You may now configure this as per your requirement.

To delete a profile, click **Delete**.

The default Message Profiles cannot be deleted.

To edit a Message Profile,

- In **Profile Name**, you may configure the name of the Message profile you want. By default, it is *Message Profile xx* where xx is the Message Profile Number from 01 to 24.

Message Leave Settings

Message Leave Settings are applicable only when the external caller reaches to your mailbox or internal caller replies to your message.

- Select the **Play Personal Greeting** check box to allow the personal greetings to be played to the caller. The Personal Greetings will be played as per the timezone - Working Hours, Break hours or Non-Working hours.

If the personal greeting is not recorded or is unavailable, no prompt will be played.

- Select the **Play Conditional Greeting** check box to allow the conditional greetings to be played to the caller. The conditional greetings will be played when the call forward is set to VMS for Busy, No Reply or Unconditional.

If the conditional greeting is not recorded or is unavailable, no prompt will be played.

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the caller to dial to stop the recording of the message. The system will play this to the caller along with the prompt before he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code. In this case, you may dial any digit to stop recording.

- Select the **Message Verification** check box to allow the recorded message to be verified by the caller before storing it in the Personal Mailbox or to re-record it. You must configure the “Message Leave Options”.

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

- In **Message Type**, select the option according to which you want the system to store the received messages as per the priority. You may select — Set as Normal, Set as Urgent or Ask caller.

  - Select **Set as Normal**, if you want the system to store the messages received in the mailbox as Normal. The messages left by the external callers or replied by the internal callers will be considered as normal messages.

  - Select **Set as Urgent**, if you want the system to store the messages received in the mailbox as Urgent. This is useful when you want the messages left by the external callers or replied by the internal callers to be considered as urgent messages with higher priority.

  - Select **Ask Caller**, if you want the system to ask the caller to select the message type before leaving the recorded message on your extension. If the call gets disconnected before the caller selects the message type, the message will be set as normal by default. You must configure the “Message Leave Options”.

  While retrieving, the Urgent and Normal messages will be played separately.

- In **Message Sensitivity**, select the option according to which the system will decide whether the messages left by the external callers or replied by the internal callers are allowed to be forwarded or not.

  - Select **Set as Normal**, if you want the system to store the messages received in the mailbox as Normal. The messages left by the external callers or replied by the internal callers will be considered as normal messages.

- Select **Set as Private**, if you want the system to store the messages received in the mailbox as Private. The message left by the external callers or replied by the internal callers will be considered as confidential and forwarding the message will be restricted.

- Select **Ask Caller**, if you want the system to ask the caller to select the message sensitivity before storing the recorded message in the mailbox. If the call gets disconnected before the caller selects the message sensitivity, the message will be set as normal by default. You must configure the "Message Leave Options".

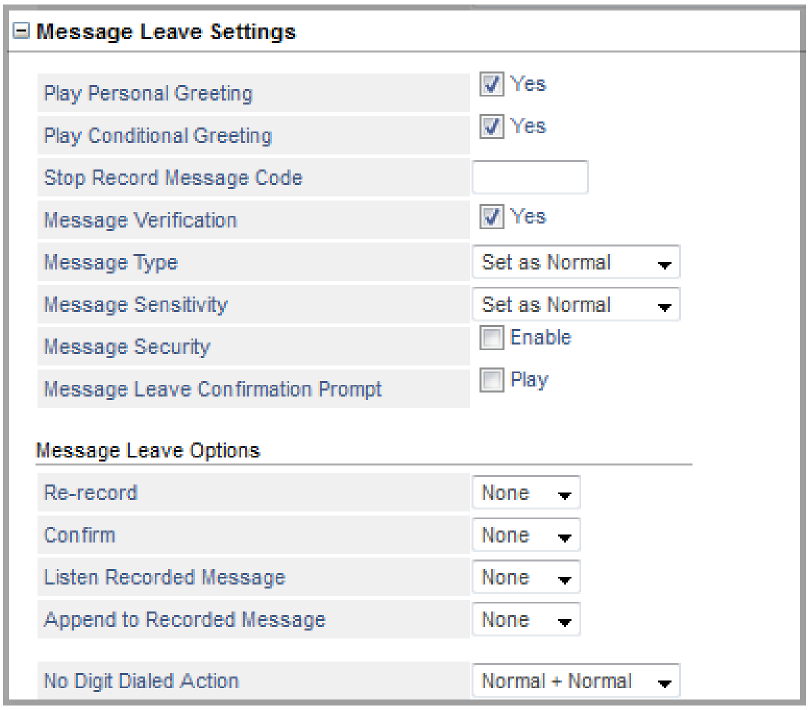
- The **Message Security** parameter is reserved for future use. Keep this check box disabled.

- Select the **Message Leave Confirmation Prompt** check box if you want the system to play the prompt to the caller after leaving the message. The Message Type and Message Sensitivity that you configured will be played.

  If you disable the check box, the system will not play the Message Leave Confirmation prompt to the caller after leaving the message.

Message Leave Options

Message Leave Options will be played based on the configuration of the parameters — Message Verification, Message Type and Message Sensitivity.



You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the caller record a new message again. This is applicable only if you have enabled the Message Verification check box.
- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.
- **Urgent**: The system will save the recorded message as Urgent message. This is applicable only if you have selected the Ask Caller as the Message Type option.
- **Private (Confidential)**: The system will save the recorded message as Private message. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.
- **Listen Recorded Message**: The system will play the recorded message to the caller and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.
- **Append to Recorded Message**: The system will let the caller record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

- *System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the caller to append the recorded message.*
- *The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.*

- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

  Based on the configuration, you may select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

Message Playback Settings

Message Playback Settings allows you to configure the settings applicable when extension user accesses his/her Personal Mailbox and plays the messages.

- In **Message Playback Direction**, select the option according to which you want the system to play the messages to the extension user. You may select — Play oldest recorded message first or Play new recorded message first.
  - Select **Play oldest recorded message first**, if you want the system to play the old messages first and then play the new messages.
  - Select **Play new recorded message first**, if you want the system to play the new messages first and then play the old messages.
- In **New Message Playback**, select the option defining the priority according to which you want the system to play the new messages to the extension user. You may select — Play Normal Messages and thereafter Urgent Messages or Play Urgent Messages first and thereafter Normal Messages.
  - Select **Play Normal Messages and thereafter Urgent Messages**, if you want the system to first play the new messages with message type as Normal and then play the new messages with the message type as Urgent.
  - Select **Play Urgent Messages first and thereafter Normal Messages**, if you want the system to first play the new messages with message type as Urgent and then play the new messages with the message type as Normal.
- In **Old Message Playback**, select the option defining the priority according to which you want the system to play the old messages to the extension user. You may select — Play Normal Messages and thereafter Urgent Messages or Play Urgent Messages first and thereafter Normal Messages.
  - Select **Play Normal Messages and thereafter Urgent Messages**, if you want the system to first play the old messages with message type as Normal and then play the old messages with the message type as Urgent.
  - Select **Play Urgent Messages first and thereafter Normal Messages**, if you want the system to first play the old messages with message type as Urgent and then play the old messages with the message type as Normal.
- **Date Playback Format:** Select the Date Playback Format — DD-MM-YYYY or MM-DD-YYYY— you want the system to play while playing the message details to the extension user.

*This Date Playback format is applicable only for **Message Profile**.*

- In **Message Type and Sensitivity**, select the option according to which you want the system to play the Message Type and Sensitivity of the stored messages to the extension user. You may select — Do Not Play, Play before Message or Play after Message.

  - Select **Do Not Play**, if you do not want the system to play the Message Type and Sensitivity.

  - Select **Play before Message**, if you want the system to play the Message Type and Sensitivity before playing the message.

  - Select **Play after Message**, if you want the system to play the Message Type and Sensitivity after playing the message.

- In **Message Details**, select the option according to which you want the system to play the Message Details of the stored messages to the extension user. You may select — Play before Message, Play after Message or Play on Demand.

  - Select **Play before Message**, if you want the system to play the Message Details before playing the message.

  - Select **Play after Message**, if you want the system to play the Message Details after playing the message.

  - Select **Play on Demand**, if you want the system to play the Message Details only when the caller wants. You must select the desired option to listen to the message details.

The System will first play the Message Type and Sensitivity and then play the Message Details.

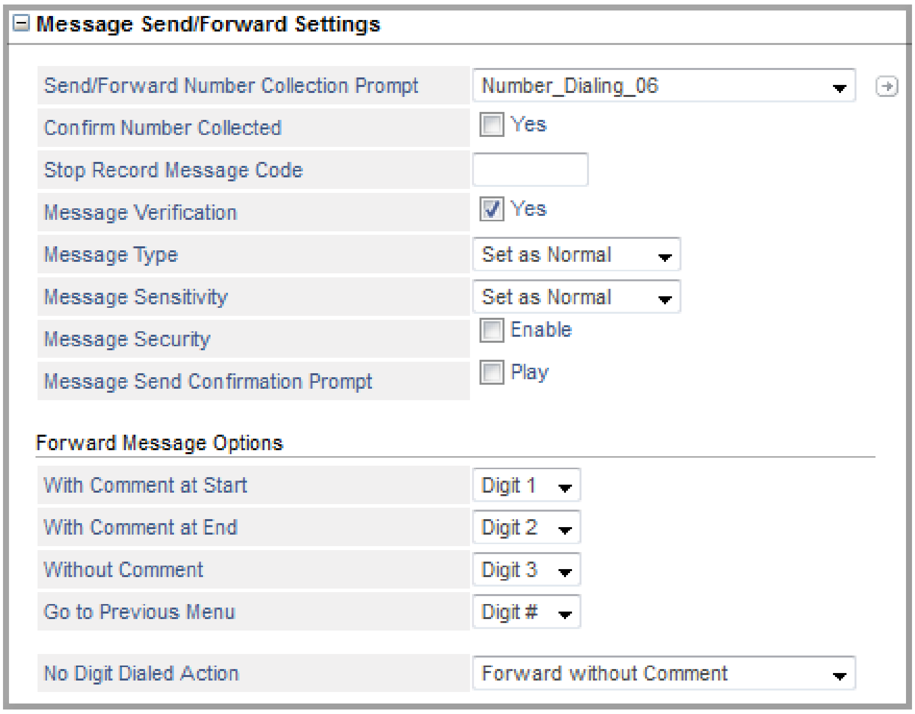
- Select the **Message Count** check box if you want the system to play the Message sequence number to the extension user before playing the message. The message count is just to differentiate between various messages played to the caller.

The message count has no interaction with the number of old/new/total messages present in the mailbox.

If you disable the check box, the system will not play the Message sequence number before playing the message to the extension user.

Message Send/Forward Settings

Message Send/Forward Settings allows you to configure the settings for the messages you send or forward.



- In **Send/Forward Number Collection Prompt**, select the prompt which you wish to play when you have selected the Send Message or Forward Message option. This is to prompt the extension user for entering the destination number/s for sending/forwarding the message.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Send/Forward Number Collection Prompt.

- Select the **Confirm Number Collected** check box to allow the system to check the Destination Number/s dialed by the user. Configure the “Destination Number Confirmation” settings.

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the extension user to dial to stop the recording of the message. The system will play this to the extension user along with the prompt before s/he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code. In this case, you may dial any digit to stop recording.

- Select the **Message Verification** check box to allow the recorded message to be verified by the extension user before storing it in the Personal Mailbox or re-record it.

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

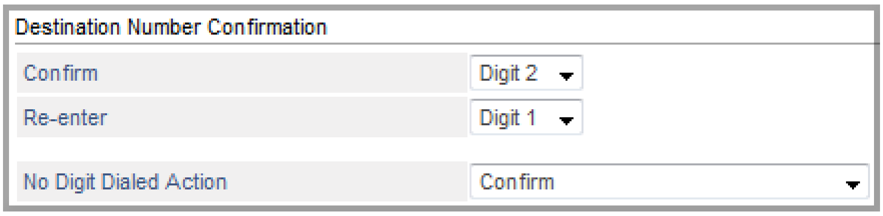
- In **Message Type**, select the option according to which you want the system to send/forward the recorded messages. You may select — Set as Normal, Set as Urgent or Ask caller.
    - Select **Set as Normal**, if you want the system to send/forward the recorded message in the mailbox as Normal.
    - Select **Set as Urgent**, if you want the system to send/forward the recorded message in the mailbox as Urgent with higher priority.
    - Select **Ask Caller**, if you want the system to ask the extension user to select the message type before storing the recorded message in the mailbox. If the call gets disconnected before the message type is selected, the message will be set as normal by default. You must configure the "Message Send Options".

  While retrieving, the Urgent and Normal messages will be played separately.
- In **Message Sensitivity**, select the option according to which you want the system to prioritize the messages as per their importance. You may select — Set as Normal, Set as Private or Ask caller.
    - Select **Set as Normal**, if you want the system to send/forward the recorded message in the mailbox as Normal.
    - Select **Set as Private**, if you want the system to send/forward the recorded message in the mailbox as Private. The message received is considered as confidential and forwarding the message is restricted.
    - Select **Ask Caller**, if you want the system to ask the extension user to select the message sensitivity before storing the recorded message in the mailbox. If the call gets disconnected before the message sensitivity is selected, the message will be set as normal by default. You must configure the "Message Send Options".
- The **Message Security** parameter is reserved for future use. Keep this check box disabled.
- Select the **Message Send Confirmation Prompt** check box if you want the system to play the prompt to the extension user after sending the message on the destination number/s i.e. to the recipients. The Message Type and Message Sensitivity that you configured will be played.

  If you disable the check box, the system will not play the Message Send Confirmation prompt to the extension user after sending the message.

Destination Number Confirmation

This option is applicable only if you have selected the **Confirm Number Collected** check box.



You can assign the Digits (0-9, *, # or None) to each of the options — Confirm, Re-enter.

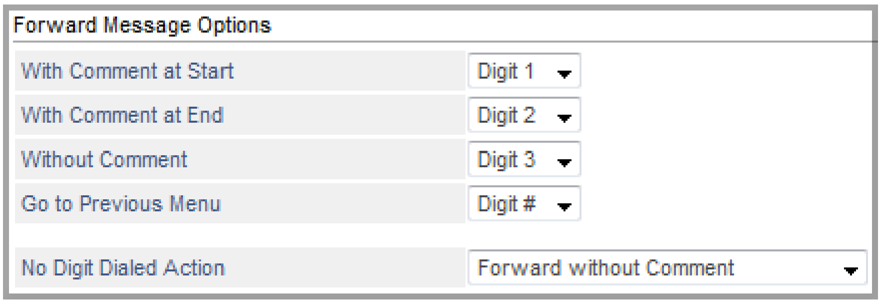
Make sure you do not assign the same digit to both the options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is a brief description of each option:

- **Confirm**: The system will save the destination number/s collected considering they are valid.
- **Re-enter**: The system will clear all the destination numbers collected and ask the extension user to enter the destination number/s again.
- In **No Digit Dialed Action**, select the option — Confirm or Re-enter. This is applicable when the extension user has not dialed any digit for Destination Number Confirmation. The system will function as per the option you select.

Forward Message Option



You can assign the Digits (0-9, *, # or None) to each of the options — With Comment at Start, With Comment at End, Without Comment or Go to Previous Menu.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is a brief description of each option:

- **With Comment at Start**: The system will provide an option to the extension user for recording a message in addition to the existing recorded message before forwarding the message to the destination number/s. The additional message will be added as the initial part of the message i.e. will be added at the beginning of the message.

- **With Comment at End**: The system will provide an option to the extension user for recording a message in addition to the existing recorded message before forwarding the message to the destination number/s. The additional message will be added as the final part of the message i.e. will be added at the end of the message.

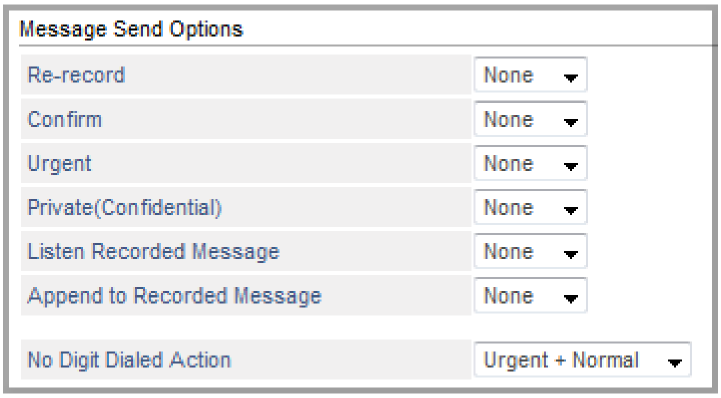
- **Without Comment**: The system will provide an option to the extension user for sending the recorded message to the destination number/s without adding any additional comment to it.

- **Go to Previous Menu**: The system will provide an option to the extension user to go back to the Mailbox Access Main Menu.

- In **No Digit Dialed Action**, select the option — With Comment at Start, With Comment at End or Without Comment. This is applicable when the extension user has not dialed any digit for Forward Message Option. The system will carry out the function when no digit is dialed as per the option you select.

Message Send Options

Message Send Options will be played based on the configuration of the parameters — Message Verification, Message Type and Message Sensitivity.



You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the extension user record a new message again.
  This is applicable only if you have enabled the Message Verification check box.

- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

- **Urgent**: The system will save the recorded message as Urgent message. This is applicable only if you have selected the Ask Caller as the Message Type option.

- **Private(Confidential)**: The system will save the recorded message as Private message. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.

- **Listen Recorded Message**: The system will play the recorded message to the extension user and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.

- **Append to Recorded Message**: The system will let the extension user record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

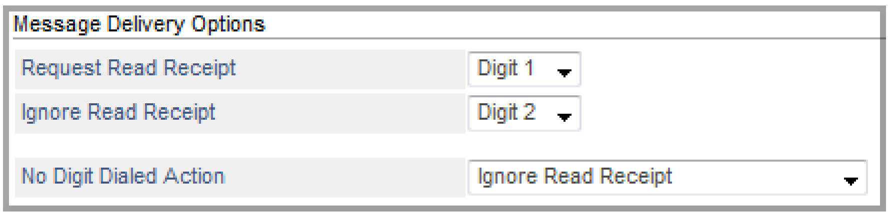
  - *System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the extension user to append the recorded message.*
  - *The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.*

- In **No Digit Dialed Action**, if no digit is dialed by the extension user, the system will take action as per the Message Type and Message Sensitivity configured.

  This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.

  Based on the configuration, you may select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

Message Delivery Options



You can assign the Digits (0-9, *, # or None) to each of the options — Request Read Receipt or Ignore Read Receipt.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the extension user to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- **Request Read Receipt**: The system will provide an option to the extension user for requesting the read receipt to be played whenever the message is read by the destination number.

  As soon as the destination number reads the message, the extension user will receive a read receipt in form of a voice mail. To retrieve the read receipt, access your voice mail. The read receipt includes the name, destination number and the first five seconds of the message sent or forwarded.

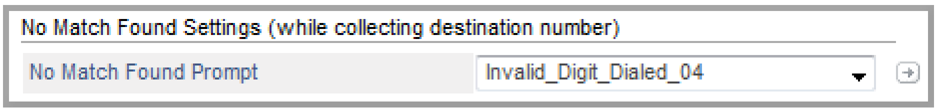
  In case of multiple destination numbers, the extension user will receive multiple read receipts in form of a voice message after the message has been read by the destination numbers.

- **Ignore Read Receipt**: The system will provide an option to the extension user for ignoring the read receipt i.e. no read receipt will be played back to the extension user.

- In **No Digit Dialed Action**, select the option — Request Read Receipt, Ignore Read Receipt. This is applicable when the extension user has not dialed any digit for Message Delivery Option. The system will carry out the function when no digit is dialed as per the option you select.

No Match Found Settings (while collecting destination number)

No Match Found Settings are applicable when no valid match is found as the Destination Number.



- In **No Match Found Prompt**, select the prompt which you wish to play when the extension user has dialed an invalid destination number/s for sending/forwarding the message.

  Select **None**, if you do not wish to play any prompt when extension user has dialed an invalid destination number.

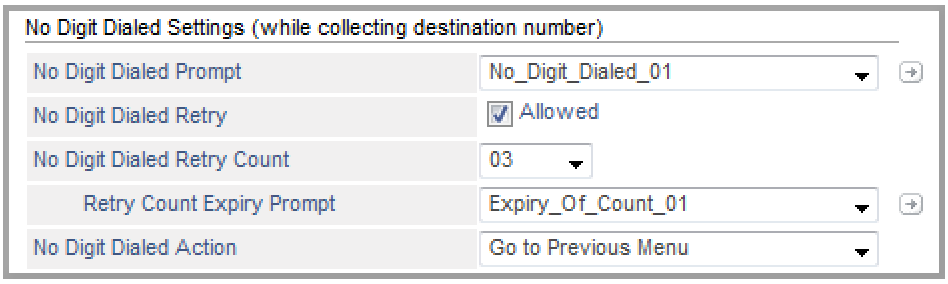
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Match Found Prompt.

No Digit Dialed Settings (while collecting destination number)

No Digit Dialed Settings are applicable when extension user has not dialed the destination number and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt which you wish to play when extension user has not dialed any digit for destination number.

  Select **None**, if you do not wish to play any prompt when extension user has not dialed any digit.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompt the extension user to dial the Destination Number again.

  - In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the extension user for dialing the destination number. The prompt will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings**. The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, select — Go to Previous Menu or Disconnect.

  - Select **Go to Previous Menu**, if you want the system to provide an option to the extension user to go back to the Mailbox Access Main Menu.

  - Select **Disconnect**, if you want the system to disconnect the call.

Call Transfer Profile

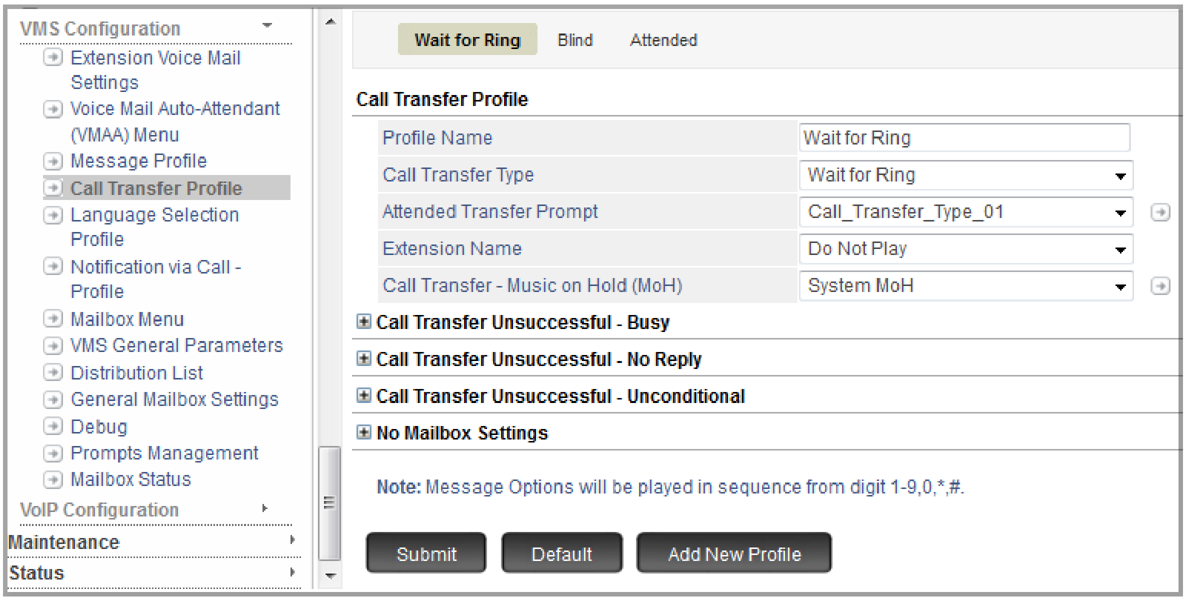
Call Transfer Profile allows you can select the Call Transfer Type — Blind, Wait for Ring, Wait for Answer, Screened, None — and customize parameters related to it as per your requirement.

VMS supports a maximum of 64 Call Transfer Profiles.

By default, three Call Transfer Profiles — Wait for Ring, Blind and Attended — are provided to you. These three profiles cannot be deleted but you may edit their settings as per your requirement.

How to Configure

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Call Transfer Profile**.



You may add a new profile, edit the default profile or delete a Call Transfer Profile.

To add a new Call Transfer Profile,

- Click **Add New Profile.** A new *Transfer Profile xx* will be created. You may now configure this as per your requirement.

To delete a profile, click **Delete**.

The default Call Transfer Profiles cannot be deleted.

To edit a Call Transfer Profile,

- In **Profile Name**, you may configure the name of the Call Transfer profile you want. By default, it is *Transfer Profile xx* where xx is the Call Transfer Profile Number from 01 to 64.
- In **Call Transfer Type**, you can select — None, Blind, Wait for Ring, Wait for Answer or Screened.
  - If you select **None**, the caller will be transfered to the transfer target's mailbox directly. You must configure the "Leave Voice Mail" parameters.
  - If you select **Blind**, the caller will be transfered to the transfer number directly.
  - If you select **Wait for Ring,** the caller will be transfered to the transfer number after the number starts ringing.
  - If you select **Wait for Answer,** the caller will be transfered to the transfer number only after the Transfer target answers the call.
  - If you select **Screened**, the caller will be transfered only after the transferor confirms to speak to the caller. You must configure the "Call Transfer Type - Screened" parameters.
- In **Attended Transfer Prompt**, select the prompt you want the system to play while the call is being transfered. This is applicable only if you have selected Wait for Ring or Wait for Answer as the Call Transfer Type option.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the prompts are uploaded these appear as options for Attended Transfer Prompts.
- In **Blind Transfer Prompt**, select the prompt you want the system to play when the call is being transfered. This is applicable only if you have selected Blind as the Call Transfer Type option.

  Select **None**, if you do not wish to play any prompt. You may also add a new Prompt. To do so, follow the same steps as given in Attended Transfer Prompt.
- In **Screened Transfer Prompt**, select the prompt you want the system to play when the call is being transfered. This is applicable only if you have selected Screened as the Call Transfer Type option.

  Select **None**, if you do not wish to play any prompt. You may also add a new Prompt. To do so, follow the same steps as given in Attended Transfer Prompt.
- In **Wait for Answer Timer (sec)**, configure the time for which you want the system to wait before transferring the call. This is applicable only if you have selected Wait for Answer or Screened as the Call Transfer Type option.

If the Wait for Answer Timer value is greater than Ring Back Tone timer, then Ring Back Tone Timer will expire first and the next action will be taken as per the ***Call Transfer Unsuccessful - Unconditional*** *configuration.*

- In **Extension Name**, you can select — Do not play or Play Always.

  Select **Play Always**, if you want the system to play the recorded Extension Name as well as the respective Transfer prompt during the transfer.

  Select **Do not Play**, if you want the system to only play the respective Transfer prompt and not the Extension Name during the transfer.

- In **Call Transfer - Music On Hold (MOH)**, you may select System MoH or add a new MoH. By default, System MoH will be played. This parameter is applicable only if you have selected Wait for Ring or Wait for Answer as the Call Transfer Type.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded, these appear as options for Call Transfer - MoH.

- In **Action after Leaving Voice Mail**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group or Go to VMAA Menu. This parameter is applicable only if you have selected None as the Call Transfer Type.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**,

    - Select the respective Operator number/Department Group number or enter the desired extension number.

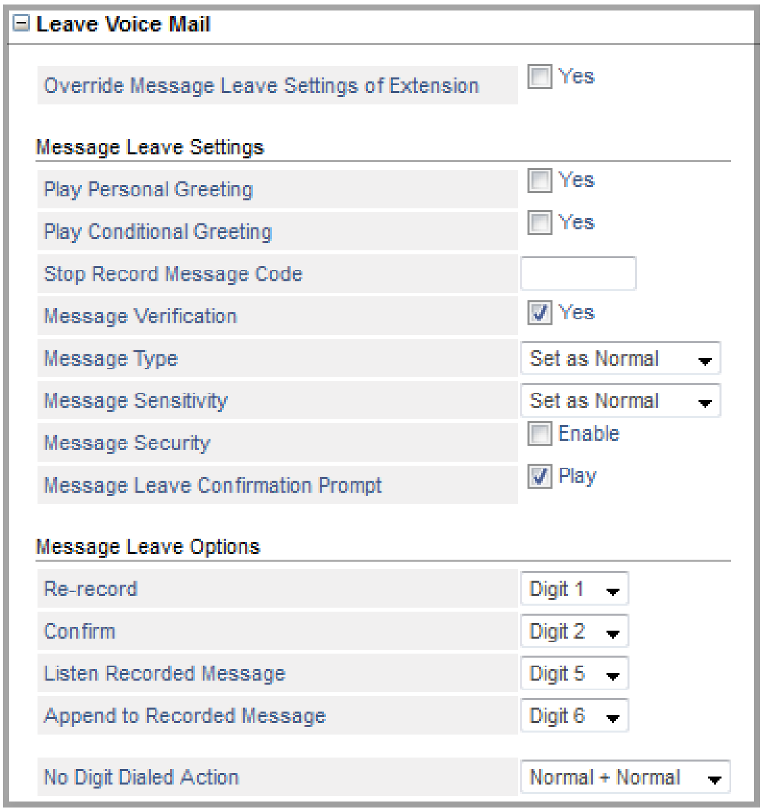
    - In Call Transfer Profile, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

Leave Voice Mail

This option is applicable only if you select None as the Call Transfer Type.



- Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply the Leave Voice Mail Settings configured on this page.

  Clear the check box, if you want the system to apply the Leave Voice Mail settings configured for the extension.

Message Leave Settings

Message Leave Settings allows you to configure the settings for the messages you receive from the caller i.e the message left for you by the caller.

- Select the **Play Personal Greeting** check box to allow the personal greetings to be played to the caller.

- *If the personal greeting is not recorded or is unavailable, no prompt will be played.*
- *Personal Greetings will not be applicable if your mailbox is defined as the destination for Call Tapping or Conversation Recording messages.*

- Select the **Play Conditional Greeting** check box to allow the conditional greetings to be played to the caller.

- *If the conditional greeting is not recorded or is unavailable, no prompt will be played.*
- *Conditional Greeting will be applicable only if your mailbox is defined as the destination for Call Transfer Unsuccessful.*

- In **Stop Record Message Code**, enter the digits (0-9,* or #) that you want the caller to dial to stop the recording of the message. The system will play this to the caller along with the prompt before he starts recording the message. You may configure a maximum of upto 3 digits.

If you do not assign any digit and keep it blank, the system will play the prompt without the stop code.

- Select the **Message Verification** check box to allow the recorded message to be verified by the caller before storing it in the Personal Mailbox or re-record it. You must configure the "Message Leave Options".

  If you disable the check box, the system will directly store the recorded message in the Personal Mailbox.

- In **Message Type**, select the option according to which you want the system to store the recorded messages. You may select — Set as Normal, Set as Urgent or Ask caller.

  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.

  - Select **Set as Urgent**, if you want the system to store the recorded message in the mailbox as Urgent.

  - Select **Ask Caller**, if you want the system to ask the caller to select the message type before storing the recorded message in the mailbox.

- In **Message Sensitivity**, select the option according to which you want the system to prioritize the messages as per their importance. You may select — Set as Normal, Set as Private or Ask caller.

  - Select **Set as Normal**, if you want the system to store the recorded message in the mailbox as Normal.

  - Select **Set as Private**, if you want the system to store the recorded message in the mailbox as Private. The message received is considered as confidential and forwarding the message is restricted.

  - Select **Ask Caller**, if you want the system to ask the caller to select the message sensitivity before storing the recorded message in the mailbox.

- Select the **Message Security** check box, if you want the system to restrict the forwarding and downloading of the messages.

Message Security has higher priority than Message Type and Message Sensitivity set for the message.

  If you disable the check box, the system will not set any security for the message.

- Select the **Message Leave Confirmation Prompt** check box if you want the system to play the Message Leave Confirmation prompt to the caller after leaving the message. The Message Leave Confirmation prompt plays the Message Type, Message Sensitivity and Message Security that you configured.

  If you disable the check box, the system will not play the Message Leave Confirmation prompt to the caller after leaving the message.

Message Leave Options

You can assign the Digits (0-9, *, # or None) to each of the options — Re-record, Confirm, Urgent, Private(Confidential), Listen Recorded Message and Append to Recorded Message.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Given below is the brief description of each option:

- **Re-record**: The system will clear the recorded message and let the caller record a new message again. This is applicable only if you have enabled the Message Verification check box.
- **Confirm**: The system will save the recorded message as per the option set by you in No Digit Dialed Action. This is applicable only if you have enabled the Message Verification check box or selected the Ask Caller as the Message Type and/or Message Sensitivity option.
- **Urgent**: The system will save the recorded message along with the Message Type as *Urgent*. This is applicable only if you have selected the Ask Caller as the Message Type option.
- **Private(Confidential)**: The system will save the recorded message along with the Message Sensitivity as *Private*. This is applicable only if you have selected the Ask Caller as the Message Sensitivity option.
- **Listen Recorded Message**: The system will play the recorded message to the caller and then play the message leave options again. This is applicable only if you have enabled the Message Verification check box.
- **Append to Recorded Message**: The system will let the caller record a new message again and add it to the existing recorded message. This is applicable only if you have enabled the Message Verification check box.

- *System will check — Maximum Message Length, New Message Delivery Option in Mailbox Full Condition etc — before allowing the caller to append the recorded message.*
- *The Message, that is, the recorded message along with the appended message cannot exceed the Maximum Message Length. Hence, the system will allow recording of the appending message only till it reaches the Maximum Message Length.*

- In **No Digit Dialed Action**, if no digit is dialed by the caller, the system will take action as per the Message Type and Message Sensitivity configured.

  If you have configured *Ask Caller* option in Message Type and/or Message Sensitivity, you can select the desired action — Normal+Normal, Urgent+Normal, Normal+Private or Urgent+Private.

Call Transfer Unsuccessful - Busy

If the called party is busy and has set Call Forward-Busy to VMS, the Call Transfer Unsuccessful - Busy parameters will be applicable. You can customize these parameters as per your requirement.

The VMS allows you to:

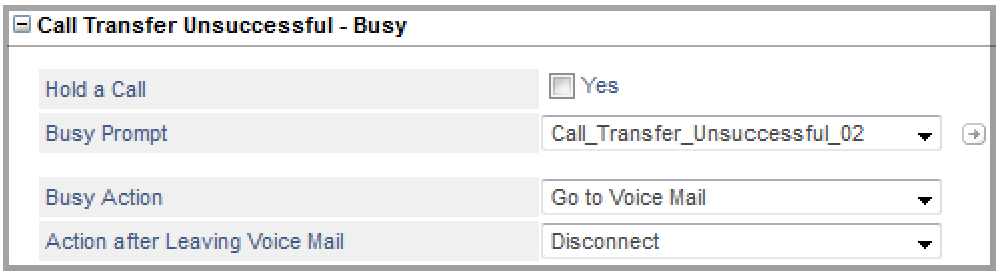
- process the call immediately for the busy condition as per the options you select.
  or
- process the call after being held for a certain duration as per the options you select.

Configure the following parameters, if you want the VMS to process the call immediately when the desired extension is busy,

- Busy Prompt
- Busy Action
- Action after Leaving Voice Mail

Configure the following parameters, if you want the VMS to put the call on hold when the desired extension is busy,

- Busy Prompt
- Busy Hold Prompt
- Provide Advanced Options during Hold
- Busy Extension Status check time interval (sec) and Time Interval Expiry Prompt
- Busy Extension Status check retry count and Retry Count Expiry Prompt
- Busy Hold - Music on Hold
- Busy Action
- Action after Leaving Voice Mail



- Select **Hold a Call** check box, if you want the call to be held. This option is applicable only for internal calls.

- In **Busy Prompt**, select the prompt you wish to play to the caller when the dialed extension is busy.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Prompt.

- In **Busy Hold Prompt**, select the prompt you wish to play to the caller when put on hold as the dialed extension is busy.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Hold Prompt.

    - Select **Provide Advanced Options during Hold** check box, if you want the system to play Advanced options to the caller.

You can assign the Digits (0-9, *, # or None) to each of the options — Leave Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate/Mobile Number, Dial Extension Number, Go to Main Menu, Go to Previous Menu and Disconnect.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- **Leave Voice Mail:** After you assign the digit, click **Settings** to configure the parameters.
  - Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply Message Leave Settings customized on this page and not that of the extension or department group number.

    Clear the check box if you want the system to apply Message Leave Settings configured for the extension or department group number.
- **Transfer to Operator:** After you assign the digit, click **Settings** to configure the parameters.
  - In **Operator Group**, select the desired Operator Group Number. Make sure you have configured the parameters for this group. For detailed instructions, see "Extension Voice Mail Settings". By default, As Configured in Extension Settings is selected, that is, the system will use the Operator Group configured for the transfer number.
  - In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. By default, *As configured for Transfer Number* is selected.
    - Select **As configured for Transfer Number** if you want the VMS Auto Attendant to use the Call Transfer Profile which is assigned to the dialed extension.
    - Select **Wait for Ring** if you want the VMS Auto Attendant to wait for the extension to start ringing and then transfer the call.
    - Select **Blind** if you want the VMS Auto Attendant to transfer the call on the extension without checking whether it is busy or free.
    - Select **Attended** if you want the VMS Auto Attendant to transfer the call when the extension answers (goes OFF-Hook).

      Click **Settings** to configure the parameters of the selected profile.

When the Call Transfer Profiles are added, or any existing profile is edited, these changes will be reflected in the options displayed.

- **Transfer to Assistant**: If you assign a digit for this option, make sure you have programmed the Assistant Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

Click **Settings** to configure the parameter.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

  Click **Settings** to configure the parameters of the selected profile.

- **Transfer to Alternate/Mobile Number**: If you assign a digit for this option, make sure you have programmed the Alternate/Mobile Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

- **Dial Extension Number:** After you assign the digit, click **Settings** to configure the parameters.

  - In **Dial Extension Number Prompt**, select the prompt you wish to play to the caller to dial the desired extension number.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Dial Extension Number Prompt.

    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Dialed Extension* is selected.

      Click **Settings** to configure the parameters of the selected profile.

No Match Found Settings: No Match Found Settings are applicable when a Extension Number is found invalid, the system will play the *No Match Found* prompt.

- In No **Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

No Digit Dialed Settings: No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.

- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompts the caller to dial the Extension Number again.

- In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

- **Go to Main Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.

- **Go to Previous Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Previous Menu.

- **Disconnect:** Assign a digit for this option if you want the system to provide the option to disconnect the call.

- In **Busy Extension Status check time interval (sec)**, enter the interval of time after which you want the system to inform the caller that the extension number is busy. The system will play the Time Interval Expiry Prompt.

  - In **Time Interval Expiry Prompt**, select the prompt you wish to play when the time interval expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Time Interval Expiry Prompt.

- In **Busy Extension Status check retry count**, select the number of times you wish the system to check if the transfer number is free. The prompt will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **Busy Hold - Music on Hold**, select the prompt you wish to play. This option is applicable only is you have enabled the Hold a Call check box.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Busy Hold - Music on Hold.

- In **Busy Action,** you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Voice Mail, Give Advanced Options.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

  - If you select **Give Advanced Option**, see "Advanced Options"

- In **Action after Leaving Voice Mail**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu. This option is applicable only is you select Go to Voice Mail or Give Advanced options as the Busy Action option.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

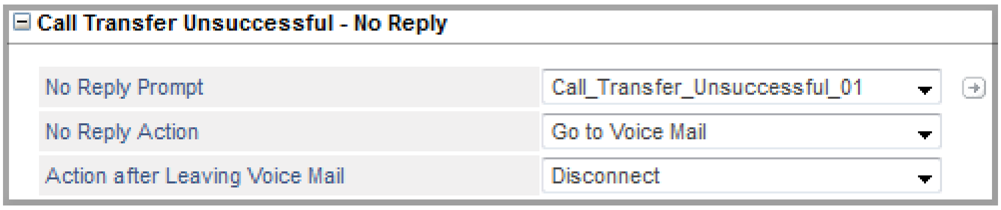
    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

Call Transfer Unsuccessful - No Reply

If the called party has set Call Forward-No Reply to VMS, the Call Transfer Unsuccessful - No Reply parameters will be applicable. You can customize these parameters as per your requirement.



- In **No Reply Prompt,** select the prompt you wish to play to the caller if the dialed extension does not reply.
- In **No Reply Action,** you can select — Go to Voice Mail, Give Advanced Options, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

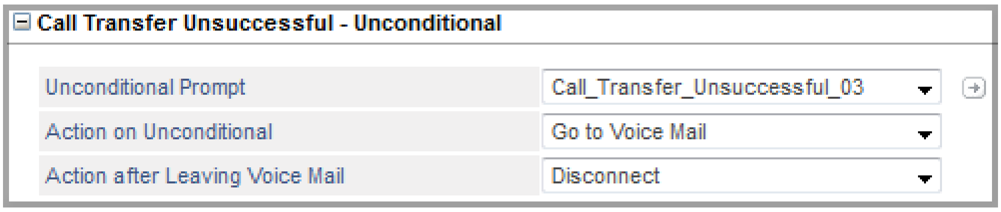
    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.
  - If you select **Give Advanced Option**, see "Advanced Options"
- In **Action after Leaving Voice Mail**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect. This option is applicable only is you select Go to Voice Mail or Give Advanced options as the No Reply Action option.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.

Call Transfer Unsuccessful - Unconditional

If the called party has set Call Forward-Unconditional to VMS, the Call Transfer Unsuccessful - Unconditional parameters will be applicable. You can customize these parameters as per your requirement.



- In **Unconditional Prompt,** select the prompt you wish to play to the caller.
- In **Action on Unconditional,** you can select — Go to Voice Mail, Give Advanced Options, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

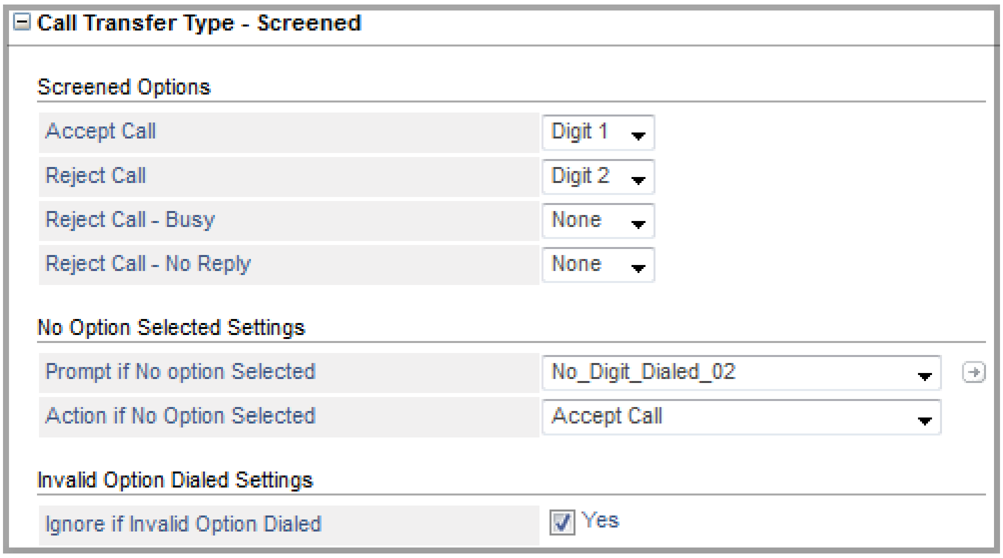
    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.
  - If you select **Give Advanced Option**, see “Advanced Options”
- In **Action after Leaving Voice Mail**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect. This option is applicable only is you select Go to Voice Mail or Give Advanced options as the No Reply Action option.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.
  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.
  - If you select **Disconnect**, the system will disconnect the call.

Call Transfer Type - Screened

Screened Transfer is when the VMS connects the caller to the transfer number after confirmation from the transfer number. If you have selected Call Transfer Type as Screened, you may customize the Call Transfer Type - Screened parameters as per your requirement.



Screened Options

The system will play Screened Options — Accept Call, Reject Call, Reject Call - Busy, Reject Call - NoReply — to the caller. You can assign the Digits (0-9, *, # or None) to each of the options.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- Accept Call - When this option is selected, the system will transfer the call to the desired number.
- Reject Call - When this option is selected, system will consider it as Call Transfer Unsuccessful - Unconditional and process the call further as per the settings of “Call Transfer Unsuccessful - Unconditional”.
- Reject Call - Busy - When this option is selected, system will consider it as Call Transfer Unsuccessful - Busy and process the call further as per the settings of “Call Transfer Unsuccessful - Busy”.
- Reject Call - No Reply - When this option is selected, system will consider it as Call Transfer Unsuccessful - No Reply and process the call further as per the settings of “Call Transfer Unsuccessful - No Reply”.

No Option Selected Settings

While the Screened Options are being played, if the caller does not select any options, the following parameters will be applicable.

- In **Prompt if No Option Selected**, select the prompt you wish to play to the caller, if no Screened Option is selected by the caller.

Select **None**, if you do not wish to play any prompt.

- Click **Settings** . The **Prompts Management** page opens.

  You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Prompt if No Option Selected.

- In **Action if No Option Selected,** you can select — Accept Call, Reject Call, Reject Call with Busy, Reject Call with No Reply, Disconnect.

Invalid Option Dialed Settings

While the Screened Options are being played, if the caller dials a digit which is not programmed, the following parameters will be applicable.

- Clear the **Ignore if Invalid Option Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.

  - In **Prompt if Invalid Option Dialed,** select the prompt which you wish to play when the caller has dialed an invalid digit.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

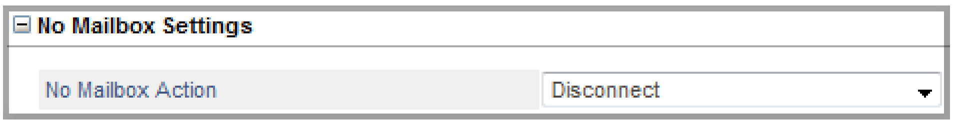
      You may add a new prompt from here. For further details, see “Prompts Management”.

    Once the files are uploaded these appear as options for Prompt if Invalid Option Selected.

  - In **Action if Invalid Option Dialed,** you can select — Accept Call, Reject Call, Reject Call with Busy, Reject Call with No Reply, Disconnect.

No Mailbox Settings

If for any option *Leave Voice Mail* is selected and a mailbox is not assigned to the number, the system will play a prompt to the caller informing the caller that a mailbox is not assigned. The system will then proceed further as per the action you select here.



- In **No Mailbox Action**, you can select — Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Disconnect.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

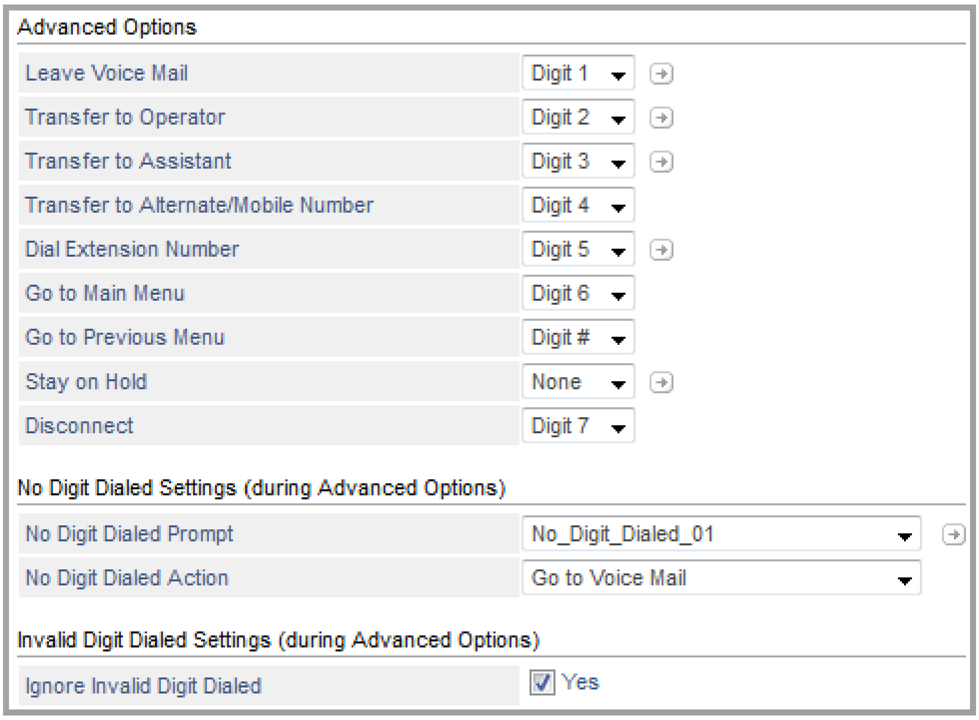
    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

Click **Settings** to configure the parameters of the selected profile.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

Advanced Options

The system will play Advanced options— Leave Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate/Mobile Number, Dial Extension Number, Go to Main Menu, Go to Previous Menu, Stay on Hold, Disconnect — to the caller. You can assign the Digits (0-9, *, # or None) to each of the options.



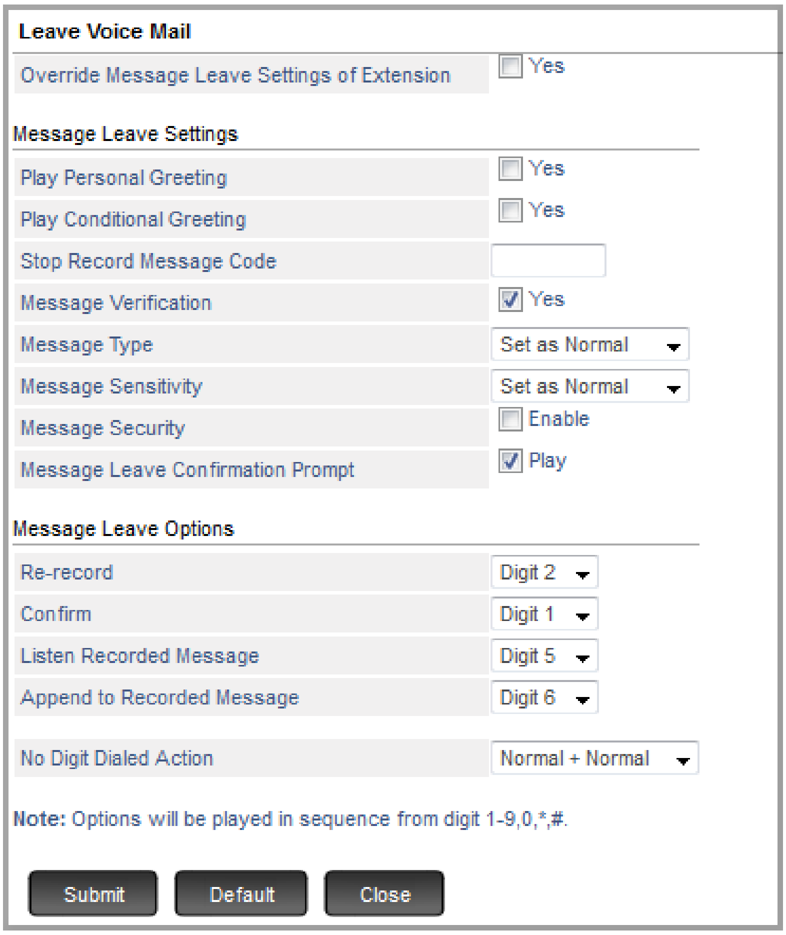
Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- Leave Voice Mail
- Transfer to Operator
- Transfer to Assistant
- Transfer to Alternate Number/Mobile Number
- Dial Extension Number
- Go to Main Menu
- Go to Previous Menu
- Stay on Hold
- Disconnect

Leave Voice Mail



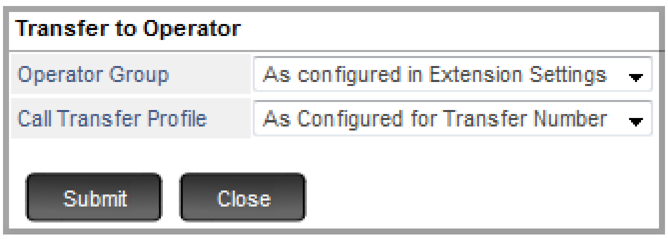
After you assign the digit, click **Settings** to configure the parameters.

- Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply Message Leave Settings customized on this page and not that of the extension or department group number.

  Clear the check box if you want the system to apply Message Leave Settings configured for the extension or department group number.

  For details, refer “Leave Voice Mail”

Transfer to Operator



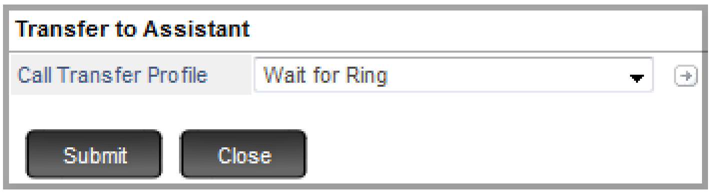
After you assign the digit, click **Settings** to configure the parameters.

- In **Operator Group**, select the desired Operator Group Number. Make sure you have configured the parameters for this group. For detailed instructions, see "Extension Voice Mail Settings". By default, As Configured in Extension Settings is selected, that is, the system will use the Operator Group configured for the Transfer Number.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As Configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile.

Transfer to Assistant



If you assign a digit for this option, make sure you have programmed the Assistant Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

Click **Settings** to configure the parameter.

- In **Call Transfer Profile**, select the desired Call Transfer Profile you want the system to use. Make sure you have configured the parameters for this profile. By default, Wait for Ring is selected, that is, the system will use the Call Transfer profile configured for transfer number.

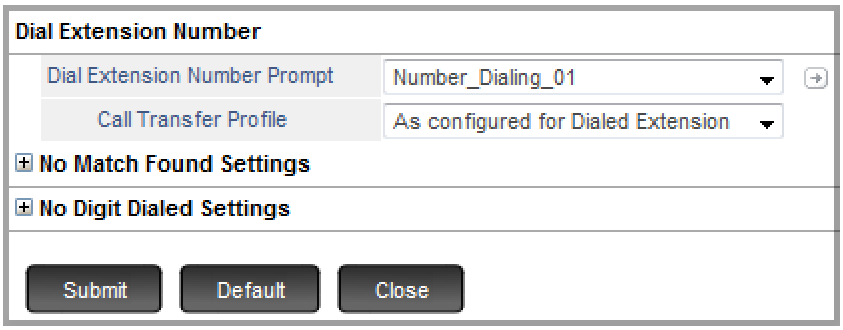
  - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *Wait for Ring* is selected.

    Click **Settings** to configure the parameters of the selected profile.

Transfer to Alternate/Mobile Number

If you assign a digit for this option, make sure you have programmed the Alternate/Mobile Number. For detailed instructions, refer "Number Programming (Assistant/Personal)" in "Mailbox Menu".

Dial Extension Number



After you assign the digit, click **Settings** to configure the parameters.

- In **Dial Extension Number Prompt**,select the prompt you wish to play to the caller to dial the desired extension number.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Dial Extension Number Prompt.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Dialed Extension* is selected.

  Click Settings to configure the parameters of the selected profile.

No Match Found Settings

No Match Found Settings are applicable when the caller has dialed a digit or a number which does not match the Extension Number list present in the system, the system will play the *No Match Found* prompt.

- In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed a digit that does not match the Extension Number List.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

  Click **Settings** to configure the parameters of the selected profile.

- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.

No Digit Dialed Settings

No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.

- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompts the caller to dial the Extension Number again.
- In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.
  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.
  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

- If you select **Disconnect**, the system will disconnect the call.

Go to Main Menu

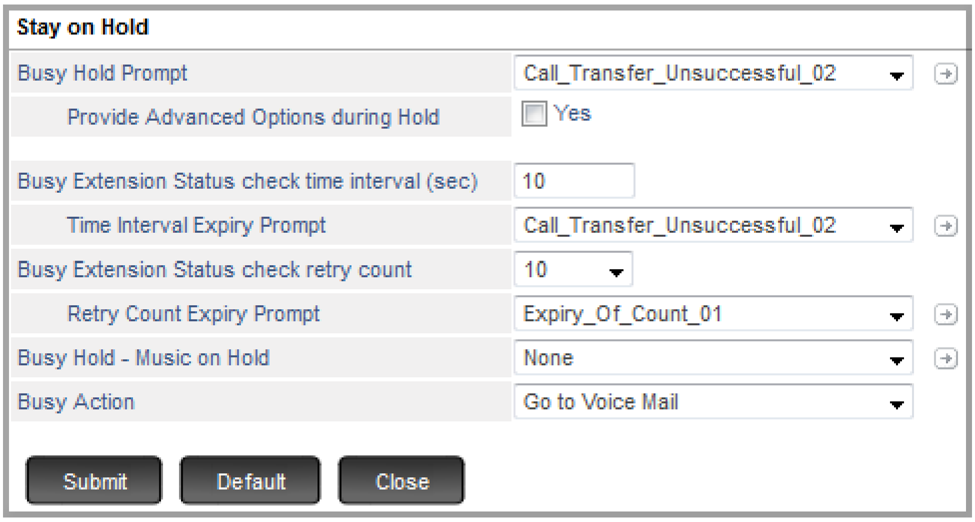
Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.

Go to Previous Menu

Assign a digit for this option if you want the system to re-direct the caller to the Previous Menu.

Stay on Hold

This option is applicable for Call Transfer Unsuccessful - Busy only.



After you assign the digit, click **Settings** to configure the parameters.

- In **Busy Hold Prompt**, select the prompt to be played to the caller when put on hold.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings**. The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Busy Hold Prompt.

  - Select **Provide Advanced Options during Hold** check box, if you want the system to play Advanced options to the caller.

    You can assign the Digits (0-9, *, # or None) to each of the options — Leave Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate/Mobile Number, Dial Extension Number, Go to Main Menu, Go to Previous Menu and Disconnect.

Make sure you do not assign the same digit to multiple options.

The digit you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the Digit selected is None will not be played.

After the digits have been assigned, the system will proceed further as per the options selected by you. Brief description of each option is explained below:

- **Leave Voice Mail:** After you assign the digit, click **Settings** to configure the parameters.
  - Select the **Override Message Leave Settings of Extension** check box, if you want the system to apply Message Leave Settings customized on this page and not that of the extension or department group number.

    Clear the check box if you want the system to apply Message Leave Settings configured for the extension or department group number.
- **Transfer to Operator:** After you assign the digit, click **Settings** to configure the parameters.
  - In **Operator Group**, select the desired Operator Group Number. Make sure you have configured the parameters for this group. For detailed instructions, see “Extension Voice Mail Settings”. By default, As Configured in Extension Settings is selected, that is, the system will use the Operator Group configured for the Transfer Number.
    - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

      Click **Settings** to configure the parameters of the selected profile.
- **Transfer to Assistant**: If you assign a digit for this option, make sure you have programmed the Assistant Number. For detailed instructions, refer “Number Programming (Assistant/Personal)” in “Mailbox Menu”.

  Click **Settings** to configure the parameter.
  - In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.
    Click **Settings** to configure the parameters of the selected profile.
- **Transfer to Alternate/Mobile Number**: If you assign a digit for this option, make sure you have programmed the Alternate/Mobile Number. For detailed instructions, refer “Number Programming (Assistant/Personal)” in “Mailbox Menu”.
- **Dial Extension Number:** After you assign the digit, click **Settings** to configure the parameters.

  Select **None**, if you do not wish to play any prompt.
  - Click **Settings** . The **Prompts Management** page opens.
    You may add a new prompt from here. For further details, see “Prompts Management”.

Once the files are uploaded these appear as options for Dial Extension Number Prompt.

- In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Dialed Extension* is selected.

  Click **Settings** to configure the parameters of the selected profile.

No Match Found Settings: No Match Found Settings are applicable when a Extension Number is found invalid, the system will play the *No Match Found* prompt.

- In **No Match Found Prompt**, select the prompt which you wish to play when caller has dialed an invalid extension number.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Match Found Prompt.

- Select the **No Match Found Retry** check box to prompt the caller to dial the Extension Number again.

- In **No Match Found Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Match Found Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.
    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

No Digit Dialed Settings: No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.

- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

- Select the **No Digit Dialed Retry** check box to prompts the caller to dial the Extension Number again.

- In **No Digit Dialed Retry Count**, select the number of times you wish to prompt the caller for dialing the extension number. The prompt will be played repeatedly till the Retry Count expires.

- In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **No Digit Dialed Action**, you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

    Click **Settings** to configure the parameters of the selected profile.

  - If you select **Go to VMAA Menu**, select the desired VMAA Menu.

  - If you select **Disconnect**, the system will disconnect the call.

- **Go to Main Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Main Menu.

- **Go to Previous Menu:** Assign a digit for this option if you want the system to re-direct the caller to the Previous Menu.

- **Disconnect:** Assign a digit for this option if you want the system to provide the option to disconnect the call.

- In **Busy Extension Status check time interval (sec)**, enter the interval of time after which you want the system to inform the caller that the extension number is busy. The system will play the Time Interval Expiry Prompt.

  - In **Time Interval Expiry Prompt**, select the prompt you wish to play when the time interval expires.

    Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Time Interval Expiry Prompt.

- In **Busy Extension Status check retry count**, select the number of times you wish the system to check if the transfer number is free. The prompt will be played repeatedly till the Retry Count expires.

  - In **Retry Count Expiry Prompt**, select the prompt you wish to play when the Retry Count expires.

    Select **None**, if you do not wish to play any prompt.

    - Click **Settings** . The **Prompts Management** page opens.

      You may add a new prompt from here. For further details, see "Prompts Management".

    Once the files are uploaded these appear as options for Retry Count Expiry Prompt.

- In **Busy Hold - Music on Hold**, select the prompt you wish to play. This option is applicable only is you have enabled the Hold a Call check box.

  Select **None**, if you do not wish to play any prompt.

  You may add a new Prompt. To do so,
  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see "Prompts Management".

  Once the files are uploaded these appear as options for Busy Hold - Music on Hold.

- In **Busy Action,** you can select — Disconnect, Transfer to Operator, Transfer to Extension, Transfer to Department Group, Go to VMAA Menu, Go to Voice Mail, Give Advanced Options.

  - If you select **Transfer to Operator** or **Transfer to Department Group** or **Transfer to Extension**, select the respective Operator number/Department Group number or enter the desired extension number.

    In **Call Transfer Profile**, select the desired Call Transfer Profile as per your requirement. By default, *As configured for Transfer Number* is selected.

Click **Settings** to configure the parameters of the selected profile.

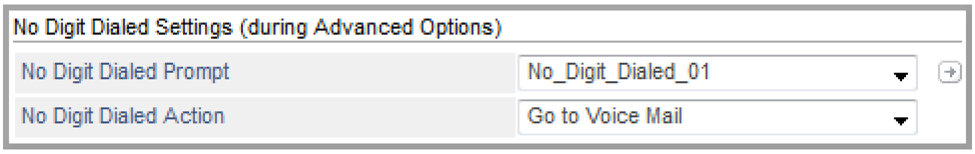
- If you select **Go to VMAA Menu**, select the desired VMAA Menu.
- If you select **Disconnect**, the system will disconnect the call.
- If you select **Give Advanced Option**, see “Advanced Options”.

Disconnect

Assign a digit for this option if you want the system to provide the option to disconnect the call.

No Digit Dialed Settings (during Advanced Options)

No Digit Dialed Settings are applicable when caller has not dialed the extension number and the first digit wait timer has expired.



- In **No Digit Dialed Prompt**, select the prompt which you wish to play when caller has not dialed any digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

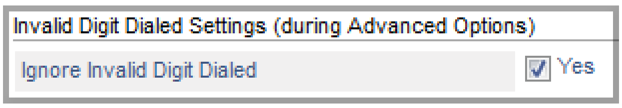
    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for No Digit Dialed Prompt.

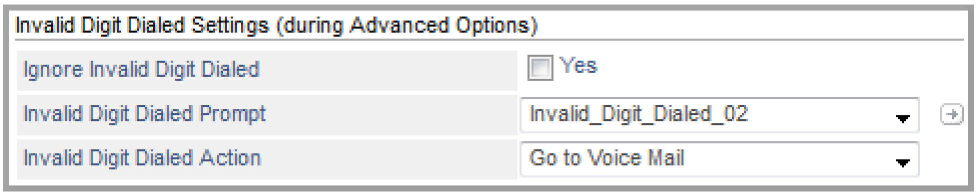
- In **No Digit Dialed Action**, you can select — Disconnect, Go to Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate Number, Go to Main Menu, Go to Previous Menu, Stay on Hold. Refer to the details given above under “Advanced Options”.

Invalid Digit Dialed Settings (during Advanced Options)

Invalid Digit Dialed Settings are applicable when caller has dialed an invalid digit — a digit which is not configured as the Advanced Options.



- Clear the **Ignore Invalid Digit Dialed** check box, if you do not want the system to ignore the invalid digit dialed and prompt the caller for the same. By default, it is enabled.



- In **Invalid Digit Dialed Prompt**, select the prompt which you wish to play when the caller has dialed an invalid digit.

  Select **None**, if you do not wish to play any prompt.

  - Click **Settings** . The **Prompts Management** page opens.

    You may add a new prompt from here. For further details, see “Prompts Management”.

  Once the files are uploaded these appear as options for Invalid Digit Dialed Prompt.

- In **Invalid Digit Dialed Action**, you can select — Disconnect, Go to Voice Mail, Transfer to Operator, Transfer to Assistant, Transfer to Alternate Number, Go to Main Menu, Go to Previous Menu, Stay on Hold.

  Refer to the details given above under “Advanced Options”.

Notification via Call-Profile

The VMS supports Notification via Call to inform the extension users about the arrival of new messages in their mailbox.

Extension users can receive new message notification calls on a phone number of their choice. This number may be another extension number or an external number. You can set the Type of notification calls as:

- **Immediate:** Users will receive notifications as soon as a new message arrives in their mailbox.
  Or
- **Scheduled:** Extension users will receive notification at specified time intervals.

You can set the preferred time slots in a day during which notification calls should be made to extension users. In addition to the time slot preference, you can also choose to receive notification calls on a Holiday.

Message Notification via Call for Department Group will not work if the destination number is an external number.

How it works

For this feature to work, you must do the following configuration for the extension:

- Select the type of Notification call.
- Define the preferred time slots by configuring Time Zones. You can configure four different Time Zones, defining the Start Time and End Time for each Time Zone.
- Configure the phone number to which the notification call is to be made. If the number is an external number, configure the Trunk Access Code to be used for making the calls.

When **Immediate** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.

- The system checks the preferred start and end time of the time zones configured for the extension. If the message has arrived within the preferred time slot (Start and End Time) it immediately makes the notification call on the number configured for the user.

  If the number is an external number, the system dials out the number using the Trunk Access Code (TAC) assigned for making notification calls.

- When the call is answered, the extension user gets connected to the VMS and can listen to the message.

- If the notification call is not answered, by default, the system makes three attempts (Message Notification Retry Count; programmable) at an interval of 5 minutes (Message Notification Interval; configurable) between each attempt.

- If the notification call remains unanswered after the third attempt, the system will not make any more attempts to place this notification call. The next notification call will be made only when another new message arrives in the mailbox of the user between the start and end time of the configured time zone.

When **Schedule** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.
- The system checks the start time of the time zone(s) configured and the notification call will be made on the number at the subsequent start time.

  If the number is an external number the system dials out the number through the Trunk Access Code (TAC) assigned for making notification calls.
- When the call is answered the extension user gets connected to the VMS and can listen to the message.
- If the notification call is not answered, the system makes three attempts (Message Notification Retry Count; configurable) at an interval of 5 minutes (Message Notification Interval; configurable) for each time zone. The system will continue to make attempts to place the notification call till the call is answered.

Thus, when Notification type is Immediate, notification call is made for each message that is received within the start and end time configured in the time zone.

When Notification type is Scheduled, notification call is made for all messages received before the start time configured in the time zone. Where multiple time zones are configured, notification call will be made at the start time of the next time zone.

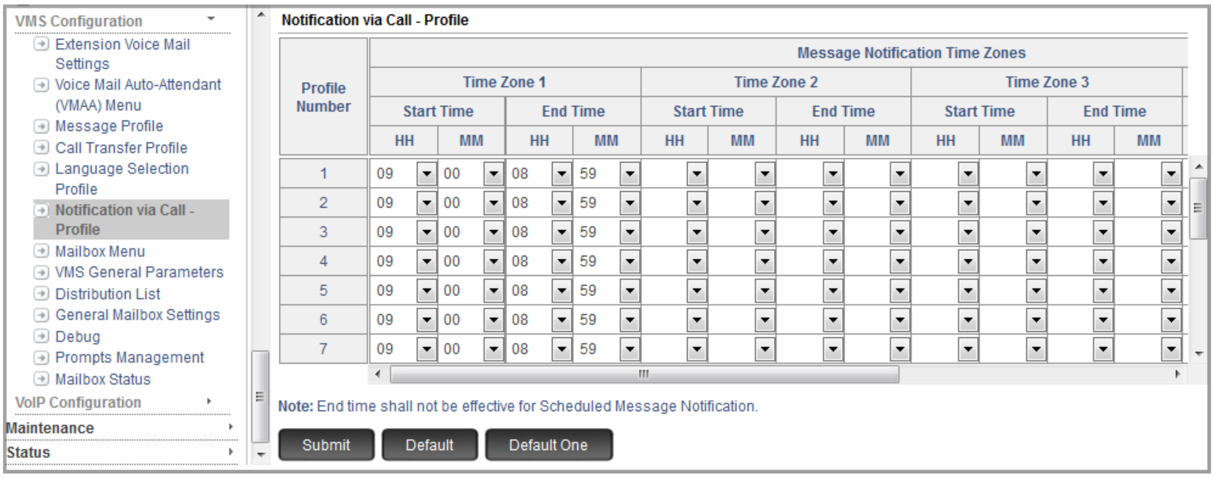
How to configure

For Message Wait Notification via Call, you need to configure:

- the parameters for **Message Wait Notification via Call** under Message Wait Settings in Extension Voice Mail Settings.
- select the desired profile in Schedule Profile. For instructions to configure the profile parameters, see “Configuring Notification via Call - Profile”.
- Make Message Notification call using TAC for calls to be made to external numbers. See “Configuring VMS General Parameters”.
- if required, the Message Notification Retry Count, Message Notification Interval and Message Notification Ring. See “System Timers and Counts”.

Configuring Notification via Call - Profile

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Notification via Call - Profile**.



- Select the **Profile Number** which you want to assign to Message Wait Notification via Call.

  The Message Wait Notification Profile determines how notification calls are to be made to the desired numbers. You can configure upto 16 different profiles. In each profile, you can set different time zones according to the user preferences.

  Configure the following parameters against the Profile Number you select:

  - For Each Time Zone, **Time Zone 1** to **4**, configure the **Start Time** and **End Time**. The valid range is 00:00 to 23:59.
  - If you want to receive notifications on a holiday, select the **Notify on Holiday** check box.
  - Click **Submit**.
  - Now, assign the profile numbers to the desired extensions. Make sure the Message Wait Notification via Call parameters have been configures in the Extension Voice Mail Settings of these extensions.

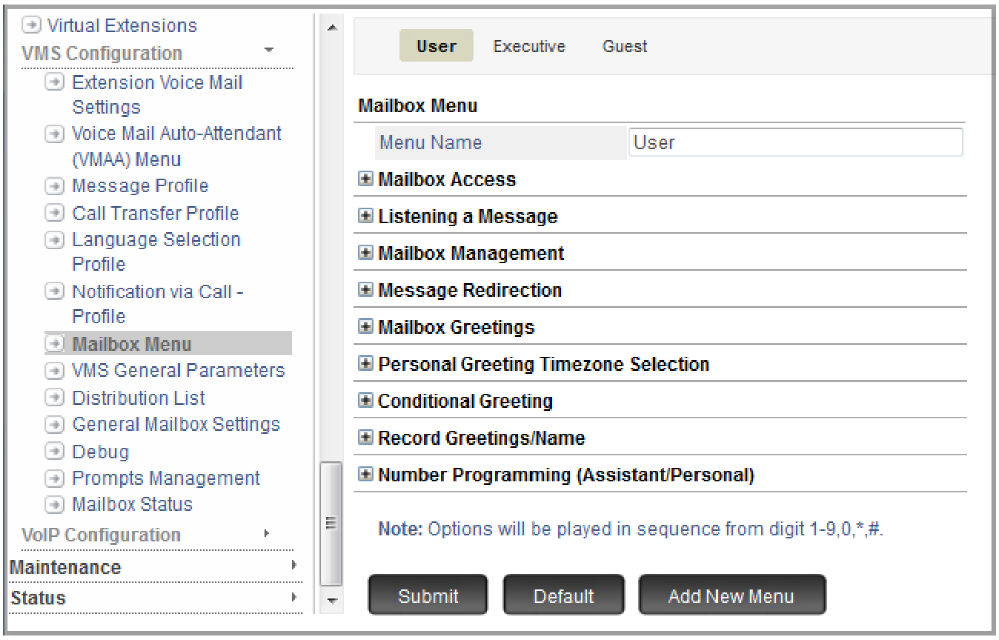
Mailbox Menu

Mailbox Menu offers you a group of parameters that will be used when the caller accesses his/her personal mailbox. VMS supports a maximum of 12 Mailbox Menus.

By default, three mailbox Menus — User, Executive and Guest— are provided to you. These three mailbox menus cannot be deleted but you can edit their settings as per your requirement.

How to Configure

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Mailbox Menu**.



You may add a new menu, edit the default menus or delete a Mailbox menu.

To add a new Mailbox Menu,

- Click **Add New Menu**. A new *Mailbox Menu xx* will be created. You may now configure this as per your requirement.

To delete a menu, click **Delete**.

The default Mailbox Menus cannot be deleted.

To edit a Mailbox Menu,

- In **Menu Name**, configure the name of the Mailbox Menu you want. By default, it is *Mailbox Menu xx* where xx is the Mailbox Menu Number from 01 to 12.

Mailbox Menu Features

You can assign the Digits (0-9, *, # or None) to each of the options given under the features — Mailbox Access, Listening a Message, Mailbox Management, Message Redirection, Mailbox Greetings, Personal Greetings Timezone Selection, Conditional Greeting, Record Greetings/Name and Number Programming (Assistant/ Personal).

Make sure you do not assign the same digit to multiple options under the respective feature. For example, the digits assigned to various options under Listening a Message feature must be unique.

Select the **Play** check box to play the respective option.

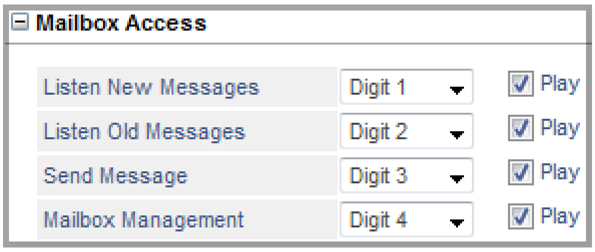
If you assign the digit to an option but do not select the Play check box for the same, then the option will not be played to the caller.

The digits you configure will be played as the digit to be dialed by the caller to select the respective option. The options will be played in the sequence — 1-9, 0, *, #. The option for which the digit selected is None will not be played.

Brief description of each option in the respective features are given below:

Mailbox Access

Mailbox Access is the main menu which will be played to the caller whenever s/he accesses his/her Personal Mailbox.



- **Listen New Messages**: The system will play the new(unread) messages present in the caller's personal mailbox. The new messages will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer "Message Playback Settings".

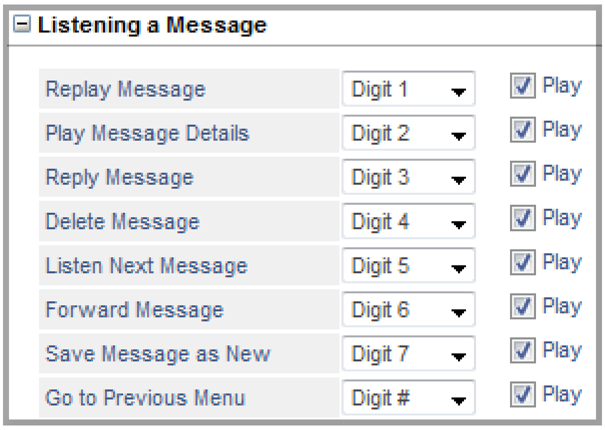
- **Listen Old Messages**: The system will play the old(read) messages present in the caller's personal mailbox. The old messages will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer "Message Playback Settings".

- **Send Message**: The system will play a prompt asking the caller to enter the Destination Number of the recipient for sending the message. The Destination Number may include a Distribution List. The Message will be send as per the *Message Send/Forward Settings* configured in the Message Profile. For details, refer "Message Send/Forward Settings".

- **Mailbox Management**: The system allows user to manage his/her personal mailbox. For further information, see "Mailbox Management".

Listening a Message

Listening a Message is a menu which will be played to the caller after every message (old/new).



- **Replay Message**: The system will play the message again.

- **Play Message Details**: The system will play the message details — Message Date/Time and the Name of sender. The message details will be played as per the *Message Playback Settings* configured in the Message Profile. For details, refer “Message Playback Settings”.

- **Reply Message**: The system will allow the caller to reply the sender with a message. The Message will be sent as per the *Message Leave Settings* configured in the Message Profile. For details, refer “Message Leave Settings”.
  After sending the reply message, the Mailbox Access menu will be played again.

  Reply Message will not be played if:
  - the message does not include the calling party number.
  - the message includes the calling party number but no Mailbox is assigned.
  - the message includes the calling party number as an external number.
  - it is a Broadcasted or a Call Tapping message.

- **Delete Message**: The system will delete the message and play the next message(old/new).

- **Listen Next Message**: The system will play the next message(old/new) as per the last message played.

  After playing the next message, the *Mailbox Access menu* will be played again.

- **Forward Message**: The system will play a prompt asking caller to enter the Destination Number of the recipient for forwarding the message.

  The Destination Number may include a Distribution List. A maximum of 10 Destination Numbers can be added. The Message will be sent as per the *Message Send/Forward Settings* configured in the Message Profile. For details, refer “Message Send/Forward Settings”.

  After forwarding the message, the *Mailbox Access menu* will be played again.

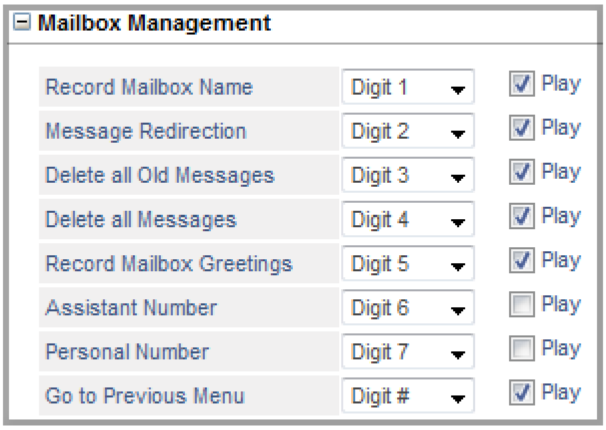
  The message will not be forwarded if the *Message Sensitivity* is set as *Private*. For details, refer “Message Profile”.

- **Save Message as New**: The system will save the message as a New Message keeping all the message properties — Date and Time, Caller, Sensitivity and Security — unchanged.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the "Mailbox Access" menu.

Mailbox Management

Mailbox Management is a menu used to manage the Personal Mailbox.



- **Record Mailbox Name**: The system will allow the caller to record, play or erase the Mailbox Name.

  The *Record Greetings/Name* options will be played where the caller can choose whether s/he want to Record, Play or Erase the Name. For details, refer "Record Greetings/Name".

- **Message Redirection**: The system will allow the caller to set or cancel Message Redirection. Message Redirection is used when the caller wants to forward all the messages to the other user's personal mailbox.

  The *Message Redirection* options will be played where caller can choose if s/he wants to set or cancel this feature. For details, refer "Message Redirection".

- **Delete all Old Messages**: The system will delete all the old messages from the caller's Personal Mailbox.

- **Delete all Messages**: The system will delete all the messages (old/new) from the caller's Personal Mailbox.

- **Record Mailbox Greetings**: The system will allow the caller to record, play or erase the Mailbox Personal and Conditional Greetings.

  The options will be played in the following sequence:

  - The *Mailbox Greetings* options will be played asking the caller to select the type of greetings — Personal or Conditional — which he wishes to Record, Play or Erase. For details, refer "Mailbox Greetings".

    - If the caller selects *Personal*, the *Personal Greeting Time zone Selection* options will be played. The caller must select the timezone — Working Hour, Break Hour or Non-Working Hour for which s/he wishes to Record, Play or Erase the personal greeting. For details, refer "Personal Greeting Timezone Selection".

        - If the caller selects *Conditional*, the *Conditional Greetings* options will be played. The caller must select the type of Conditional Greeting — Busy, No-Reply or Unconditional for which s/he wishes to Record, Play or Erase the greetings. For details, refer “Conditional Greeting”.

    - The *Record Greetings/Name* options will be played where the caller can choose whether s/he wants to Record, Play or Erase the Mailbox Greetings. For details, refer “Record Greetings/Name”.

- **Assistant Number**: The system will allow the caller to enter, play or clear the Assistant Number.

  The caller can call this number when the user is not available to answer the call.

  The *Number Programming (Assistant/Personal)* options will be played from where the caller can choose whether s/he wants to Enter, Play or clear the Assistant Number. For details, refer “Number Programming (Assistant/Personal)”

- **Personal Number**: The system will allow the caller to enter, play or clear the Personal Number.

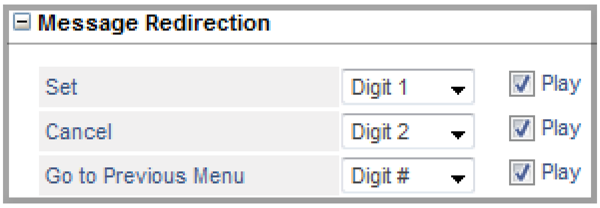
  The Personal Number is used as an alternate number. The caller can call this number when the user is not available to answer the call.

  The *Number Programming (Assistant/Personal)* options will be played from where the caller can choose whether s/he want to Enter, Play or clear the Personal Number. For details, refer “Number Programming (Assistant/Personal)”.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Access” menu.

Message Redirection

Message Redirection is a menu which is applicable if the caller selects the *Message Redirection* option in “Mailbox Management”.



- **Set**: The system will set the Message Redirection feature and play a prompt to the caller for entering the Message Redirect Number.

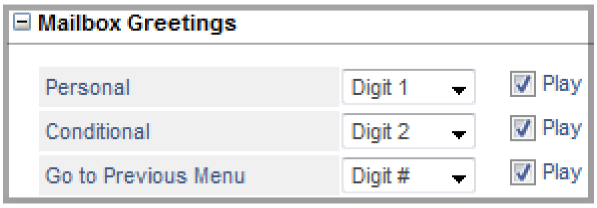
  The Message Redirect Number must be a valid Extension Number or Department Group Number. A Mailbox must be assigned to this Number.

- **Cancel**: The system will clear the Message Redirect Number programmed by the caller.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Management” menu.

Mailbox Greetings

Mailbox Greetings is a menu which is applicable if the caller selects the *Record Mailbox Greetings* option in the Mailbox Management.



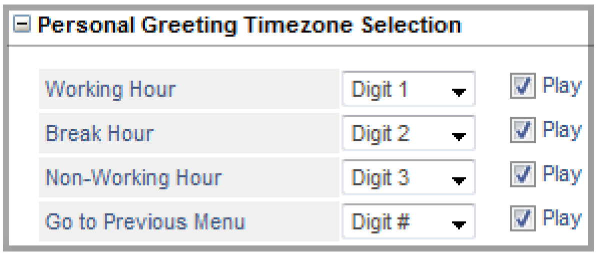
- **Personal**: The system will play the *Personal GreetingTimezone Selection* options. The caller must select the timezone — Working Hour, Break Hour or Non-Working Hour for which s/he wishes to Record, Play or Erase the Personal greetings. For details, refer "Personal Greeting Timezone Selection".

- **Conditional**: The system will play the *Conditional Greetings* option. The caller must select the type of Conditional Greeting — Busy, No-Reply or Unconditional for which s/he wishes to Record, Play or Erase. For details, refer "Conditional Greeting".

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the "Mailbox Management" menu.

Personal Greeting Timezone Selection

Personal Greeting Timezone Selection is a menu which is applicable if the caller selects the *Personal* Greetings option in the Record Mailbox Greetings. You can record the personal greetings like "Good Morning", "Good Afternoon" or "Good Evening" to greet the caller.



The various Timezones are:

- **Working Hour**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Working Hour Timezone.

- **Break Hour**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Break Hour Timezone.

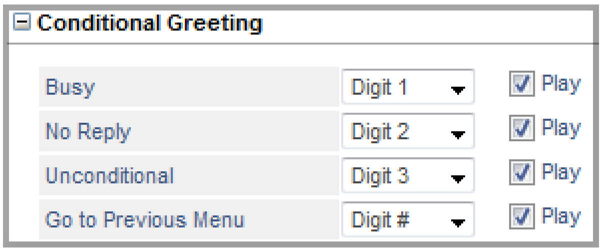
- **Non-Working Hour**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase a personal greeting for the Non-Working Hour Timezone.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Record Mailbox Greetings Menu.

  For details, refer "Record Greetings/Name".

Conditional Greeting

Conditional Greeting is a menu which is applicable if the caller selects the *Conditional* option in the Record Mailbox Greetings. Conditional Greetings are applicable if any call forward — Busy, No Reply or Unconditional — is set on the calling extension.



The various Conditional Greeting types are:

- **Busy**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — Busy.

- **No-Reply**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — No-Reply.

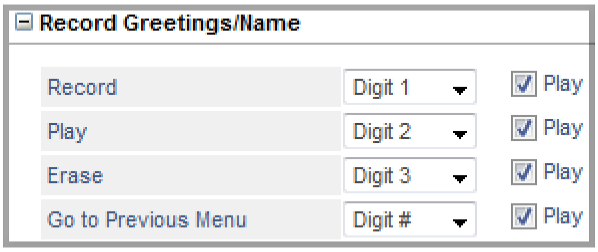
- **Unconditional**: The system will play the *Record Greetings/Name* options and allow the caller to Record, Play or Erase the Conditional Greeting of type — Unconditional.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Record Mailbox Greetings Menu.

  For details, refer "Record Greetings/Name".

Record Greetings/Name

Record Greetings/Name is a menu which allows the caller to Record, Play or Erase the Mailbox Name or the Personal/Conditional Greetings.



- **Record**: The system will allow the caller to Record the Mailbox Name or the Personal/Conditional Greeting. Maximum allowed length for recording a name/greeting is 120 seconds.

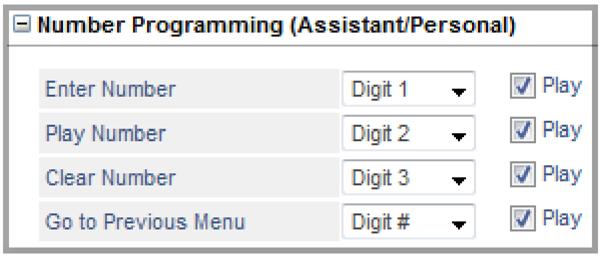
- **Play**: The system will Play the Mailbox Name or the Personal/Conditional Greeting to the caller.

- **Erase**: The system will clear the recorded Mailbox Name or the Personal/Conditional Greeting.

- **Go to Previous Menu**: The system will provide an option to the caller to go back to the Previous Menu — *Record Mailbox Name, Personal Greeting Timezone Selection* or *Conditional Greeting*.

Number Programming (Assistant/Personal)

Number Programming (Assistant/Personal) is a menu which allows the caller to program the Assistant Number or Personal Number.



- **Enter Number**: The system will allow the caller to program the Assistant Number or Personal Number. For Assistant Number, only the Assistant Number List present in the system will be allowed.

Make sure the Assistance Number is an extension number and the Personal Number is an external (Mobile) number.

Make sure you do not enter the Department Group Mailbox Number as the Assistant Number or the Personal Number.

- **Play Number**: The system will play the Assistant Number or Personal Number to the caller.
- **Clear Number**: The system will clear the Assistant Number or Personal Number programmed by the caller.
- **Go to Previous Menu**: The system will provide an option to the caller to go back to the “Mailbox Management”.

Distribution List

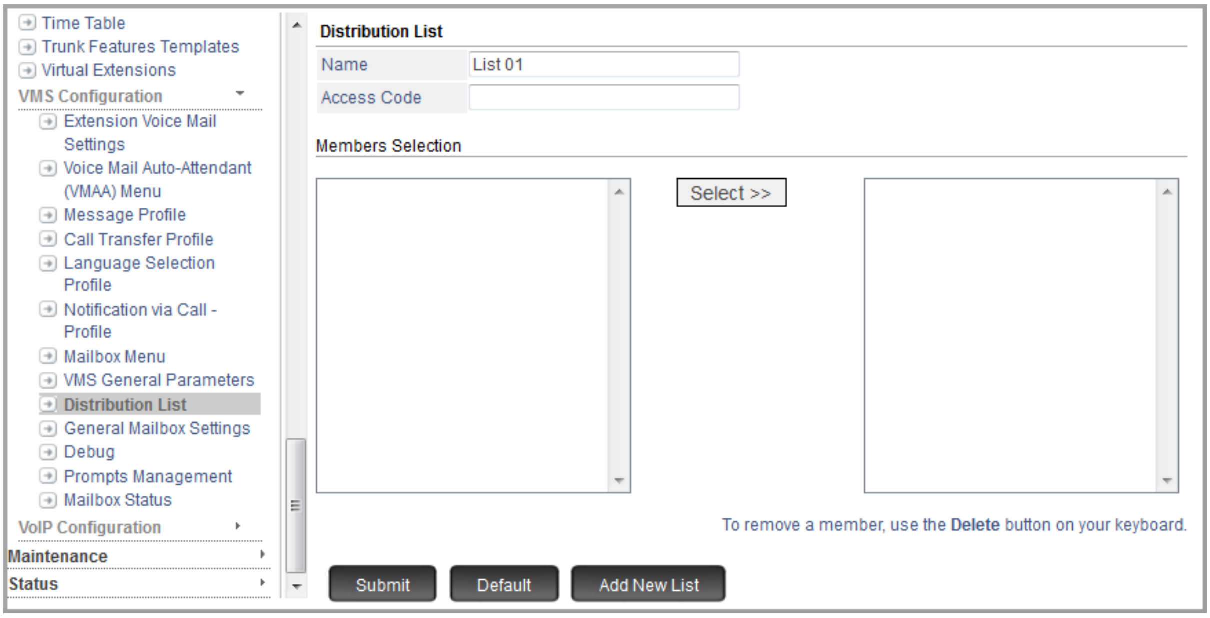
A Distribution List enables extension users to send the same message to a group of extensions at the same time.

Any extension with a mailbox can be included in a Distribution List. You can create upto 30 Distribution Lists of 50 members each.

How to configure

To configure Distribution List,

- Login as System Engineer.
- Under **Configuration,** click **VMS Configuration**.
- Click **Distribution List**.



You may add a new list, edit the default list or delete a Distribution List.

To add a new List,

- Click **Add New List**. A new *List xx* will be created. You may now configure this as per your requirement.

To delete a list, click **Delete**.

The default Distribution List cannot be deleted.

To edit a Distribution List,

- In **Name**, configure the name of the Distribution List. By default, it is *List xx* where xx is the Distribution List Number from 01 to 30.

- In **Access Code**, configure the Access Code you wish to assign to the respective Distribution List. The caller can send or forward the message to a Distribution List by dialing the respective access code. It can be a maximum of 6 digits.

Make sure, the Access Codes assigned to the Distribution Lists do not conflict with the existing access codes. System will not save the configured Distribution List access code if the same code is already assigned.

Members Selection

- All the Extension Numbers appear in the left side box arranged sequentially in the increasing order.
    - To select the members for the Distribution List,
        - Select the desired Extension and click **Select>>** button. The selected extension will appear on the right side box. It is also possible to select a range of extensions at a time by pressing 'SHIFT' and 'Down' arrow keys.

          A maximum of 50 extensions can be selected.
    - To delete the members from the Distribution List,
        - Place your cursor on the desired Extension and click delete key from your keyboard. The selected extension will be deleted. It is also possible to select a range of extensions at a time by pressing 'SHIFT' and 'Down' arrow keys.
- Click **Submit**.

Recording Voice Messages

The VMS of ANANT UCS supports voice messages for different functions, which are broadly classified as:

- **System Greetings**: These messages are played to the caller when a new call lands on the VMS. Callers are greeted according to the time of the day - morning, afternoon, evening (Time Zone). You can customize the Time Zones as per your requirement. For detailed instructions, see "Greeting Message Time". A different System Greeting can also be played to callers on holidays.

  System Greetings are played to callers when the VMS Auto Attendant feature is enabled on trunks.

- **Personal Greetings:** These messages are played to callers when they are diverted to the extension user's mailbox to leave a message. Extension users can record personal mailbox greeting messages of their choice.

- **Conditional Greetings:** These messages are played to callers when they are diverted to the extension user's mailbox for certain conditions—busy, no reply or unconditional/unregistered Call Forward. Extension users can record a different message for each condition.

- **Welcome Messages:** These messages are played to callers who call the VMS. Welcome messages help the callers navigate through the VMS. Welcome messages are played according to the time of the day, that is, the Time Zone configured in the system.

  Welcome Messages are played to callers when the VMS Auto Attendant feature is enabled on trunks.

- **Holiday Messages:** These messages are played to callers who call the VMS on a Holiday. These messages are played in place of the Welcome messages and help the callers navigate through the VMS. A different message can be played for each holiday.

  Holiday Messages are played to callers when the VMS Auto Attendant feature is enabled on trunks and the Holiday Table is configured.

- **Prompts/Responses:** These are voice guidance messages that are played to the caller in response to the action taken (that is, when the caller dials a digit).

For all of these message types, audio files containing the appropriate recorded voice guidance messages are loaded in the configuration of the VMS. The VMS plays the messages related to the function it is performing.

For example, if Voice Mail Auto Attendant (the VMS Auto Attendant feature) is enabled on a trunk, the VMS plays messages relevant to the Voice Mail Auto Attendant Menu configured for the current Time Zone. This helps the caller navigate through various options as the VMS plays the related message, as explained below:

- The VMS plays the default System Greeting and Welcome message to the caller according to the time of the day, e.g.: *"Good Morning". "Welcome! Please dial the extension number or To dial by name press '6', To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#' (hash)".*

- As the caller navigates, the VMS plays the pre-recorded voice messages related to the particular option selected by the caller. If the caller dials 6 to dial by name, the VMS plays relevant voice message, e.g.: "Please enter first three letters of the name" and then plays "More than one match found. Matching Names will be played one by one. To Select the name press '1', to Skip the name press '2', To Repeat the last name press '3'."

- If the caller dials 1 to select the name, the VMS plays the prompt: "To confirm press '1', to Re-enter press '2'."

- If the caller dials 1 the VMS transfers the call as per the transfer type assigned to the selected station.

In the same way, when you set a voice guided alarm, the VMS plays the alarm-related voice prompts, like: *"Enter the time, HH MM in twenty four hour format*". Thus, for every voice mail related function or feature, the VMS plays the appropriate voice message.

The VMS gives you the option of either using the default voice guidance messages or recording custom messages that better suits your purpose.

All VMS default voice messages are in English only. If you want, you may record voice messages in your local language.

No special configuration is required for using the default voice messages. However, if you want to use custom messages, you must first:

- record the message (Greetings, Welcome Messages, Holiday Messages, Voice Guidance prompt).
- upload the new recorded message file in the VMS configuration.

Recording Voice Messages

When you record messages of your choice, consider these important points:

- The custom messages must be in wav format.

- Make sure the message files have the following attributes:
  - Audio Format: CCIT u-law
  - Channel: 1 (mono)
  - Sampling Frequency: 8 KHz
  - Audio Sample Size: 8 bit

- You must record the custom messages from an external source.

- You must verify the message and then these messages must be uploaded on to the VMS configuration files.

- Extension users can record their personal mailbox greetings on their own. Refer the topic "Recording Personal Greetings" to know more.

- Extension users can record the conditional greetings on their own. Refer the topic "Recording Conditional Greetings" to know more.

Uploading Custom Voice Messages

As mentioned earlier, when you record voice messages from an external source, make sure that:

- The audio file is recorded in the prescribed format (.wav) and attributes.
- ANANT UCS is connected to a computer (standalone or LAN).
- You can upload a single prompt or the entire folder. For detailed instructions, see "Prompts Management".

Prompts Management

Prompts Management allows you to view all the prompt folders present in the system for each of the configured languages. You may upload or download a prompt file to/from the VMS Prompt folders for any language. The default prompts recorded in English language are provided to you. See "VMS Prompts" for details.

You may overwrite the default prompts or upload new prompts. While uploading the prompts for other languages, you must record them with reference to the default prompts listed under the topic ***VMS Prompts***.

There are 21 Prompt folders present in the system. The first 11 folders are flexible while the others are fixed.

The Flexible Folders are:

- Greeting
- Auto-Attendant
- Number Dialing
- No Digit Dialed
- Invalid Digit Dialed
- Expiry of Count
- Call Transfer Type
- Call Transfer Unsuccessful
- Information
- MoH
- Disconnect

The Fixed Folders are:

- Language
- Dial by Name
- Call Transfer
- Message Record
- Message Send Forward
- Mailbox Access
- Mailbox Access Menu
- Number and Month
- Alarm
- Miscellaneous

The Flexible Folders include the prompt files which the system uses according to the configuration done by you. You may upload new prompt files, download and delete the existing files from the system.

The Fixed Folders include the prompt files which the system uses internally and are non-configurable. You may overwrite or download the existing files but cannot delete these from the system.

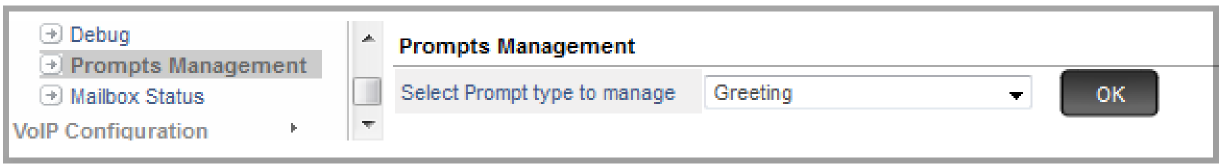
The following table defines the maximum number of prompt files supported in each folder.

Folder Name	Maximum Prompt files supported	Maximum length allowed for each prompt (seconds)
Greeting	24	180
Auto-Attendant	99	180

Folder Name	Maximum Prompt files supported	Maximum length allowed for each prompt (seconds)
Number Dialing	12	120
No Digit Dialed	12	120
Invalid Digit Dialed	12	120
Expiry of Count	12	120
Call Transfer Type	12	120
Call Transfer Unsuccessful	12	120
Information	24	180
MoH	12	180
Disconnect	12	120
Language	01	120
Dial by Name	13	120
Call Transfer	16	120
Message Record	19	120
Message Send Forward	30	120
Mailbox Access	29	120
Mailbox Access Menu	50	120
Number	46	120
Alarm	53	120
Miscellaneous	24	120

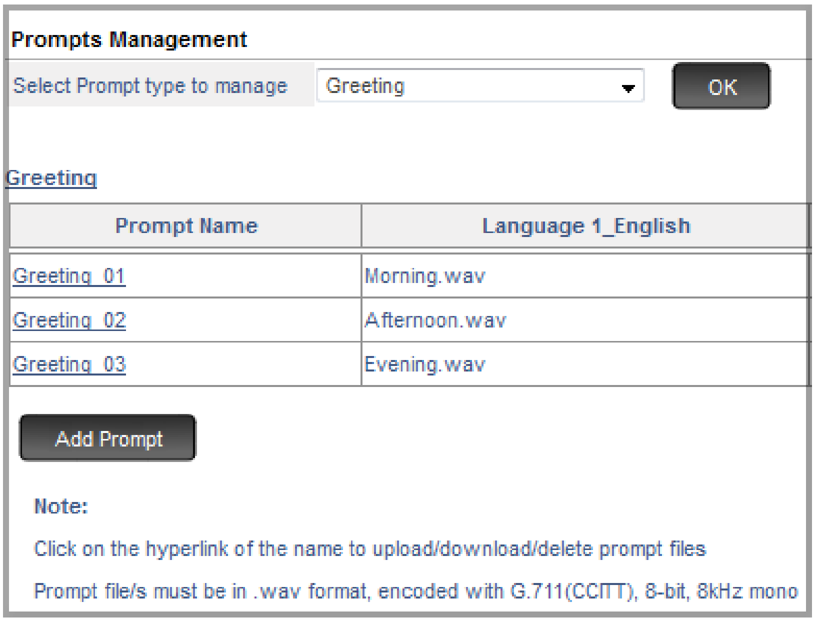
How to Configure

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Prompts Management**.



- In **Select Prompt type to manage**, select the desired folder.
- Click **OK**.

  All the prompt files present in the respective language folders of the selected Prompt Type will be displayed.



You can upload or download the entire folder for the respective Prompt Type for each of the languages.

To add a new prompt in a Flexible Folder,

- Click the **Add Prompt** button at the bottom of the page.

  The **Add/Edit Prompt** window opens.

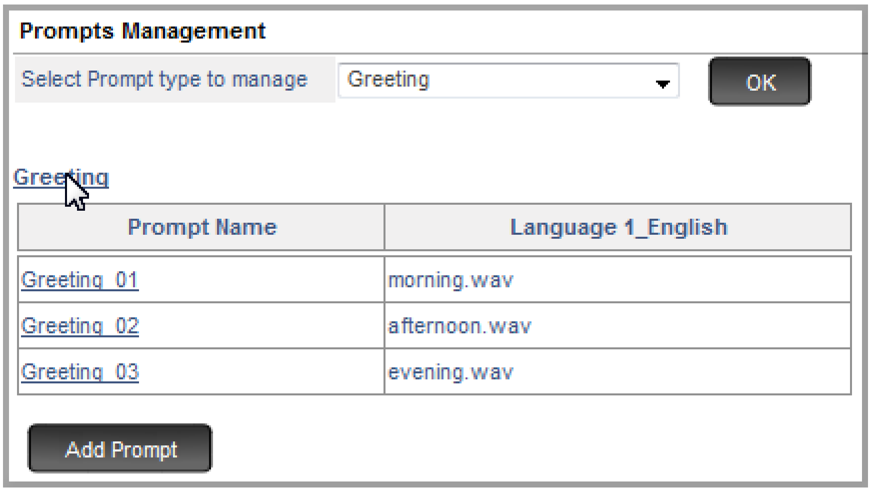
- Click the **Browse** button to reach the location on the local disk where the respective prompts are stored in your PC.

- Click  to upload.

- *Make sure the folder you upload is a zip file.*
- *Make sure the prompt files within the folder are in .wav format and have the following attributes:*
  - *Audio Format: u-law*
  - *Channel: 1 (mono)*
  - *Sampling Frequency: 8 KHz*
  - *Audio Sample Size: 8 bit*

To Upload or Download the entire Prompt Type folder,

- Click on the **Prompt Type** link.



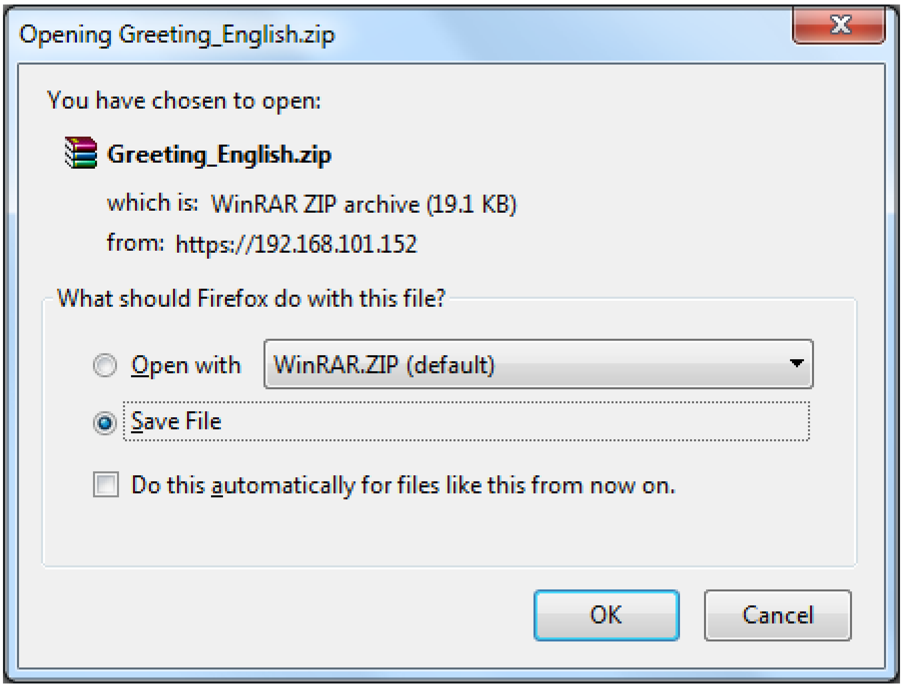
The **Upload/Download Prompt Type** Window opens.

- The language-wise folders will be displayed.
- Click the **Browse** button of the desired language to reach the location on the local disk where the respective prompt folder is stored in your PC.
- Click to upload.

- *Make sure, the file to be uploaded is a .wav file.*
- *The filename can be a maximum of 64 characters.*
- *The allowed characters for the filename are A to Z, a to z, 0 to 9, - and _.*

To download all the prompts of the respective prompt type and language selected, click .

The **Opening Prompt Type_Language.zip** window will open.

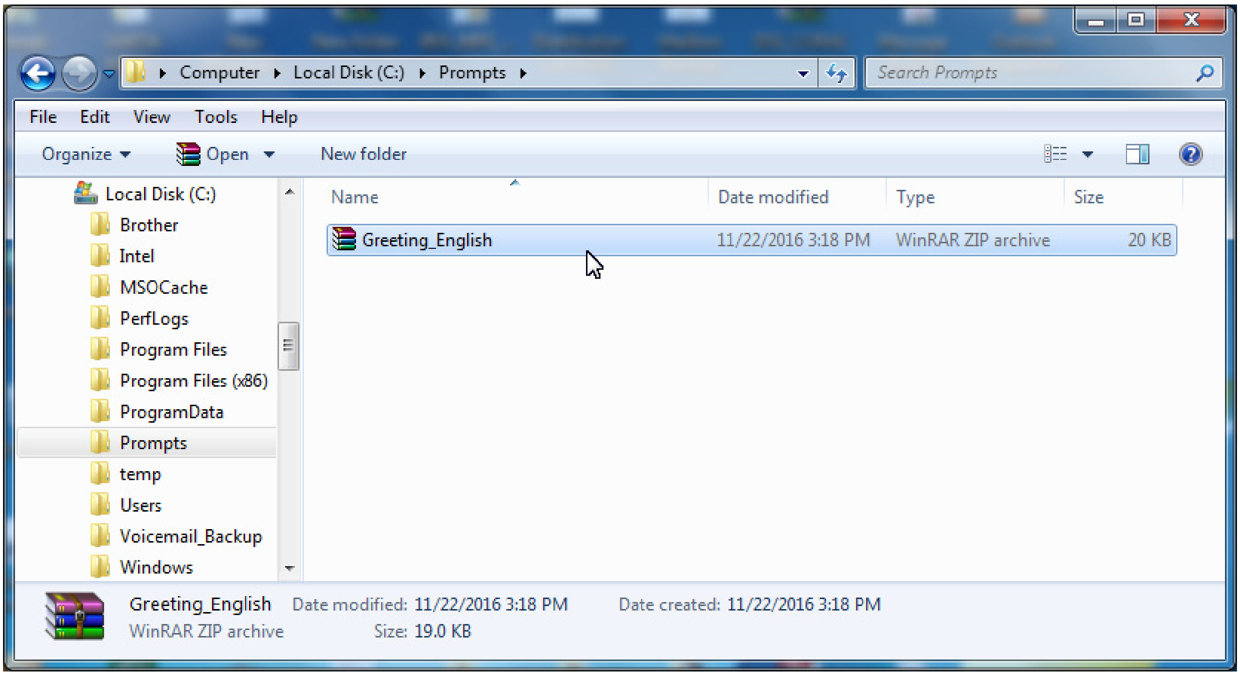


- You can either open the zip file or save the file to a location.

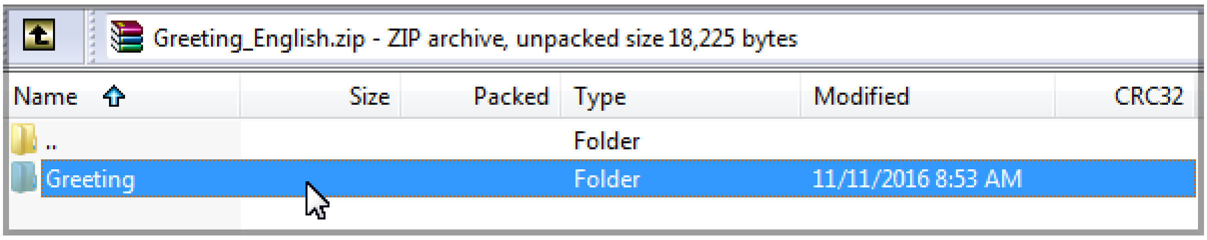
- *The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
  OR
  If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

- *If you are using Mozilla Firefox (version 3.5.1 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

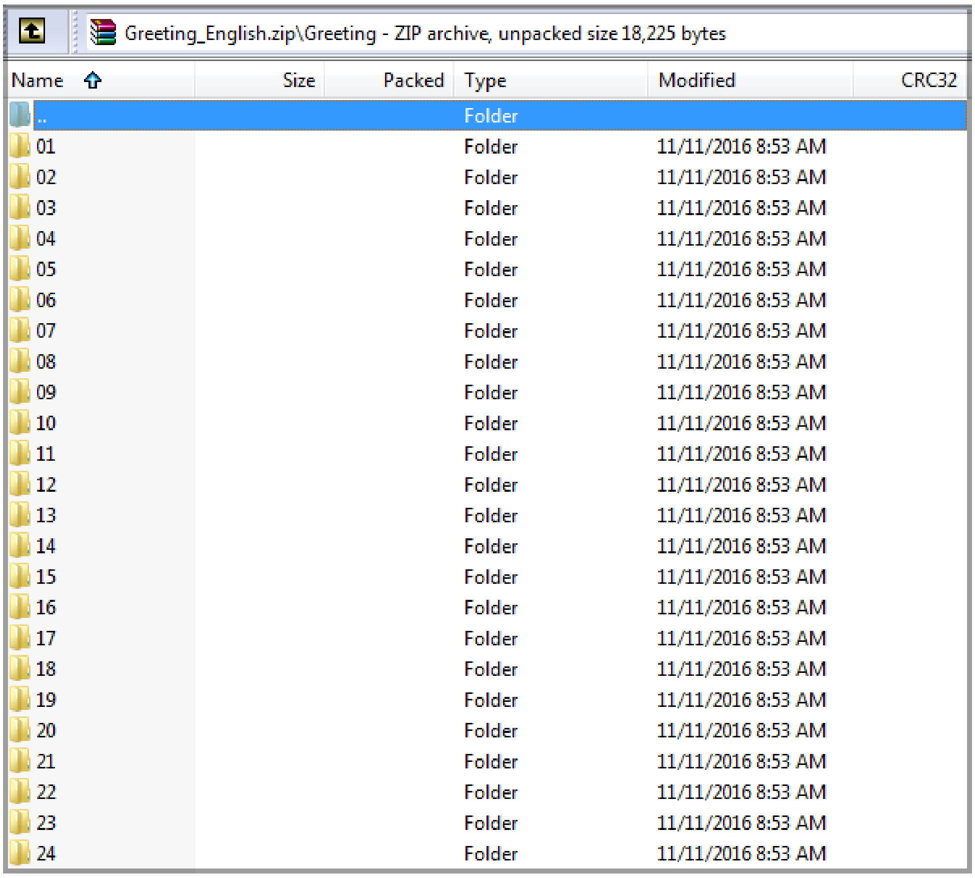
- Save the file on the local disk.



- To view the files,
  - Open the **Prompt Type_Language** folder.



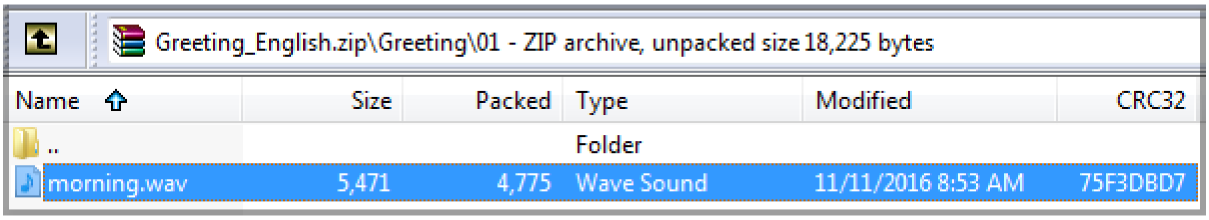
- Now, open the **Prompt Type folder** to view all the subfolders containing the prompts.



Greeting_English.zip\Greeting - ZIP archive, unpacked size 18,225 bytes

Name	Size	Packed	Type	Modified	CRC32
..			Folder		
01			Folder	11/11/2016 8:53 AM	
02			Folder	11/11/2016 8:53 AM	
03			Folder	11/11/2016 8:53 AM	
04			Folder	11/11/2016 8:53 AM	
05			Folder	11/11/2016 8:53 AM	
06			Folder	11/11/2016 8:53 AM	
07			Folder	11/11/2016 8:53 AM	
08			Folder	11/11/2016 8:53 AM	
09			Folder	11/11/2016 8:53 AM	
10			Folder	11/11/2016 8:53 AM	
11			Folder	11/11/2016 8:53 AM	
12			Folder	11/11/2016 8:53 AM	
13			Folder	11/11/2016 8:53 AM	
14			Folder	11/11/2016 8:53 AM	
15			Folder	11/11/2016 8:53 AM	
16			Folder	11/11/2016 8:53 AM	
17			Folder	11/11/2016 8:53 AM	
18			Folder	11/11/2016 8:53 AM	
19			Folder	11/11/2016 8:53 AM	
20			Folder	11/11/2016 8:53 AM	
21			Folder	11/11/2016 8:53 AM	
22			Folder	11/11/2016 8:53 AM	
23			Folder	11/11/2016 8:53 AM	
24			Folder	11/11/2016 8:53 AM	

- Open the subfolder containing the prompt.



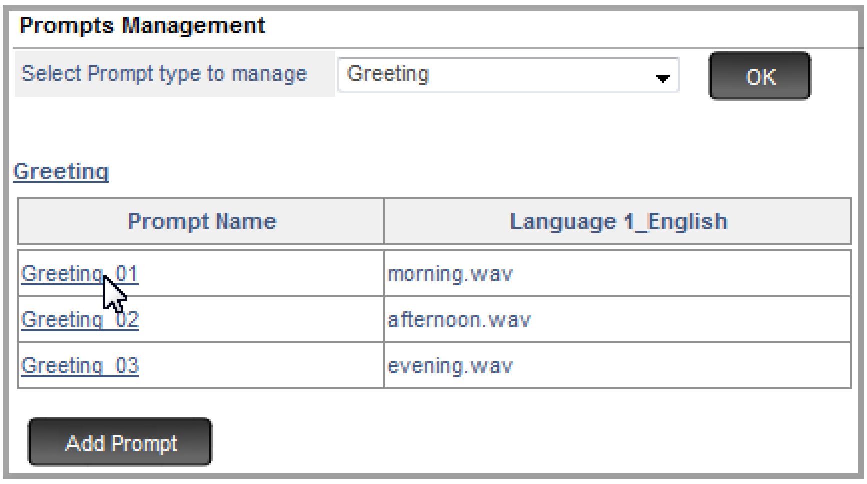
Greeting_English.zip\Greeting\01 - ZIP archive, unpacked size 18,225 bytes

Name	Size	Packed	Type	Modified	CRC32
..			Folder		
morning.wav	5,471	4,775	Wave Sound	11/11/2016 8:53 AM	75F3DBD7

You may now upload, download or delete a prompt file to/from a folder. The files present in the Fixed Folders cannot be deleted or added, but the existing files can be overwritten by a new one.

To Upload or Download a Prompt,

- Click on the respective **Prompt Name** link.



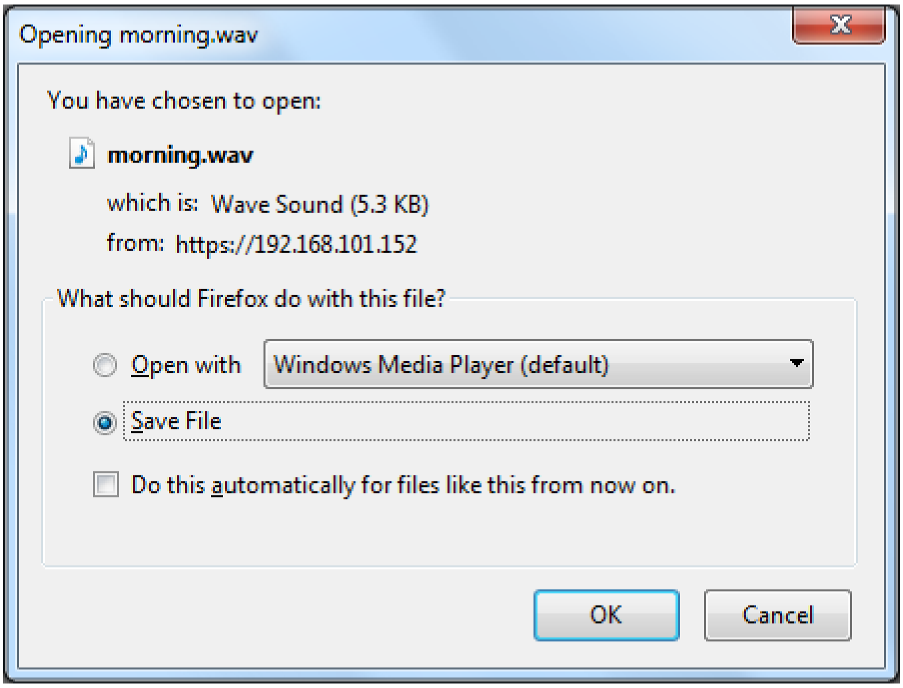
The **Add/Edit Prompt** window will open.

- To upload or overwrite a prompt,
    - Click the **Browse** button to reach the location on the local disk where the respective prompts are stored in your PC.
    - Click to upload.

- *Make sure, the file to be uploaded is a wav file.*
- *The filename can be a maximum of 64 characters.*
- *The allowed characters for the filename are A to Z, a to z, 0 to 9, - and _.*

- To download a prompt, click .

The **Opening Prompt Name_xx.wav** window will open; where xx signifies the prompt number.

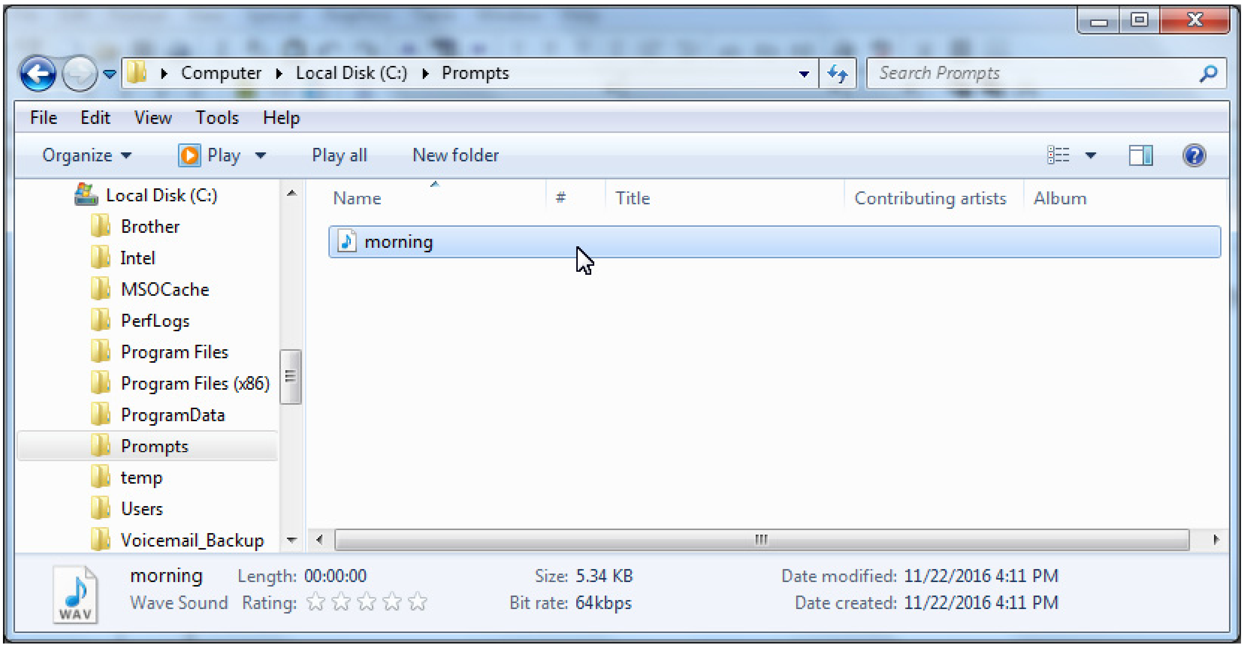


- You can either open the wav file or save the file to a location.

- *The above window display depends upon the browser you are using. Check the* ***Download Settings*** *of your browser and set the Download path accordingly.*
  OR
  If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

- *If you are using Mozilla Firefox (version 3.5.1 recommended), before you save the configuration files, set the* ***Downloads*** *option of your browser as* ***Always ask me where to save files****.*

- Save the file on the local disk.



- To delete a prompt, click .

For Firmware Versions earlier than V2.3,

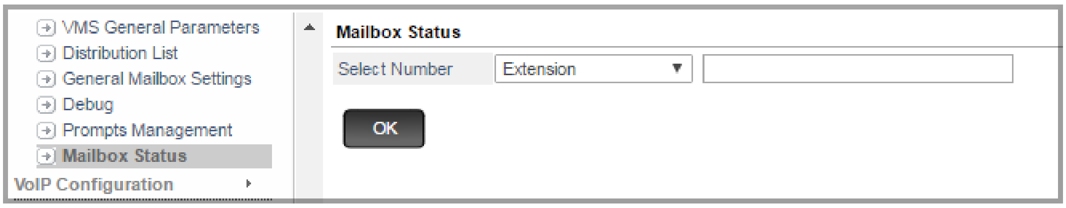
- *If you wish to upload customized prompts in the Miscellaneous folder, follow the same instructions as mentioned above to download the folder and copy the customized prompts in the desired folders.*
- *You also need to make sure that the Folder number 01 has only one file you need, delete the other file. Only then you will be able to upload the folder again with customized prompts.*

Mailbox Status

The Mailbox Status displays the mailbox details of the respective Extension, Department Group or General Mailbox.

Viewing Mailbox Status

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click **Mailbox Status**.



OR

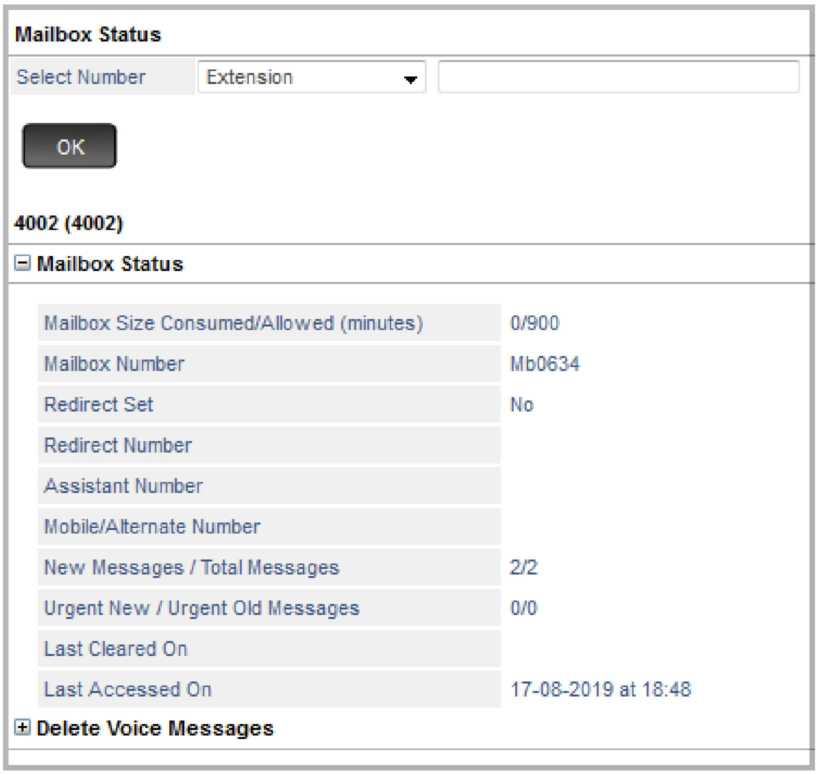
- Login as System Administrator.
- Under **Voice mail**, click **Mailbox Status**.

- **Select Number:** You may select — Extension, Department Group or General Mailbox — whose mailbox status you wish to view.
  - For **Extension**, enter the Extension Number/Name and then select the same from the drop down list.

- For **Department Group**, select the desired Department Group Number from the drop down list.

- Click **OK**.

- Click **Mailbox Status** to expand.



The respective mailbox parameters along with their status will be displayed.

- **Mailbox Size Consumed / Allowed (minutes)**: It displays the minutes consumed by the Mailbox messages out of the maximum minutes allowed for the Mailbox.

- **Mailbox Number**: It displays the mailbox number assigned to the respective Extension or Department Group or the General Mailbox.

- **Redirect Set**: It displays 'Yes' if the Redirect is set and 'No' if the Redirect is not set for the respective mailbox user.

- **Redirect Number**: It displays the Redirect number, if configured, for the mailbox user.

- **Assistant Number**: It displays the Assistant number, if configured, for the mailbox user.

- **Mobile/Alternate Number**: It displays the Mobile/Alternate number, if configured, for the mailbox user.

- **New Messages / Total Messages**: It displays the number of New Messages out of the Total Messages available in the respective mailbox.

- **Urgent New / Urgent Old Messages**: It displays the number of New Urgent Messages as well as the Old Urgent Messages available in the respective mailbox.

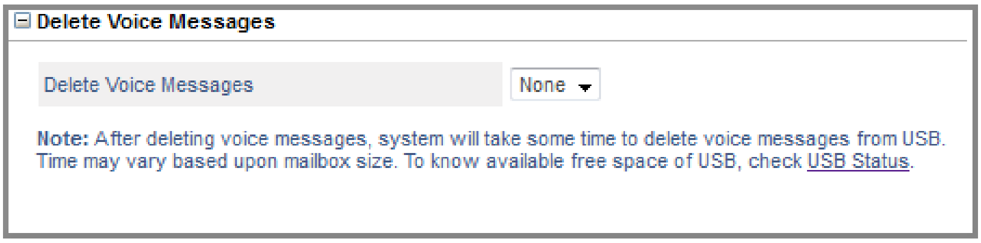
- **Last Cleared On**: It displays the date and time when all the Mailbox Messages (old and new) were last cleared by the user or SE.

- **Last Accessed On**: It displays the date and time when the Mailbox was last accessed.

Delete Voice Messages

***Delete Voice Messages** will be displayed only when the Mailbox assigned to Extension, Department Group or General Mailbox contains old or new voice messages.*

- Click **Delete Voice Messages** to expand.



- From the **Delete Voice Messages** list, select the type of messages you want to delete — None, All, New or Old. By default, None is selected.

- Click **Delete**.

  A confirmation message is displayed. Click **OK** to delete the messages.

- ***Delete voice messages** is not applicable for Personal Greetings, Conditional Greetings and Extension Name.*

- *After deleting voice messages, system will take some time to clear them. The time taken to clear the messages will vary depending upon the mailbox size.The USB Status will be updated only after these messages have been cleared.*

- *You can also view the Mailbox Status from the SE Mode. To view, click the **Mailbox Status** under the **Status** link and follow the steps described above.*

Voicemail Backup

The Voicemail Backup allows you to store the Backup of the desired — Extensions, Department Groups, General Mailbox— voicemail messages on the Network Drive. For details regarding the Network Drive configuration, see “Network Drive Settings”.

ANANT UCS allows you to take the voicemail backup of a single extension, range of extensions, all extensions or selected extensions.

You can take Voicemail Backup, either

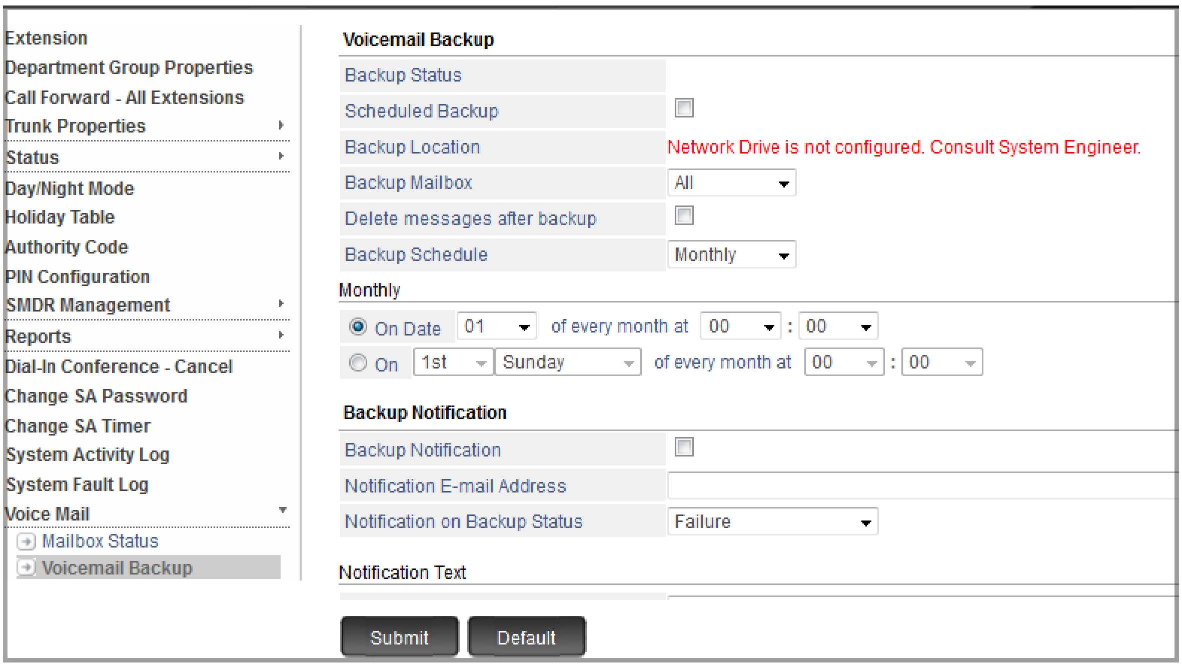
- Manually: The backup is taken whenever desired.
  OR
- Scheduled: The backup is taken at a particular Day & Time.

- *If you want to store the backup files in a PC having Windows as the Operating System, make sure it has IPV4 address.*
- *Backup is not possible in Apple PCs.*

How to Configure

Scheduled Backup

- Login as System Administrator.
- Under **Voice Mail**, click **Voicemail Backup.**



To schedule automatic Voicemail Backup configure the following:

- **Backup Status:** Displays the last scheduled/manual backup status (Successful or Failed) with date, time, type of backup done and location at which backup is stored. By default, this field is blank. When the Voicemail Backup is on-going, this displays the progress status of the backup. However, if you wish to abort the backup midway, you may click the **Abort** button.

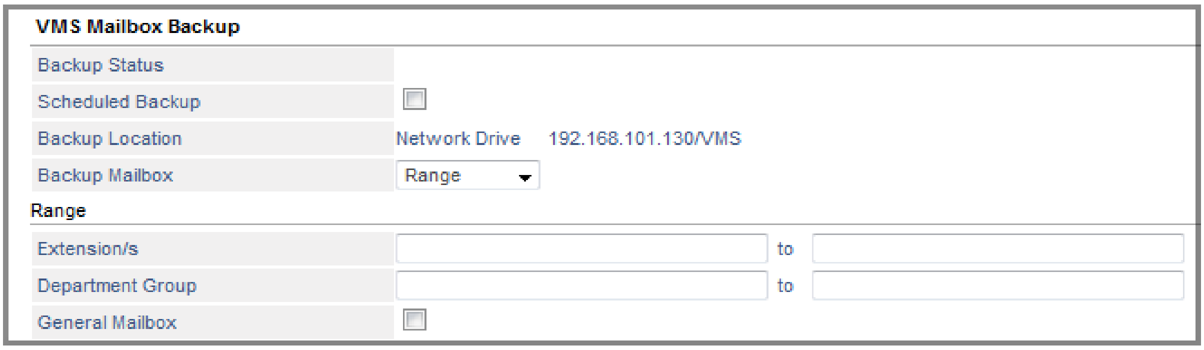
__Abort__ button will be visible only when the Voicemail Backup is in progress.

- **Scheduled Backup:** Allows you to schedule automatic Voicemail Backup on a specific day, date and time. To set a scheduled backup, enable this check box. By default, it is disabled.

- **Backup Location:** Displays the path of Network drive with the Folder Name configured in the Network Drive Settings through SE mode. If the parameters have not been configured for the Network Drive, it will display the error message here.

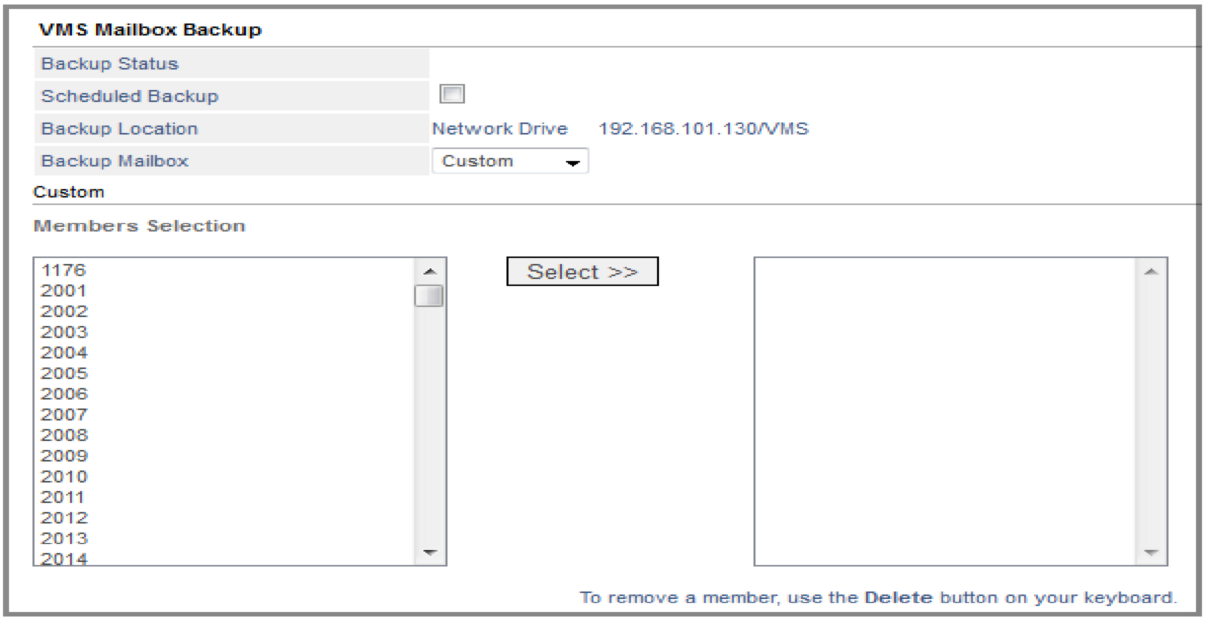
If storage capacity of the Network drive is full, then backup will not be taken and an error will be displayed.

- **Backup Mailbox:** This allows you to select the mailbox/es for which the backup needs to be taken.
    - To backup voicemail of all extensions, select **All** Extensions. This would take the backup of all mailboxes — Extensions, Department Groups and General mailbox. By default, All is selected.
    - To backup voicemail of a single extension, select **Extension**. Enter the Extension Number/Name then select the same from the drop down list.
    - To backup voicemail of multiple extensions in sequence, select **Range**.



- To backup voicemail of a range of Personal mailboxes, enter the starting and ending **Extension** Numbers and then select the same from the drop down list.
- To backup voicemail of a range of **Department Group** mailboxes, enter the starting and ending Department Group Numbers and then select the same from the drop down list.
- To backup voicemail of the **General Mailbox**, enable this check box.

- To backup voicemail of randomly selected extensions from the list, select **Custom**.



  - Select the required Extension numbers, Department Group or General mailbox from the **Member Selection** list and click **Select**. The selected mailboxes will be displayed in the right-side panel.

  - To remove a selected member, click on the desired extension in the right-side panel and press Delete key on your keyboard.

- Select the **Delete messages after backup** check box if you want the system to delete the respective extension/s voicemail messages after their backup has been stored on the network drive.

- To set an automatic **Backup Schedule**, select the desired option — Hourly, Daily, Weekly or Monthly.

  Voicemail Backup will be taken only when the System's time matches with the configured time in the **Backup Schedule.**

To receive Backup Notification, configure the following:

- You will receive **Backup Notification** only if this check box is enabled. By default, it is disabled.

- If you have opted for Backup Notification, then enter the desired email id in **Notification E-mail Address** to which the notification should be sent.

Voicemail Backup E-mail notification will be sent only when SMTP account is configured in System Log Notification. For instructions, see *"System Log Notification"* *and* *"SMTP Settings"**.*

- In **Notification of Backup Status**, select the desired status for which you require notification — Failure, Success or Success + Failure. By default, Failure is selected.

- In **Notification Text: Success**, enter the text you would like to receive as subject line in email when the Voicemail Backup has been successful. By default, the text is **Voicemail Backup completed successfully on <date> at <time>.**

- In **Notification Text: Failure**, enter the text you would like to receive as subject line in email when the Voicemail Backup has failed. By default, the text is **Voicemail Backup failed on <date> at <time>.**

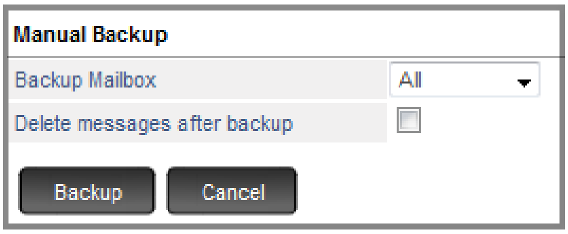
Date-Time will be replaced with the date and time of the system when manual/scheduled Voicemail Backup was taken.

Manual Backup

If the Network drive is not configured, then Manual Backup option will not be displayed.

To take the Voicemail backup manually,

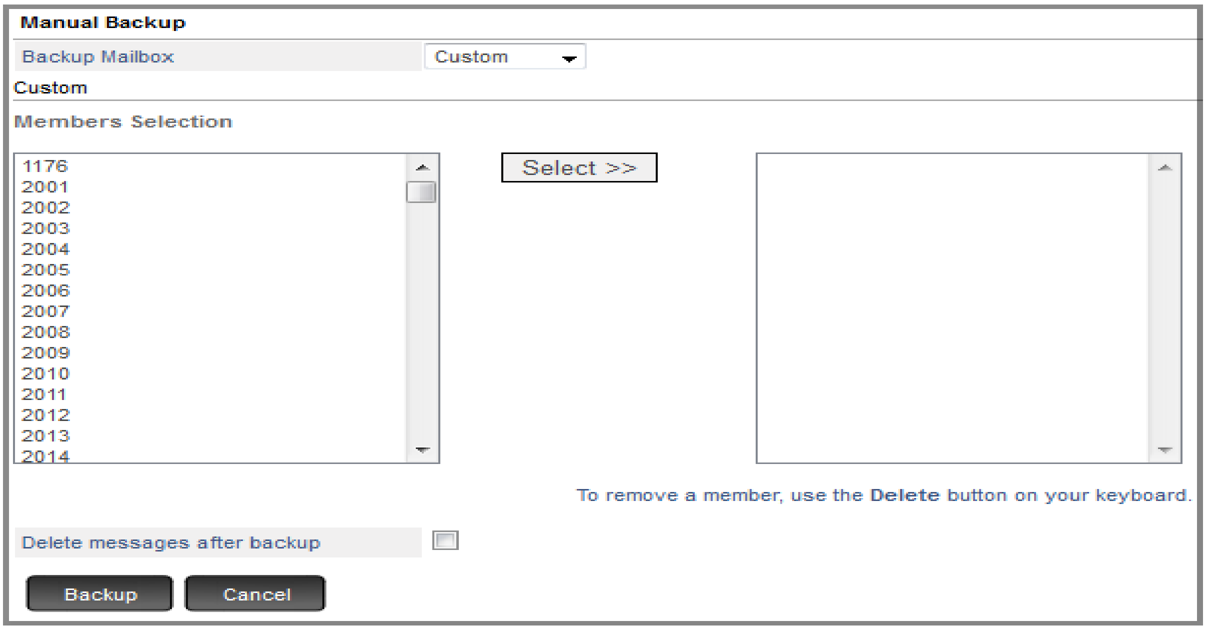
- Click on **Manual Backup** button. A new window for Manual Backup is opened.



- **Backup Mailbox** allows you to select the mailbox/es of which the backup needs to be taken.
  - To backup voicemail of all extensions, select **All** Extensions. This would take the backup of all mailboxes — Extensions, Department Groups and General mailbox. By default, All option is selected.
  - To backup voicemail of a single extension, select **Extension**. Enter the Extension Number/Name and then select the same from the drop down list.
  - To backup voicemail of multiple extensions in sequence, select **Range**.

    - To backup voicemail of a range of Personal mailboxes, enter the starting and ending **Extension** Numbers and then select the same from the drop down list.
    - To backup voicemail of a range of **Department Group** mailboxes, enter the starting and ending Department Group Numbers and then select the same from the drop down list.
    - To backup voicemail of the **General Mailbox**, enable this check box.

- To backup voicemail of randomly selected extensions from the list, select **Custom**.



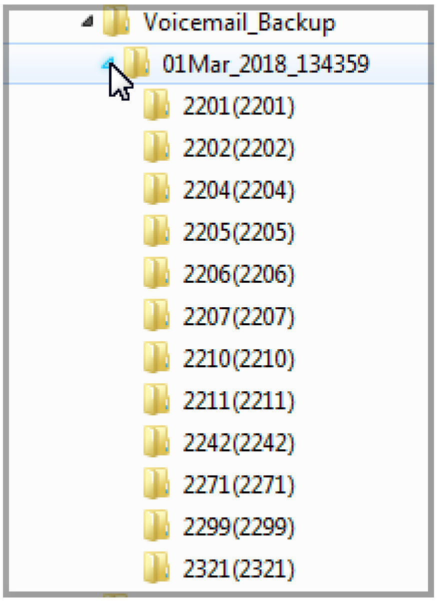
  - Select the required Extension numbers, Department Group or General mailbox from the **Member Selection** list and click **Select**. The selected mailboxes will be displayed in the right-side panel.
  - To remove a selected member, click on the desired extension in the right-side panel and press Delete key on your keyboard.
- Select the **Delete messages after backup** check box if you want the system to delete the respective extension/s voicemail messages after their backup has been stored on the network drive.

*If you click **Backup** and a Scheduled Backup is in progress, then Manual Backup will not be taken and an error is displayed.*

Folder Structure

The Voicemail Backup of the respective Extension/s will be stored in the *Shared Folder* you configure in Network Drive Settings.

The folder structure will be as shown below:



The **Voicemail_Backup** folder consists of the folders with the folder names defined as per the Date and Time of the backup.

The Folder Names will be as per the Date and Time Format you configure in *VMS General Parameters*.

Supported formats for folder names are DDMonthname_YYYY_HHMMSS and DDMonthname_YYYY_HHMMSS_PM/AM.

Where,
DDMonthname_YYYY represents the Date, *HHMMSS* represents the Time in 24 Hour format and *HHMMSS_PM/AM* represents the Time in 12 Hour format.

The sub-folders with the Extension Names contain the respective extension’s voicemail files.

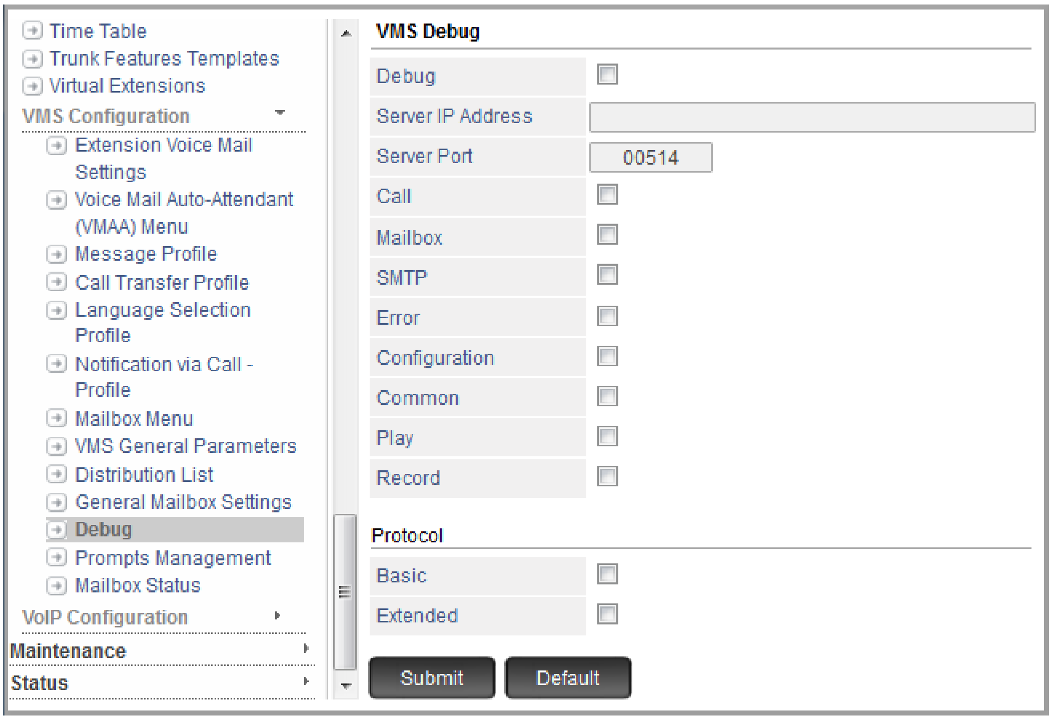
VMS Debug

The VMS supports debug for the VMS Application and SMTP. You can view debug messages on the Syslog Server[63].

To be able to use Syslog for debug, you will need to configure the Syslog Server Address and the Server Port on which Syslog will listen for the debug messages.

Configuring VMS Debug

- Login as System Engineer.
- Under **Configuration**, click **VMS Configuration**.
- Click the **Debug**.



- Select the **Enable Debug** check box if you wish to view the VMS Debug
  - In **Server IP Address**, configure the IP Address of the Syslog Server.
  - In **Server Port**, configure the address of the Listening Port of the Syslog Server. Valid port range is: 1025 to 65535; 514. By default, the remote server port address is '514'.

If you disable the check box, the system will not send the VMS Debug to the Syslog Server.
The system supports following VMS Debug Levels. You may select the respective checkbox of the debug levels you wish to include in the VMS Debug.

63. *ANANT UCS supports Syslog Client, which enables the VMS to send debug messages in syslog format to the remote 'Syslog Server' on the IP network. You can view the system debug messages on the remote Syslog server or any other application which can capture the Syslog debug messages.*

- Select the **Call** check box to view the debug for all types of call flow.
- Select the **Mailbox** check box to view the debug for all operations related to Mailbox.
- Select the **SMTP** check box to view the debug related to the SMTP Client handled by VMS.
- Select the **Error** check box to view the debug for all types of errors.
- Select the **Configuration** check box to view the debug for configuration update.
- Select the **Common** check box to view the debug.
- Select the **Play** check box to view the debug related to play from VMS application as well as layer side.
- Select the **Record** check box to view the debug related to the records from VMS application as well as layer side.

Protocol

- Select the **Basic** check box to view the debug for data received with its IE.
- Select the **Extended** check box to view the debug for data received in raw hex format.
- Click **Submit**.

*You can also configure the VMS Debug from the **Maintenance** link. To do so, click VMS under Debug.*

CHAPTER 8 Features and Facilities

Abbreviated Dialing

Abbreviated Dialing is the use of short codes (abbreviated numbers), to dial out long-digit numbers. It is also referred to as Memory Dialing.

Abbreviated Dialing allows you to dial quickly and easily, frequently called, long-digit numbers.

This feature requires you to store the frequently called, long-digit numbers[64] and their corresponding short codes in special lists, known as 'directories'. These directories may be 'personal' or 'global'.

ANANT UCS supports two types of Abbreviated Dialing based on the type of directory used: Personal Abbreviated Dialing and Global Abbreviated Dialing.

Abbreviated Dialing forms the basis of two other features of the ANANT UCS: "Dialed Number Directory" and "Quick Dial".

Personal Abbreviated Dialing

This variation of Abbreviated Dialing makes use of the Personal Directory.

Personal Directories can be programmed and assigned to groups of extensions. The use of Personal Directories is limited to the extensions to which they are assigned.

A personal directory accommodates 25 numbers. Each number may be up to 16 digits long. A personal directory has Index numbers from 0001 to 0025 against which the frequently dialed telephone numbers are stored along with their corresponding names, email id and trunk access code ID.

As many as 50 different personal directories, numbered from 01 to 50 can be created and assigned to extensions.

With a personal directory assigned to an extension, the extension user simply dials out the Feature Access Code for Abbreviated Dialing and the Index Number at which the desired number is stored in the personal directory.

For example: personal directory number 02 is assigned to extension 2001. The number 02652630555 is stored at Index number 16 of this directory. The extension user 2001 can call this number by simply dialing '8' (feature access code) followed by '0016' (the index number).

The system will automatically dial out the number using the trunk access code ID specified for this number in the personal directory.

64. These may be numbers of your branch offices, your clients, as also numbers of emergency services such as fire, police.

- *When an extension user dials an abbreviated number from the Personal Directory, the system first checks OG Trunk Bundle Group (OGTBG) and Toll Control Level (Call Privilege) of that extension and then dials out the number.*
- *Each extension can access only the personal directory assigned to it.*
- *Personal Directory can be programmed by the System Engineer, as well as extension users. Extension users can add contacts to the Personal Directory assigned to them from their IP Phones.*

Global Abbreviated Dialing

This variation of Abbreviated Dialing makes use of a system-wide list of numbers stored in the memory of ANANT UCS, known as the Global Directory.

Being a system-wide list, the Global Directory can be accessed by any extension connected to the system.

The Global Directory has the capacity to store up to 2999 numbers of a maximum of 16 digits each. The Global Directory is divided into three parts:

- Part 1 - contains Memory Location codes 0100 to 2399.
- Part 2 - contains Memory Location codes 2400 to 2699.
- Part 3 - contains Memory Location codes 2700 to 2999.

The Global Directory has Memory Location codes starting from 0100 to 2999. The telephone numbers along with their corresponding names are stored against Memory Location codes.

Whenever extension users of the system want to use Global Abbreviated Dialing, all they needs to do is dial the feature access code ('8' or '6') and the Memory Location code at which the desired number is stored.

For example: the number 02652630566 is stored at Memory Location 0102 of the Global Directory. Now, extension users of the system can call this number by simply dialing the '8' or '6' (feature access code for Abbreviated Dialing) followed by '0102' (Memory Location code at which the desired number, 02652630566, is stored).

The system will dial out the number using any of the trunks in the "OG Trunk Bundle Group" assigned to it in the Memory Location.

Extensions can use Global Abbreviated Dialing only if this feature is included in the "Class of Service (CoS)" allowed to them.

- *Further, an extension can access only that part of the Global Directory which is allowed to it in the CoS. For instance, if extension 2001 is allowed Global Directory Part 1 in its CoS, the user of extension 2001 can dial out only those numbers contained in Global Directory Part 1.*
- *So, to be able to access the entire Global Directory, extensions must be assigned all three parts of the directory in their Class of Service. By default, only Global Directory Part 1, 2 and 3 are not included in the CoS of all extensions.*
- *Global Directory can be programmed by the System Engineer, and by SIP extension users who have Global Directory Programming allowed to them in the Class of Service.*
- *While the System Engineer can program all three parts of the Global Directory, SIP extension users who are allowed Global Directory Programming in their Class of Service can configure only Global Directory Part 1.*

How to configure

For both Personal and Global Abbreviated Dialing to work, the System Engineer must:

- Configure the Personal Directories and the Global Directory.
- Assign Personal Directory to the desired extensions.
- Enable Global Directory Part 1/2/3 as desired in the Class of Service (CoS) group allowed to the extensions.
- Enable 'Global Directory Part 1' and 'Global Directory Programming' in the CoS group of the extension users, who are to be allowed to program (add, delete, edit) contacts in the Global Directory Part 1 from their IP Phones.

All the above parameters can be configured by the System Engineer using Jeeves.

Preparing Numbers Lists for Personal and Global Directories

In consultation with the extension users, you may:

- Find out the number of personal directories that need to be configured.
- Make a list of numbers frequently dialed by the extension.
- Ask the extension users the numbers they would like to be included in the personal directory of their extension.
- Make separate lists of numbers along with their corresponding names, email id, trunk access codes, for each personal directory. You may draw five-column tables on paper and enter the Numbers and corresponding names, email ids and trunk access codes against each Index number. For example:

Personal Directory 01

Index No.	Number	Name	Email ID (optional)	TAC ID
01				
02				
:	:	:		:
25				

Personal Directory 02

Index No.	Number	Name	Email ID	TAC ID (optional)
01				
02				
:	:	:		:
25				

- Compile the numbers to be included in the global directory. Numbers that are commonly dialed by all extensions can be included in the global directory.
- Draw a five-column table on paper and enter the numbers along with their names, email id, the Outgoing Trunk Bundle Group (OGTBG) and alternate number group at each Memory location. For example:

Global Directory

Memory Location	Number	Name	Email ID (optional)	OGTBG	Group (optional)
0100					
0101					
:	:	:	:		
2999					

- Prepare the Global Directory keeping in mind that it is divided into three parts: Part 1 (0100 to 2399), Part 2 (2400 to 2699), and Part 3 (2700 to 2999).

Uploading Personal and Global Directory Contacts

If you have personal and global directory contacts database in an excel sheet, you can convert the same into a CSV file and upload it through Jeeves.

For Personal directory contacts, the format of the CSV file must be as follow: For example 21, 9867985489, Sean Gilbert, Sean@hotmail.com, 0

Where,
21 is the Index Number at which you want the entry to be stored (mandatory)
9867985489 is the Contact Number (mandatory)
Sean Gilbert is the Contact Name (optional)
Sean@hotmail.com is the Email ID (optional)
0 is the Trunk Access Code (TAC) to route the call (optional)

To upload Personal Contacts CSV file using Jeeves, see "Upload Personal Directory CSV files"

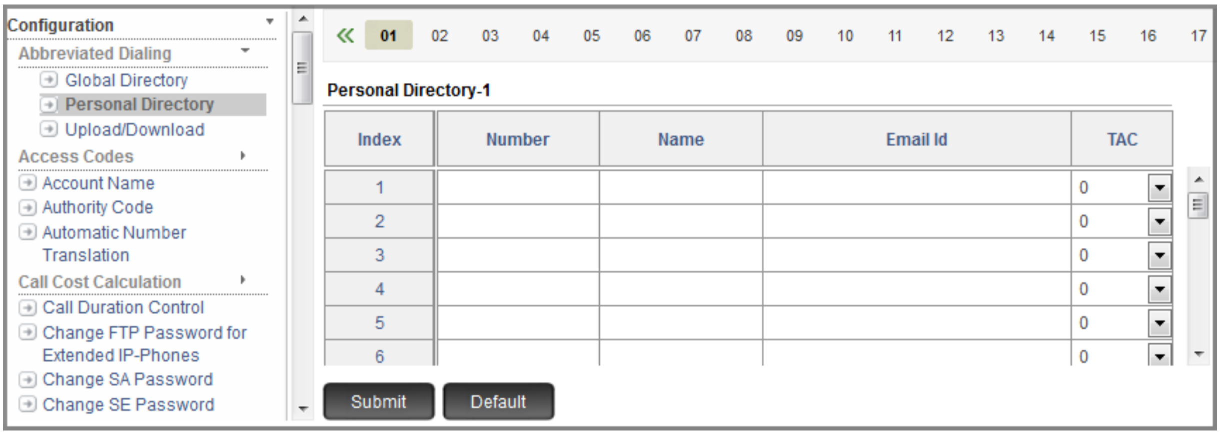
For Global directory contacts, the format of the CSV file must be as follow: For example 0121, 9867985489, Sean Gilbert, Sean@hotmail.com, 01, 000

Where,
0121 is the Index Number at which you want the entry to be stored (mandatory)
9867985489 is the Contact Number (mandatory)
Sean Gilbert is the Contact Name (optional)
Sean@hormail.com is the Email ID (optional)
01 is the Outgoing Trunk Bundle Group (OTBG) to route the call (optional)
000 is the Alternate Number Group (optional)

To upload Global Contacts CSV file using Jeeves, see "Upload Global Directory CSV file".

Configuring Personal Directory

- Login as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Personal Directory.**



You can configure upto 50 Personal Directories. Select the directory number by clicking the required number tab above the table.

For each directory configure the following parameters:

- Enter the **Number** you wish to store against an Index Number. Enter the contact's **Name** and **Email Id** against the number.

  The length of the Number field is limited to 16 digits. The length of the Name field is limited to 18 alphanumeric characters. All ASCII characters except <, >, :, ", /, \, |, ?, * are allowed.

*The **Name** can be configured upto 18 characters using Jeeves and CSV file only.*

From the Extended Phones — SPARSH VP210 and SPARSH VP510, you will be able to view 18 characters Name configured from the Server as well as you will be able to add or edit Name of contact to a maximum of 18 characters.

From all other Extended Phones, you will be able to view 18 characters Name configured from the Server. But from these Phones you will be able to add or edit Name of contact to a maximum of 12 characters only.

  The Email ID can be a maximum of 64 characters.Ensure that the number, name and email id are programmed within this limit.

  Each directory has a limit of 25 entries. You may enter upto 25 Numbers and Names in each Personal Directory.

  Change the **TAC** Index (TAC ID), if required.

- Click **Submit**.
- Repeat the above steps to program each Personal Directory.

Keep a print of each personal directory for your record and for the record of the extension user to whose phone the personal directory is assigned. This will also help you take care of overlaps and include some of the numbers that are dialed by all users in the Global Directory instead of the Personal Directory.

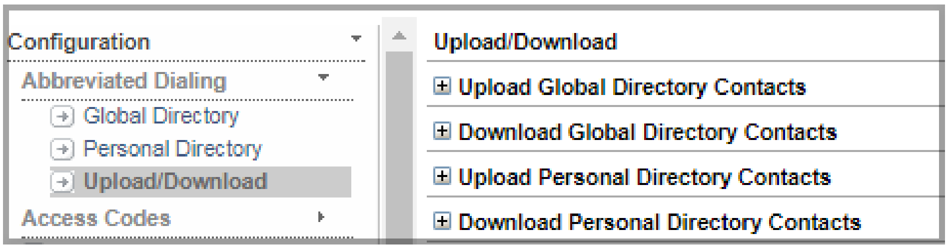
Assigning Personal Directories to Extensions

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings.**
- Select the software port number of the SIP extension you want to assign the Personal Directory.
- Click the **Advance** button.
- Scroll to Personal Directory, and select the **Personal Directory** number you want to assign to this extension.
- Click **Submit.**

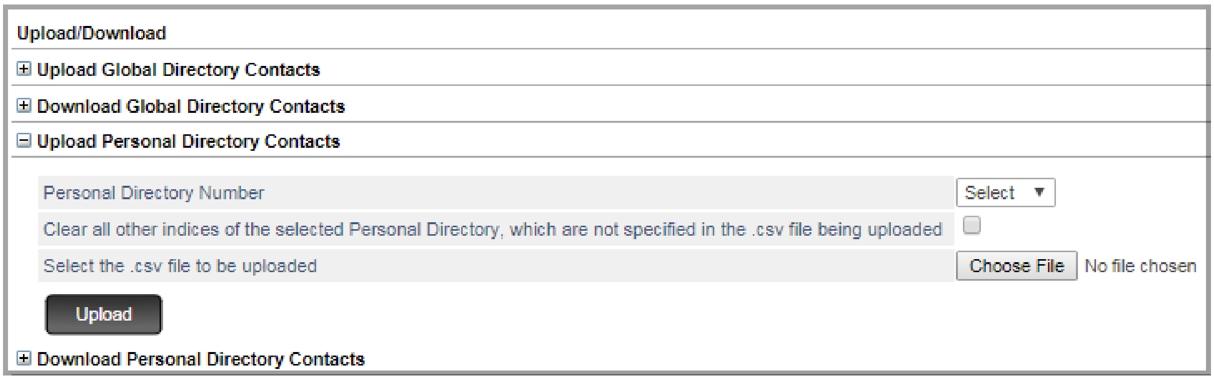
It is possible to assign the same personal directory to multiple extensions.

Upload Personal Directory CSV files

- Login as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download.**



- Click **Upload Personal Directory Contacts** to expand.



- In **Personal Directory Number**, select the number of the personal directory in which you want to upload the contacts from the CSV file.

- Select the **Clear all other indices of the selected Personal Directory, which are not specified in the .csv file being uploaded** check box, to overwrite the existing contacts in the Personal directory with the contacts of CSV file. Default: Disabled.

- Click the **Browse** button to **Select the .csv file to be uploaded** from the location on the local disk.

- Click the **Upload** button.

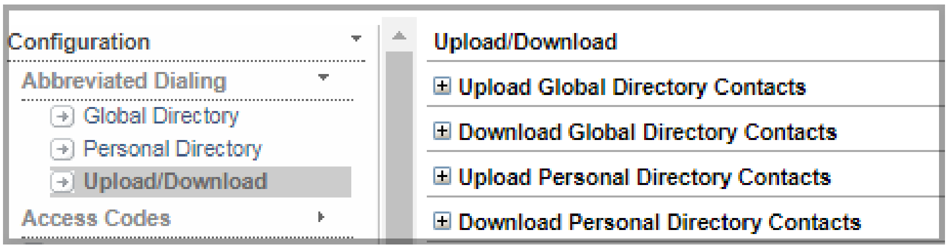
- All the contacts of the CSV file will be uploaded in the selected Personal Directory. To view, click the respective Personal Directory link.

Download Personal Directory CSV files

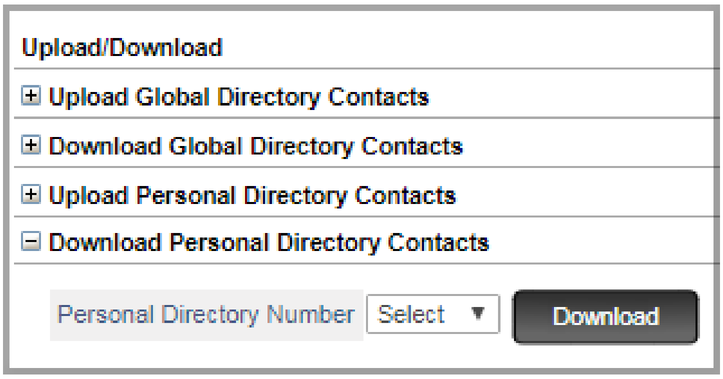
- Login as System Engineer.

- Under **Configuration**, click **Abbreviated Dialing**.

- Click **Upload/Download**.

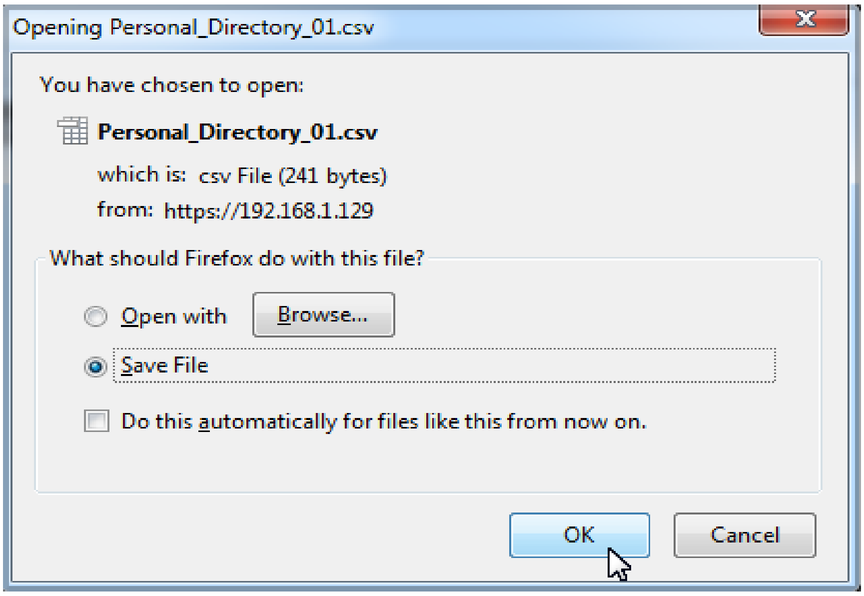


- Click **Download Personal Directory Contacts** to expand.



- In **Personal Directory Number**, select the number of the personal directory from which you want to download the contacts.

- Click the **Download** button.



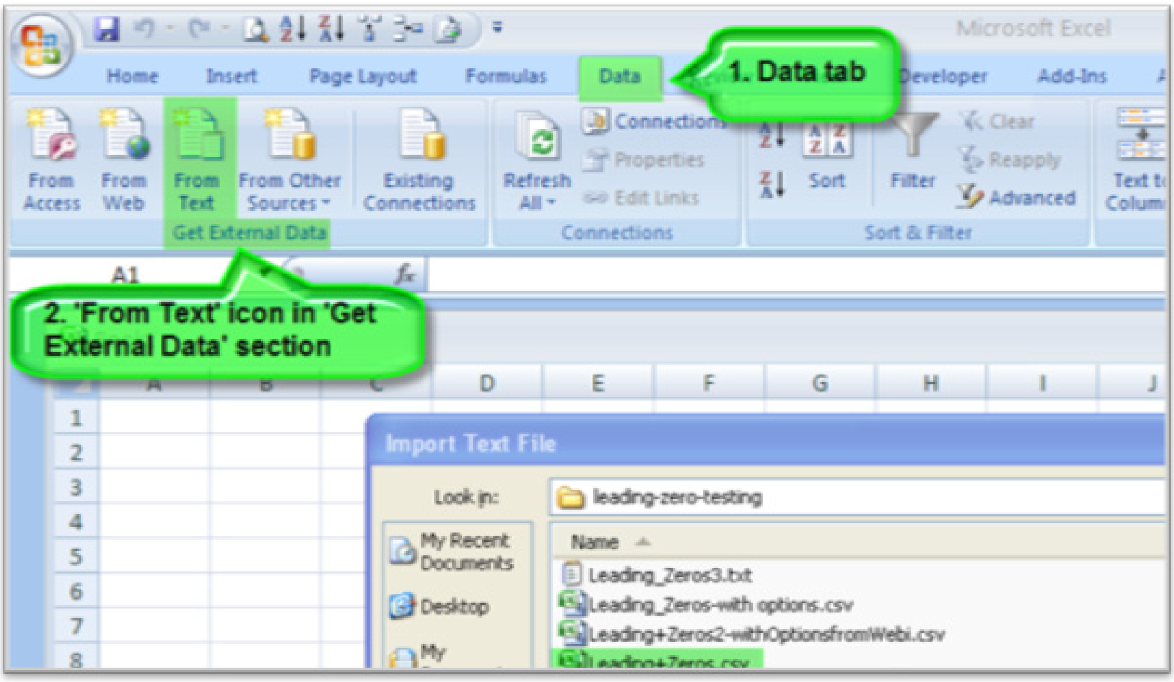
- You will get a prompt with an option to open the **Opening Personal_Directory_01.csv** file or save the file to a location. Save the file on the local disk.

- To Open the **Personal_Directory_01.csv** file from the location on the local disk, follow the steps given below:

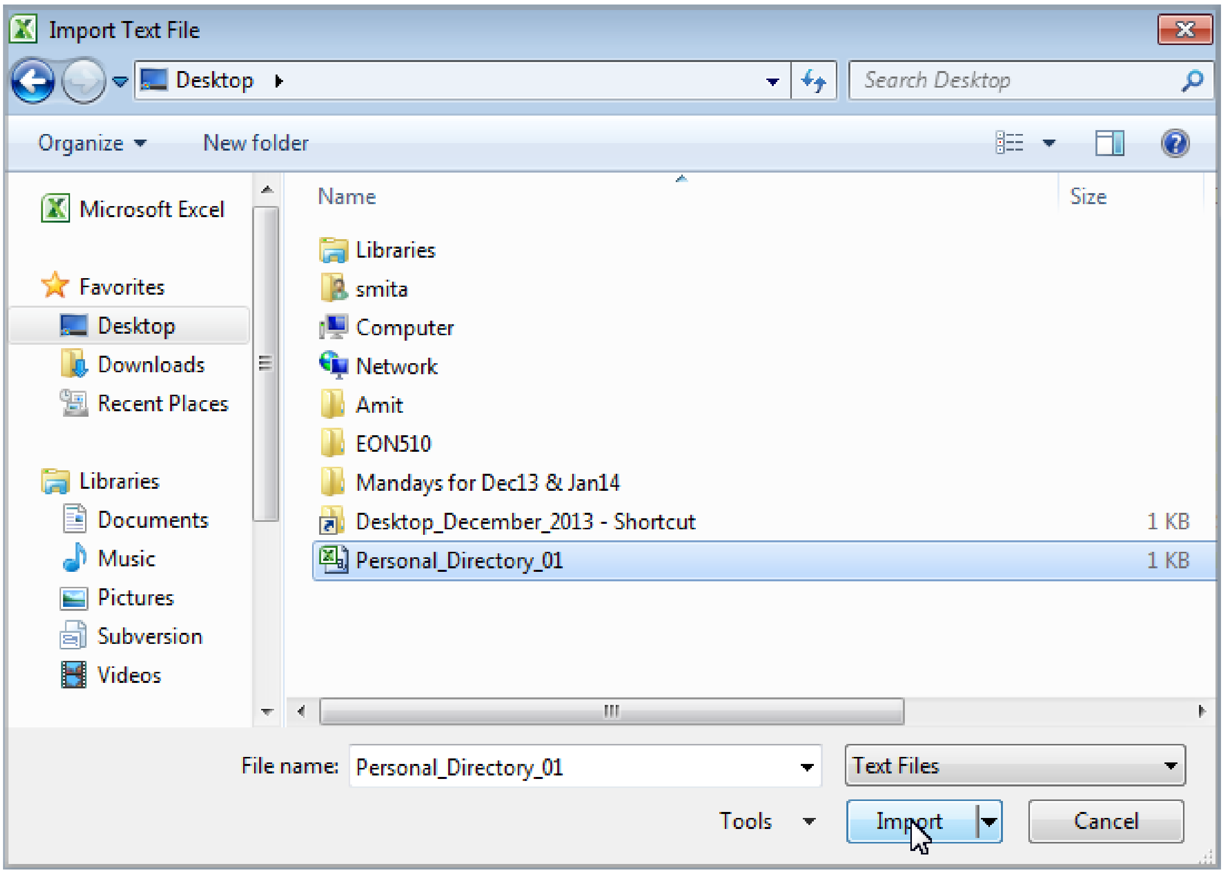
  - **DO NOT OPEN THE CSV FILE DIRECTLY WITH EXCEL**!

  - Open a **New** worksheet in Excel.

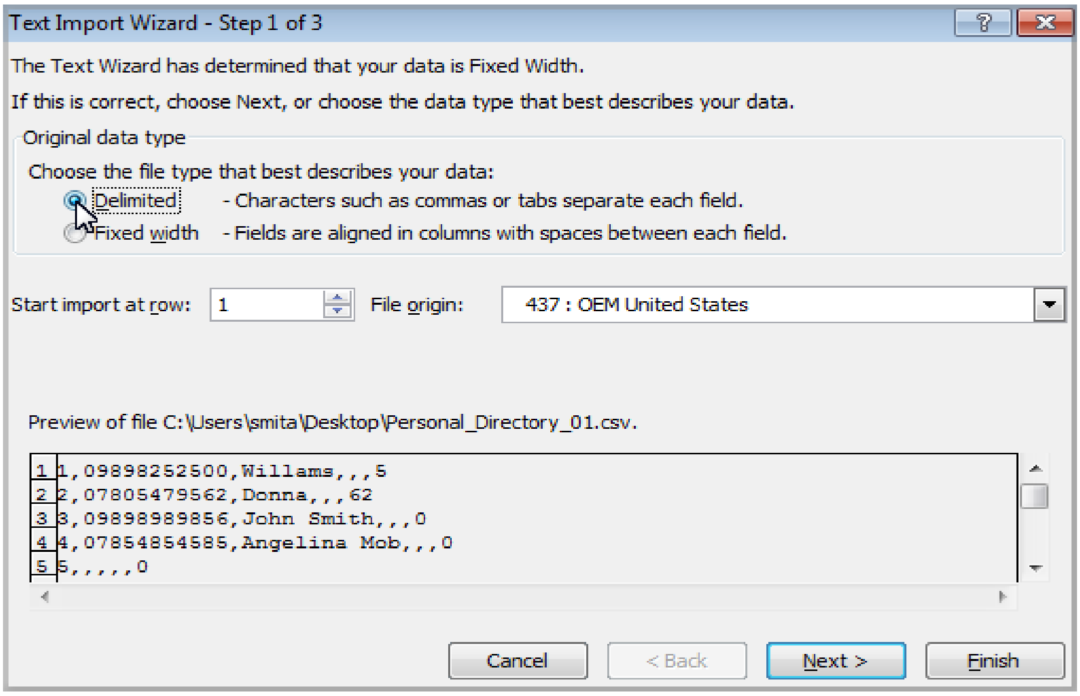
- Open the **Data** tab and select **From text button in the Get External Data**.



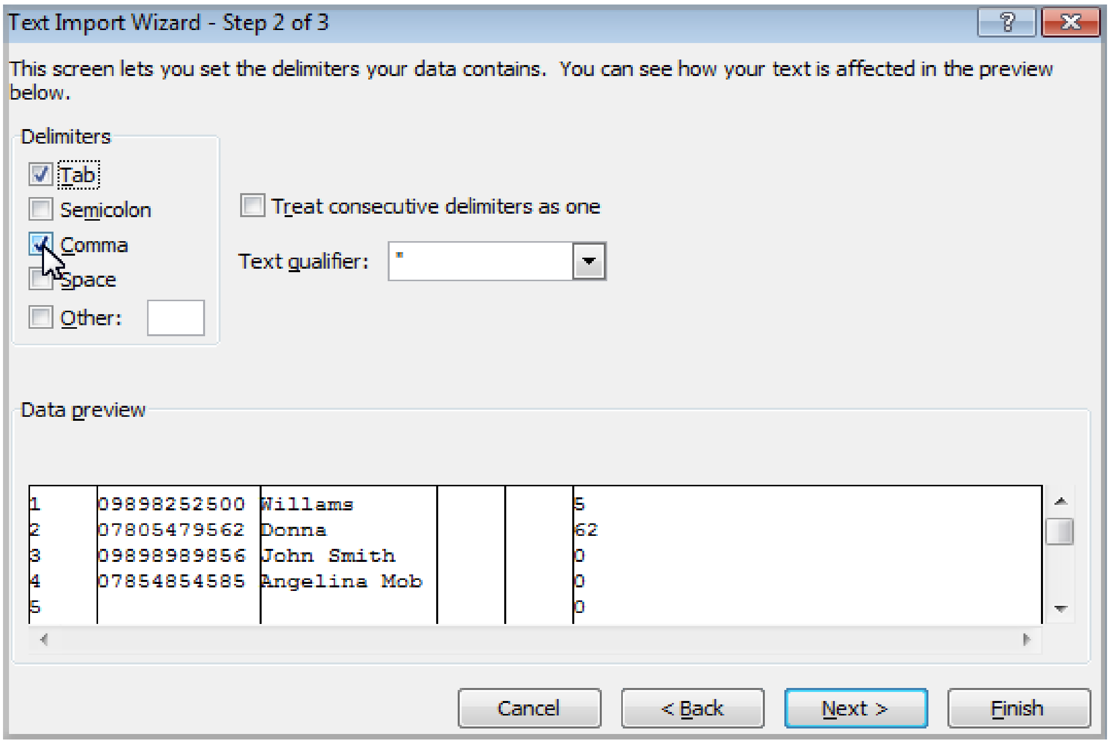
- Select your CSV file from the location on the local disk to import.



- In **Original data type** section, select **Delimited** radio button and click **Next**.

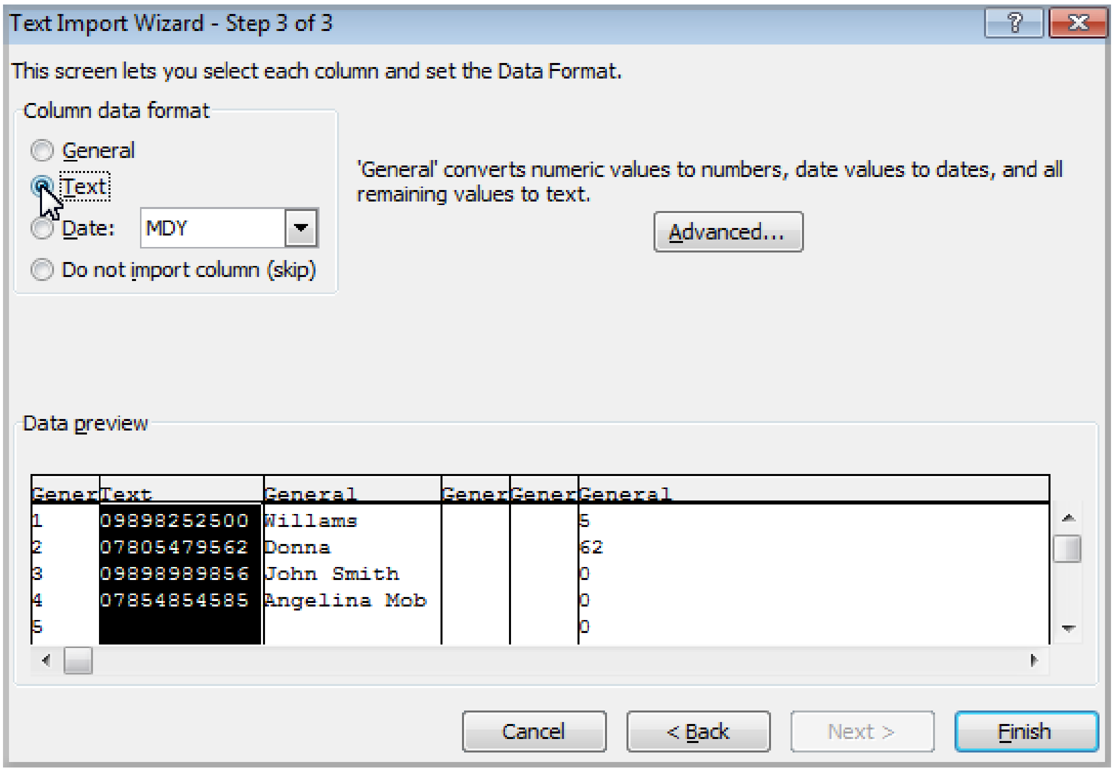


- In **Delimiters** select the **Comma** check box (column dividers will appear in preview) and click **Next**.



- Select the column with leading zeros and in **Column data format** select the **Text** radio button.

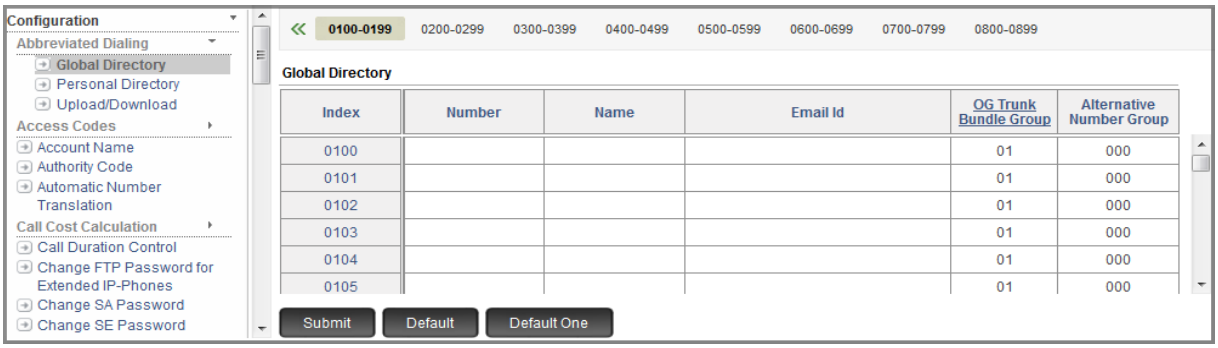
You will have to do this for each column where the data contains leading zeros.



- Click **Finish**.
- The leading zeros will still be there in the new worksheet with the imported data.

Configuring Global Directory

- Login as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Global Directory.**



Each page has 100 entries. To go to the next 100 entries click the links above the table '0200-0299'

- Enter the **Number** you wish to store against a Memory Location Code. Enter the contact's **Name** and **Email Id** against the number.

The length of the Number field is limited to 16 digits. The length of the Name field is limited to 18 alphanumeric characters. All ASCII characters except <, >, :, ", /, \, |, ?, * are allowed.

*The **Name** can be configured upto 18 characters using Jeeves and CSV file only.*

From the Extended Phones — SPARSH VP210 and SPARSH VP510, you will be able to view 18 characters Name configured from the Server as well as you will be able to add or edit Name of contact to a maximum of 18 characters.

From all other Extended Phones, you will be able to view 18 characters Name configured from the Server. But from these Phones you will be able to add or edit Name of contact to a maximum of 12 characters only.

The Email ID can be a maximum of 64 characters Ensure that the number and the name are programmed within this limit.

- Change the **OG Trunk Bundle Group,** if required. Default: 01. See “OG Trunk Bundle Group”.

*While routing the Global Directory number, the system will use the OGTBG configured in the Station Advanced Feature Template assigned to the Extension that is dialing a Global Directory number. So, if you want the system to route Global Directory numbers using OGTBG you configured here, make sure you have selected **OGTBG configured in the Global Dir.** in the **Route Global Directory Calls using** parameter of the Station Advanced Feature Template.*

- Configure the Alternate Number Group, if required.

- Click **Submit**.

Applying Global Directory to Extensions

To apply the Global Directory to Extensions,

- Make sure that the feature Global Directory is enabled in the CoS of the extensions to which you are assigning the Global Directory.

  If the entire directory is to be assigned to all the extensions, you may simply enable 'Global Directory Part 1, Part 2 and Part 3 in the default CoS group 01 in the default Station Basic Feature Template 01 assigned to the extensions.

  However, if selected extensions are to be allowed Global Directory Part 2/Part 3, follow these steps:

  - Define a CoS group with Global Directory Part 2/Part 3 enabled.
  - Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
  - Assign this template to the extensions to which Global Directory Part2 /Part 3 are to be allowed.

  Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for further instructions.

- Decide which of the Extension users are to be allowed ‘Global Directory Programming’ (of Global Directory Part 1) and allow this feature in their Class of Service.

  By default, Global Directory Programming is disabled in the default CoS group 01 in the default Station Basic Feature Template 01 assigned to all extensions of the system. This means none of the extensions can program Global Directory.

If you want to allow Global Directory Programming to all the extension users, simply enable this feature in the CoS group of the Station Basic Feature Template assigned to them.

If you want to allow Global Directory Programming to only selected extensions, then follow these steps:

- Define a CoS group with Global Directory Programming enabled.
- Make sure this CoS also has Global Directory Part 1 enabled.
- Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
- Assign this template to the extensions to which Global Directory Programming is to be allowed.

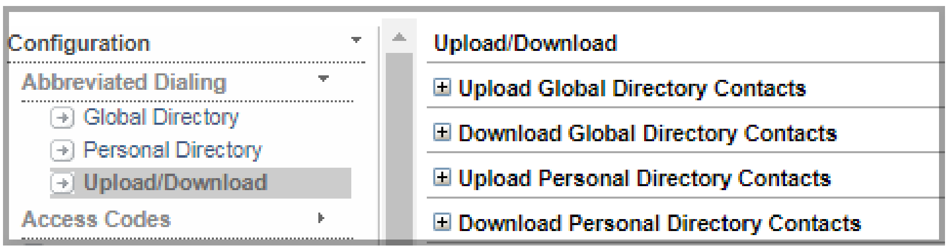
Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for programming instructions.

- *If Global Directory Part 1, 2 or 3 is assigned to an extension user, the system will not check for Toll Control.*
- *When you assign Global Directory Programming to an extension user, the user can program any number in Global Directory Part 1, this includes numbers denied to the extension user in the Call Privilege defined in the Toll Control level of this extension user.*
- *Since the system does not check for Toll Control for numbers dialed out from Global Directory Part 1, there is a possibility of extension users programming numbers not allowed to them in their Toll Control level in the Global Directory Part 1, inadvertently or intentionally.*
- *Hence, the System Engineer is advised to exercise caution when allowing this feature to the extension users.*

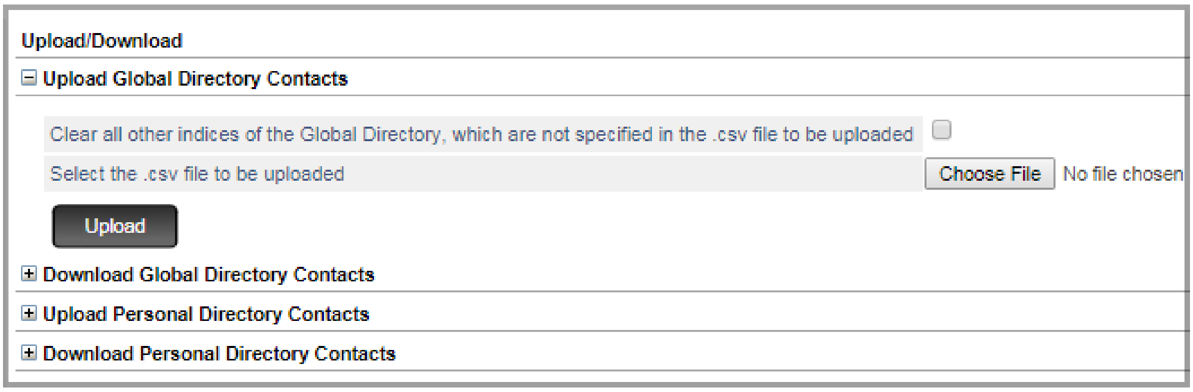
- Click **Submit.**

Upload Global Directory CSV file

- Login as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download.**



- Click **Upload Global Directory Contacts** to expand.

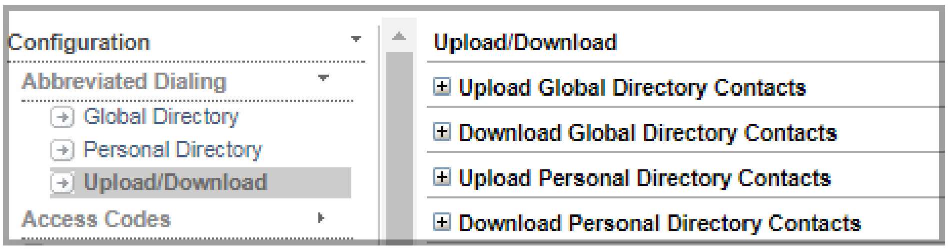


- Select the **Clear all other indices of the Global Directory, which are not specified in the .csv file being uploaded** check box, to overwrite the existing contacts in the Global directory with the contacts of CSV file. Default: Disabled.
- Click the **Browse** button to **Select the .csv file to be uploaded** from the location on the local disk.
- Click the **Upload** button.
- All the contacts of the CSV file will be uploaded in the Global Directory. To view, click the Global Directory link.

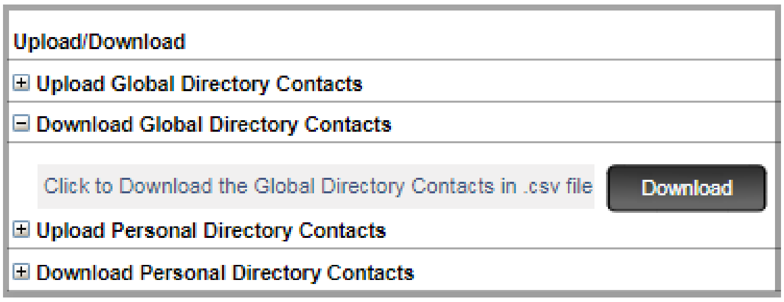
- *If all the Global Directories are selected for LDAP, then contacts will not be uploaded through .csv file.*
- *The contacts will be uploaded through .csv file only in the Global Directory that is not synchronized for LDAP.*

Download Global Directory CSV files

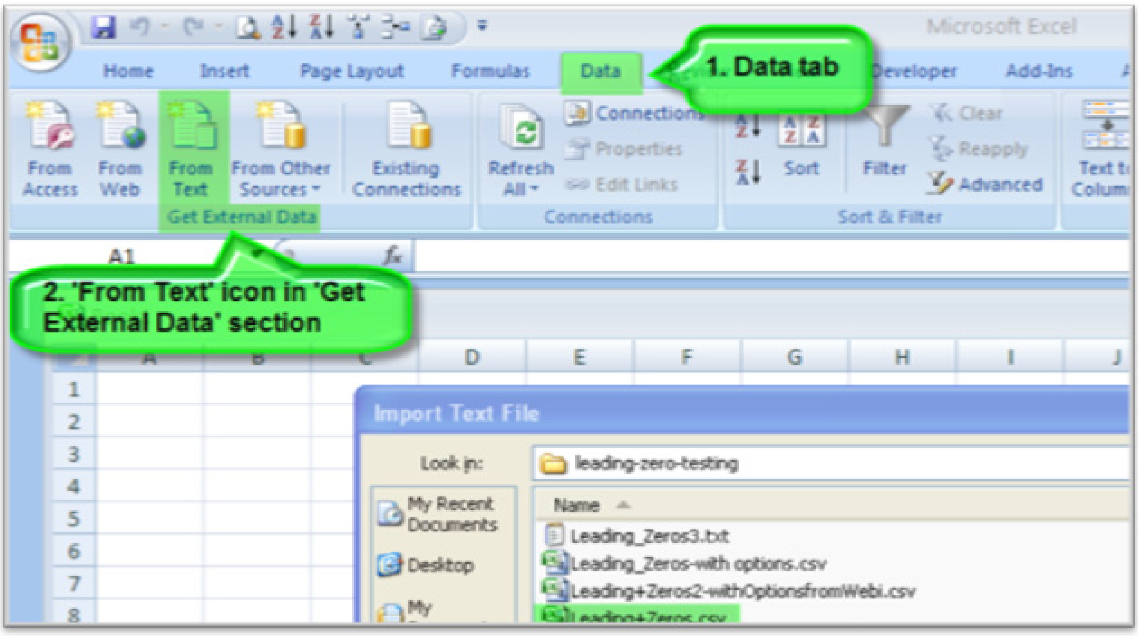
- Login as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.
- Click **Upload/Download**.



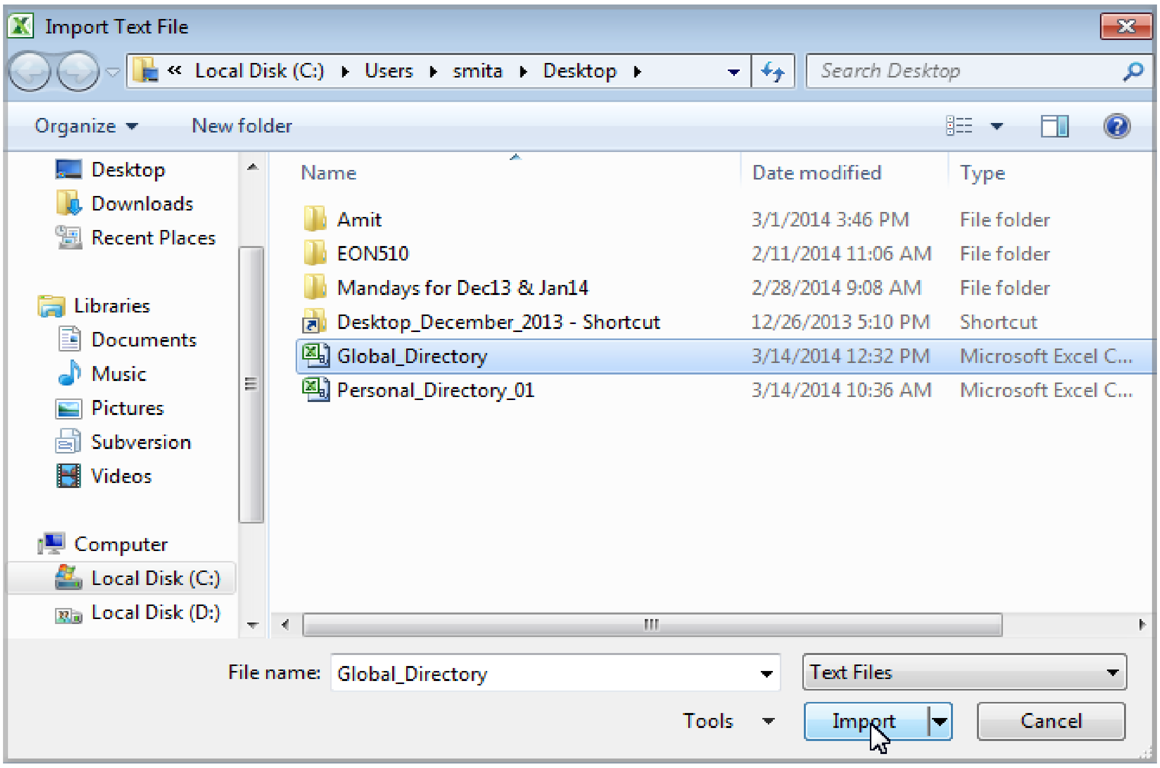
- Click **Download Global Directory Contacts** to expand.



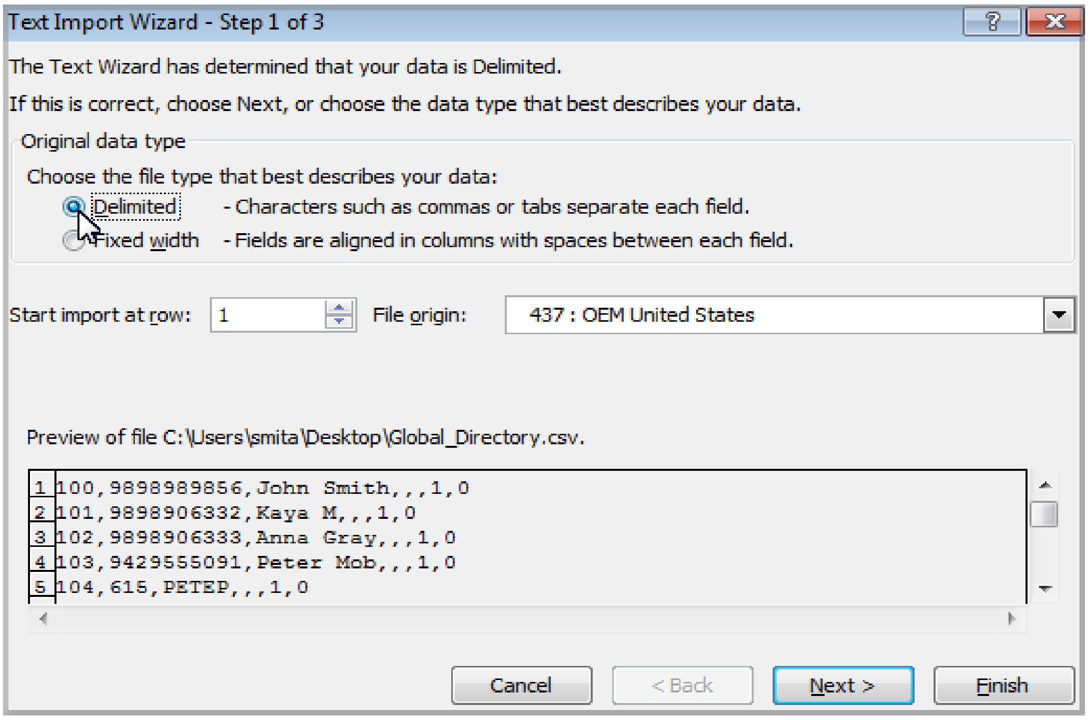
- Click the **Download** button.
- You will get a prompt with an option to open the **Opening Global_Directory.csv** file or save the file to a location. Save the file on the local disk.
- To Open the **Global_Directory.csv** file from the location on the local disk, make sure you follow the steps given below:
  - **DO NOT OPEN THE CSV FILE DIRECTLY WITH EXCEL**!
  - Open a **New** worksheet in Excel.
  - Open the **Data** tab and select **From text button in the Get External Data**.



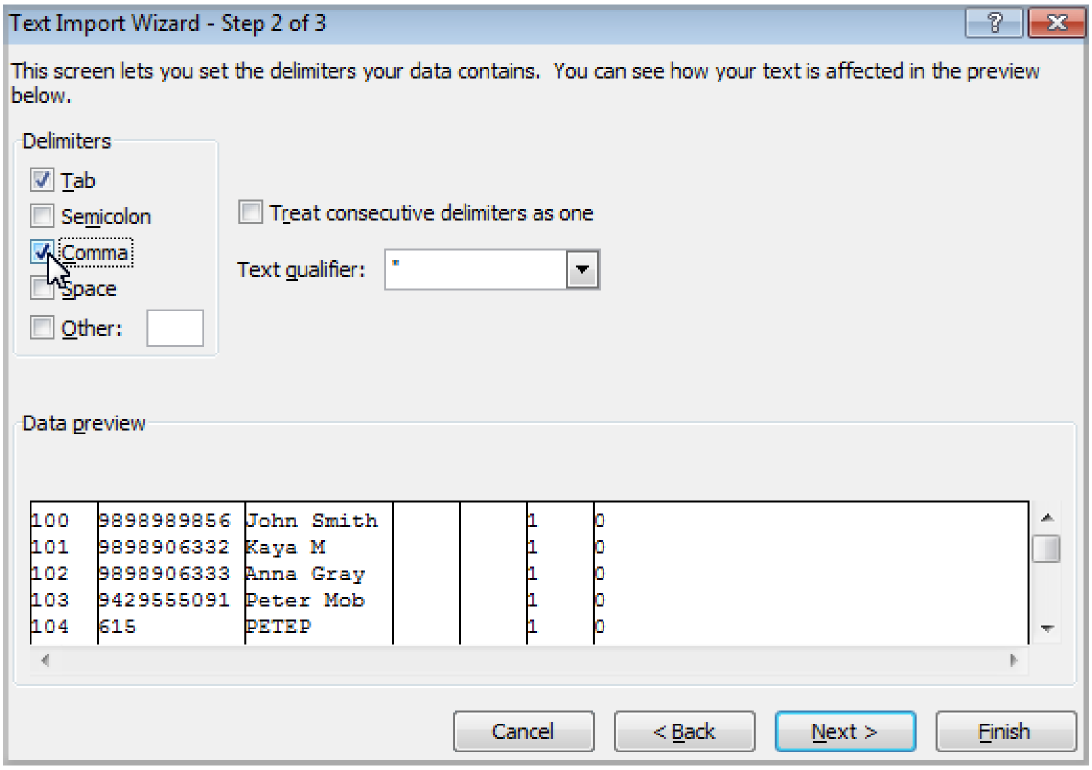
- Select your CSV file from the location on the local disk to import.



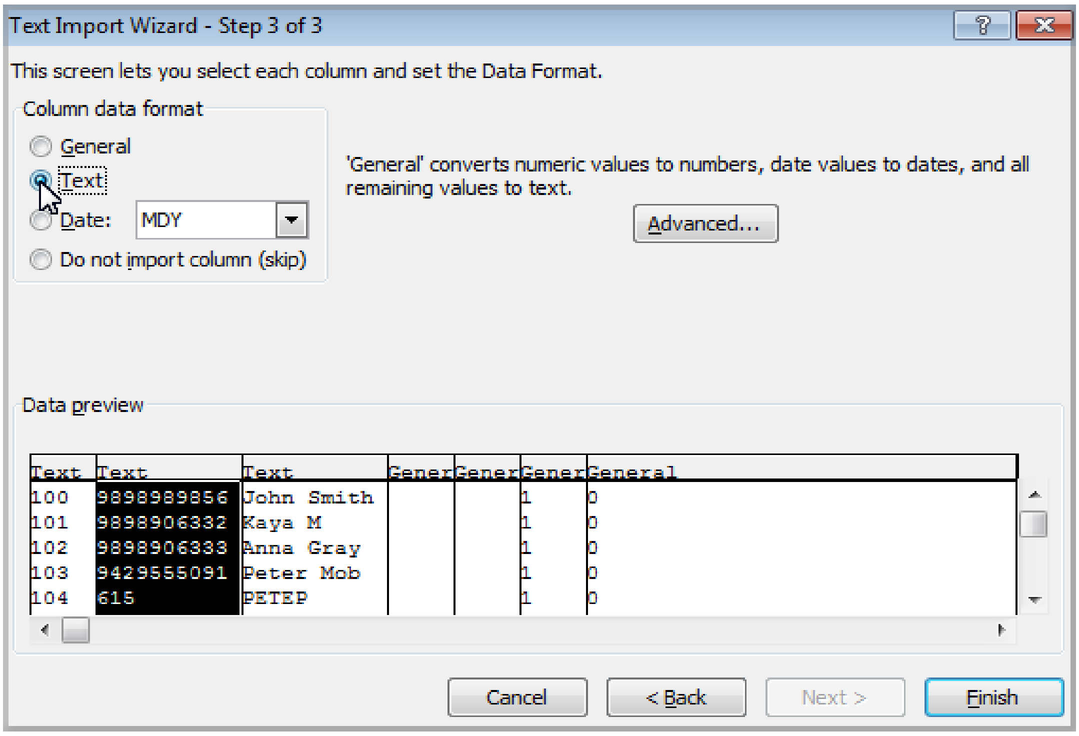
- In **Original data type** section, select **Delimited** radio button and click **Next**.



- In **Delimiters** select the **Comma** check box (column dividers will appear in preview) and click **Next**.



- Select the column with leading zeros and in **Column data format** select the **Text** radio button. You will have to do this for each column where the data contains leading zeros.



- Click **Finish**.

- The leading zeros will still be there in the new worksheet with the imported data.

Personal Directory Programming by Extension Users

Extension users can program their own Personal Directories using their extension phones.

For Extended IP Phone Users

- Press DSS Key assigned to Personal Directory (if programmed).
  OR
- Dial **1071**.
- Enter Personal Memory Index (01 to 25)
- Enter Number of the contact (max. 16 digits).
- Press 'Enter' key.
- Enter Name of the contact (max. 12 characters).
- Press 'Enter' key.
- Enter Trunk Access Code.
- Press 'Enter' key.
- You get confirmation tone and the message on your phone's display

Global Directory Configuration by Extension Users

Extension users can add, delete and edit contacts in Global Directory Part 1 using their extension phones, provided that:

- Global Directory Part 1 is allowed to them in their Class of Service.
- Global Directory Programming is allowed to them in their Class of Service.

- *Extension users can only add, delete and edit names and numbers of contacts in Global Directory Part 1. However, they cannot program the Outgoing Trunk Bundle Group (OGTBG) for the contacts in the directory.*
- *If LDAP is enabled, then the contacts stored in the Global Directory synchronized with LDAP cannot be edited or deleted from the system.*
- *When an extension user programs Global Directory Part 1, the system will automatically assign the number and name to a free Memory Location. The system will use the OGTBG assigned to that Memory Location by the System Engineer to dial out the number added by the extension user.*
- *By default, OGTBG 01 is assigned to all Memory Location Codes in the Global Directory.*
- *To simplify configuration for both the System Engineer and the extension users, the System Engineer is recommended to assign the same OGTBG number uniformly to all Memory Location Codes, and enable Least Cost Routing on this OGTBG.*
- *If no OGTBG has been assigned to a Memory Location in the Global Directory (that is, the field is blank), and an extension user adds a contact to this Memory Location, the number will not be dialed out*

To program Global Directory Part 1 from the Extended IP Phone, follow these steps:

- Press Enter key to enter Phone Menu.
- Scroll to ‘Contacts’ and press Enter Key.
- You will get the following options:
  - Add
  - Edit
  - Delete

Adding a contact

To add a contact, select ‘Add’ and press Enter key.

- Enter your contact’s name on the prompt: ‘Name:’
  A maximum of 12 characters are allowed.
- Press Enter key to save name.
- Enter your contact’s number on the prompt: ‘Number:’
  A maximum of 16 digits are allowed.
- Press Enter key to save number.
- You will get the confirmation tone and the confirmatory message: “Stored at Index xxx”.

Editing a contact

To edit a contact,

- Scroll to ‘Contacts’ in the phone menu and press Enter key.
- Select ‘Edit’ and press Enter key.
- You get the prompt: 'Name:'
- Enter the initial letters of the contact's name.
- The number of matching entries that will appear at a time on your phone's display will vary according to your phone's LCD display capacity.
- Scroll with the Up/Down navigation keys to reach the desired contact's name on the list.
- Press 'Enter' key to select the name.
- The system displays the name you selected.
- To delete a character, use the Back/Forward navigation key to place the cursor under the character you want to delete.
- Press the ‘Cancel’ key to delete the character you selected with the cursor.
- To enter a character, use the Back/Forward navigation key to place the cursor in the position you want to enter the character.
- Enter the desired character by pressing the relevant digit pad keys in quick succession.
- After you have finished editing the name/ number, press Enter key.
- The number of the contact whose name you edited will be displayed.
- Repeat the same steps as you did for editing the name.
- After you have finished editing the number, press Enter key.
- You will get the confirmation tone and the confirmatory message: “Stored at Index xxx”.

Deleting a contact

To delete a contact,

- Scroll to ‘Contacts’ in the phone menu and press Enter key.
- Select ‘Delete’ and press Enter key.
- You get the prompt: 'Name'
- Enter the initial letters of the contact's name.
- The number of matching entries that will appear at a time on your phone's display will vary according to your phone's LCD display capacity.
- Scroll with the Up/Down navigation keys to reach the desired contact's name on the list.
- Press 'Enter' key to delete the name.
- You will get the confirmation tone and the confirmatory message: ’Deleted’.

How to use

Personal Abbreviated Dialing

For Extended IP Phone Users

- Press DSS Key assigned 'Abbreviated Dialing' function (if programmed).
  OR
- Dial 8 (users world wide)
  OR
- Dial **6** (users in USA)
- You get the message 'Give Index'
- Enter Personal Directory Index number: 0001 to 0025.
- The desired number will be dialed out.

Global Abbreviated Dialing

For Extended IP Phone Users

- Press DSS Key assigned 'Abbreviated Dialing' function (if programmed).
  OR
- Dial **8** (users world wide)
  OR
- Dial **6** (users in USA)
- You get the message 'Give Index'
- Enter Global Directory Index number: 0100 to 2999.
- The desired number will be dialed out.

Access Codes

Access codes are short digit sequences dialed from an extension phone to instruct the System to perform a function such as:

- Calling an extension.
- Calling a group of extensions (“Department Call”).
- Grabbing a trunk line or any trunk line from a group of trunks (“OG Trunk Bundle Group”).
- Invoking a feature. Activating or deactivating a feature.

Accordingly Access Codes are classified into:

- **Station Codes:** Codes used for calling extensions. These codes are also commonly referred to extension numbers, phone numbers. For the purpose of this document, station codes are referred to as Flexible Numbers.
- **Logical Group Codes:** Codes used for calling a group of extensions as in a Department group, a group of trunks as in Outgoing Trunk Bundle Group.

  Default logical group codes: the factory-set codes for Department Numbers start from 3901, 3902…. Outgoing Trunk Bundle Groups from 61, 62, etc.
- **Feature Codes:** Codes used for invoking a feature.

  Default feature codes: there are different feature codes for every feature/function of the ANANT UCS, e.g.: '2' for Auto Call Back, '5' for Raid, '13' for Call Forward, etc.

  Access codes may consist of single digits or a sequence of a maximum of 6 digits.

  You can change the default access codes to codes of your choice. For example: the default Operator code '9' can be changed to '0'; the default trunk access code for dialing Trunk Group1 can be changed from '61' to '5'.

How it works

Whenever an access code is dialed from an extension, the system matches each digit in the code with the access codes programmed within the system to determine the instruction, that is, whether it is an extension it must call, or a trunk line it must grab, a port it has to activate, etc. The system processes the instruction when a match is found.

For example:

- An extension user dials 131 to set Call Forward.
- When the first digit '1' is dialed, the system finds a match. As several default access codes begin with '1' the system waits for the next digit to be dialed.
- When the second digit '3' is dialed, the system finds a match for '13'.

- As '13' is common for all Call Forward options[65], the system waits for the next digit to be dialed

- When the user dials the third digit '1', the system finds a match for '131'.

- If there is more than one access codes matching with '131', e.g. '1311', '1314', '1315' the system will wait for the next digit to be dialed.

- If no further digit is dialed on expiry of the Inter Digit Wait Timer, the system understands the instruction as 'Call Forward - Unconditional' and waits for the destination phone number to be dialed.

Access Codes are related to various phases of a call. When a call is processed by a System, it goes through a number of pre-defined phases.

Typically a call passes through the different phases as shown below:

Idle	Dial	Routing	Blocked	Placed	Matured 2-Way	Matured 3-Way	Denied
No activity.	Digits are pressed on the phone keypad/dialed from the rotary. Dial tone is played.	The system is processing the call. The call is neither placed nor blocked. Beeps are played.	The dialed extension is busy. Busy tone is played.	The dialed extension is ringing. Ring Back Tone is played.	Connected with the dialed extension. Two-way speech.	Connected with two extensions. Three-way speech.	No reply from dialed extension. Error Tone is played.

Different access codes are dialed at different call phases. Station Codes and Logical Group Codes are dialed in the 'Dial' phase.

As different features are invoked in each call phase, Feature Access Codes are dialed at different call phases. For example:

- Call Forward code is dialed at the 'Dial' phase'.

- DND Override code is dialed at the 'Routing' phase'

- Auto Call Back code is dialed at the 'Blocked' phase' as well as 'Placed' phase.

- Three-party Conference code is dialed at the 'Matured 2 way' phase.

'Idle' phase is when no code is dialed. In the 'Denied Phase' no code is allowed to be dialed.

Each access code in a single call phase may be of different lengths, but must be unique. For example, the same access code cannot be used for two different features like Call Forward and Redial, since both these features are invoked in the 'Dial' phase.

65. *Call forwarding options: Unconditional, When Busy, When No Reply, When Busy or No Reply.*

However, the same access code can be used for features in different call phases. For example, '4' is the default feature access code for DND Override (Routing Phase), Call Pick-Up-Group (Dial Phase) and Barge-In (Blocked Phase).

Similarly, Station and Logical Group Codes too must be unique and should not match with any of the features invoked in the 'Dial' phase. Refer the topics “Flexible Numbers” and “OG Trunk Bundle Group” to know more.

How to configure

ANANT UCS provides default Access Codes for logical groups - department and trunk groups - and features.

It also provides country-specific default Access Codes which are applied automatically when you select the 'Region' to configure the system.

The default Access Codes for India are presented in the table below. The default Access Code tables also indicate the call phase in which each feature is invoked.

Feature	Access Code	Call Phases					
		Dial	Routing	Blocked	Placed	Matured 2-way	Matured 3-way
Enter SE Programming Mode	1#91	Y				Y	
Enter SA Programming Mode	1#92	Y				Y	
Call Pickup - Group	4	Y				Y	
Call Pickup - Selective	12	Y				Y	
Auto Call Back - Set	2			Y	Y		
Auto Call Back - Cancel	102	Y				Y	
Redial	7	Y				Y	
Auto Redial - Set	17	Y				Y	
Auto Redial - Cancel	1070	Y				Y	
Personal Directory Programming	1071	Y				Y	
Abbreviated Dialing	8	Y				Y	
Operator	9	Y				Y	Y
Call Forward	13	Y				Y	
Dynamic Lock	14	Y				Y	
Hotline	15	Y				Y	
Alarm	161	Y				Y	
Do Not Disturb	18	Y				Y	
Interrupt Request	3			Y			
Barge-In	4			Y			
Raid	5			Y			
Call Toggle	1						Y

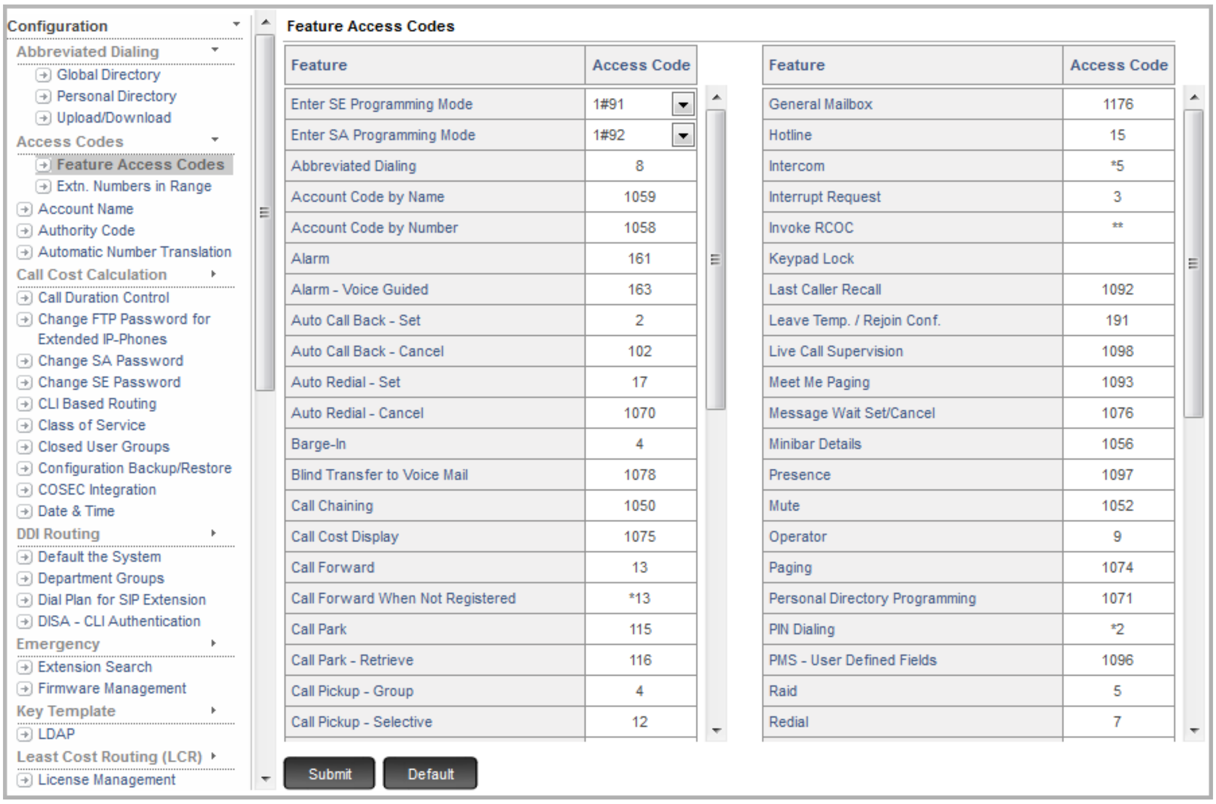
Feature	Access Code	Call Phases					
		Dial	**Routing**	**Blocked**	**Placed**	**Matured 2-way**	**Matured 3-way**
Conference	*3						Y
PIN Dialing	*2	Y				Y	
Dial-In Conference	*19	Y				Y	
Call Park	115					Y	
Call Park - Retrieve	116	Y				Y	
Room Monitor	1073	Y				Y	
Last Caller Recall	1092	Y				Y	
Walk-In Class of Service	111	Y				Y	
Change User Password	114	Y				Y	
Paging	1074	Y				Y	
DISA Login	1079	Y					
Cancel all Feature Settings	1051	Y				Y	
Selective Port Access	69	Y				Y	
Flashing on Trunk	*	Y				Y	
User Absent/Present	104	Y				Y	
Account Code by Number	1058	Y				Y	
Account Code by Name	1059	Y				Y	
Meet Me Paging	1093	Y				Y	
Do Not Disturb Override	4		Y				
Presence	1097	Y					
Conversation Recording	1095					Y	
Forced Release	#*			Y			
Transfer	F					Y	
Live Call Supervision	1098	Y				Y	
Forced Answer	5				Y		
Change Room Clean Status	1054	Y					
Guest Number Prefix	1055	Y					
Minibar Details	1056	Y					
Mute	1052	Y				Y	
Emergency Conference	1177	Y					
Self Ring Test	1057	Y					
Call Chaining	1050					Y	
SA Command Prefix	1072	Y				Y	
COSEC Door Open	*7	Y					

Feature	Access Code	Call Phases					
		Dial	Routing	Blocked	Placed	Matured 2-way	Matured 3-way
Floor Service	38	Y				Y	
Keypad Lock	-	Y					
CLI Restriction	103	Y					
Call Cost Display	1075	Y					
Reminder	162	Y				Y	
Alarm - Voice Guided	163	Y					
Reminder - Voice Guided	164	Y					
Blind Transfer to Voice Mail	1078					Y	
Message Wait Set/Cancel	1076	Y				Y	
Retrieve New Message	1077	Y					
PMS - User Defined Fields	1096	Y					
Invoke RCOC	**	Y					
Scheduled Call Forward	1175	Y				Y	
Department Call Forward	1179	Y				Y	
General Mailbox	1176	Y				Y	
Intercom	*5	Y				Y	
Terminate Conference	190						Y
Leave Temp. / Rejoin Conf.	191						Y
Call Forward - When Not Registered	*13	Y					
Trunk Access Code 1(TAC1)	0	Y					
Trunk Access Code 2(TAC2)	5	Y					
Trunk Access Code 3(TAC3)	61	Y					
Trunk Access Code 4(TAC4)	62	Y					
Trunk Access Code 5(TAC5)	63	Y					
Trunk Access Code 6(TAC6)	64	Y					
Voice Mail	3931	Y					

You can either use the default Access Codes or change them to suit your preferences.

Changing Feature Access Codes

- Login as System Engineer.
- Under **Configuration**, click **Access Codes**.
- Click **Feature Access Codes.**



- Change the Access Codes as per your preference. A maximum of 6 digits.
- Click **Submit.**

  If you enter a code that is already assigned to a extension or a feature, the system will not accept the duplicated code. The value will remain unchanged.

- To disable an access code, delete the existing code and leave the field blank.
- Click **Default** to default all Access Codes.

To change Station Access Codes (for extensions), Department Groups, Trunks and Trunk Groups, and General Mailbox Access Code refer the topics “Flexible Numbers”, “Department Call”,“OG Trunk Bundle Group” and “General Mailbox Settings”.

Account Codes

Account Codes are very useful feature for organizations such as business consultants, law firms, advertising and media agencies, and the like that cater to several clients, interacting with third parties on behalf of their clients. Such organizations need to keep track of calls made to and on behalf of each client.

An 'Account Code' is a unique three-digit number that an organization can assign to each of its clients. Each Account Code may be given a name and programmed in the Account Name List.

Doing so, whenever calls are made to the client or to a third party on behalf of the client,

- The extension user dials the Account Code or Name assigned to the client.
- The Account Code may be dialed,
  - before dialing the external number.
    Or
  - when in speech with the client/third party
- Details of these calls are recorded by the Account Code dialed in the Station Message Detail Recording Report (SMDR) for Outgoing Calls.
- The SMDR report can be printed using the Account Code as filter.

  This way, the organization can know the details of calls made to and on behalf of each client.
- ANANT UCS supports as many as 999 Account Codes.

How it works

For example, an advertising media agency makes nearly 100 calls every day to and on behalf of its clients that includes 'Midas Business Solutions', 'Jet-Set Holidays', 'Bacchus Vineyard'.

- Assign a three-digit account code to Midas Business Solutions, for instance '001' and the name code 'Midas Biz' in the Account Name List.
- Assign a three-digit account code to Jet-Set Holidays, for instance '002' and the name code 'Jet' in the Account Name List.

Case 1: Applying same Account Code

- A, a person from advertising media agency makes a call to Midas Business Solutions and talks to the secretary B. During an ongoing conversation A dials the account code 001. In between, A needs to consult the manager C. Therefore, A presses the *Transfer* key to put B on **Consultation hold** and dials the number of C. Account code 001 will be applicable to the second call made to C also.

Case 2: Applying different Account Codes

- A, a person from advertising media agency makes a call to Midas Business Solutions and talks to the secretary B. During an ongoing conversation A dials the account code 001. In between, A needs to talk to another client, say Jet-Set Holidays D. Therefore, A pressed the *Call Hold* key to put B on **Exclusive/**

Global hold and dials the number of D. While in speech with D, A dials account code 002. Here, Account Code 001 will be applicable to Midas Business Solutions and Account Code 002 will be applicable to Jet-Set Holidays.

Forced Account Code

ANANT UCS can be programmed to prompt extension users to dial the Account Code whenever they grab a trunk to dial out a number. Hence the feature name Forced Account Code.

To apply Forced Account Code, this feature must be enabled on the extensions and trunks from which calls using Account Codes are to be made. When Forced Account Code is enabled on an extension, and the extension user dials out a number or the Trunk Access Code to grab a trunk, the system will play an error tone. If the extension is an extended IP Phone, the system will flash a message on the phone's display that Account Code is required to make the outgoing call.

- *Account Codes are applicable for external calls only.*
- *To use Account Codes, this feature must be included in the Class of Service (CoS) group allowed to the extensions.*
- *If you want to use Account Names, you must program the Account Name List.*
- *When the Forced Account Code is enabled on an extension and trunk, the system will ask the user to enter the account code irrespective of the method of dialing: Global Abbreviated dialing, Personal abbreviated dialing, Least Cost Routing.*
- *However, if Forced Account Code is enabled on the selected trunk, and the number is dialed using OGTBG and the call is routed through the trunk which has forced account enabled the system will dial out the number using Store and Forward dialing.*

  In the case of Abbreviated Dialing or Direct Dialing, if the extension user fails to dial the Account Code, an error message will be displayed on the extension user's IP Phone.

How to configure

For Account Code to work, you must:

1. Enable Account Codes feature in the Class of Service (CoS) of the extensions to which this feature is to be allowed.
2. Prepare and program the Account Name List, if it is to be used.
3. If Forced Account Code is to be used, you must enable:
   - *'Do not allow outgoing calls without Account Code'* check box in the "Station Advanced Feature Template" applied to the extensions from which calls using account codes are to be made.
   - *'Forced Account Code' check box in* the "Trunk Feature Template" applied on the trunks through which calls using account codes are to be made.

Preparing Account Name List

In consultation with the Users, you may prepare the Account Name list. You may do the following

- Draw a two-column table on a paper.
- Write Account Codes on one column. Account codes may be any three-digit number between 001 and 999.
- Write the Account Names, that is, names of the clients on the second column, against their respective Account Codes.
- The names must not exceed 12 characters. All ASCII characters except <, >, :, ", /, \, |, ?, * are allowed. For example:

Account Code	Client Account Name
001	Midas Biz
002	Jet Set
:	
010	Bacchus

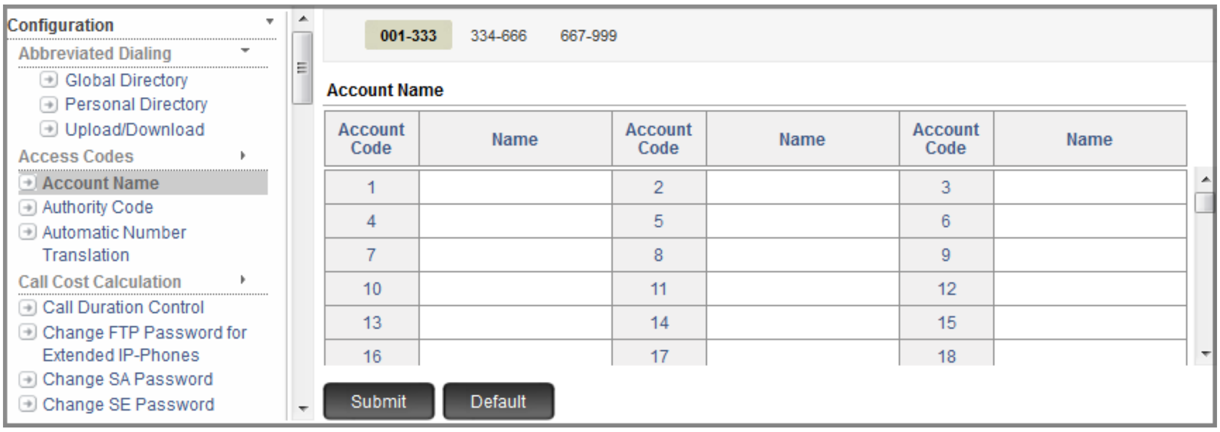
You need not follow a cardinal numbering sequence when assigning Account Codes.

You may assign any code to any client. For instance, you can assign code '111' to Midas Business Solutions, '222' to Jet-Set Holidays, '333' to Bacchus Vineyard.

Configuring Account Codes

After preparing the Account Name list,

- Login as System Engineer.
- Under **Configuration**, click **Account Name**.



- Enter the Names of the clients against the account codes you have assigned to them. Refer the paper with the two-column table you created.
- Click **Submit.**

Now, include the feature **Account Codes** in the **Class of Service** of the extensions.

By default, Station Basic Feature Template Number 01 is assigned to all extensions of the system. Template 01 has the feature Account Codes in the default CoS Group (Number 01). So, all extensions of the system can use this feature.

If Account Codes is to be allowed only to selected extensions, follow these steps:

1. Define a CoS group with 'Account Code' enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
3. Assign this new Template to the extensions to which Account Codes is to be allowed.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for detailed instructions.

If Forced Account Code is to be used, enable '*Do not allow outgoing calls without Account Code'* check box in the "Station Advanced Feature Template" of the extensions and *'Forced Account Code' check box* in the "Trunk Feature Template" assigned to the trunks.

When you enable 'Do not allow outgoing calls without Account Code' check box in a template, this feature will be enabled on all SIP extensions that are assigned this template. If necessary, create a separate template with this feature and assign this template only to those extensions that are to be assigned this feature.

To enable **Forced Account Code** check box on Trunks,

- Go to **SIP Trunk Parameters** to configure SIP Trunk.
- Click **Others** to expand.
- Click the **Trunk Feature Template**. The feature template page opens.

  By default, Trunk Feature Template 01 is assigned.

  If you want to enable this feature on all trunks, enable it in Trunk Feature Template 01.

  If you want to enable this feature only on selective trunks, program a different Template number with this feature.
- Click **Miscellaneous** to expand, select the **Forced Account Code** check box in the template number assigned to the trunk.
- Click **Submit**.
- Return to the SIP trunk parameters page.
- Now change the Trunk Feature Template number of the trunk you want to configure. This number should be the same as the template in which you have enabled the Forced Account Code check box.
- Click **Submit**.

How to use

Account Codes can be dialed in two ways: by Number and by Names.

Account codes, that is, number and names, can be dialed:

- before making the call,
- during the call,
- when grabbing a trunk (if Forced Account Code check box is enabled).

Print and hand out copies of the Account Code List to everyone in the organization for reference while making calls.

Dialing Account Code by Number

For Extended IP Phone Users

To enter Account Code Number before making the call:

- Press DSS Key assigned to 'Account Code by Number'.
  OR
- Dial **1058**
- Enter Account Code
- Dial Trunk Access Code
- Dial the number of the client.

To enter Account Code Number during the call:

- Press ‘Transfer’ Key
- Press DSS Key assigned to Account Code by Number.
  OR
- Dial **1058**
- Enter Account Code
- Speech will be resumed.

To enter Account Code Number when Forced Account Code check box is enabled:

For Extended IP Phone Users

- Press DSS Key assigned to Account Code by Number.
  OR
- Dial **1058**
- Enter the Account Code Number.
- You get dial tone.
- Dial Trunk Access Code followed by the number of the client.

  If you dial the Trunk Access Code to grab a trunk, without dialing the Forced Account Code, you will get an error tone. Go ON-hook and then go OFF-hook. Now follow the same steps in the sequence mentioned above.

Dialing Account Code by Name

For Extended IP Phone Users

To enter Account Code Name before making the call:

- Press DSS Key assigned to Account Code by Name.
  OR
- Dial **1059**.
- Enter the initial letter of the client's name.
  The Account Name List will be displayed on your Extended IP Phone, alphabetically with the corresponding account codes.
- Scroll to select the desired client name and press Enter key.
- Dial Trunk Access Code.
- Dial the client's number.

To enter Account Code Name during the call:

- Press Hold key to put the called party on hold.
- Press DSS Key assigned to Account Code by Name.
  OR
- Dial **1059**.
- Dial the initial letter of the client's name.
  The Account Name List will be displayed on your phone, alphabetically with the corresponding account codes.
- Scroll to select the desired client name and press Enter key.
  Speech will be resumed with the called party.

Dialed the wrong account code or name?

If you have dialed the wrong account code or name while in the middle of a call, you can correct it by pressing 'Hold' again and following the steps described above. The system will override the previously dialed account code or name.

Printing Call Reports of Clients

You can print the call details of your clients using their account codes as filter.

Refer the section "Station Message Detail Recording-Report", for more detailed instructions on printing reports using filters.

Alarms

Alarms are an efficient and user-friendly feature available to all extensions of ANANT UCS.

Alarms can be set and canceled by the extension users for themselves.

Alarms can be set as:

- *Once Only* - A one-time call, where the extension phone rings at the set time.
- *Daily* - A repeat call, where the extension phone rings at the set time everyday.

Alarms can be served as:

- *Personalized* - The Operator greets the extension user to serve the alarm request.
- *Automated* - The system serves the alarm request by playing a voice message or music.

How it works

Personalized Alarm

When the Alarm serving mechanism is configured as 'Personalized',

- The Operator phone rings first[66], displaying the number of the extension to which the alarm is to be served.
- When the Operator answers this call, a call is placed on the extension on which the alarm is set.
- The extension rings for the duration of the Alarm Ring Timer.
- When the extension user answers the call, the Operator greets the extension user with the time and alarm message.
- This event is recorded in the Hotel-Motel Activity Log as 'Wake-up Alarm of <HH:MM> Answered on <phone number>'.
- If the extension user does not answer the call till the Alarm Ring Timer has elapsed, the Operator phone will display a text message notifying 'No Reply' from the extension. The Alarm is now considered as served.
- This event is recorded in the Hotel-Motel Activity log as 'Wake-up Alarm of <HH:MM> No Reply on <Phone Number>'.
- If the extension is busy the Operator phone will display a text message notifying that the extension number is 'Busy'.

66. *The Operator phone rings for the duration of the Alarm Ring Timer. If the Operator does not answer the call, the system will make two more Alarm Attempts at an Alarm Attempt Interval of 5 minutes to call the Operator.*

- The Operator can now choose to
  - inform the extension user about the alarm in person or send someone to do it.
  - try the busy extension again.
  - set “Auto Call Back (ACB)”.

Automated Alarm

When the Alarm serving mechanism is configured as 'Automated',

- The extension phone rings at the set time till the end of the Alarm Ring Timer. The Alarm message will appear on its display.
- When the extension user answers the call, s/he may be played music-on-hold, or a pre-recorded voice message or be connected to a routing group, depending upon the Alarm Notification Type configured by the System Engineer.

  The System Engineer may consult with the Enterprise to decide which of these options is to be configured as the Alarm Notification Type.
- If the extension user does not answer the alarm call, the system makes two more attempts (in all, 3 attempts) at an interval of 5 minutes between each attempt, to call the extension. (Each attempt is recorded in the Hotel-Motel Activity log as 'Wake-up Alarm of <HH:MM> No Reply on <Phone Number>'.
- If all Alarm attempts go unanswered, the system places the call on the Operator phone. The Operator phone rings till the end of the Alarm Ring Timer. The Operator phone displays the extension number with the message 'No Reply'. The Alarm is now considered as served. (This event is recorded as "Alarm Notification to Front Desk for <Phone Number>").
- If the extension phone is busy the system will continue to make Alarm Attempts at the Alarm Interval configured. When all Alarm Attempts go unanswered, the system will place a call on the Operator phone. The Operator phone will display the number of the extension phone with the message 'Busy'.

Snooze

The Snooze function can be added to Automated-Alarms to ensure that the extension user answers the call. Snooze is a system-wide feature; when set, this function will be added to all Automated Alarms.

When Snooze is activated,

- The extension phone rings for the Number of Alarm Attempts configured, at set Alarm Attempt Intervals.
- The extension stops ringing, when the extension user answers the call and dials the Code '0' to acknowledge the Alarm. Please note that this Alarm Acknowledgement Code is non-configurable.

Consider you have set an alarm with snooze enabled and Number of Alarm Attempts set as three (configurable). If this alarm call is not acknowledged by the extension user at the first alarm attempt and due to some reason, the system restarts, then the pending two alarm attempts will not be served. However, this alarm will be displayed under the pending alarm list.

*The system now considers this as a new alarm and will serve the same on the next day at the same time. Also, the number attempts made by the system will be as per value configured in the parameter **Number of Alarm Attempts**, provided it is not acknowledged by the extension user.*

Alarm Status Report

The Operator can know the status of Alarms (details of Alarms that have not been served) from the WakeUp Alarms Reports from SA mode or by pressing the DSS key assigned to Wakeup Call Log.

The WakeUp Alarms Report is useful when Operators change shifts.

Using SA Mode

The Jeeves displays the status of Alarms set by Operator as well as extension users appears on this report, with details of time (hours and minutes), type (once only, daily), and serving mechanism (personalized, automated). The Alarm Report generated by the system can be printed or sent to a computer.

Using DSS Key

The Extended IP Phone users can also view the Wakeup Call Log Status on the phone LCD using the DSS key assigned to Wakeup Call Log. For instructions to assign a DSS Key to Wakeup Call Log, see “Configuring Alarm Parameters”.

The Wakeup Call Log[67] is the log of:

- Unanswered Alarm Calls - This log will display both Alarms Calls that have not been answered as well as unacknowledged Alarm Calls.
- Pending Alarm Calls - This log will display Alarm Calls which have been set for a later time and/or date.
- Served Alarm Calls - This log will display Alarm Calls which have already been served.

You can check Alarms set (pending), served and unanswered for last 25 hours. Altogether maximum 500 entries will be displayed. Each Alarm Call will display the details of time (hours and minutes), date and type (once only, daily).

The LED of the DSS key assigned to Wakeup Call Log glows in Red to indicate Unanswered Alarm calls or it glows in Blue to indicate Pending Alarm calls.

If there are both, Unanswered and Pending Alarm calls, the LED of the DSS key will glow in Red. After you view the Unanswered Alarm Calls, the LED will glow in Blue to indicate Pending Alarm Calls. The LED will glow in Blue till all the Pending Alarm calls have been served. If any Daily Alarm has been set the LED of the DSS key will glow in Blue till the alarm is canceled.

67. *The logs will also display the Reminders.*

To view the log from any Extended IP Phone,

- Press the DSS Key assigned to Wakeup Call Log.
- The phone displays the logs — Unanswered Alarm Calls, Pending Alarm Calls, Served Alarm Calls.
- Select the desired log to view the details in the respective log.
- The acknowledged call is removed from the Alarms Log and is logged into the System Activity Log with the details of the extension that acknowledged the call.

- *ANANT UCS can register as many as 960 Alarm requests set by the Operator and extension users.*
- *Multiple Alarms can be set for an extension by the Operator and/or by the extension user. For example, Daily Alarm at 09:00am is set for an extension. The extension user wants to change the alarm time to 08:30am for a day. The extension user/Operator can set another alarm, that is, a Once Only Alarm, at 08:30am without disturbing the daily alarm. Both the Alarms will ring at the set time.*
- *When multiple alarm requests have been set on an extension, if the Operator/extension user cancels an alarm set for an extension, the system cancels all alarms set for the extension. It is not possible to cancel any of these alarms selectively.*
- *It is not possible to modify an alarm request. Instead, the alarm request should be canceled and a new one should be made.*
- *The duration of Alarm Ring Timer, the Number of Alarm Attempts and the Alarm Attempt Interval are programmable.*
- *Alarms can be set for all extensions of the system, including the Operator phone also.*
- *All the Alarm events are logged in the "Hotel-Motel Activity Log”.*
- *Alarm settings will be retained in the system during power down and system upgrades.*

How to configure

The following parameters play an important role in the functioning of the Alarm feature. These parameters carry default values. The default values have been selected keeping the larger user base in mind. However, these values can be changed by the System Engineer at the time of installation or afterwards as per users' requirements.

1. **Alarm Ring Timer (sec)** - The duration for which the system rings the extension to serve an Alarm call. By default, the Alarm Ring Timer is set to 45 seconds. This timer can be set between 001 to 255 seconds. This timer also signifies the duration for which the Operator phone rings to notify that an Alarm call has not been answered or the extension phone is busy.
2. **Number of Alarm Attempts** - Number of times the system attempts to place an Alarm call on the extension phone before notifying the Operator that the call is not answered or the phone is busy. By default, the Number of Alarm Attempts is set to '3'. The Number of Alarm Attempts can be set between 1 and 9.
3. **Alarm Attempt Interval (mins)** - The time period between each Alarm Call attempt. By default, the Alarm Attempt Interval is set to 5 minutes. The Alarm Attempt Interval can be set between 1 and 9.

4. **Use Alarm with Snooze** - Snooze is a functionality which forces the extension user to acknowledge the Alarm call. With snooze enabled, the system expects the user to answer the Alarm call by going OFF-Hook and dial Acknowledgement code '0'. With snooze disabled, the system considers the Alarm as answered when the extension user simply answers the alarm call by going OFF-Hook (dialing Acknowledgement code is not mandatory). Users may choose whether or not to enable snooze. By default, snooze is disabled.

5. **Configurable Alarm Type** - When the Operator and extension user set an Alarm call request, the system gives them the choice of setting 'Once Only' or 'Daily' Alarm calls.

   User experience however, shows that 'Once Only' Alarm call requests are more common than 'Daily' Alarm requests. So, the system allows you the flexibility of setting 'Once Only' as the default Alarm Type, by disabling the 'Configuring Alarm Type' check box.

   When this check box is disabled the system will prompt the Operator/Extension user to enter the Time of the Alarm call and consider the Alarm Type as 'Once Only'.

   By default, this check box is disabled.

6. **Configurable Alarm Category** - When the Operator sets an Alarm call for an extension, the system prompts the Operator to select an Alarm Type (Once Only or Daily) and to select the alarm serving mechanism - 'Automated or Personalized'.

   If the Enterprise wishes to offer only 'Automated' Alarms to its extension users, the system allows the flexibility to set 'Automated' as the default Alarm call serving mechanism. This can be done by disabling the 'Configurable Alarm Category' check box.

   When this check box is disabled, the system will consider the Alarm call serving mechanism as 'Automated' and will prompt the Operator only for the Time of the Alarm call.

   By default, this check box is disabled.

   .

   - *When both the check-boxes 'Configurable Alarm Type' and 'Configurable Alarm Category' are disabled, the system will set and serve 'Once Only - Automated' alarms only.*
   - *If the 'Configurable Alarm Type' check box is disabled, but the 'Configurable Alarm Category' check box is enabled, the system will set 'Once Only' alarm calls, but give the option of selecting 'Automated' or 'Personalized' as the serving mechanism.*
   - *Similarly, if the 'Configurable Alarm Type' check box is enabled, but the 'Configurable Alarm Category' check box is disabled, the system will allow both 'Once Only' and 'Daily' alarms to be set, but the serving mechanism will be 'Automated'*

7. **Voice Guided Alarm Verification:** For Voice-guided Alarms, the VMS of the system allows you to enable/disable the Alarm Verification for alarms and reminders, allowing extension users who want to use alarms and reminders to confirm the Time set for an alarm and Date and time set as a reminder. By default, this check box is enabled.

The ‘Configurable Alarm Type’ and ‘Configurable Alarm Category’ check boxes are not applicable for Voice-guided Alarms. In the case of Voice-guided Alarms, the Operator/Extension user will be prompted to select the Alarm type and serving mechanism, each time, even when both above mentioned check boxes are disabled.

8. **Alarm Notification Type** - This is the means of notifying the extension user about the Alarm call. The extension user can be played Music-On-Hold, Live Music, Pre-recorded Voice Message, Weather information, Date and Time, etc. The system supports four types of Alarm Notifications:

   - *Music-On-Hold:* Selecting this option would play music-on-hold to the extension user when s/he answers the Alarm call.

   - *Routing Group:* Selecting this option would connect the extension user to the stations programmed in the Alarm Notification Group. The System Engineer may connect a device which can play customized alarm greetings with date, time, weather conditions, traffic conditions, a marketing message, etc. on the stations programmed in the Alarm Notification Group.

If Voice Mail Auto Attendant is selected as a Routing Group member, the system will place the call on the Voice Mail System.

   - *Voice Mail:* Selecting this option would connect the extension user to the Voice mail System.

9. **Macros** - This is a short code for simulating the Alarm call. When the DSS Key assigned to Alarm is pressed the IP Phone send a fixed string to the system. The system interprets this string and translates it into a string that can be understood by the system. For example, the IP Phone has a DSS key for Alarm calls which sends the string '51' to the system. The system can be configured to translate '51' into the feature access code for Alarm calls, '*161'.

   All the above listed parameters can be configured using Jeeves.

Configuring Alarm Parameters

- Login as System Engineer.

- To configure Alarm parameters, go to “System Parameters”.

- To view the Alarm Report Status on your phone LCD, you must assign a DSS Key to Wakeup Call Log. Refer the topic “DSS Keys Programming”, “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510” for instructions.

- To select Alarm Notification Type for extensions, under *Configuring Extensions*, see“Station Advanced Feature Template”.

  - If you select Music-On-Hold as Alarm Notification type, no further configuration is required.

  - If you select Voice Mail as the Alarm Notification Type, no further configuration is required.

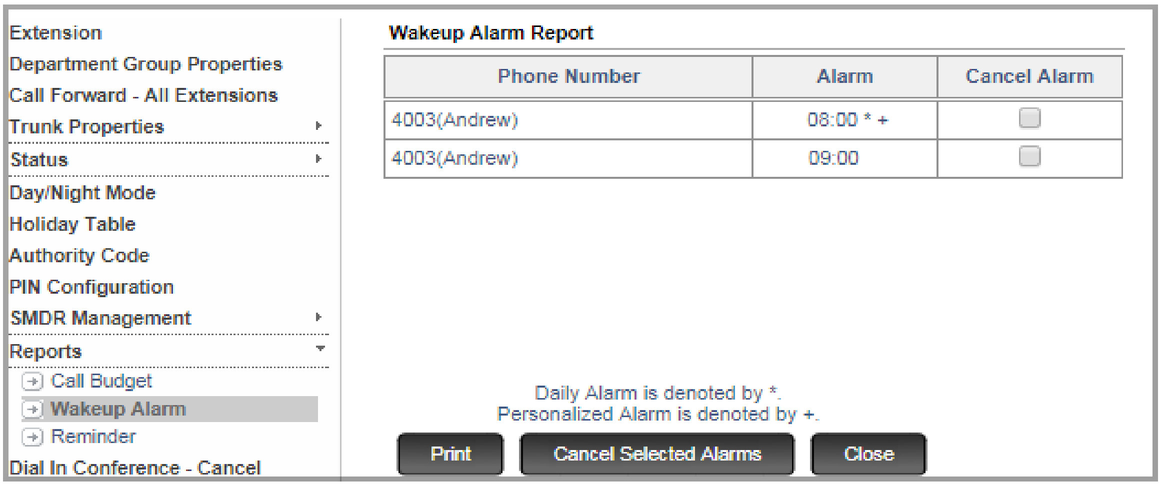
  - If you have selected Routing Group as Alarm Notification Type, you must create a Routing Group and assign this Routing Group number in the Station Advanced Feature Template of the extensions. See “Routing Group” and “Station Advanced Feature Template” for instructions on applying the template to SIP Extensions and Virtual Extensions.

- To create macro and Assign the DSS key for IP Phones, see “Macros”.

Viewing and Printing Alarm Reports

The Operator can view the status of Alarms that are yet to be served from the System Administrator mode and if required print them.

- Login as System Administrator.
- Click **Reports**.
- Under **Reports**, click **Wakeup Alarm**.



- The unserved Alarm calls will appear on your screen.
- You can cancel any of the unserved Alarm calls by selecting the check-box and clicking the **Cancel Selected Alarms.**
- You can also print this page by clicking the **Print.**
- Click **Close** to exit the page.
- Click **Submit.**

How to use

Alarms can be set by the extension users by themselves. The extension users can also ask the Operator to set the alarm for them.

Alarms set/canceled by Operator

The Operator can set/cancel non-voice guided Alarms from System Administrator mode.

Operator using Extended IP Phone

Using DSS Key:

To set Alarm for the extension user,

- Press the key assigned the 'Remote Alarm' function.
- Enter the Extension Number.
- Enter Time in HH:MM
- Select 'Once Only' or 'Daily'.
- Press 'Enter' key.
- Select 'Personalized' or 'Automated'.
- Press 'Enter' key to set Alarm.
- You get a confirmation tone and a text message with the phone number for which the alarm is set.
- Go Idle or you get dial tone after 3 seconds.

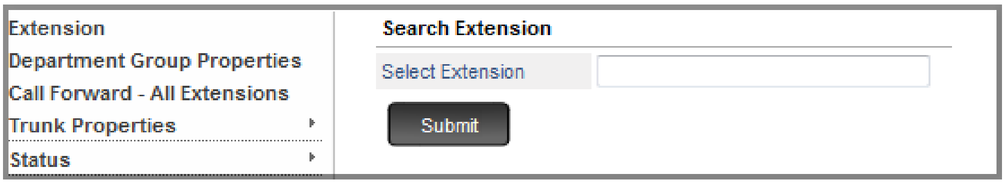
To cancel Alarms,

- Press key assigned the 'Remote Alarm' function.
- Enter Extension Number.
- Select 'Cancel All'.
- Press 'Enter' Key.

Using Jeeves:

- Login as System Administrator.

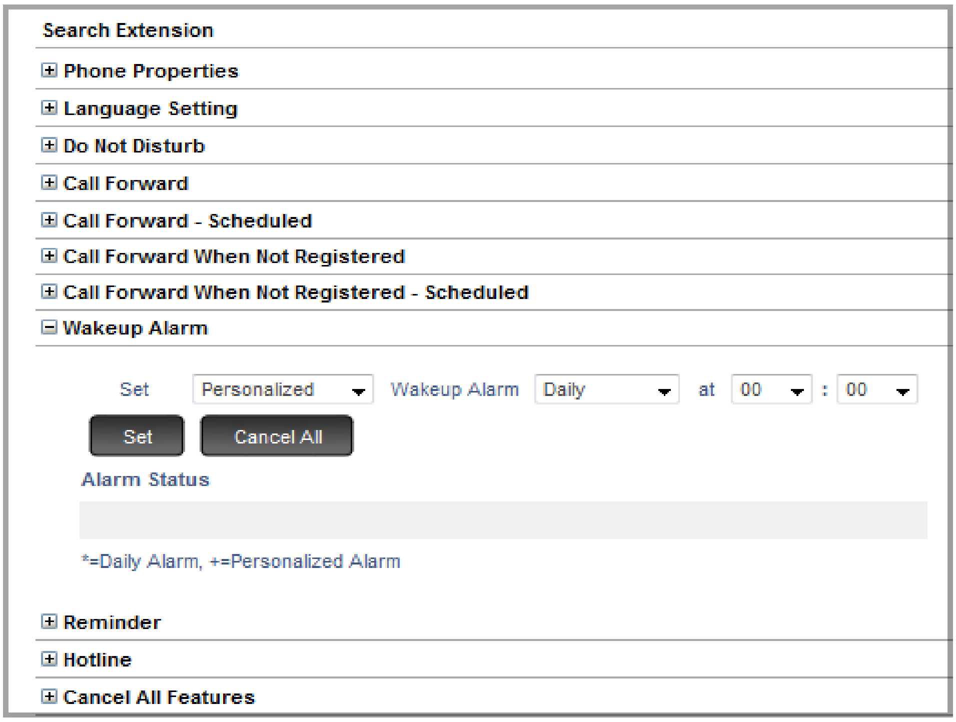
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set the alarm.

- Click **Submit**.

- The page of the Extension will open.



- Now set the desired type of alarm on this extension.

- Click **Submit**.

- Repeat the same to set alarm on another extension number.

Alarms set/cancel by Extension Users

Extension Users using Extended IP Phone

Using DSS Key:

To set Alarm,

- Press the key assigned the 'Alarm' function.
- Enter Time in HH:MM
- Select 'Once Only' or 'Daily'.
- Press 'Enter' key.
- You get a confirmatory text message and confirmation tone.
- Go Idle or you get dial tone after 3 seconds.

To cancel Alarms,

- Press the key assigned the 'Alarm' function.
  OR
- Dial 161
- Select 'Cancel All'.
- Press 'Enter' Key.

Dialing Commands:

To set Alarm,

- Pick up the handset.
- Dial ***161***
- Enter Time in HH:MM (24-hours format)
- Dial 1 for Once Only or Dial 2 for Daily.
- Press 'Enter' key.
- You get a confirmatory text message and confirmation tone.
- Replace handset or you get dial tone after 3 seconds

To cancel Alarms,

- Pick up the handset.
- Dial ***161***
- Dial #.
- You get a confirmatory text message and confirmation tone.
- Replace handset or you get dial tone after 3 seconds.

- *Extension users can set only automated alarms from their phones. For personalized alarms, they must request the Operator.*
- *If there are multiple alarms set, alarms cannot be canceled selectively. Only the Operator can cancel alarms selectively from SA mode using Jeeves.*
- *Alarms set on an extension will be served, even if DND is also set on the same extension.*

Alternate Number Dialing

Alternate Number Dialing allows you to dial different phone numbers in an attempt to reach a person whose line is busy.

Alternate Number Dialing is useful when the person or organization you are trying to reach has more than one number, where they may be reached. The system dials out different phone numbers of the same party, saving your time and effort of dialing each of these numbers manually.

How it works

This feature works as an extension of the features “Last Number Redial” and “Auto Redial”. It requires you to configure Alternate Number Groups in the Global Directory first. With the alternate numbers configured in the Global Directory, all you need to do is to use Last Number Redial or Auto Redial, every time you want the system to try Alternate Number Dialing.

For example: Midas Business Solutions has four telephone numbers: 2640459, 2631235, 2635589 and 2565590. To be able to use Alternate Number Dialing, you must first configure all four numbers as Alternate Number Group in the Global Directory.

Now, when you dial one of these numbers, '2640459', and get a busy tone, you can either initiate Last Number Redial or set an Auto Redial request.

When you initiate Last Number Redial,

- The system will dial an alternative number for the dialed number.
- If the redialed number is busy, you can set Last Number Redial again.
- The system will dial a second alternative number.
- If the second alternative number is also busy, you can set Last Number Redial again.
- The system will dial a third alternative number.
- This process will be repeated each time you set Last Number Redial, until the call gets through.

When you set an Auto Redial request on busy tone,

- The system will dial an alternative number.
- If the alternative number is busy, the system will redial another alternative number.
- The system will dial a different (alternative) number on each auto redial attempt[68], until the call gets through.
- (for the number of redial attempts configured), until the call gets through.

---

68. *The number of auto redial attempts depends on the Auto Redial Count configured in the system. By default, the system will make 5 redial attempts if Auto Redial 'normal' is set. If Auto Redial 'Priority' is set, the system will make 20 redial attempts.*

- When any of the alternate numbers gets through, the system will give a ring on your extension.

- *Alternate Number Dialing will work only on extensions that are allowed the features "Last Number Redial" in their "Class of Service (CoS)"*
- *Also, Alternate Number Dialing will work only for those numbers that exist in the Global Directory assigned to each extension. The Global Directory is divided into three parts, 0100-2399 (Part 1), 2400-2699 (Part 2), and 2700-2999 (Part 3). If an extension is assigned only Global Directory Part 2, Alternate Number Dialing will work only for those numbers grouped as Alternate Number Groups in Global Directory Part 2.*
- *Alternate Number Dialing will work also with "Abbreviated Dialing". For example, an extension user dials the abbreviated code 80100, and the dialed out number is busy. When the extension user sets Redial or Auto Redial, the system will try the alternate numbers related to 80100, if configured.*

How to configure

For Alternate Number Dialing to work, the System Engineer must:

1. Make a List of Alternate Numbers.
2. Create Alternate Number Groups.
3. Configure Alternate Number Groups in the Global Directory.
4. Enable the features 'Last Number Redial', 'Global Directory', in the Class of Service (CoS) group of the extensions to which Alternate Number Dialing facility is to be provided. If desired, 'Auto Redial', 'Auto Redial Priority' may also be enabled in the CoS of these extensions.

All of the above parameters can be configured using Jeeves.

- *To create Alternate Number Groups, the alternate numbers must exist in the Global Directory. If any of the alternate numbers do not exist in the Global Directory, first configure the numbers in the directory, before you begin creating Alternate Number Groups. Refer the topic "Abbreviated Dialing" for instructions on configuring the Global Directory.*
- *As Alternate Number Dialing works only for the Alternate Number Groups in the Global Directory assigned to each extension, ensure that the relevant Global Directory with the Alternate Number Groups is allowed in the CoS of the extensions.*

Preparing Alternate Number List

In consultation with the user, you may:

- Draw a two-column table on a paper.
- Write the name of the contact on one column and the Alternate Numbers for the contact on the other column.

- Make a list of the numbers which need to be grouped as alternate numbers. For example:

Name of the Contact	Alternate Numbers
Midas Business Solutions	2640459, 2631235, 2635589, 2565590
Jet Set Holidays	022281110001, 022281110002
Bacchus Vineyard	2640075, 2640076
GoodLife Inn	2788856, 2788896

You may include as many alternate numbers as required by the user.

Creating Alternate Number Groups

- Assign the alternate numbers of each of contacts to an 'Alternate Number Group'.
- Each group must be assigned a number between 000 to 255.
- Taking the above example further, the Alternate Number Groups on the list may be numbered as follows:

Name of the Contact	Alternate Numbers	Alternate Number Group (No.)
Midas Business Solutions	2640459, 2631235, 2635589, 2565590	001
Jet Set Holidays	022281110001, 022281110002	002
Bacchus Vineyard	2640075, 2640076	003
GoodLife Inn	2788856, 2788896	004

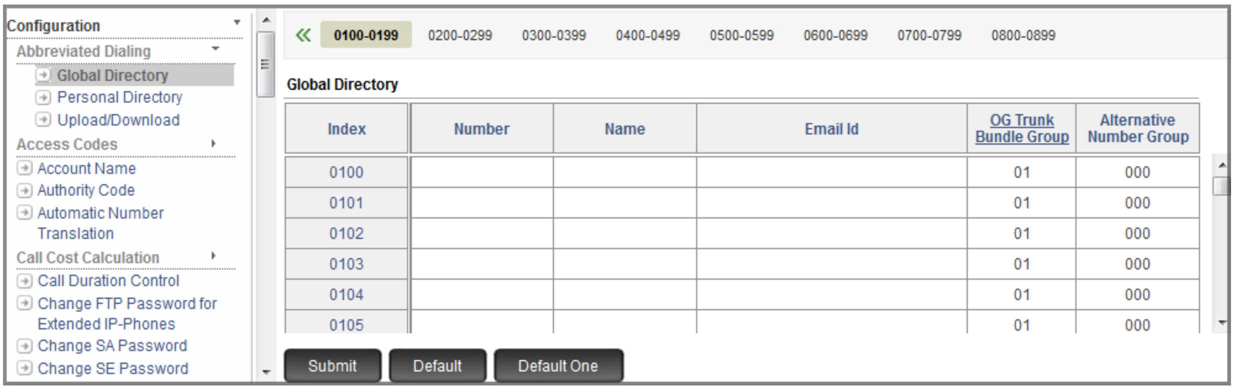
Configuring Alternate Number Groups in Global Directory

Make sure that the numbers on the list are also programmed in the Global Directory. If any of these numbers do not exist in the Global directory, program them in the Global Directory first. Refer the topic “Abbreviated Dialing” for instructions on programming the Global Directory.

To create Alternate Number Groups and program them in the Global Directory,

- Login as System Engineer.
- Under **Configuration**, click **Abbreviated Dialing**.

- Click **Global Directory**.



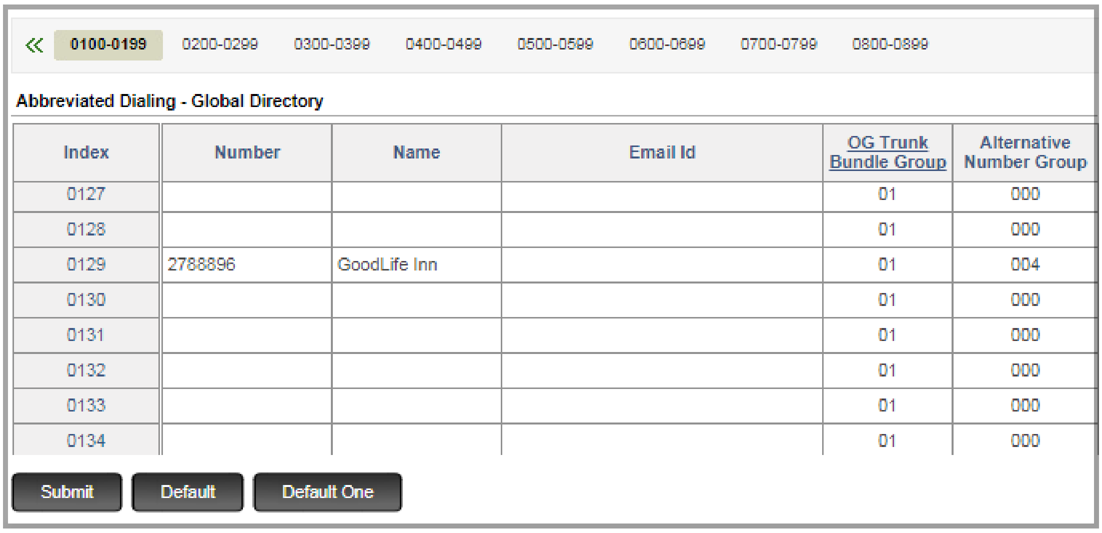
- Enter the number of the **Alternate Number Group** in the last column of the page.

  For example, you have assigned Alternate Number Group '001' to all the numbers of the contact Midas Business Solutions, enter this number against each number belonging to this contact.

  Similarly, enter Alternate Group number '004' against the numbers belonging to the 'GoodLife Inn' to which it is assigned.

Memory Location	Outgoing Trunk Bundle Group	Number	Name	Alternate Number Group
0100	01	2640459	Midas Biz	001
0101	01	2631235	Midas Biz	001
0102	01	2635589	Midas Biz	001
0103	01	2565590	Midas Biz	001
0104	01	2788856	GoodLife Inn	004
0105	01	022281110001	Jet Set	002
0106	01	022281110002	Jet Set	002
0107	01	033298765432	R. Mendez	000
0108	01	2640075	Bacchus	003
0109	01	2640076	Bacchus	003
:	:	:	:	:
0129	01	2788896	GoodLife Inn	004

The numbers of the contacts may not necessarily appear alphabetically or in a sequence. It is possible that the numbers of the same contact may be programmed at different memory locations in the Global Directory.



Index	Number	Name	Email Id	OG Trunk Bundle Group	Alternative Number Group
0127				01	000
0128				01	000
0129	2788896	GoodLife Inn		01	004
0130				01	000
0131				01	000
0132				01	000
0133				01	000
0134				01	000

In the above example, one number of the GoodLife Inn is configured at memory location Index 0104 and the other on Index 0129. Since these two numbers are grouped and assigned the number alternate group number '004', this number must be entered against the GoodLife Inn numbers at the respective memory location Index.

- After assigning Alternate Number Groups, click **Submit.**

- Enable the features **Last Number Redial** and **Global Directory**, in the Class of Service (CoS) group of the extensions to which Alternate Number Dialing facility is to be provided. If desired, **Auto Redial**, **Auto Redial Priority** may also be enabled in the CoS of these extensions.

By default, Station Basic Feature Template Number 01 is assigned to all extensions of the system. The default CoS Group 01 in this template has 'Redial' enabled in the set of 'Basic Features', so all extensions of the system can use Last Number Redial.

However, the default CoS Group 01 has only Global Directory Part 1 enabled.

Recall that Alternate Number Dialing will work only for those numbers that exist in the Global Directory assigned to each extension. So, the Global Directory Part containing the Alternate Number Groups must be allowed to the extensions in their Class of Service. For example, if Alternate Number Groups are programmed in Global Directory Part 2, extensions must have Global Directory Part 2 in their Class of Service.

If all extensions are to be allowed the Alternate Number Dialing facility, simply enable the Global Directories containing Alternate Number groups in the default CoS group 01.

However, if Alternate Number Dialing is to be allowed to select extensions only, define a new CoS group and prepare a new Template with this CoS group and apply it to the desired extensions.

Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for detailed instructions.

The Station Basic Feature Template 01 does not have the features Auto Redial and Auto Redial Priority in the default CoS group 01. If these features are also to be allowed to the extensions, enable them in the CoS you prepare.

How to use

Confirm with your System Engineer that:

- Alternate Number Groups are programmed in the Global Directory allowed to your extension.
- 'Basic Features' (these include Redial) are enabled in the Class of Service allowed to your extension.

Now, follow the instructions for using the feature “Last Number Redial”.

Apple Push Notification Service Support

Apple Push Notification Service (commonly referred to as Apple Notification Service or APNs) is a platform notification service created by Apple Inc. that enables third party application developers to send notification data to their applications installed on Apple devices.

Previously, VoIP applications needed to maintain a persistent connection in order to receive calls. Keeping a connection open in the background, drains the battery as well as causes all kinds of problems when the application crashes or is terminated by users.

In iOS 8 Apple has introduced PushKit as part of their effort to improve battery life, performance and stability for VoIP applications such as Skype, WhatsApp, etc. PushKit offers high-priority push notification with a large payload. The VoIP application receives the notification in the background, sets up the connection and displays a local notification to the user.

ANANT UCS supports PushKit for VARTA AMP100 Application only. Push Notifications will be sent for calls, new messages as well as for voicemail. Push Notifications will be sent to the MATRIX VARTA AMP100 Application only if it is in the background and when there is persistent internet connection. You will receive the Push Notifications even after you exit the application provided the **Calls and Messages after exit** check box is enabled in the VARTA AMP100 Application. For details refer to the *VARTA AMP100 User Guide*.

How it works

Pre-requisites for Push Notifications:

- Make sure that the server has a persistent internet connection.
- Make sure the Date and Time of the server is synchronized with the NTP Server.
- To receive IM and IM notifications make sure the application is registered at Location 1. For more details, refer "Configuring Matrix VARTA ADR100/AMP100 UC Clients"

Let us see how the notifications will be sent by the server when MATRIX VARTA AMP100 application is registered with the server as a SIP Extension and it is in the background. There is an incoming call or message:

- You can check the status of the SIP Extension user. It will display Registered (as the device is in the background) and under the respective Contact 1, 2, 3, it will display the time remaining for the expiry of the VARTA Client Inactivity Timer. The default value of the VARTA Client Inactivity Timer is 10 days. To configure this timer, refer to "System Timers and Counts".

- The server will send a Push Notification to the MATRIX VARTA AMP100 application (client).

- The server will wait for 15 seconds after sending the Push Notification:

  - if the client registers with the server within this time, the call will be connected or the message will be delivered. The status of the SIP Extension will display Registered and under the respective Contact 1, 2, 3 it will display the SIP ID, IP Address and the Registration Expiry Timer.

  - if the client does not register with the server within this time, the call will be disconnected or the message will be rejected.The status of the SIP Extension will display Registered and under the respective Contact 1, 2, 3 it will display the time remaining for the expiry of the VARTA Client Inactivity Timer.

- The server maintains a configurable timer, VARTA Client Inactivity Timer which is set as 10 days. Till the expiry of the timer the server will send Push Notifications to the application.

- If for this duration, the server does not receive any registration request from the application and the timer expires, the server will consider the application as unregistered and will stop all Push Notifications to the application.The status of the SIP Extension will display Not Registered and under the respective Contact 1, 2, 3 the details will be cleared. Calls and messages will be rejected.

  The server will start sending notifications to the application in the background after the application is bought in the foreground once and a registration request is received by the server.

Feature Interactions when the Application is in the Background

Call Forward when not Registered:

If the application user is a member of any Routing Group or Department Group, the Call Forward functionality will not be applicable.

To know more, see "Call Forward-When Not Registered".

Handover and Handoff:

VARTA AMP100 users will be able to use Handover but Handoff will not be possible. To know more, see "Handover and Handoff".

System Restart

After System Restart the VARTA Client Inactivity Timer will be reset.

APNS Connectivity

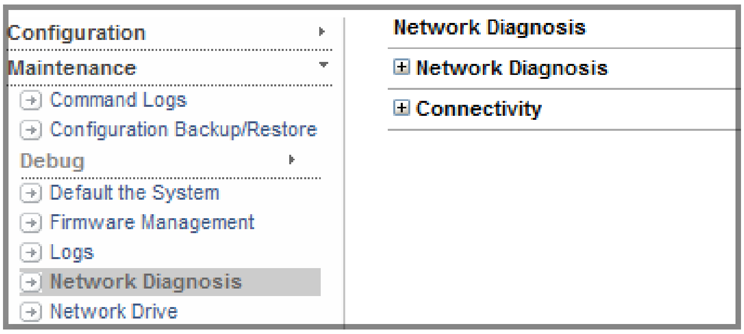
A connectivity between the system and the APNS Server is required so that the Push notifications can be sent to the clients.

To check the APNS connectivity status,

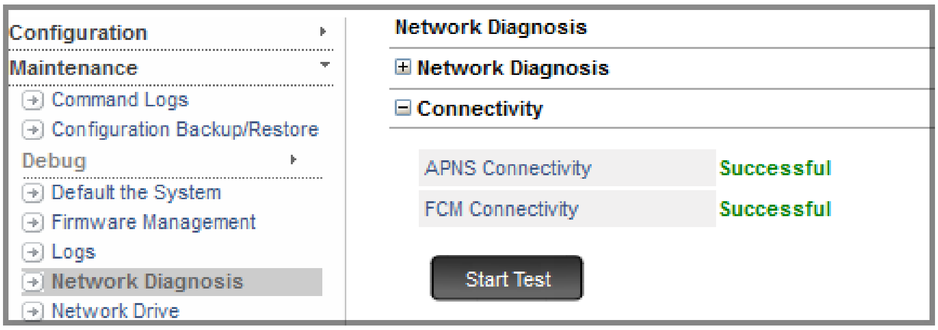
- Login as System Engineer.

- Click **Maintenance**.

- Under Maintenance, click **Network Diagnosis.**



- Click **Connectivity** to expand.



- Click **Start Test**.

- In **Connectivity**, the status is displayed as:
  - Successful - if the connectivity between the system and the APNS/FCM Server is established
  - Timeout - if there is no connectivity.

*If the **Connectivity** Test of either of the servers (APNS or FCM) with ANANT UCS is not successful, then Push Notifications will not be sent to the Mobile Clients — MATRIX VARTA ADR100 / AMP100 application.*

Authority Codes

Authority Code is a unique password-protected code with an associated Class of Service, Toll Control and Call Budget, which can be assigned to extension users. With Authority Codes, extension users of the system can make calls or access features from any other extension of the system as per the Class of Service, Toll Control and other features / facilities assigned to their code.

This feature is useful when you want a group to extension users to use a single extension, but at the same time you want to keep an account of the calls made by each user. If required, you can assign a call budget to each Authority code to control call cost.

To make outgoing calls or access features, extension users must 'Walk-In' from any SIP extension and then dial their Authority Code and Password.

How it works

An 'Authority Code' is a unique three-digit number, protected by a password. The password can be a minimum of 4 digits and a maximum of upto 10 digits. Valid digits are 0 to 9, * and #.The default Authority Password is ***1111***. To be able to use an Authority Code, the password must be changed to another value. To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential.

The ANANT UCS supports as many as 999 Authority Codes.

Each Authority Code has an associated Class of Service and Toll Control, which is configured in the Station Basic Feature Template assigned to the Code.

To make calls using an Authority Code,

User A is assigned 222 as Authority Code and is provided a unique 4-digit Authority Password.

To access features or make calls as per the assigned Authority Code 222, User A must do the following:

- Dial the feature code for Walk-In Class of Service (default 111) from any extension of the system.
- Dial the code for Walk-In by Authority followed by the Authority Code and then the Authority Password.

  To make calls or access features according to the Authority Code dial 111, the feature code for 'Walk-In Class of Service' from any extension of the system.
- Select the option ‘Walk in by Authority Code’ or dial ‘2’.
- A dials the Authority Code assigned to him/her followed by the Password.

  If the user enters a wrong password, to prevent any attempt to misuse the Authority Code, by default, the system allows the user only three attempts to re-enter the password. If the user fails to enter the correct password at the third attempt, the system will set the password to default, 1111. With the default password the user will not be able to use the Authority Code.

  The number of attempts to re-enter can be configured by setting the parameter **Retry counts for Authority Code** password in the “System Timers and Counts”.

- User A can now make outgoing calls.
- If the Walk Out mode set for A is One Call, A will automatically be logged out from the extension after one call.
- If the Walk Out mode set for A is Multiple Calls, A can make as many calls as desired, and remains 'walked-in' until the A dials the feature code to 'Walk-Out', or until another extension user walks into the same extension.
- If a call budget has been assigned to A's Authority Code, A will be able to make calls till the assigned amount is consumed.
- Details of the calls made by A are recorded by the Authority Code in the Station Message Detail Recording Report (SMDR) for Outgoing Calls.
- The SMDR report can be printed using the Authority Code as filter.

  This way, the organization can know the details of calls made by each user as well as have control over the expenses.

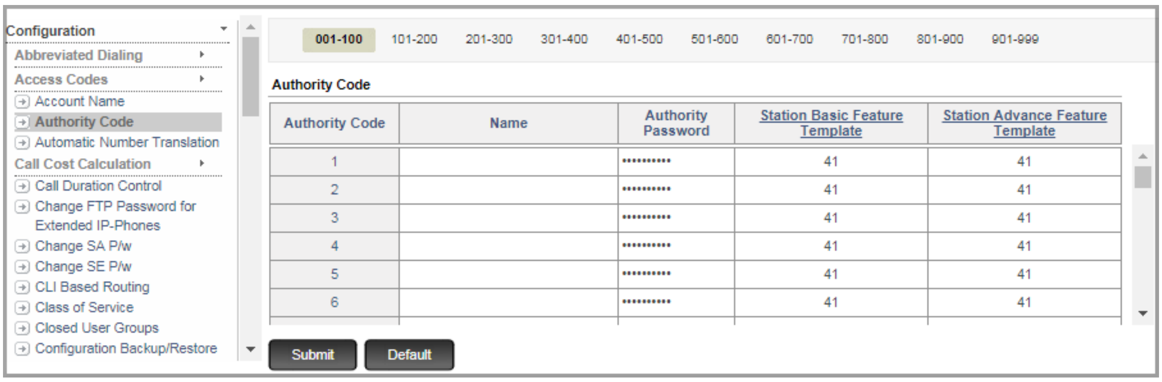
How to configure

For Authority Codes to work, you must:

- Configure the Authority Code Table.
- Assign a Station Basic Feature Template to the Authority Code, with the desired Class of Service, Toll Control and features/facilities.
- Enable the parameter *Store Outgoing Calls* in the Template assigned to the Authority Code, if you want details of outgoing calls made using Authority Codes.
- Assign a Station Advanced Feature Template to the Authority Code with the desired Walk-Out Mode
- Assign each user a Call Budget, if required.
- Change the Retry Counts for the Authority Code, if required.

Configuring Authority Code Table

- Make a list of users you want to assign Authority Codes to and their respective passwords.
- Login as System Engineer.
- Under **Configuration**, click **Authority Code**.



Against each **Authority Code**,

- Assign an **Authority Password**. The password can be a minimum of 4 digits and a maximum of upto 10 digits. Valid digits are 0 to 9, * and #. Default: 1111.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential.

- Enter the **Name** of the user. The name acts as an identifier. Default: Blank.

- By default, **Station Basic Feature Template** number 41 and **Station Advanced Feature Template** number 41 are assigned to all Authority codes.

  If you want to change the calling permission, allow/deny features to users, you must customize the Class of Service, Toll Control and other features/facilties in the Template according to your requirement.

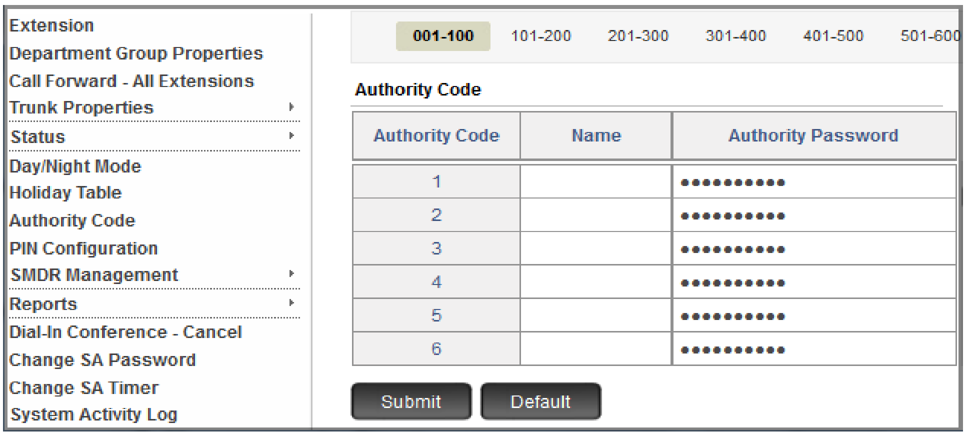
  You may set the 'Walk-Out Mode' in as One Call or Multiple Calls in the Advanced Feature Template. Refer to “Station Basic Feature Template” and “Station Advanced Feature Template” for detailed instructions.

- Click **Submit.**

You can assign/change the Name and Password from the SA mode also. To do this,

- Login as System Administrator.

- Click **Authority Code**.



Against each **Authority Code**,

- Assign/change **Authority Password**. The password can be a minimum of 4 digits and a maximum of upto 10 digits. Valid digits are 0 to 9, * and #. Default: 1111.

  To avoid unauthorized access, we recommend you to change the password regularly. Make sure it is strong and is kept confidential.

- Enter the **Name** of the user against the authority code you have assigned to them. The name acts as an identifier. Default: Blank.

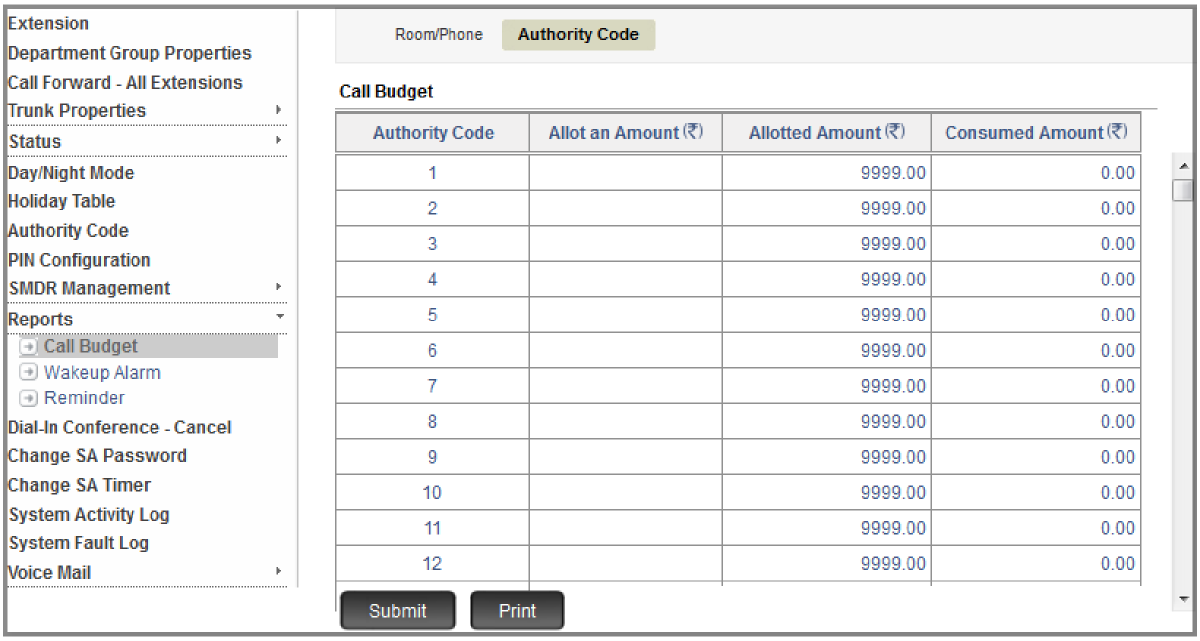
- Click **Submit.**

Assigning a Call Budget to Authority Codes

Call Budget is a cost control feature that allows you to keep a track on the total cost of each user. With this, each user can be allotted a 'budget' limit for outgoing calls, which is automatically reloaded at the start of every month.

To assign a Call Budget to the Authority Code,

- Login as System Administrator.
- Under **Reports**, click **Call Budget**.
- Click the **Authority Code** tab.



For each Authority Code assigned to the user,

- In the **Allot an Amount**, enter the amount you want to assign to the user as budget limit for outgoing calls. The amount you allot here will be displayed as Allotted Amount.
- *If you are re-assigning a new amount before the previous balance is consumed, make sure you add the available balance to the new amount. Enter this amount in **Allot an Amount**.*
- *For example, you have allotted an amount of Rs.1000 and the consumed amount is Rs.600. The available balance is Rs.400. Now, if you want to assign a new amount of Rs.500. In Allot an Amount you must enter 900 (Available Balance + New = 400 + 500).*
- The **Allotted Amount** column displays the amount allotted to the user for making outgoing calls.
- The **Consumed Amount** column displays the call budget amount consumed by the user.
- Click **Submit**.

Setting the Retry Counts for Authority Code

By default, the system allows three attempts to re-enter the Authority Code Password. If you want to increase or decrease the number of attempts, set the parameter ***Retry Counts for Authority Code*** to the desired value. For instructions, see "System Timers and Counts".

How to use

For Extended IP Phone Users

To use Authority Code from any extension:

- Go OFF-Hook.
- Press DSS Key assigned to 'Walk-in Class of Service'.
  OR
- Dial **111**
- Select the 'Walk-in by Authority Code' option and press the Enter key.
- Dial the Authority Code, followed by the Password.
- You will hear the confirmation tone, followed by dial tone.
- Dial the desired number.

Printing Reports of Outgoing Calls made using Authority Codes

You can print call details of users who made outgoing calls using Authority Codes. For this, you will need to:

- enable **Store Outgoing Calls** in the Station Basic Feature Template of the extension user.
- set the **Calls made using Authority Code** filter in Outgoing Call Report.
- configure the **destination port** for SMDR-Outgoing Call Report.

Refer the section "Station Message Detail Recording-Report", for detailed instructions on printing reports using filters.

Auto Attendant

Auto Attendant allows external callers to reach an extension directly without the intervention of the Operator.

If Auto Attendant is enabled on a trunk, whenever an external call lands on that trunk, the *Voice Mail Auto Attendant* of ANANT UCS greets and prompts the caller to dial the desired extension number. The call is then placed to the extension number dialed by the caller.

ANANT UCS offers *Delayed Auto Attendant*, whereby incoming calls routed to the Operator or the Trunk Landing Group, can be answered by the Voice Mail Auto Attendant, if none of the landing extensions answers the call within a certain time period.

Regular callers who know the extension numbers, can use the Auto Attendant to reach the desired extensions without Operator assistance. Thus, this reduces the call traffic on the Operator extension, saves caller's time for call set-up and transfer. The Auto Attendant is particularly useful during non-working hours and holidays, and it helps project a professional image of the organization.

How it works

Auto Attendant can be configured on the trunks, for the three time zones (working hours, break hours and non-working hours).

Voice Mail Auto Attendant

ANANT UCS supports 64 simultaneous VMS calls[69]. These calls can be made by extension users to access their mailbox or incoming calls landing on a trunks that has Voice Mail Auto Attendant enabled.

If all channels are used by extension users, the external incoming calls on the Voice Mail Auto Attendant enabled trunks remain unanswered. To make sure that the incoming calls on the trunks are answered you can reserve channels of the VMS as per your requirement.

The extension users will not have access to the channels reserved for incoming calls on the trunks to access their mailbox.

If all the reserved channels are busy and there is an incoming call on the trunk, it will land on the unreserved channel if free.

To reserve a channel, see "Configuring VMS General Parameters".

When the Voice Mail Auto Attendant of ANANT UCS is selected as the destination for incoming calls on a trunk, this is how it will work:

- A call lands on a Trunk.
- The Voice Mail System (VMS) answers the call.
- The VMS greets the caller with the Welcome message and the Greeting Message selected for the current time zone (working hours, break hours and non-working hours).
- If the system detects the day as a holiday, the VMS plays the Holiday Message. To know more, see "Holiday Table".
- The VMS plays prompts to the caller to process the call further.

69. For the VMS calls, the number of channels that will be supported would be as per the license you purchase.

Delayed Auto Attendant

You can use Delayed Auto Attendant to have incoming calls that are not answered by the landing destinations—the Operator and the Trunk Landing Group—within a certain time period, to be handled either by the Voice Mail Auto Attendant.

When you use Delayed Auto Attendant,

- as a call lands on a trunk, the system checks the incoming call routing configured for the current time zone for the trunk.
- on finding *Delayed Auto Attendant* enabled, the system rings on the destination extensions (Operator and Trunk Landing Group) for the duration of time defined for ringing the extensions (default: 10 seconds).
- if no reply is received from the extensions, the system routes the call to the Voice Mail Auto Attendant.
- the call is processed further by the VMS.

How to configure

To use the **Voice Mail Auto Attendant** on trunks, do the following:

1. Make a list of the SIP trunks and their port numbers on which you want to use the Voice Mail Auto Attendant.
2. In the "Trunk Feature Template" assigned to these trunks,
   - enable the **Voice Mail Auto Attendant** check box for the desired time zones.
   - select the desired **Voice Mail Auto Attendant (VMAA) Menu**.
3. Complete the VMS related configuration.
   - Configure Welcome and Greeting messages. You may either use the default, pre-recorded welcome messages of the VMS, or record the custom welcome messages that meet your requirements, in .WAV file format.
   - Configure the Voice Mail Auto Attendant (VMAA) Menu parameters.

   For more information and instructions, see the "Configuring Voice Mail System".

To use **Delayed Auto Attendant** on trunks, do the following:

1. Make a list of the SIP trunks and their port numbers on which you want to enable Delayed Auto Attendant.
2. In the "Trunk Feature Template" assigned to these trunks, set the desired time option in **Auto Attendant - Delay Auto Attendance** and complete the voice mail related configurations.

For more information and instructions, see the "Configuring Voice Mail System".

Auto Answer

Auto Answer allows incoming calls to be answered without any manual interventions by the extension users.

This feature is particularly useful for Operators in high call traffic settings, as it saves the effort of picking up the handset or pressing the speaker key repeatedly.

How it works

With Auto Answer set on an extension and whenever a call lands on that extension.

- the extension rings for the duration of the Auto Answer Timer[70]. This timer is configurable, and by default it is set to 1 second.
- the phone goes OFF-Hook to answer the call, without any intervention by the extension user such as picking up the handset or pressing the speaker or the headset key.

> - *If the extension is an Extended IP Phone, no beep will be played to the user after the phone goes off-hook to answer the call. So, Auto Answer is a sensitive feature and must be used in accordance with the local privacy laws.*
> - *Matrix Comsec is not responsible for any mis-/abuse of this feature by the users.*

- If a headset is connected, and headset connectivity is enabled, the incoming speech audio will be diverted to the headset automatically.
- the extension user can talk to the caller.

Auto Answer works only if the phone is in idle state; the phone must not be busy with an active call or using a feature.

How to configure

For Auto Answer to work, you are required to do the following:

1. Enable Auto Answer in the extensions.
2. Change Auto Answer Timer, if required. The range of this timer is 1 to 9 seconds. By default, the Auto Answer Timer is set to 1 second.
3. Enable Headset Connectivity, if headset is to be used for Auto Answer.

All of the above can be configured by the System Engineer using Jeeves.

The extension users can also configure the above parameters using the Phone Menu. See "How to use" Auto Answer later in this topic.

70. *This timer defines the time in seconds that the phone should wait before going OFF-Hook to answer incoming calls.*

Configuring Auto Answer

To configure auto answer for Extended IP Phones, see "Configuring Matrix SPARSH VP248" and "Configuring Matrix SPARSH VP310".

How to use

Extension users can set/cancel Auto Answer and enable Headset connectivity from their Extended IP Phone or by navigating the Phone Menu.

To set Auto Answer:

- Press the DSS Key assigned to Auto Answer[71].
  OR
- Press 'Enter' Key.
- Scroll down to select 'Phone Settings'; press Enter Key.
- Enter Your User Password (default: 1111).
- Scroll down to select 'Call Answer Type'; press Enter Key.
- You get the options:
  Manual Call Answer
  Auto Call Answer
- Scroll to select Auto Answer and press Enter key.
- Now select the Timer for Auto Answer from any of the options:
  Answer After 1 sec (default)
  Answer After 2 sec
  :
  Answer After 9 sec
- Press Enter Key.

To cancel Auto Answer:

- Repeat the above steps.
- Select 'Manual Answer' as the Call Answer Type.

It is recommended that Auto Answer Timer be set to at least 2 seconds.

To enable Headset Connectivity:

- Press the DSS Key assigned to Headset function[72].
  OR
- Press 'Enter' Key.
- Scroll down to select 'Phone Settings'; press Enter Key.
- Enter your User Password (default: 1111).
- Scroll down to select 'Headset Connectivity'; press Enter Key.
- You get the options:
  Headset Not Connected
  Headset Connected
- Scroll to select 'Headset Connected' and press Enter key.

71. *This function must have been programmed by the System Engineer on a DSS Key of the phone. Refer "DSS Keys Programming"- for instructions.*
72. *This function must have been programmed by the System Engineer on a DSS Key of the phone. Refer "DSS Keys Programming"- for instructions.*

Auto Call Back (ACB)

If the extension number you have dialed is busy or is not responding, you may use the Auto Call Back feature, instead of repeatedly dialing the number. Similarly, when you dial a code to access a trunk and the trunk is busy, you may set Auto Call Back. ANANT UCS allows you to set a maximum of 300 Auto Call Backs.

How it works

When you set Auto Call Back,

- ANANT UCS will queue your call attempt.
- As soon as both extensions, yours and the remote extension, are available, the system will ring first on your extension for the duration of the Auto Call Back Ring Timer. This timer is set by default to 30 seconds and can be configured.
- When you go OFF-Hook, the system will ring on the remote extension (provided it is also available at that moment) for the duration of the Auto Call Back Ring Timer.
- When the remote extension user goes OFF-Hook, your call will get connected.

  However, if the remote extension gets busy before the system can ring on it, the system will continue to try again.

  Auto Call Back set for a busy trunk works the same way. As soon as the busy trunk port you are trying to access is available, the system will ring your extension. When you go OFF-Hook you will be connected to the trunk port.

- *Each extension of ANANT UCS can set only one Auto Call Back request at a time. If you set another Auto Call Back request, before the first one has been served, the system will override the first request and serve the second.*
- *ANANT UCS has the capacity to serve 300 Auto Call Back requests from its extensions at a time. The service duration for each request is 60 minutes. Requests that are not served within 60 minutes are automatically canceled by the system. Also, the system will not serve any more requests if all the 300 requests are pending. In such a case, the system will play an error tone, when an extension attempts to make a request.*

Auto Call Back request set by you will be cleared by the system if:

- it was successfully served, that is, your extension was connected to the remote extension or the trunk you were trying to reach.
- you do not answer the Auto Call Back ring, before the expiry of the Ring Timer, that is, within 30 seconds (default setting).
- the remote extension does not answer the Auto Call Back ring before the expiry of the Ring Timer.
- it has not been served within 60 minutes.

Auto Call Back works for internal calls and for accessing trunk ports only.

How to configure

Auto Call Back is a Class-of-Service dependent feature. An extension user can set/cancel Auto Call Back only if it is enabled in the extension's Class of Service.

The only configuration involved in this feature is enabling/disabling Auto Call Back in the Class of Service and changing the duration of the Auto Call Back Ring Timer, if required.

Both these can be configured using Jeeves.

Configuring Auto Call Back

By default, Station Basic Feature Template Number 01 is assigned to all extensions of ANANT UCS. Station Basic Feature Template 01 has the features 'Auto Call Back Busy' and 'Auto Call Back No Reply' enabled in the default CoS group 01. So, all extensions of the ANANT UCS can set/cancel Auto Call Back if the called number is busy or does not reply.

However, if Auto Call Back Busy/No Reply is to be denied to any of the extensions, follow these steps:

1. Define a CoS group with Auto Call Back Busy/No Reply disable.

2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

3. Assign this new Template to the selected extensions to which Auto Call Back is to be denied.

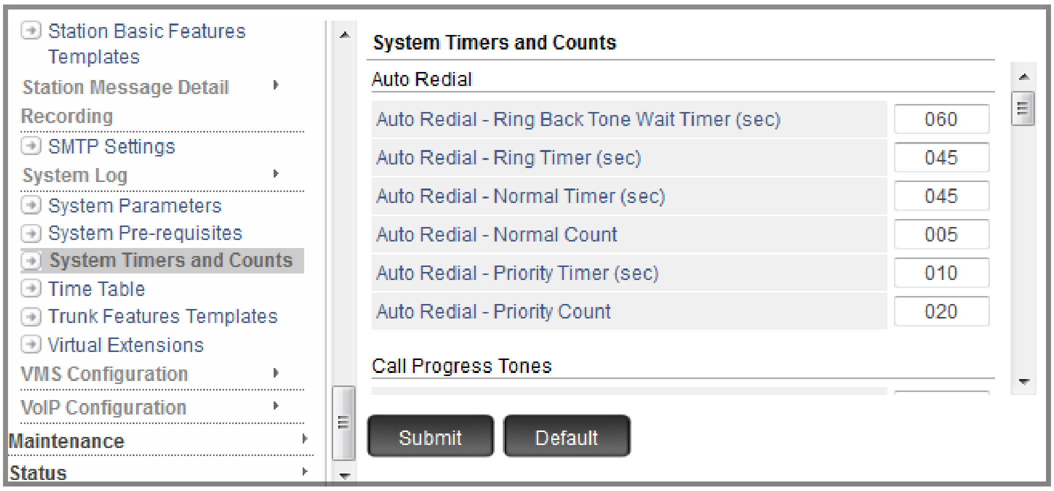
Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for instructions on how to enable/disable a feature in a CoS group, how to prepare a Station Basic Feature Template with a new CoS group and assign the new template to SIP extensions using Jeeves.

If the user wants to increase or decrease the duration of the of the Auto Call Back ring on both extensions, that is, the extension requesting Auto Call Back and the destination extension, configure the 'Auto Call Back Ring Timer', according to user preference.

- Login as System Engineer.

- Under **Configuration**, click **System Timers and Counts**.

- Scroll to **Other Features**.



- Set the **Auto Call Back Ring Timer (sec)** to the desired duration.

- Click **Submit.**

How to use

Extension users can set two types of Auto Call Back:

- Auto Call Back on Busy - when the extension/trunk they are trying is Busy.
- Auto Call Back on No Reply - when there is no reply from the extension they are trying.

Auto Call Back on Busy

For Extended IP Phone Users

Using DSS Key:

To set Auto Call Back on Busy:

- Press the Call Back' Key / DSS Key assigned to Auto Call Back on Busy Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back on Busy:

- Press the Call Back' Key / DSS Key assigned to Auto Call Back again.
- You get confirmatory message "Auto Call Back Canceled" on the phone's display. The LED of the DSS Key will be turned off.

- Go idle or you get dial tone after 3 seconds.

Using Command:

To set Auto Call Back on Busy:

- Dial 2 on Busy Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key assigned to Auto Call Back will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back on Busy:

- Dial 102.
- You get confirmatory message 'Auto Call Back Canceled' on the phone's display. The LED of the DSS key assigned to Auto Call Back will be turned off.
- Go idle or you get dial tone after 3 seconds.

Auto Call Back on No Reply

For Extended IP Phone Users

Using DSS Key:

To set Auto Call Back on No Reply:

- Press the 'Call Back' Key / DSS Key assigned to Auto Call Back on Ring Back Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back on No Reply:

- Press the 'Call Back' Key / DSS Key assigned to Auto Call Back again.
- You get confirmatory message "Auto Call Back Canceled" on the phone's display. The LED of the DSS Key will be turned off.
- Go idle or you get dial tone after 3 seconds.

Using Command:

To set Auto Call Back on No Reply:

- Dial 2 on Ring Back Tone.
- You get confirmatory message "Auto Call Back Set" on the phone's display. The LED of the DSS Key assigned to Auto Call Back will be turned on.
- Go idle or you get dial tone after 3 seconds.

To cancel Auto Call Back:

- Dial 102.
- You get confirmatory message 'Auto Call Back Canceled' on the phone's display. The LED of the DSS key assigned to Auto Call Back will be turned off.
- Go idle or you get dial tone after 3 seconds.

If you hear an error tone while setting an Auto Call Back request, it is likely that the system already has 300 pending requests and is unable to accept yours.

Auto Redial

The Auto Redial feature retries a call automatically if the dialed number is busy. It repeatedly checks the busy line till it is free. When the called number is no longer busy, the extension of the caller rings.

Auto Redial saves time and the effort of repeatedly dialing the entire phone number over and over until the called party gets off the phone.

The Auto Redial feature is supported for external numbers only. Maximum 50 Auto Redials can be set by extension users.

How it works

When an extension user dials a number and gets a busy tone, s/he may set Auto Redial. When Auto Redial is set,

- ANANT UCS checks for a free trunk to dial the number.
- ANANT UCS will dial out the requested number and will wait until the 'Ring Back Tone Wait Timer[73]' expires to sense the Ring Back Tone from the SIP Trunk Service Provider. This timer is configurable and is set to 60 seconds as default.
- If the system does not detect Ring Back Tone for 60 seconds, it releases the trunk and tries again after some time. If the system detects a busy tone, it releases the trunk and redials the number automatically after some time. This process is repeated until the system detects the Ring Back Tone.
- When ANANT UCS detects the Ring Back Tone instead of the Busy Tone, it will ring on the extension that set Auto Redial. The extension will ring for the duration of the 'Redial Ring Timer[74]'. This timer is configurable and is set to 45 seconds as default.
- The extension must go OFF- Hook to get connected to the remote party.
- If the extension is in the middle of any activity such as dialing, ringing or speech, ANANT UCS will suspend Auto Redial until the extension becomes idle again. After which it dials the requested number again.

Two types of Auto Redial are supported by ANANT UCS - Auto Redial (normal) and Auto Redial 'Priority' - that differ from each other in terms of the number of redial attempts and the interval between attempts.

- **Auto Redial (normal):** The system is configured by default to make 5 attempts to redial at an interval of 45 seconds (default) between each attempt. Both, the number of attempts as well as the duration of the interval can be changed to match the user preference, like decreasing the number of attempts to 3 and increasing the interval to 60 seconds.
- **Auto Redial 'Priority':** The system makes a greater number of attempts to redial and the duration of the interval between each attempt is less. By default, the system is configured to make 20 redial attempts at intervals of 20 seconds. The number of attempts as well as duration of the interval are configurable; for instance, the number of attempts can be set to 30 and the interval to 15 seconds.

---

73. *Time for which ANANT UCS waits to sense the RBT from the SIP Trunk Service Provider after dialing the requested number. Valid range of the timer: 000 to 255 seconds. Default: 060 seconds.*
74. *Time for which the extension that has requested Auto Redial should ring. Valid range of the timer: 000 to 255 seconds. Default: 045 seconds.*

To change the number of redial attempts and the interval between them, the SE must configure Auto Redial Count and the Auto Redial Timer respectively. In addition to these, the system has two other related timers, which can be configured to match user preference:

- Auto Redial Ring Back Tone Wait Timer.
- Auto Redial Ring Timer.

- *An extension user can request Auto Redial for multiple numbers at a time from the same extension and more than one extension can attempt auto redial simultaneously.*
- *The system uses the same OG Trunk Bundle Group you used. If you dialed the number on group code 30, the system grabs one of the free trunks from group code 30 for Auto Redial.*
- *If the number was dialed the first time using selective trunk access, the system will use the same trunk to execute Auto Redial.*
- *If the extension is configured for 'Dynamic Lock', and you have set the 'Auto Redial', the system will check the Toll control as per dynamic lock level.*

How to configure

For Auto Redial to work, the System Engineer must:

1. Enable the features 'Auto Redial' and 'Auto Redial Priority' in the Class of Service (CoS) group of the extensions to which this feature is to be allowed.
2. Change the 'Auto Redial Normal/Priority Count' and the 'Auto Redial Normal/Priority Timer' to match user preference. This will change the number of redial attempts made by the system and the interval between them.
3. If required, also change other related Timers such as Auto Redial Ring Back Tone (RBT) Wait Timer, Auto Redial Ring Timer.

   All the above parameters can be configured using Jeeves.

Configuring Auto Redial

By default, Station Basic Feature Template Number 01 is assigned to all extensions of the system. The Station Basic Feature Template 01 does not have the feature Auto Redial and Auto Redial Priority in the default CoS group 01. Thus none of the extensions of ANANT UCS have this feature.

If the user wants to allow all extensions the Auto Redial and /or the Auto Redial Priority feature, then you can simply enable this feature in the default CoS group 01.

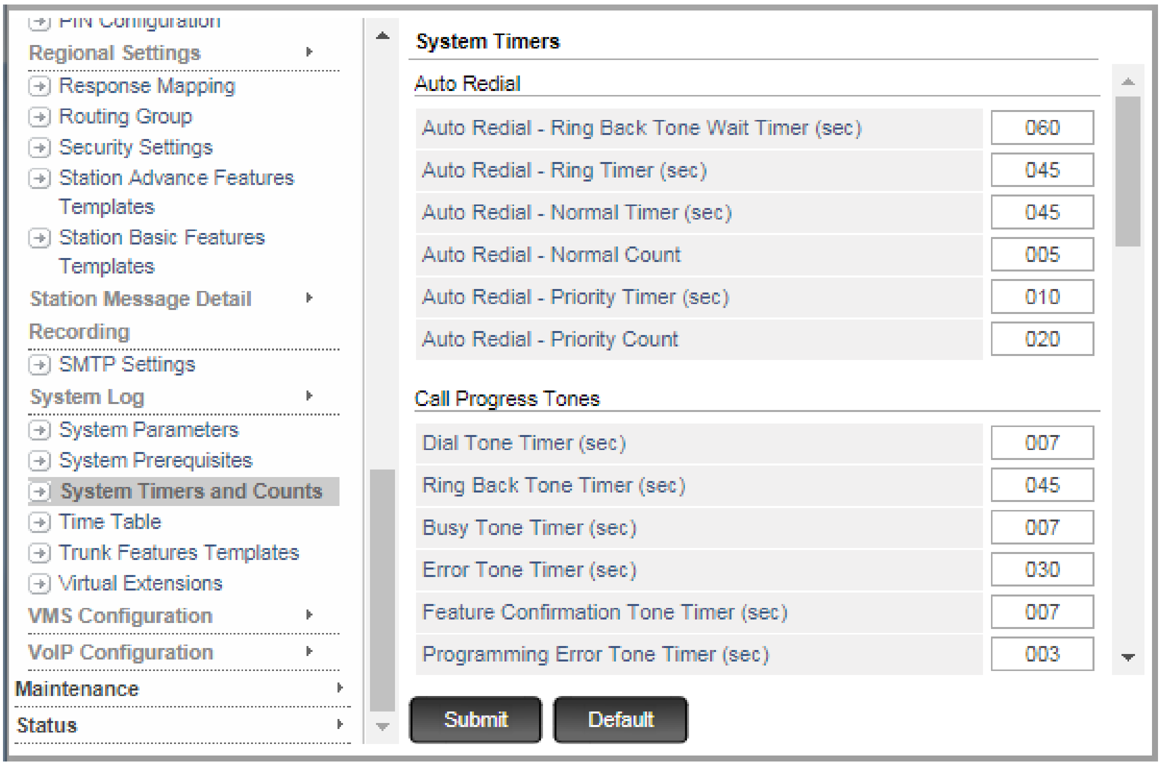
However, if Auto Redial/Auto Redial Priority is to be allowed on only selected extensions, follow these steps:

1. Define a CoS group with Auto Redial/Auto Redial Priority enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
3. Assign this new Template to the extensions to which Auto Redial/Auto Redial Priority is to be allowed.

   Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for detailed instructions.

To change Auto Redial Counts and Timers:

- Login as System Engineer.
- Under **Configuration**, click **System Timers and Counts**.



- Below **Auto Redial**, change the Count and Timer of the type of Auto Redial - Normal or Priority - as per your requirement.
- You may change any of the related timers - Auto Redial Ring Back Tone (RBT) Wait Timer, Auto Redial Ring Timer - as per your preferences on this page.
- Click **Submit**.

- *If the ANANT UCS is installed in Australia, please ensure that:*
  i. *The Timer for Auto Redial Normal as well as Priority must be set to more than 5 seconds.*
  ii. *The Auto Redial Priority Count should be set to less than 15.*

How to use

Auto Redial can be set/canceled from Extended IP Phone.

For Extended IP Phone Users

Using DSS Key

To set Auto Redial:

- When the external number you are trying is busy,
- Go ON-Hook on Busy Tone.
- Press the 'Redial' Key.

- Go Idle.

To cancel Auto Redial:

- Press the ‘Redial’ Key again.

Using Command:

To set Auto Redial:

- When the external number you are trying is busy,
- Go ON-Hook on Busy Tone.
- Dial **17**.
- Go Idle or you get dial tone after 3 seconds.

To cancel Auto Redial:

- Pick up the handset.
- Dial **1070**.
- Go Idle or you get dial tone after 3 seconds.

Automatic Number Translation

ANANT UCS offers connectivity to VoIP networks and these may have different plan. For example, the ITSP requires area codes to be dialed also for local numbers.

When ANANT UCS is connected with such VoIP networks, outgoing calls need to be routed through these trunks. However, when extension users dial local number they cannot be expected to dial numbers with area codes.

The Automatic Number Translation feature of ANANT UCS takes care of this. It modifies/manipulates dialed numbers or part thereof to match with specific route numbering plan understood by the VoIP network. This includes adding or stripping off of country codes and area codes.

For example, when an extension user dials a local landline number, if Automatic Number Translation is configured, ANANT UCS will prefix the number with the appropriate country-area code before routing the call through the VoIP network.

How it works

Automatic Number Translation makes use of the Automatic Number Translation (ANT) Table. The ANT Table consists of three columns:

- **Dialed Number:** This column contains the numbers you expect the users to dial.
- **Strip Digit:** This column contains the number of digit(s) to be stripped off by the system from the Dialed Number string before dialing it out.
- **Add Prefix:** This column contains the digit(s) which are to be added as prefix to the Dialed Number string by the system before dialing it out.

This table is applied on the desired trunk, through which outgoing calls are made.

Upto 8 different ANT tables can be configured and each table can accommodate upto 32 strings.

Here is an example of how this table is to be configured and used:

You want

- All 10-digit numbers to be dialed out after adding the prefix ‘1’.
- All 7-digit numbers, starting with 2 to be dialed out after adding the prefix ‘1315’.
- All numbers beginning with 91 to be stripped off the first 2 digits and ‘0’ to be added as prefix.

When you do not want to specify any numeric digits in the numbers to be modified, use the character **X**. This character represents any numeric digit from 0 to 9. For example, a 10-digit number (having the numeric digits from 0 to 9) can be represented using this character as **XXXXXXXXXX**.

Thus, the entries you will need to make in the ANT table will be as follows:

Dialed Numbers	Strip Digit	Add Prefix
XXXXXXXXXX	0	1
2XXXXXX	0	1315
91	2	0

- The entry for 10-digit numbers to be dialed out after adding the prefix ‘1’ will be as shown in the first row of this table. The 10-digit number is represented with the **X** character in the Dialed Numbers column. Since no digit is to be stripped off, ‘0’ is entered in the Strip Digit column. As the prefix ‘1’ is to be added, this number is entered in the Add Prefix column. The system will add 1 as prefix before dialing out numbers from 0000000000 to 9999999999.

- Similarly, the entry for 7-digit numbers starting with 2 to be dialed out after adding the prefix ‘1315’ will be as shown in the second row of this table. The system will add 1315 as prefix before dialing out numbers from 2000000 to 2999999.

- The entry for all numbers beginning with 91 to be stripped off the first 2 digits and ‘0’ to be added as prefix will be as shown in the third row of this table. When users dial numbers beginning with 91, the system will strip off the first two digits and add 0. For example, when a user dials the number 919925801882, the system will dial out 09925801882.

  Automatic Number Translation also forms the basis of “Multi-Stage Dialing”.

How to configure

To apply Automatic Number Translation on a trunk,

- Decide which of the SIP trunks that are to be assigned the Automatic Number Translation (ANT) feature.

- Decide the number of ANT tables you need. You can program 8 different tables, with a maximum of 32 entries in each.

- Make the tables on a piece of paper by drawing three-column tables. In the first column of a the table, write the dialed numbers that need to be modified before being dialed out from the trunk. For each dialed number in the first column, enter the number of digits you want the system to strip off (if required) from this number in the second column. In the third column enter the number you want the system to add as prefix (if required) before dialing out the number.

- Configure the Automatic Number Translation Table in the system using the tables you prepared.

- Enable the Automatic Number Translation check box in the Outgoing Trunk Bundle (OGTB) of the trunk.

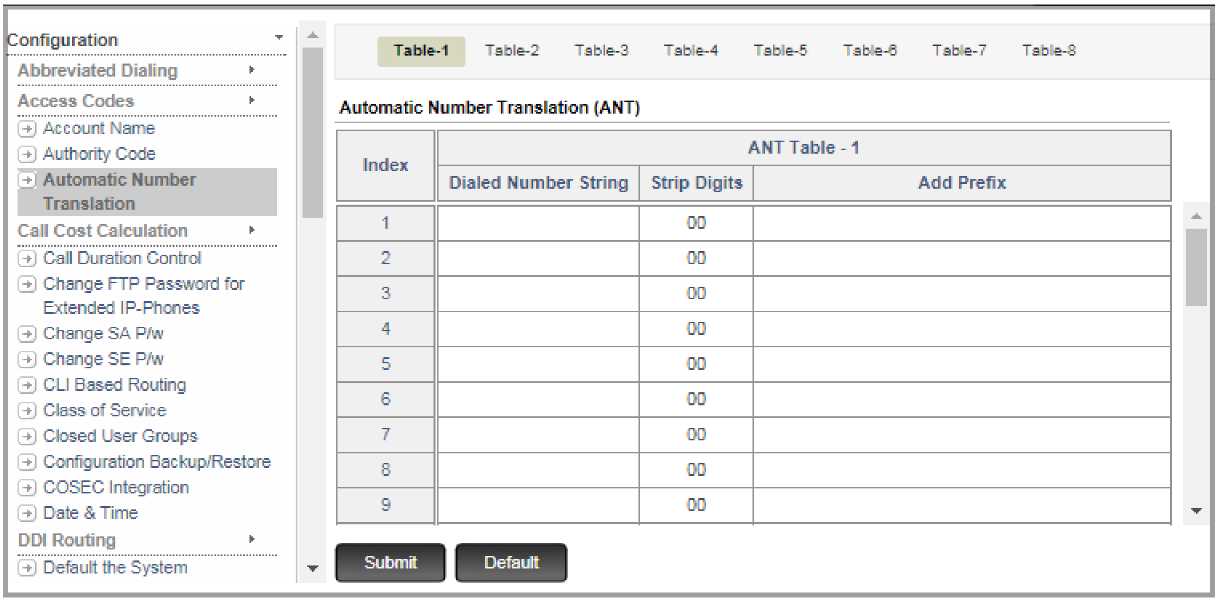
- Assign the Automatic Number Translation Table you configured to the OGTB of the trunk.

Configuring Automatic Number Translation

- Login as System Engineer.
- Click **Configuration**.

Configuring the ANT Table

- Click **Automatic Number Translation**.



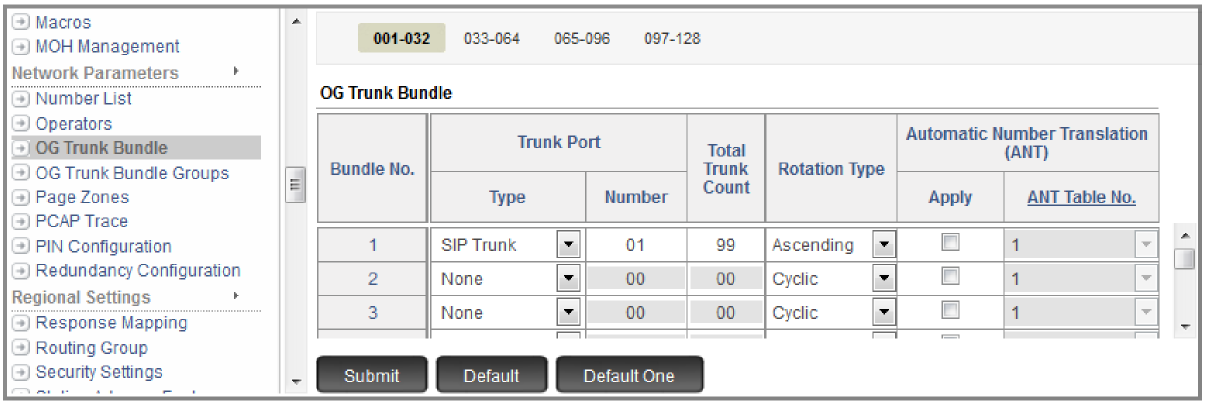
- Choose any table number from 1 to 8.
- In the **Dialed Number** column of the table you chose, configure the numbers you expect the extension users to dial. The Dialed Numbers can be a maximum of 16 characters. Default: Blank.

  When you want to specify number-length without specifying the numeric digits, use the character **X** to represent the digit. **X** represents any number from 0 to 9.

- In the **Strip Digit** column, enter the number of digits you want the system to strip off from the Dialed Number before the dialing out this number. The valid range is from 00 to 16. Default: 0.
- In the **Add Prefix** column, enter the number you want the system to add as prefix to the Dialed Number before the system dials out this number. The Prefix can be a maximum of 40 characters. Default: Blank.
- Click **Submit.**

Enabling Automatic Number Translation on Trunks

- Under **Configuration**, click **Outgoing Trunk Bundle** to open the page.
- In the OG Trunk Bundle of the desired trunk, select the **Automatic Number Translation (ANT) Apply** check box to enable.



- In the **ANT Table No.** list, select the table number you configured for this trunk.
- Repeat the steps to apply the feature on another OGTB.
- Click **Submit.**

Barge-In

Barge-In allows you to break into an on-going conversation between two extension users, between an extension user and an external caller as well.

Barge-In can be used by Operators to transfer Incoming calls to busy extensions. The Operator can put the caller on hold, barge into the busy extension to inform about the call, and then transfer the call.

Barge-In can be used by a Boss to interrupt the secretary's busy extension.

ANANT UCS offers flexibility to allow/deny Barge-In feature to an extension user, that is, allow the extension user to barge into on-going conversations. It also provides the flexibility to prevent conversations of extension users from being barged in, referred to as Privacy against Barge-In.

How it works

- A, B and C are users of the system.
- A and B are talking to each other.
- C calls A.
- C gets busy tone.
- C dials Barge-In feature code.
- C gets Ring Back tone (RBT) and A gets beeps indicating a new call.
- C gets RBT and A gets beeps for Barge-in timer. (By default, 10 seconds)
- During the beeps, A may press 'Flash' to answer the call.
- If A does not respond till the end of the Barge-In Timer (set to 10 seconds, by default), A gets connected to C. B is put on hold and is given hold-on music.

Feature Interactions

- **Call States:**
  - Barge-In works only if the dialed extension is busy. The dialed extension may be busy with another extension or trunk (external number).
  - Barge-In cannot be used when accessed trunk is busy.
  - Barge-In works only if the user about to be barged in is in a two-way normal speech with another user or external party. However, it will not work if the conversation is being recorded.
  - It will not work if the busy signal is due to the user being OFF-Hook, or in the middle of dialing, or accessing a feature of the System.

- **“Call Toggle”:** Once A and C comes in speech with each other, A can toggle between B and C using Call Toggle feature.
- **Privacy against Barge-In:** If the feature 'Privacy against Barge-in is enabled for an extension, it cannot be barged into.
- **“Priority”:** No Interaction with Barge-In. If 'A' has lower priority than 'B' but has Barge-In enabled; A can barge in B.
- **“Do Not Disturb (DND)”:** Barge-In will not work if the called user has set DND. If 'A' has set DND. A is busy with C. B calls A. B cannot barge in A.
- **DND Override:** Barge-In will work if the calling user is allowed DND-Override. If 'A' has set DND. A is busy with C. B calls A. On busy signal, B dials the Barge-In code. Barge-In will be successful only if B has DND-Override enabled.
- **“Call Taping”**: Barge-In will not work when the two-way conversation between the users is being taped.

How to configure

The functioning of this feature is controlled by three parameters, 'Barge-In', 'Privacy against Barge-In' and 'Barge-In Timer'.

Barge-In and Privacy against Barge-In

First decide which of the extensions are to be allowed Barge-In and the extension that are to be protected against Barge-In.

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS. The Station Basic Feature Template 01 is assigned CoS group 01. The default CoS group 01 has both Barge-In and Privacy from Barge-In are disabled. So, none of the extensions of the ANANT UCS can use these features.

If you want to allow Barge-In to the all extensions, simply enable Barge-In in the default CoS group 01.

However, if Barge-In is to be allowed on only selected extensions then follow these steps:

a. Define a CoS group with Barge-In enabled.
b. Prepare an extension Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
c. Assign this new Template to the extensions to which Barge-In is to be allowed.

Repeat the above steps to allow 'Privacy from Barge-In' in the CoS of extensions that are to be exempted from Barge-In.

Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for detailed instructions on programming.

Barge-In Timer

Barge-In Timer is the time after which the caller gets connected to the called party. By default the Timer is set to 10 seconds.

Changing Barge-In Timer

- Login as System Engineer.
- Under **Configuration**, click **System Timers and Counts.**
- Scroll to **Other Features**.
- Go to the parameter **Barge-In Timer (sec)**.
- Set the desired value for the Timer.
- Click **Submit.**

How to use

For Extended IP Phone Users

- Dial an extension.
- If the extension is busy, you get Busy Tone.
- Press DSS Key assigned to 'Barge-In' function.
  OR
- Dial **4**[75].
- You get Ring Back Tone.
- Wait for the system to connect you to the called extension.
- Talk.
- Replace the handset after the conversation has ended.

75. *This default feature access code can be changed to suit your preference. Refer the topic “Access Codes”.*

Busy Lamp Field for Trunks

On SIP extensions, ANANT UCS supports the feature Busy Lamp Field (BLF) Subscription for Trunks, enabling SIP extension users to monitor the status of desired trunks.

Using BLF, SIP Extension users can monitor the status of different SIP Trunks.

To provide SIP extension users this feature, you must enable this feature on their SIP phones and configure the trunk to be monitored on the BLF key of their SIP phones. The number of trunks that can be monitored will depend on the number of BLF keys supported on the SIP Phones.

With BLF subscription and BLF key configured, whenever, there is a change in the state of the monitored trunk, ANANT UCS sends a NOTIFY message to the SIP Extension. The NOTIFY message contains the Call State. On receiving the NOTIFY message, the SIP Extension updates the LED indication of the BLF key on the SIP phone.

The SIP extensions will indicate the following calls states for the outgoing and incoming calls on the monitored trunks:

Outgoing Calls	
Call State	**Description**
Trying	When an outgoing call is made through the monitored trunk.
Confirmed	When the external party answers the call and speech is established with the extension user, that is, the call is matured.
Hold	When the call on the trunk has been put on hold by the extension.
Available/Idle	When the SIP Extension user disconnects the call.

Incoming Calls	
Call State	**Description**
Early	When an indication is received from the Network that the external party is ringing.
Confirmed	When the incoming call is placed on the SIP Extension as the destination and speech is established with the extension user, that is, the call is matured.
Hold	When the call on the trunk has been put on hold by the SIP extension.
Available/Idle	When the SIP Extension user disconnects the call

SIP phones may differ in the BLF indication (LED color and cadence, text message display) they provide for the Call States. Refer to the manufacturer's documentation for BLF Indication supported on the SIP phones.

How it works

- An Standard SIP Phone is registered as a SIP Extension, with the extension number 3301.
- As the user of extension 3301 wants to monitor the trunk SIP - 01, BLF subscription is enabled on extension 3301 and SIP - 01 is assigned to the BLF key on the SIP phone.

- Extension 3301 makes an outgoing call to an external number 2630555. The BLF key will indicate the current call state of the SIP - 01 Trunk as “Trying” according to the LED indication supported by the SIP phone for this call state.

  If the SIP phone supports text message display for call states, each call state will be displayed on the phone.

- When the external party answers the call, the call between the SIP - 01 and SIP Extension 3301 gets matured. The BLF key will indicate the current call state of the SIP Trunk as “Confirmed” according to the LED indication supported by the SIP phone for this call state.

- When the SIP Extension 3301 disconnects the call, ANANT UCS will disconnect the call of external number and the BLF key will display the call state of the SIP as “Terminated” according to the LED indication supported by the SIP phone for this call state.

  Similarly, the BLF key configured on the SIP Extension 3301 will display the call states of the SIP - 01 trunk for incoming calls from external numbers.

Since multiple calls can be made through a single trunk, the BLF key will indicate the status of the first call detected by the system. When the first call is terminated, the status of the second call (if ongoing) will be indicated. Similarly the status of all subsequent calls will be indicated after the previous call is terminated.

How to configure

To provide BLF to SIP extension users, you must do the following:

- Enable **Busy Lamp Field Subscription** on the SIP Extensions you want to provide this feature. For instructions, see “Configuring SIP Extensions” under “Configuring SIP Extensions”.

- Assign a BLF Key for the trunk to be monitored on the SIP Phones registered as extensions. For instructions refer to the manufacturer’s documentation (Installation Guide/User Guide) for the respective SIP Phones.

- To monitor the SIP trunks, configure the BLF Key as per the table given below:

Trunk	User ID part in SUBSCRIBE	Remarks
SIP	SIPxx	Here, xx should be a valid SIP Trunk number, that is, from 01 to 99.

Call Back on Trunk Ports

The feature Call Back on Trunk Ports is used to respond to missed calls from particular numbers on the SIP trunks of ANANT UCS.

When Call Back feature is enabled on a trunk port, and there is a missed call on that trunk port, ANANT UCS determines if the calling number is eligible for a call back or not. It calls back the same number or an alternative number configured for that number, either from the trunk on which it was received or from a different trunk, depending on the configuration.

Employees at remote locations can use this feature to have ANANT UCS installed in their office call them back, thereby saving on charges wherever applicable.

How it works

For this feature to work:

- Call Back must be enabled on the desired SIP Trunks.
- The CLI of those callers whom the system should call back must be configured in the 'Incoming Number List'.
- The 'Call Back Timer' may be configured. When the caller disconnects within the Call Back Timer, the Call Back will be applied for that number.
- You must define 'Call Back on', that is, you must select whether the number which must be called back should be the same CLI number which the call was received or an alternative number.
- The number on which call back is to be made must be configured in the 'Outgoing Number List', if it is not the same CLI number or if it is an alternative number.
- You must select whether the call back should be made using the same trunk port on which the call was received or an Outgoing Trunk Bundle Group (OGTBG). If you select OGTBG, you must also configure the OGTBG.
- You may enable Least Cost Routing (LCR) on the OGTB if you want the system to select the least cost trunk for calling back the missed call number. Configure LCR accordingly.
- Select a 'Call Back Mode', that is, how the call should be routed when the call back is answered by the remote party; whether it should be routed through DISA or Operator.

Following is an example of a Call Back on a SIP Trunk, when the above parameters are configured.

- Caller A calls SIP Trunk -1.
- The system checks if the Call Back check box is enabled on the SIP Trunk -1.
- The check box is enabled.
- The system matches the CLI of A with the Call Back Incoming Number List assigned to SIP Trunk -1 to determine if the calling number is eligible for a call back.

- A match is found on Index 15 of the Call Back Incoming Number List.[76]
- The system waits for the period of the Call Back Timer (configurable, default: 10 seconds).
- A must disconnect before the expiry of the Call Back Timer so that the system can treat it as a Missed Call.
- If A disconnects within the Call Back Timer, the system applies Call Back for A’s number.
- The system checks the ‘Call Back on’ parameter, whether it has to call back the same number or an alternative number.
- If an alternative number is configured as ‘Call Back on’, the system checks the Outgoing Call Back Number List for the alternative number. As the CLI of A matches with the number on Index 15 of the Call Back *Incoming* Number List, the system checks Index 15 of the Call Back *Outgoing* Number List for the corresponding alternative number to this number.
- The system checks if the number is to be called from the same port or an OGTBG.
- If the same port is configured, the system will make a call to the number using SIP Trunk -1.
- If OGTBG is configured, the system will check if Least Cost Routing is enabled in the OGTBG and make the call back accordingly.
- When A answers the call,
- The system checks the type of Call Back Mode enabled on SIP Trunk -1 (the port on which the call back request was made).

  Following scenarios are possible:

  1. 'Pin Authentication - Multiple Calls' or 'CLI Authentication - Multiple Calls' is enabled as Call Back Mode on SIP Trunk -1.
     - A gets dial tone of ANANT UCS.
     - A can now reach any station or trunk of ANANT UCS from DISA Mode.
  2. 'CLI Authentication - Single Call’ is enabled as Call Back Mode on SIP Trunk -1.
     - A gets dial tone of another trunk of ANANT UCS.
     - A can make calls from the trunk.
  3. ‘Operator’ is enabled as Call Back Mode on SIP Trunk -1.
     - A gets Ring Back Tone.
     - The system lands the call on the Operator extension assigned to SIP Trunk -1.

  Read the topics “Auto Attendant”,“Direct Inward System Access (DISA)” and “Configuring 'Operator'” to know more about the call respective call logic.

76. *If the system does not find a match for the CLI of the caller in the Call Back Incoming Number List, the 'Call Back' feature will not be applicable and the call will be processed according to the normal incoming call logic.*

- *Since this feature is essentially for callers, they must be aware of its functioning to be able to use it, that is, disconnect the call within the Call Back Timer. If the caller does not disconnect within the Call Back Timer, the call will be processed according to the normal incoming call logic.*
- *ANANT UCS supports only one call back request at a time, for one trunk port. The second incoming call on that trunk port will be processed by the system as per normal incoming call routing.*
- *For call back requests made from an OGTBG, if any of its trunks is busy, ANANT UCS will support only the last call back request in the OGTBG. Previous requests will be processed as per the normal incoming call management logic.*

How to configure

For this feature to function, you must configure the following parameters on the SIP Trunks on which you want to use this feature:

- **Enable Call Back:** Select this check box on the desired SIP trunk port on which you want to activate the Call Back on Trunk Port feature. By default, this check box is disabled for all SIP trunks.
- **Call Back Timer:** This is the duration for which the system waits for the caller to disconnect the call after the system has found a matching number for the caller's CLI in the Call Back Incoming Number List.

  When the caller disconnects within Call Back Timer, the system applies Call Back on the trunk. If the caller does not disconnect within the Call Back Timer, the incoming call management logic is applied for the call on the trunk.

  The range of this timer is from 01 to 99 seconds. By default, it is set to 10 seconds.
- **Call Back Mode:** Select from the following options how a 'Call Back' call answered by the remote party should be routed:
  - **PIN Authentication-Multiple Calls:** The system will process the call as per DISA call logic - allow remote party to enter DISA mode with PIN-Authentication. On successful authentication (DISA Login) the user is allowed to make calls or use features as allowed to him/her.
  - **CLI Authentication-Multiple Calls:** The system will process the call as per DISA call logic, allowing the remote party to enter DISA mode with CLI Authentication-Multiple calls as authentication method and level of access.
  - **CLI Authentication-Single Call:** The system will process the call as per DISA call logic, allowing the remote party to enter DISA mode with CLI Authentication-Single call as authentication method and level of access. Refer the feature description for "Direct Inward System Access (DISA)".
  - **Operator:** When the remote party answers the Call Back call, the system will route the call to the Operator[77].

  By default, Operator is selected as the Call Back Mode.

77. *'Operator' is the station which is assigned to the SIP Trunk in the Trunk Feature Template. Refer "Trunk Feature Template" to know more.*

- **Call Back on:** For each Trunk you have set the Call Back feature, you must define ‘Call Back on’, that is, you must select whether the number which must be called back should be the same number from which the call was received or a different number.

  When missed call is eligible for call back (matches with Incoming Number list), the 'Call Back on' parameter determines the number on which the call back is to be made, that is, whether on the same number from which the missed call is received or on a different number.

  In countries where CLI received on trunks can be dialed out without any modification, you may select ‘CLI Number’ as ‘Call Back on’ option.

  In countries where CLI received on trunks can be dialed only after appropriate modification, you may select “Alternate Number’ as the ‘Call Back on’ option. You may also select ‘Alternate Number’ as Call Back on when you want the call back to be made to a different number.

- Call Back **Incoming Number List:** This is the list of numbers that are eligible for Call Back. The system checks the CLI of the caller with this list to determine if the caller is eligible for a call back.

  The system compares the number string configured in the Call Back Incoming List with the number string received as CLI.

  Number string configured in the 'Call Back Incoming Number List' shall be compared with the actual received CLI.

  The number string configured in the Call Back Incoming Number List may be shorter than the number string received as CLI, but only if the programmed number string completely matches with the received CLI from the right towards left, the system will consider it as a complete match.

  For example, if the configured string is 263055 and the number string received in the CLI is 2652630555, the system will consider it a complete match. If the received CLI 912652630555, the system will consider this caller too as eligible for a call back. Thus any CLI received with 263055 as the last 7 digits will be considered as match found.

  By default, ‘Number List’ 15 is assigned to all trunk port types as Call Back Incoming Number List. You may configure this list for all port types, or you may configure another Number List and assign it to the particular SIP trunk.

  Refer the topic “Number Lists” for instructions on how to configure the Number List.

- Call Back **Outgoing Number List:** When the system finds a missed call eligible for a call back, it will make the call back on the basis of the Call Back on option you selected and the Outgoing Number List you configured.

  If you selected ‘CLI Number’ as “Call Back on’ option, you do not need to configure the corresponding outgoing number for the CLI received.

  However, if the CLI received needs to be modified before being dialed out, then configure the modified CLI in the Outgoing List as the corresponding outgoing number for the CLI received.

  The modified CLI or the Alternate number should be configured at the same index number as the index number at with the received CLI is configured in the Call Back Incoming Number List. For example, for the received CLI number string configured at Index 15 in the Call Back Incoming Number List, the corresponding modified CLI/Alternate number string should be configured at the same Index, 15, in the Call Back Outgoing Number List.

When the CLI received matches with the number string configured at Index 15 of the 'Call Back Incoming Number List', the call back will be made using the (modified/Alternate) number configured at Index 15 of the 'Call Back Outgoing Number List'.

By default, ‘Number List’ 16 is assigned to all trunk port types as Call Back Outgoing Number List. You may configure this list for all SIP Trunks, or you may configure another Number List and assign it to the particular trunk.

Refer the topic “Number Lists” for instructions on how to configure the Number List.

If you have selected ‘Alternate Number’ as ‘Call Back on’ option, but do not want to provide alternative numbers to call back particular callers (that is, CLI received), in such a case, configure the CLI of these callers in the Incoming Number List but keep the corresponding index numbers in the Outgoing Number Lists blank.

- **Call Back from:** This parameter determines the trunk port to be used to make call back.The call back can be made using the same port or an Outgoing Trunk Bundle Group (OTGTBG). Select ‘Same port’ if you want the call back to be made using the same trunk on which the missed call was received. If you select OGTBG, the call back will be made using the OGTBG, which you have defined.
- **OGTB Group** for Call Back**:** If you selected OGTBG for making the call back in the previous parameter, you must assign the OGTBG that must be used in this parameter.

  By default, OGTBG 01 is selected for Call Back.

  If you want the system to select the lowest cost trunk for making the call back, enable Least Cost Routing on the OGTBG that you define here for Call Back.

To know more, refer “Call Back” in “Configuring SIP Trunks”.

Call Budget on Extension

Call Budget is a cost control feature that allows you to keep a tab on the total cost of phone call made by extension users.

With this feature, each extension can be allotted a 'budget' limit for outgoing calls, which is automatically reloaded at the start of every month.

Long distance calls form a major part of the increased cost of telephone calls. Though excessive use or misuse of long distance dialing can be restricted using Toll Control, there may be extension users whose nature of work requires them to make long distance calls. Instead of denying them the facility, their telephone bill can be limited to a certain amount using Call Budget.

With a Call Budget allotted to the extension, the user is free to make calls as long as s/he does not cross the budget limit. Once the user exceeds the budget limit, the extension can be denied access to long distance dialing.

The extension user can be assigned a fresh budget, after which s/he can resume making long distance calls.

Call Budget can be enabled on all the extensions as well as on selected extensions. Each extension can be assigned a different amount depending on user requirement.

How it works

When an extension allotted Call Budget makes a call,

- The system checks the current call budget amount of the extension.
- If the consumed amount is within the budget limit allotted to the extension,
    - The system allows the extension to make the call as per the "Toll Control Levels" assigned to it.
    - After the call ends, the system calculates and adds the call amount to the extension's account. Thus it calculates and updates the total cost of calls made from the phone.
- If the consumed amount exceeds the budget limit allotted to the extension,
    - The system considers this as Call Budget exhausted.
    - The system allows the extension to make the call as per the Calls allowed when Call Budget is consumed assigned to the extension.
    - After the call ends, the system calculates and adds the call amount to the extension's account.
- Until a new Call Budget is allocated to the extension user, the extension user can make calls only as per Toll Control assigned for the Call Budget Consumed state.
- Once a new Call Budget is allocated, the extension user can make calls as per the "Toll Control" assigned to the extension.
- If the budget exceeds anytime during the month, and if no fresh budget amount is allotted, the system allows calls to be made as per the Calls allowed when Call Budget is consumed till the end of the month.

From the 1st day of the following month, the system automatically reloads the budget amount. The extension can now make calls.

- The Call Budget allotted to extension is valid for one month. The system automatically reloads the budget at the start of every month.
- The budget amount can be changed or allotted afresh to extensions from the System Administrator (SA) mode, at any time. The Call Budget allotted by the SA will be reloaded in the following month.

- *Call Budget is not based on real time (online) call cost calculation. ANANT UCS calculates the call cost only after the call has ended.*
- *So, if the Call Budget allotted to an extension user gets exhausted in the middle of a call, the call will not get disconnected, though the budget exceeds. To prevent this from occurring, the System Engineer may program the "Call Duration Control (CDC)" feature.*
- *Call Budget is dependent on precise Call Cost Calculation. So, SMDR parameters and long distance codes must be programmed properly to prevent errors in calculation.*
- *ANANT UCS will calculate cost of phone calls made by extension phones even when no call budget is allocated[78].*

How to configure

The working of this feature is controlled by the parameters: **Call Budget** check box, **Calls allowed when Call Budget is consumed** and **Preset Call Budget Amount** (this parameter is applicable only for the Hotel Mode).

Call Budget check box and Calls allowed when Call Budget is consumed

To enable Call Budget feature on an extension, the System Engineer must enable the Call Budget check box and define the Calls allowed when Call Budget is consumed in the Station Basic Feature Template assigned to the extensions.

In the default Station Basic Feature Template 01 assigned to all extensions of the ANANT UCS, the Call Budget check box is disabled and Calls allowed when Call Budget is consumed is set to 'No Calls'.

If Call Budget is to be allowed to all extensions, simply enable the check box in the default Station Basic Feature Template 01 and select the desired option for Calls allowed when Call Budget is consumed.

However, if Call Budget is to be allowed to selected extensions, then prepare a separate Station Basic Feature Template with the Call Budget check box enabled and the Calls allowed when Call Budget is consumed set to the desired option. Now, apply this template on extensions that are to be allowed this feature.

Refer the topic "Station Basic Feature Template" for detailed instructions for programming a feature in the template and assigning templates to extensions.

78. *Based on the feature "Call Cost Calculation (CCC)"*

Preset Call Budget Amount[79]

The amount of '9999' is set as default Call Budget in the system. This value may be changed by the System Engineer according to the organization's practices. For example, if the organization wants to allocate a fixed amount of $10 to all extension users, the Call Budget value can be set to '10'.

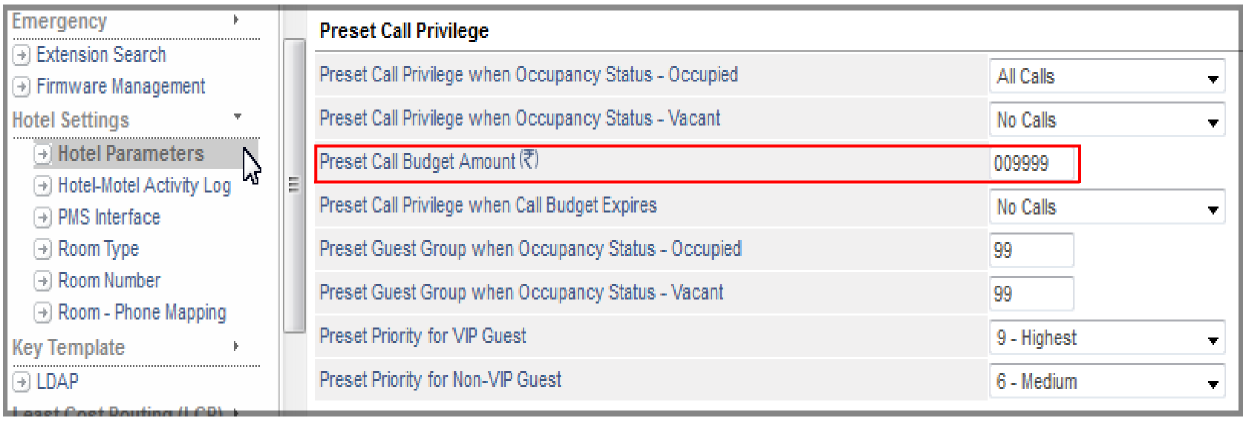
The new Call Budget set by the System Engineer will be considered as the Preset Call Budget amount. This amount will be allocated at the start of every month to all extensions having Call Budget feature in their Station Basic Feature Template.

Further, the System Administrator (SA) can override the Preset Call Budget amount set by the System Engineer, and allot call budgets on an extension-by-extension basis. For example: allotting higher amount to extensions of senior managers, Marketing, Sales, Exports departments, and lower amount to extensions that are less likely to make long distance calls frequently.

The amount may be greater or lesser than the default amount set by the System Engineer. The Call Budget amount allotted by the SA will be reloaded at the start of every month on the extension. For instructions refer 'How to Use' later in this section.

Changing the Preset Call Budget Amount

- Login as System Engineer.
- Under **Configuration**, click **Hotel Settings**.
- Click **Hotel Parameters**.
- Scroll to **Preset Call Privilege** and change the **Preset Call Budget Amount** to the required value.



- Click **Submit**.

- *The amount programmed as Preset Call Budget is to be considered as the local currency.*
- *At the time of installation, when the SE selects the Region Code (country code) and defaults the system, the related Currency Code is applied.*
- *The currency symbol will not be displayed on the Operator's phone, on account of the limited number of characters that can be displayed.*
- *The local currency symbol will appear at the relevant places in the outgoing SMDR reports.*

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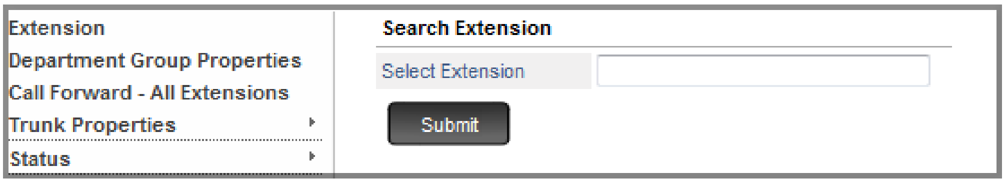
79. *Applicable only for the Hotel Mode.*

How to use

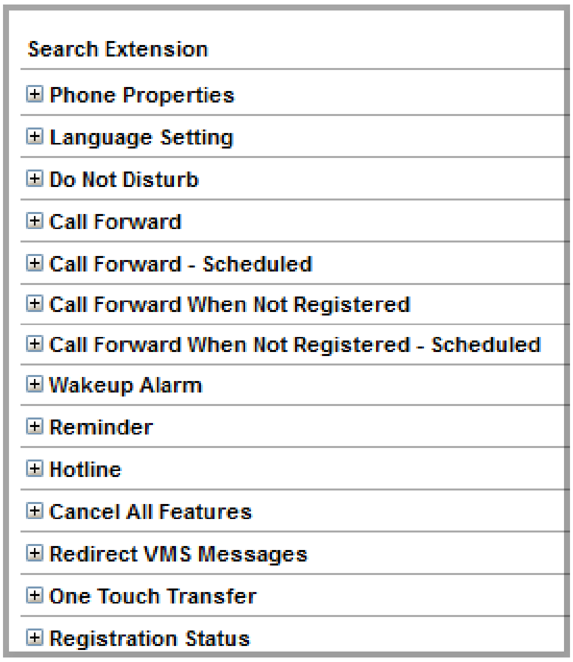
Call Budget amount can be allotted to extensions from the System Administrator mode.

Assigning Call Budget to Extensions

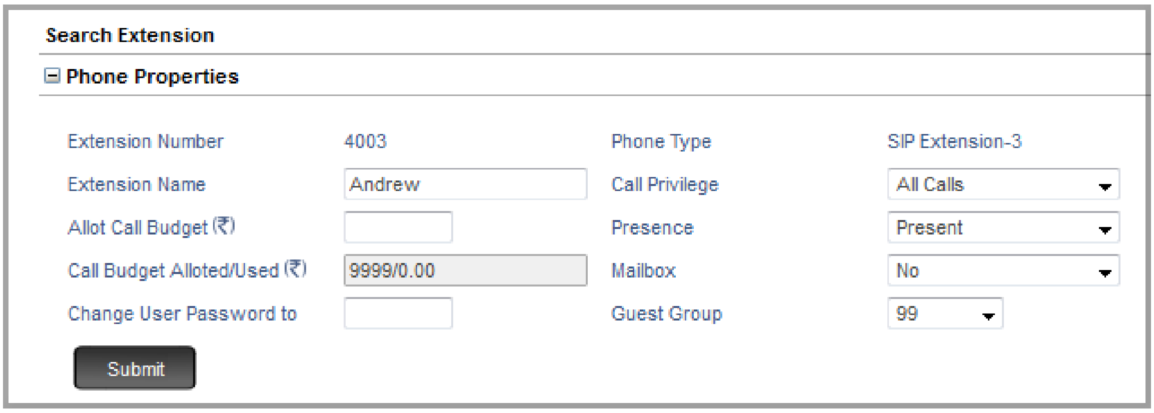
- Login as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.
- The searched extension user details appear on your screen.



- Click **Phone Properties** to expand.



- In **Allot Call Budget**, enter the amount you want to assign to the user as budget limit for outgoing calls.

  To re-assign a new amount before the previous balance is consumed, make sure you add the available balance to the new amount. Enter this amount in Allot a Call Budget.

  For example, if you have allotted an amount is Rs.1000 and the consumed amount is Rs.600. The available balance is Rs.400. Now, if you want to assign a new amount of Rs.500. In Allot a Call Budget you must enter 900 (Balance + New = 400 + 500).

- The **Call Budget Alloted/Used** displays the amount allotted to the user as well as the call budget amount consumed by the user for making outgoing calls.

- Click **Submit.**

- To allot call budget to another extension, follow the same instructions as above.

Call Budget on Trunk

Call Budget on Trunks is an expense control feature of ANANT UCS that allows you to keep track of the cost of phone calls made from the different SIP Trunks of ANANT UCS.

With this feature, each trunk can be allotted a 'budget' limit for outgoing calls. This budget limit can be programmed to be reloaded manually each time it is exceeded or at a scheduled date, either daily or at a particular date of the month.

There are three types of Call Budget limit that can be set on the trunks:

- **Amount:** In this type of Call Budget, a fixed amount is assigned to the trunk. By default the amount of 999999 (to be considered in the local currency) is set as Call Budget Amount on trunks. With Amount-based Call Budget you can control the actual expense incurred on making calls from a trunk.

- **Minutes:** In this type of Call Budget, a fixed number of Minutes are assigned to the trunk. By default, 999999 minutes are assigned as Call Budget Minutes on trunks. This type of Call Budget is useful when the Service Provider offers 'Free' minutes. For example, the Service Provider allows the customer to make calls for the first 1000 minutes every month. This offer can be availed of by programming Minutes-based Call Budget on the trunk port.

- **Number of Calls:** In this type of Call Budget, you can define the maximum number of calls that can be made from a trunk. By default, the maximum number of Call Budget - Calls is set to 9999 calls on the trunks. This type of Call Budget is useful when the Service Provider offers a certain number of free calls or a certain number of free calls for a fixed period. For instance, the Service Provider offers 150 free calls per month.

With a Call Budget allotted to a trunk, the users can make calls from the trunk as long as the budget limit set for the trunk—Amount or Minutes or Maximum number of Calls—is not crossed. Once the budget limit exceeds calls will not be routed through this trunk.

The consumed Budget can be reset, after which the trunk becomes functional again and allows outgoing calls to be made. The consumed Call Budget can be reset manually, that is, anytime, as required/desired, or on a scheduled date either daily or on a particular date of the month.

By default, *Call Budget Type - Minutes* is enabled on all the SIP trunks for *300 Minutes*. You can also change the configurations as per your requirement.

How it works

Call Budget can be enabled on the desired SIP trunks. Each trunk can be assigned a different Call Budget, depending on the requirement of the users. When Call Budget is enabled on a SIP trunk, for each outgoing call,

- The system checks the type of Call Budget set on the trunk - Amount, Minutes or number of Calls.
- It checks the Call Budget consumed.

Call Budget- Amount

- When Amount-based Call Budget is selected, the Amount should be assigned to the trunk.

- At the end of each outgoing call made from the trunk, the system will calculate the cost of the call on the basis of the Pulse Rate Type programmed. The system will thus calculate the total amount consumed after the end of each call. Refer the topic “Call Cost Calculation (CCC)” to know more.

Call Budget - Minutes

- When Minutes-based Call Budget is set, the total minutes for which calls will be allowed from the trunk port must be defined.

- With the number of Minutes defined, at the end of each call, the system will calculate the duration of the call on the basis of the units programmed in the Pulse Rate. The system will calculate the consumed minute on the basis of the duration of the call. Refer the topic “Call Cost Calculation (CCC)” to know more.

Call Budget - Number of Calls

- When the Call Budget is based on 'Number of Calls', the maximum number of calls to be allowed from the trunk port is to be defined.

- With the number of calls programmed, the system will maintain a count for the number of matured outgoing calls made from that trunk port.

- Thus for each matured call, the Number of Calls-Count is incremented, irrespective of the actual duration of the matured call.

When the assigned 'cost' or 'minutes' or 'number of calls' assigned to trunk is exhausted, ANANT UCS will:

- print 'system activity log'.
- bar or limit outgoing calls from such trunks.
- play an Error Tone to the extension users who attempt to access such trunks.
- However, incoming calls will remain unaffected, and will be allowed on these trunks.

The consumed Call Budget Amount/Minutes/Calls can be reset manually at any time from the System Administrator mode or the System Engineer mode or can be programmed to be automatically reset either daily or on a particular date of the month.

The current Call Budget Amount/Minutes/Calls limit can be changed from the System Administrator (SA) mode, at any time. If scheduled reset of consumed Call Budget is programmed, then the Call Budget allotted by the SA will be reloaded on the scheduled date.

Once a new Call Budget is allocated to the trunk, outgoing call facility is resumed on the trunk.

- *Call Budget on Trunks is not based on real time (online) call cost calculation. ANANT UCS calculates the call cost only after the call has ended.*

- *If the Call Budget allotted to a Trunk Port gets exhausted in the middle of a call, the call will not be disconnected, though the budget is exceeded.*

- *Call Budget on Trunks is dependent on precise Call Cost Calculation. So, SMDR parameters and long distance codes must be programmed properly to prevent errors in calculation.*

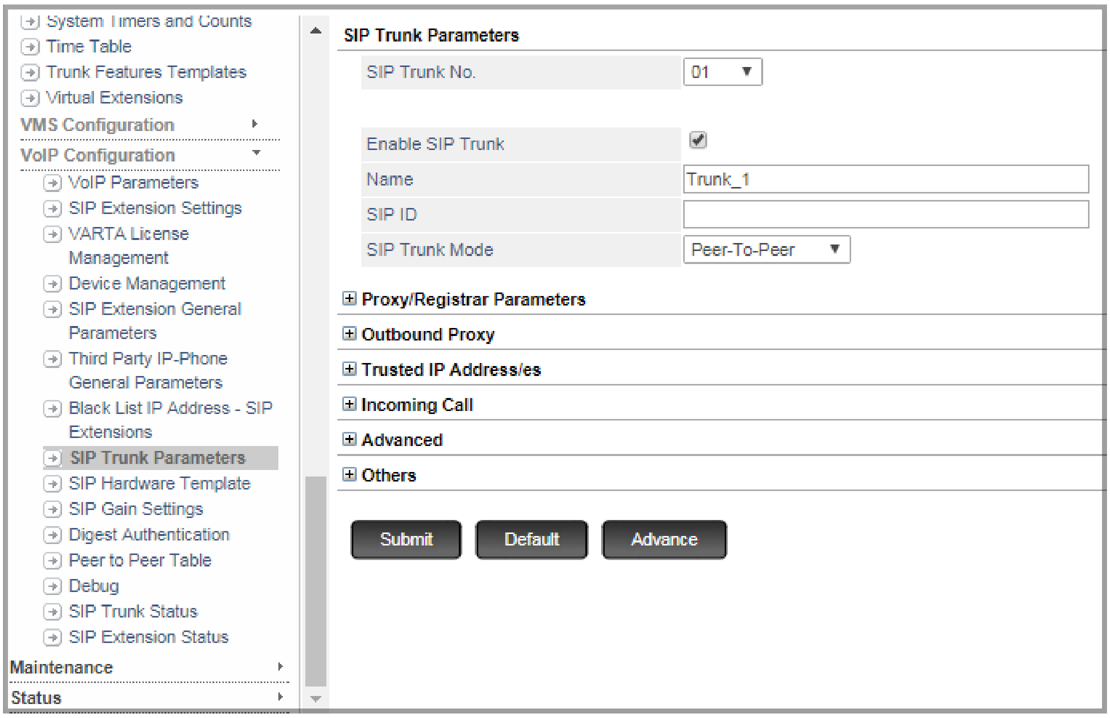
- *ANANT UCS will calculate cost of phone calls made by the trunks even when no call budget is allocated*[80].

How to configure

Call Budget on Trunks is to be programmed in the SIP Trunk Parameters on which you want to enable this feature.

Configuring Call Budget on Trunks

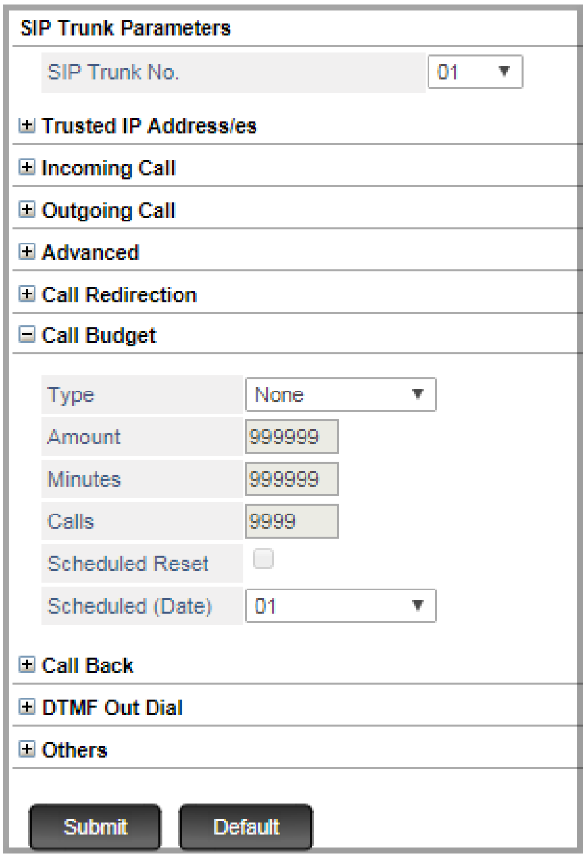
- Login as System Engineer.
- **Call Budget** parameters must be programmed in the **SIP Trunk Port Parameters** page you want to configure. For instance, to configure Call Budget on a SIP Trunk,
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Trunk Parameters**.



- Select the desired SIP Trunk number.

80. *Based on the feature "Call Cost Calculation (CCC)".*

- Click the **Advance** button and configure the following parameters:



- **Call Budget:** By default Call Budget is enabled, if you wish to change the default settings or disable it, configure the following parameters for this trunk port:
  - **Type:** Select the type of Call Budget on Trunk, that is, Amount or Minutes or Calls to be applied on this trunk port. By default, Minutes is selected as the Call Budget type.To disable select Type as None.
  - **Amount:** If you selected 'Amount' as the Call Budget Type, enter the Budget Amount in this field. By default the Amount is set to 999999.
  - **Minutes:** If you selected 'Minutes' as the Call Budget Type, enter the number of Minutes in this field. By default the number of minutes is set as 000300.
  - **Calls:** If you selected 'Calls' as the Call Budget Type, enter the number of Calls in this field. By default the number of calls is set to 9999.
  - **Scheduled Reset:** Enable this check box if you want the Call Budget Amount/Minutes/Number of Calls to be reset on a particular date of every month.
  - **Scheduled (Date):** Select the date of the month (Daily or 1-31) on which you want the Call Budget Amount/Minutes/Number of Calls to be reset every month. You may select 'Daily' if your plan suggests so.
- Click **Submit.**

The consumed Call Budget on trunk can be reset from the System Engineer mode as well as the System Administrator mode manually at any time, referred to as Manual Reset.

Manual Reset of Call Budget on Trunk

You may configure manual reset of the consumed Call Budget on the SIP trunk.

- Click **SIP Trunk Status** under **VoIP** Configuration.
- Select the **Reset Consumed Budget** check box of the SIP Trunk for which you want to reset the consumed Call Budget.

Similarly, you can reset Call Budget on Trunks from the SA mode.

*You can also view the SIP Trunk Status from the **Status** link. To view, click the SIP Trunk Status link under Status.*

Call Chaining

Call Chaining is when an external/internal call transferred by the Operator to another extension or external number is made to return to the Operator's extension after the conversation between the caller and the extension/external number to which it is transferred has ended.

Call Chaining is useful in situations where the Operator intervention is required after the transferred call has ended. For instance:

- The caller needs to take an appointment or requires some information from the Operator after talking to the desired extension.

- A marketing executive who calls his supervisor to consult on a technical problem needs to be informed about his travel itinerary and ticket booking by the Operator. The Operator can transfer the call to the supervisor, and use Call Chaining to retrieve the call once the conversation has ended to give the information to the executive.

Call Chaining can be set for multiple calls.

Also refer "Call Transfer", "Call Park".

How it works

- A is an External Caller
- B is an extension.

- A calls a Trunk of ANANT UCS.
- The Operator answers the call.
- The Operator sets Call Chaining and transfers the call to B.
- B disconnects the call with A, but A is still connected.
- The call comes back to the Operator, if the Operator is free.
- Now A is in speech with the Operator.

- If A disconnects the call with B, the call will be released. It will not return to the Operator.

- If the Operator is busy, A will be played music on hold for the duration of the Call Park Release Timer.
- If the Operator is busy and the Timer elapses, the call will be released.

Call Chaining can be performed when call is transferred from any extension to another extension or external number.

How to configure

The only configuration involved in the functioning of this feature is assigning this feature to a DSS key which has LED and configuring, if necessary, the Call Park Release Timer.

Refer the topic "DSS Keys Programming" and "Call Park" for instructions.

How to use

For Extended IP Phone Users

- Go OFF-Hook to answer the incoming call.
- The caller requests for an extension/trunk.
- Press DSS Key assigned to Call Chaining.
  OR
- Press the ‘Transfer Key’ and dial **1050**.

  You get confirmation tone and the message 'Party Chained' on your phone's display. If DSS Key is used, the LED of the key will glow.

- Dial the requested extension/external number.

- You get Ring Back Tone.
- The called party answers.
- Go ON Hook to transfer the call.
- When extension/trunk disconnects, your extension rings.
- Go OFF Hook. You get connected with the caller.
- Go Idle after the conversation ends, or you get dial tone after 3 seconds.

Call Cost Calculation (CCC)

ANANT UCS can calculate the cost in amount for the external calls made by the extension users.

The call cost is calculated according to the tariffs offered by the ITSP's. Different types of tariffs are provided by different ITSP's. These may be:

- Time Based Tariffs, for example, free calling from 11.00 p.m. to 6.00 a.m.
- Special Day Tariffs, for example, subsidized calls rates on August 15.
- Tariffs according to the Geographical Distances, for example, calling rates for UAE differ from calling rates to Canada.

Using the Call Cost Calculation feature, the system calculates the cost of the call according to the duration of the call interpreted in terms of units and the charge applicable for the duration as per the service provider.

How it works

For this feature to work,

- you must get the tariff details from the ITSP's and configure the same.
- determine the outgoing trunk for the calls according to the type of calls, namely, local calls, national calls or international calls.

ANANT UCS will calculate the cost of the call as follows:

- When the call is made from a trunk, the system checks the **Call Cost Calculation Pulse Rate Option,** 1 to 4 assigned to the trunk, on the basis of the **Call Cost Calculation Time Schedule** configured for the outgoing call.

  Each Call Cost Calculation Pulse Rate option contains a **Pulse Rate Type for the Pulse Rate**, which is assigned in the Area Code Table.

- The system matches the Number dialed by the extension user with the **Area Code Table** configured in the system. When the area code matches with an entry in the table, the system obtains the **Pulse rate type** configured for the Call Cost Calculation Pulse Rate option assigned to the trunk.

- This **Pulse Rate type** obtained from the **Area Code Table** is checked in the **Pulse Rate table** to obtain the corresponding **duration** and **cost** to be applied for the call duration. The Pulse Rate Table may be the **Normal Pulse Rate Table** or **Discounted Pulse Rate Table**, depending on the day of the call. The system uses the built-in RTC to determine the day.

- The **Pulse Rate Type** applied (duration and cost) is divided into two parts for each time zone:
  - First unit.
  - Additional units.

  - Number of Units is derived from the pulse rate at the time of the call and duration of the call. System acquires the pulse rate type and call duration with the help of RTC.

    Total Units = First Unit + Additional Unit.
    If the call duration is less than the pulse rate of the first unit then additional unit is zero.
    Call Units = (Call duration in seconds)/(Pulse rate in seconds).

- If the duration of the call is less than or equal to one unit,

  Cost of Call = Cost of First Unit + Service Charge as applied.

- If the duration of the call is more than one unit,

  *Cost of Call = [Cost of First Unit + (Number of Additional units x Cost for Additional units)] + Service Charge a*s applied.

- The system applies the rates as configured in the Normal Pulse Rate Table for all the days, except when it detects a day configured in the Discounted Pulse Rate Schedule. These are special days when special Tariffs are offered.

  For the days configured in the Discounted Pulse Rate Schedule as special days, the system checks the duration and cost of the First unit and the Additional units configured in the Discounted Pulse Rate Table.

ANANT UCS uses the **Cost of the Call** for SMDR. This cost is deducted from the Call Budget (Amount), if allotted to the trunk and also from the Call Budget, if assigned to the extension users.

The logic for call cost calculation is explained with the help of an example:

- An Outgoing Call is made using trunk, SIP Trunk-1.
- On SIP Trunk-1, Call Cost Calculation is configured as follows:
  - Call Cost Calculation Pulse Rate option = 1
  - Call Cost Calculation Schedule
    - Time Zone 1 = Start Time: 00:00, End Time: 22:00
    - Time Zone 2 = Start Time: 22:01, End Time: 23:59
  - Assign Trunk Feature Template 1 to SIP Trunk-1. Configuring the Call Cost Calculation parameters in the Trunk Feature Template as follows:

Template No.	Call Cost Calculation Pulse Rate Option	Call Cost Calculation Time Schedule											
		T1				T2				T3			
		Start Time		End Time		Start Time		End Time		Start Time		End Time	
		HH	MM	HH	MM	HH	MM	HH	MM	HH	MM	HH	MM
001	1	00	00	22	00	22	01	23	59				

  - Area Code Table is configured as:

Index	Area Code	Name	Ignore Digit Count	Pulse Rate Type for Pulse Rate			
				Option - 1	Option - 2	Option - 3	Option - 4
001	26	Local		03	06	09	10
002	09	SIP		05	03	07	08

- The Normal Pulse Rate Table is configured as:

Pulse Rate Type		Time Zone T1		Time Zone T2		Time Zone T3		Time Zone T4	
		First Unit	**Add. Unit**	**First Unit**	**Add. Unit**	**First Unit**	**Add. Unit**	**First Unit**	**Add. Unit**
01	Duration (sec)	180	180	60	30	90	30	120	60
	Cost	002.00	002.00	002.00	002.00	002.00	002.00	002.00	002.00
02	Duration (sec)	300	300	300	300	300	300	300	300
	Cost	001.00	001.00	001.00	001.00	001.00	001.00	001.00	001.00
03	Durations)	30	30	30	30	30	30	30	30
	Cost	001.00	001.00	001.00	001.00	001.00	001.00	001.00	001.00
04	Duration (sec)	45	45	45	45	45	45	45	45
	Cost	001.00	001.00	001.00	001.00	001.00	001.00	001.00	001.00
05	Duration (sec)	180	180	180	180	180	180	180	180
	Cost	003.00	003.00	003.00	003.00	003.00	003.00	003.00	003.00
:	Durations)	:	:	:	:	:	:	:	:
	Cost	:	:	:	:	:	:	:	:
32	Duration (sec)								
	Cost								

- With this configuration, ANANT UCS will calculate the call cost as follows:
  - An Outgoing Call is made by an extension user, to the number ‘2630555’ through the trunk, SIP Trunk-1 at 20:10 hours. ANANT UCS will check the Call Cost Calculation parameters assigned on the trunk and determine the Time Zone as per time of the call. The system will also check the corresponding Pulse Rate Option configured on the trunk.

    In this example, Time Zone for SIP Trunk-1 at 20:10 Hours would be Time Zone 1, and the Pulse Rate Option for SIP Trunk-1 is ‘1’.
  - ANANT UCS will match the dialed number ‘2630555’ in the Area Code table. A best match is found with the entry configured at index 001 in the Area Code Table.

    As per the Area Code Table, the Pulse Rate Type ‘03’ is programmed in ‘Pulse Rate Option 1’ for the matching entry (at Index 001).

    (However, if SIP Trunk-1 would have been assigned Pulse Rate Option ‘2’, the Pulse Rate type ‘06’ would have been selected as shown in Area Code Table)
  - Finally, for Pulse Rate Type ‘03’ ANANT UCS will check the Normal Pulse Rate Table. ANANT UCS will consider the Cost for the First Unit as 001.00 (As ` or $ as per applicable currency) for the duration of 30 seconds and for the additional unit also, the cost will be considered as 001.00 for the duration of 30 seconds. This data will be used for calculating the total cost of call based on the total duration of the call.

Similarly, when there are Special Tariffs offered on certain days, the system will check the Discounted Pulse Rate Schedule and the Discounted Pulse Rate Table.

The days on which the special rates are to be applied must be configured in the Discounted Pulse Rate Schedule. The duration and cost of the First unit and the Additional units must be configured in the Discounted Pulse Rate Table according to the Time Zones.

How to configure

To be able to use Call Cost Calculation, you must do the following:

- Define the Unit and Service Charge on the basis of which call cost is to be calculated.
- Assign Call Cost Calculation Pulse Rate Option and the Call Cost Calculation Schedule on the trunks on which you want to apply this feature.
- Configure the Pulse Rate Types for the Pulse Rate Option you assign to the trunk.
- You may configure different Pulse Rate Types:
    - For Normal Days configure the pulse rate in the Normal Pulse Rate Table.
    - For Special days configure the pulse rate in the Discounted Rate Table. If you configure the Discounted Pulse Rate Table, you must also configure the Discounted Pulse Rate schedule.
- Configure the Area Code Table.

Configuring Call Cost Calculation

To configure call cost calculation parameters

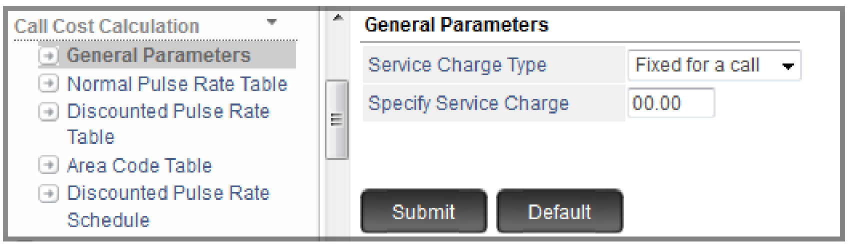
- Login as System Engineer.
- Under **Configuration,** click the **Call Cost Calculation**.

Service Charge

By default, no Service Charge is applied on call cost by the system. Service Charge on call cost is generally applied in Hotels and other organization which charge users for the calls made by them.

If you want to apply Service Charge,

- Click **General Parameters.**



- In the **Service Charge Type** field, select the type of service charge you want to apply from the options:
    - **Fixed for a call:** A fixed amount is added as service charge to every call regardless of the cost of that call.

      If you select this option, you must define the Amount to be added as service charge in the **Specify Service Charge** field.
    - **per Unit:** service charge is added to each unit of the call. For example, if a call worth 10 units was made, the service charge will be applied on each of the 10 units, instead of the one time service charge as in the case of Fixed service charge.

      If you select this option, you must define the amount to be added as service charge on each unit in the **Specify Service Charge** field.
    - **Percentage of call cost:** A percentage of the cost of the call is added as a service charge for that call.

      If you select this option, you must define the percentage in the **Specify Service Charge in %** field which appears.
- Click **Submit.**

Applying Call Cost Calculation on Trunks

- To assign **Call Cost Calculation Pulse Rate Option** and configure the **Call Cost Calculation Schedule** on the trunks, go to *Configuration*, and configure these in the "Trunk Feature Template" assigned to the SIP trunks. See "Configuring Trunks" and "Configuring SIP Trunks" under *Configuring ANANT UCS* for instructions.

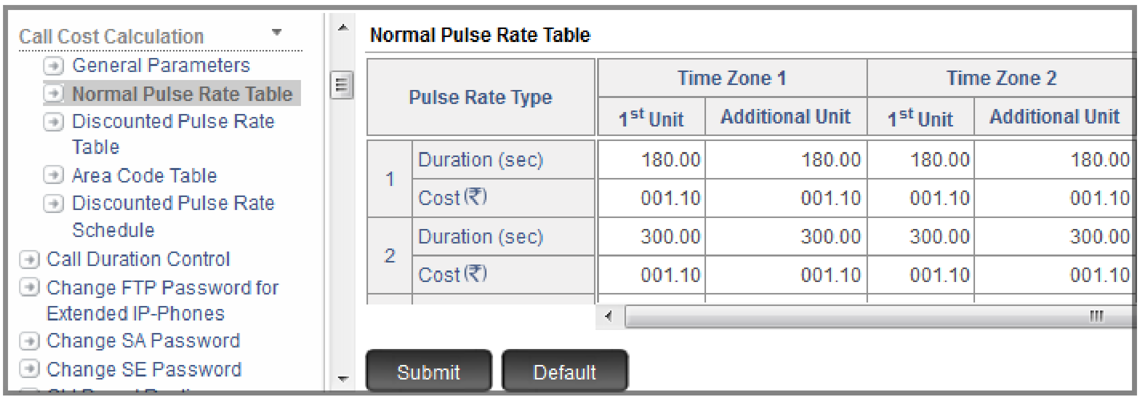
Configuring Pulse Rate Types

Generally service providers offer discounted call rates for special days. To take care of this, ANANT UCS provides two types of Pulse Rate Tables: Normal and Discounted.

If your service providers do not offer any special rates, you may skip configuring the Discounted Pulse Rate table.

To configure the Normal Pulse Rate table,

- Under **Configuration,** click **Call Cost Calculation**.
- Click **Normal Pulse Rate Table**.



- Configure the **Pulse Rate Type** with rates for the **First Unit** and the **Additional Unit**.

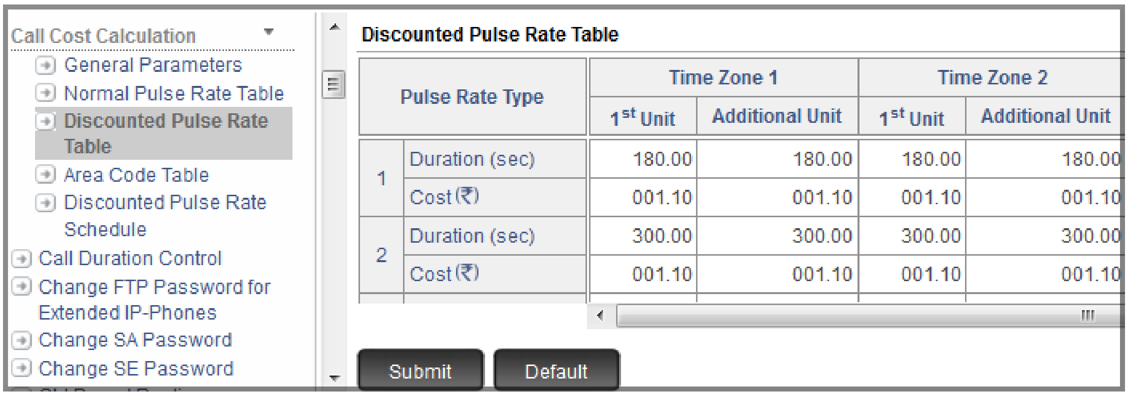
  Generally service providers offer different call rates for different types of calls, for example: local, national, international. You can configure different Pulse Rate Types for different types of calls. Thus, each Pulse Rate Type can have different rates for the First and the Additional unit.

  The Pulse Rates offered by service providers may vary according to the time of the day. In such cases, you will need to configure the **Call Cost Calculation Schedule** for the trunk, by dividing the day into Time Zones, from 1 to 4, as required, to match the time of the pulse rates offered by your service provider.
  If you have configured Time Zones for the Call Cost Schedule on a trunk, you may define the different Pulse Rate Types for each Time Zone.

- Click **Submit.**

To configure the Discounted Pulse Rate table,

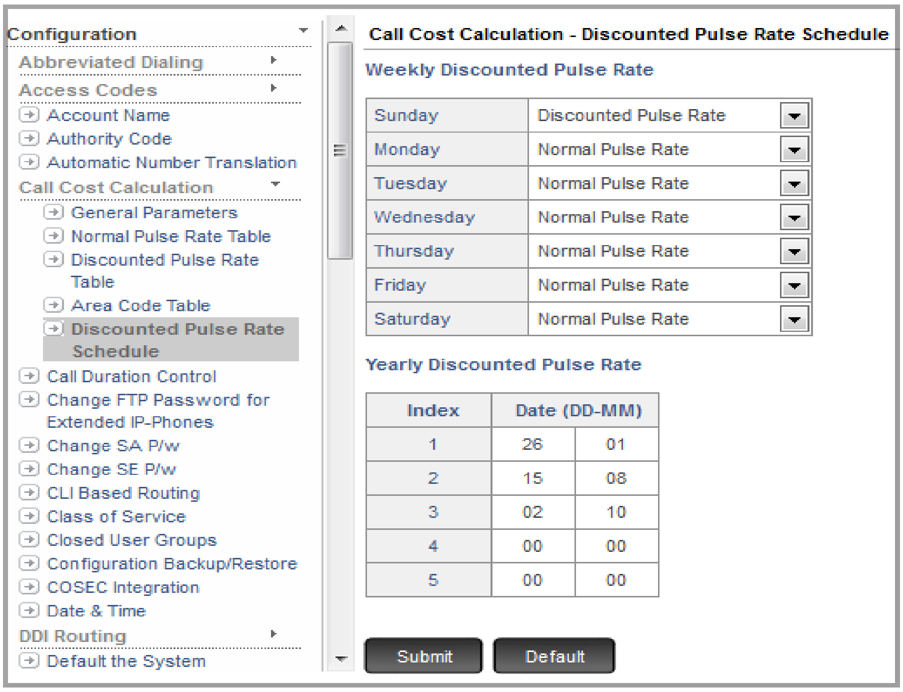
- Click the **Discounted Pulse Rate Table.**



- Configure the **Discounted Pulse Rate Type** with rates for the **First Unit** and the **Additional Unit**, as required.

- Click **Submit.**

- Click the **Discounted Pulse Rate Schedule**.



- Define **Weekly** and **Yearly** special days on this page.

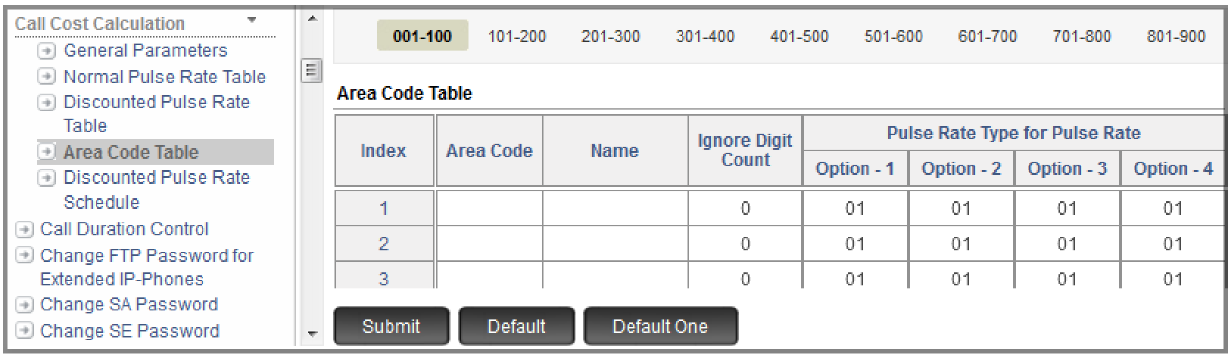
- Click **Submit.**

Configuring Area Code Table

The pulse rate of a call depends on the destination number dialed. Generally, pulse rate varies according to the type of the call: Local, national, international. ANANT UCS enables you to configure Pulse Rates according to Area codes.

To configure Pulse Rates for Area Codes,

- Click **Area Code**.



- In the **Area Code** column, enter the number strings (prefix) of the Area Codes, country codes, local numbers. You can configure as many as 999 area codes in the table.

- You may enter a **Name** to tag each Area Code.

- Do not configure the Ignore Digit Count. This parameter is relevant only for Service Provider Based Least Cost Routing.
- Different service providers offer different pulse rates for the same type of calls. To take care of this, ANANT UCS allows you to assign different Pulse Rate Options for each area code.
- For each area code, configure the Pulse Rate to be followed for the desired Pulse Rate Options in the **Pulse Rate Type for Pulse Rate (Option 1 to 4)** column.
- Click **Submit**.

The **Default Area Code Table for the Region - USA** is listed below:

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
1	1201	NJ	2	0
2	1202	DC	2	0
3	1203	CT	2	0
4	1204	Manitoba	2	0
5	1205	AL	2	0
6	1206	WA	2	0
7	1207	ME	2	0
8	1208	ID	2	0
9	1209	CA	2	0
10	1210	TX	2	0
11	1212	NY	2	0
12	1213	CA	2	0
13	1214	TX	2	0
14	1215	PA	2	0
15	1216	OH	2	0
16	1217	IL	2	0
17	1218	MN	2	0
18	1219	IN	2	0
19	1224	IL	2	0
20	1225	LA	2	0
21	1226	Ontario	2	0
22	1228	MS	2	0
23	1229	GA	2	0
24	1231	MI	2	0
25	1234	OH	2	0
26	1239	FL	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
27	1240	MD	2	0
28	1242	Bahamas	2	0
29	1246	Barbados	2	0
30	1248	MI	2	0
31	1250	BC	2	0
32	1251	AL	2	0
33	1252	NC	2	0
34	1253	WA	2	0
35	1254	TX	2	0
36	1256	AL	2	0
37	1260	IN	2	0
38	1262	WI	2	0
39	1264	Anguilla	2	0
40	1267	PA	2	0
41	1268	Antigua	2	0
42	1269	MI	2	0
43	1270	KY	2	0
44	1276	VA	2	0
45	1281	TX	2	0
46	1284	BVI	2	0
47	1289	Ontario	2	0
48	1301	MD	2	0
49	1302	DE	2	0
50	1303	CO	2	0
51	1304	WV	2	0
52	1305	FL	2	0
53	1306	Saskatchewan	2	0
54	1307	WY	2	0
55	1308	NE	2	0
56	1309	IL	2	0
57	1310	CA	2	0
58	1312	IL	2	0
59	1313	MI	2	0
60	1314	MO	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
61	1315	NY	2	0
62	1316	KS	2	0
63	1317	IN	2	0
64	1318	LA	2	0
65	1319	IA	2	0
66	1320	MN	2	0
67	1321	FL	2	0
68	1323	CA	2	0
69	1325	TX	2	0
70	1330	OH	2	0
71	1331	IL	2	0
72	1334	AL	2	0
73	1336	NC	2	0
74	1337	LA	2	0
75	1339	MA	2	0
76	1340	USVI	2	0
77	1345	Cayman	2	0
78	1347	NY	2	0
79	1351	MA	2	0
80	1352	FL	2	0
81	1360	WA	2	0
82	1361	TX	2	0
83	1386	FL	2	0
84	1401	RI	2	0
85	1402	NE	2	0
86	1403	Alberta	2	0
87	1404	GA	2	0
88	1405	OK	2	0
89	1406	MT	2	0
90	1407	FL	2	0
91	1408	CA	2	0
92	1409	TX	2	0
93	1410	MD	2	0
94	1412	PA	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
95	1413	MA	2	0
96	1414	WI	2	0
97	1415	CA	2	0
98	1416	Ontario	2	0
99	1417	MO	2	0
100	1418	Quebec	2	0
101	1419	OH	2	0
102	1423	TN	2	0
103	1424	CA	2	0
104	1425	WA	2	0
105	1430	TX	2	0
106	1432	TX	2	0
107	1434	VA	2	0
108	1435	UT	2	0
109	1438	Quebec	2	0
110	1440	OH	2	0
111	1441	Bermuda	2	0
112	1443	MD	2	0
113	1450	Quebec	2	0
114	1456	NANParea	2	0
115	1469	TX	2	0
116	1473	Grenada	2	0
117	1478	GA	2	0
118	1479	AR	2	0
119	1480	AZ	2	0
120	1484	PA	2	0
121	1500	NANParea	2	0
122	1501	AR	2	0
123	1502	KY	2	0
124	1503	OR	2	0
125	1504	LA	2	0
126	1505	NM	2	0
127	1506	NewBrunswick	2	0
128	1507	MN	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
129	1508	MA	2	0
130	1509	WA	2	0
131	1510	CA	2	0
132	1512	TX	2	0
133	1513	OH	2	0
134	1514	Quebec	2	0
135	1515	IA	2	0
136	1516	NY	2	0
137	1517	MI	2	0
138	1518	NY	2	0
139	1519	Ontario	2	0
140	1520	AZ	2	0
141	1530	CA	2	0
142	1540	VA	2	0
143	1541	OR	2	0
144	1551	NJ	2	0
145	1559	CA	2	0
146	1561	FL	2	0
147	1562	CA	2	0
148	1563	IA	2	0
149	1567	OH	2	0
150	1570	PA	2	0
151	1571	VA	2	0
152	1573	MO	2	0
153	1574	IN	2	0
154	1575	NM	2	0
155	1580	OK	2	0
156	1585	NY	2	0
157	1586	MI	2	0
158	1600	Canada	2	0
159	1601	MS	2	0
160	1602	AZ	2	0
161	1603	NH	2	0
162	1604	BC	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
163	1605	SD	2	0
164	1606	KY	2	0
165	1607	NY	2	0
166	1608	WI	2	0
167	1609	NJ	2	0
168	1610	PA	2	0
169	1612	MN	2	0
170	1613	Ontario	2	0
171	1614	OH	2	0
172	1615	TN	2	0
173	1616	MI	2	0
174	1617	MA	2	0
175	1618	IL	2	0
176	1619	CA	2	0
177	1620	KS	2	0
178	1623	AZ	2	0
179	1626	CA	2	0
180	1630	IL	2	0
181	1631	NY	2	0
182	1636	MO	2	0
183	1641	IA	2	0
184	1646	NY	2	0
185	1647	Ontario	2	0
186	1649	T&CIsland	2	0
187	1650	CA	2	0
188	1651	MN	2	0
189	1660	MO	2	0
190	1661	CA	2	0
191	1662	MS	2	0
192	1664	Montsrat	2	0
193	1670	CNMI	2	0
194	1671	GU	2	0
195	1678	GA	2	0
196	1682	TX	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
197	1684	AS	2	0
198	1700	NANParea	2	0
199	1701	ND	2	0
200	1702	NV	2	0
201	1703	VA	2	0
202	1704	NC	2	0
203	1705	Ontario	2	0
204	1706	GA	2	0
205	1707	CA	2	0
206	1708	IL	2	0
207	1709	Newfoundland	2	0
208	1710	US	2	0
209	1712	IA	2	0
210	1713	TX	2	0
211	1714	CA	2	0
212	1715	WI	2	0
213	1716	NY	2	0
214	1717	PA	2	0
215	1718	NY	2	0
216	1719	CO	2	0
217	1720	CO	2	0
218	1724	PA	2	0
219	1727	FL	2	0
220	1731	TN	2	0
221	1732	NJ	2	0
222	1734	MI	2	0
223	1740	OH	2	0
224	1754	FL	2	0
225	1757	VA	2	0
226	1758	St.Lucia	2	0
227	1760	CA	2	0
228	1762	GA	2	0
229	1763	MN	2	0
230	1765	IN	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
231	1767	Dominica	2	0
232	1769	MS	2	0
233	1770	GA	2	0
234	1772	FL	2	0
235	1773	IL	2	0
236	1774	MA	2	0
237	1775	NV	2	0
238	1778	BC	2	0
239	1779	IL	2	0
240	1780	Alberta	2	0
241	1781	MA	2	0
242	1784	St. V&G	2	0
243	1785	KS	2	0
244	1786	FL	2	0
245	1787	PrtoRico	2	0
246	1800	NANParea	2	0
247	1801	UT	2	0
248	1802	VT	2	0
249	1803	SC	2	0
250	1804	VA	2	0
251	1805	CA	2	0
252	1806	TX	2	0
253	1807	Ontario	2	0
254	1808	HI	2	0
255	1809	DomRepub	2	0
256	1810	MI	2	0
257	1812	IN	2	0
258	1813	FL	2	0
259	1814	PA	2	0
260	1815	IL	2	0
261	1816	MO	2	0
262	1817	TX	2	0
263	1818	CA	2	0
264	1819	Quebec	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
265	1828	NC	2	0
266	1829	DomRepub	2	0
267	1830	TX	2	0
268	1831	CA	2	0
269	1832	TX	2	0
270	1843	SC	2	0
271	1845	NY	2	0
272	1847	IL	2	0
273	1848	NJ	2	0
274	1850	FL	2	0
275	1856	NJ	2	0
276	1857	MA	2	0
277	1858	CA	2	0
278	1859	KY	2	0
279	1860	CT	2	0
280	1862	NJ	2	0
281	1863	FL	2	0
282	1864	SC	2	0
283	1865	TN	2	0
284	1866	NANParea	2	0
285	1867	Yukon	2	0
286	1868	Tri&Tob	2	0
287	1869	St. K&N	2	0
288	1870	AR	2	0
289	1876	Jamaica	2	0
290	1877	NANParea	2	0
291	1878	PA	2	0
292	1888	NANParea	2	0
293	1900	NANParea	2	0
294	1901	TN	2	0
295	1902	N Scotia	2	0
296	1903	TX	2	0
297	1904	FL	2	0
298	1905	Ontario	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
299	1906	MI	2	0
300	1907	AK	2	0
301	1908	NJ	2	0
302	1909	CA	2	0
303	1910	NC	2	0
304	1912	GA	2	0
305	1913	KS	2	0
306	1914	NY	2	0
307	1915	TX	2	0
308	1916	CA	2	0
309	1917	NY	2	0
310	1918	OK	2	0
311	1919	NC	2	0
312	1920	WI	2	0
313	1925	CA	2	0
314	1928	AZ	2	0
315	1931	TN	2	0
316	1936	TX	2	0
317	1937	OH	2	0
318	1939	PrtoRico	2	0
319	1940	TX	2	0
320	1941	FL	2	0
321	1947	MI	2	0
322	1949	CA	2	0
323	1951	CA	2	0
324	1952	MN	2	0
325	1954	FL	2	0
326	1956	TX	2	0
327	1970	CO	2	0
328	1971	OR	2	0
329	1972	TX	2	0
330	1973	NJ	2	0
331	1978	MA	2	0
332	1979	TX	2	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
333	1980	NC	2	0
334	1985	LA	2	0
335	1989	MI	2	0
336	0117	Kazkhstan	3	0
337	01120	Egypt	3	0
338	01127	South Africa	3	0
339	01130	Greece	3	0
340	01131	Netherlands	3	0
341	01132	Belgium	3	0
342	01133	France	3	0
343	01134	Spain	3	0
344	01136	Hungary	3	0
345	01139	VaticanCity	3	0
346	01140	Romania	3	0
347	01141	Switzerland	3	0
348	01143	Austria	3	0
349	01144	UK	3	0
350	01145	Denmark	3	0
351	01146	Sweden	3	0
352	01147	Norway	3	0
353	01148	Poland	3	0
354	01149	Germany	3	0
355	01151	Peru	3	0
356	01152	Mexico	3	0
357	01153	Cuba	3	0
358	01154	Argentine	3	0
359	01155	Brazil	3	0
360	01156	Chile	3	0
361	01157	Colombia	3	0
362	01158	Venezuela	3	0
363	01160	Malaysia	3	0
364	01161	Australia	3	0
365	01162	Indonesia	3	0
366	01163	Philippines	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
367	01164	NZ	3	0
368	01165	Singapore	3	0
369	01166	Thailand	3	0
370	01181	Japan	3	0
371	01182	Korea	3	0
372	01184	VietNam	3	0
373	01186	China	3	0
374	01190	Turkey	3	0
375	01191	India	3	0
376	01192	Pakistan	3	0
377	01193	Afghanistan	3	0
378	01194	Sri Lanka	3	0
379	01195	Myanmar	3	0
380	01198	Iran	3	0
381	011212	Morocco	3	0
382	011213	Algeria	3	0
383	011216	Tunisia	3	0
384	011218	Libya	3	0
385	011220	Gambia	3	0
386	011221	Senegal	3	0
387	011222	Mauritania	3	0
388	011223	Mali	3	0
389	011224	Guinea	3	0
390	011225	IvoryCoast	3	0
391	011226	BurkinaFaso	3	0
392	011227	Niger	3	0
393	011228	Togolese	3	0
394	011229	Benin	3	0
395	011230	Mauritius	3	0
396	011231	Liberia	3	0
397	011232	SierraLeone	3	0
398	011233	Ghana	3	0
399	011234	Nigeria	3	0
400	011235	Chad	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
401	011236	CenAfrica	3	0
402	011237	Cameroon	3	0
403	011238	CapeVerde	3	0
404	011239	SaoTome	3	0
405	011240	Equtl_Guinea	3	0
406	011241	Gabonese	3	0
407	011242	Congo	3	0
408	011243	CongoDem	3	0
409	011244	Angola	3	0
410	011245	GuineaBissa	3	0
411	011246	DiegoGarcia	3	0
412	011247	Ascension	3	0
413	011248	Seychelles	3	0
414	011249	Sudan	3	0
415	011250	Rwandese	3	0
416	011251	Ethiopia	3	0
417	011252	SomalianRep	3	0
418	011253	Djibouti	3	0
419	011254	Kenya	3	0
420	011255	Tanzania	3	0
421	011256	Uganda	3	0
422	011257	Burundi	3	0
423	011258	Mozambique	3	0
424	011260	Zambia	3	0
425	011261	Madagascar	3	0
426	011262	Reunion	3	0
427	011263	Zimbabwe	3	0
428	011264	Namibia	3	0
429	011265	Malawi	3	0
430	011266	Lesotho	3	0
431	011267	Botswana	3	0
432	011268	Swaziland	3	0
433	011269	Comoros	3	0
434	011290	StHelena	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
435	011291	Eritrea	3	0
436	011297	Aruba	3	0
437	011298	FaroeIsland	3	0
438	011299	Greenland	3	0
439	011350	Gibraltar	3	0
440	011351	Portugal	3	0
441	011352	Luxembourg	3	0
442	011353	Ireland	3	0
443	011354	Iceland	3	0
444	011355	Albania	3	0
445	011356	Malta	3	0
446	011357	Cyprus	3	0
447	011358	Finland	3	0
448	011359	Bulgaria	3	0
449	011370	Lithuania	3	0
450	011371	Latvia	3	0
451	011372	Estonia	3	0
452	011373	Moldova	3	0
453	011374	Armenia	3	0
454	011375	Belarus	3	0
455	011376	Andorra	3	0
456	011377	Monaco	3	0
457	011378	SanMarino	3	0
458	011379	VaticanCity	3	0
459	011380	Ukraine	3	0
460	011381	Yugoslavia	3	0
461	011385	Croatia	3	0
462	011386	Slovenia	3	0
463	011387	Bosnia	3	0
464	011389	Macedonia	3	0
465	011420	Czech Repub	3	0
466	011421	Slovakia	3	0
467	011423	Liechtenstein	3	0
468	011500	Falklands	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
469	011501	Belize	3	0
470	011502	Guatemala	3	0
471	011503	El Salvador	3	0
472	011504	Honduras	3	0
473	011505	Nicaragua	3	0
474	011506	CostaRica	3	0
475	011507	Panama	3	0
476	011508	St.Pierre	3	0
477	011509	Haiti	3	0
478	011590	Guadeloupe	3	0
479	011591	Bolivia	3	0
480	011592	Guyana	3	0
481	011593	Ecuador	3	0
482	011594	FrenchGuyana	3	0
483	011595	Paraguay	3	0
484	011596	Martinique	3	0
485	011597	Suriname	3	0
486	011598	Uruguay	3	0
487	011599	NethAntilles	3	0
488	011670	East Timor	3	0
489	011672	Antarctic	3	0
490	011673	Brunei	3	0
491	011674	Nauru	3	0
492	011675	PapuaNewGuin	3	0
493	011676	Tonga	3	0
494	011677	SolomonIslnd	3	0
495	011678	Vanuatu	3	0
496	011679	Fiji	3	0
497	011680	Palau	3	0
498	011681	Wallis Island	3	0
499	011682	Cook Islands	3	0
500	011683	Niuel Island	3	0
501	011684	AmerSamoa	3	0
502	011685	WSamoa	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
503	011686	Kiribati	3	0
504	011687	NewCaledonia	3	0
505	011688	Tuvalu	3	0
506	011689	FrenchPolyne	3	0
507	011690	Tokelau	3	0
508	011691	Micronesia	3	0
509	011692	MarshallIsInd	3	0
510	011850	Korea North	3	0
511	011852	Hong Kong	3	0
512	011853	Macau	3	0
513	011855	Cambodia	3	0
514	011856	Laos	3	0
515	011870	SatIndlOcn	3	0
516	011871	SatEastAtl	3	0
517	011872	SatPacific	3	0
518	011873	SatIndianOcn	3	0
519	011874	SatWestAtl	3	0
520	011880	Bangladesh	3	0
521	011960	Maldives	3	0
522	011961	Lebanon	3	0
523	011962	Jordan	3	0
524	011963	SyrianArab	3	0
525	011964	Iraq	3	0
526	011965	Kuwait	3	0
527	011966	SaudiArabia	3	0
528	011967	Yemen	3	0
529	011968	Oman	3	0
530	011971	UAE	3	0
531	011972	Israel	3	0
532	011973	Bahrain	3	0
533	011974	Qatar	3	0
534	011975	Bhutan	3	0
535	011976	Mongolia	3	0
536	011977	Nepal	3	0

Index	Area Code	Area Name	Pulse Rate Type	Ignore Digit Count
537	011992	Tajikistan	3	0
538	011993	Turkmenistan	3	0
539	011994	Azerbaijani	3	0
540	011995	Georgia	3	0
541	011996	Kyrgyzstan	3	0
542	011998	Uzbekistan	3	0
543				
544				
545				
546				
547				
:				
998				
999				

Call Cost Display

With Call Cost Display, you can view the cost of the last 10 external calls made from your extension through the SIP trunks.

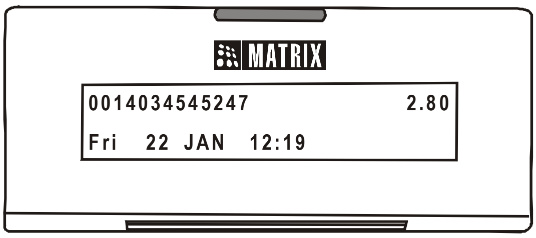
The system will display the dialed numbers and the call cost for each number that it has calculated on the LCD of Extended IP Phone. By default, this feature is enabled.

How to use

For Extended IP Phone Users

- Press DSS Key assigned to Call Cost Display feature.
  OR
- Dial **1075**.
- Scroll with the up/down navigation keys to view the cost of the last 10 calls.
- The display shows the last 10 dialed numbers and their corresponding call cost.

  For example: If the call charge is for the dialed number 0014034545247 is $2 and 80 cents' then the display will show:



- The display is retained till the extension remains OFF-Hook.
- Go Idle or you get dial tone after 3 seconds.

Call Duration Control (CDC)

Call Duration Control (CDC) allows a maximum time limit to be set on internal and external (both incoming and outgoing) calls. When the maximum call duration is reached, the calls are disconnected, after a warning tone indicating to the user that the calls in progress will be disconnected.

By limiting the duration of the conversations, CDC helps increase availability of trunks for making outgoing calls and for receiving incoming calls, which is important in high call traffic situations. Besides increasing trunk availability, CDC curbs unrelated and unproductive conversations.

By default this feature is enabled on the trunks and extensions, hence calls will be disconnected after the expiry of the timer. You can change the configurations as per your requirements.

How it works

- A is an extension user. B is an external number.

External-Outgoing Calls

- A dials B's number.
- The system checks the CDC Table applied to A.
- ANANT UCS will check whether the CDC is applicable on Extension of A as well as the Trunk used for making the outgoing call.

To check whether CDC is applicable on Extension A

- The system checks the **CDC Table** selected in the Station Advanced Feature Template assigned to A.
  - In the CDC Table, it checks whether the check box, **Apply CDC for outgoing calls made from trunk**, is enabled.
  - It matches B's number with the entries on the **Apply CDC for calls matching with numbers** list and the **Do Not Apply CDC for calls matching with numbers** list in the CDC table.

To check whether CDC is applicable on the Trunk

- The system checks the **Apply CDC for Outgoing Calls** check box in the Trunk Features Template assigned to the trunk.

The following results are possible:

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found for the number in the **Apply CDC for calls matching with numbers** List. The **Apply CDC for outgoing calls** check box is enabled in the Trunk Features Template assigned to the trunk. CDC is applied on the call.
- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found for the number in the **Apply CDC for calls matching with numbers** List. The **Apply**

CDC for outgoing calls check box is disabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in the **Do Not Apply CDC for calls matching with numbers** list. The **Apply CDC for outgoing calls** check box is enabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in the **Do Not Apply CDC for calls matching with numbers** list. The **Apply CDC for outgoing calls** check box is disabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in both the number lists. The system gives precedence to the Do Not Apply CDC for calls matching with numbers list. The **Apply CDC for outgoing calls** check box is enabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- In the CDC Table, the **Apply CDC for outgoing calls made from trunk** check box is enabled and a match is found in both the number lists. The system gives precedence to the Do Not Apply CDC for calls matching with numbers list. The **Apply CDC for outgoing calls** check box is disabled in the Trunk Features Template assigned to the trunk. CDC is not applied on the call.

- When CDC is applied to the call, the **CDC Timer** starts as soon as B has answered the call. This timer is set to 160 seconds as default, but can be configured to the desired time limit.

- At the end of the default/configured time limit of the **CDC Timer**, the Beep Timer starts (5 seconds; non-configurable) then the CDC Goodbye Timer starts. This Goodbye timer provides a grace period of 20 seconds for the user to finish the call. This Timer is non-configurable.

- At the end of the Goodbye Timer, the call is disconnected, if the **Disconnect CDC after Time**r check box is enabled.

- If this check box is disabled, the call will not be disconnected.

- Instead, the **CDC Timer** will be loaded again for the default/configured duration. The user can know how long s/he has been talking.

- A is played Warning Beeps. B cannot hear the beeps. This continues until either party disconnects.

External-Incoming Calls

CDC works similarly for incoming calls.

- B calls A.

- The system checks whether the following conditions are fulfilled,

    - the **Apply CDC for incoming calls received from trunk** check box is enabled in the CDC Table assigned to A.

    - the **Disconnect Call after CDC Timer** check box is enabled in the CDC Table assigned to A.

    - a match is found for the B’s number in the **Apply CDC for calls matching with numbers** List.

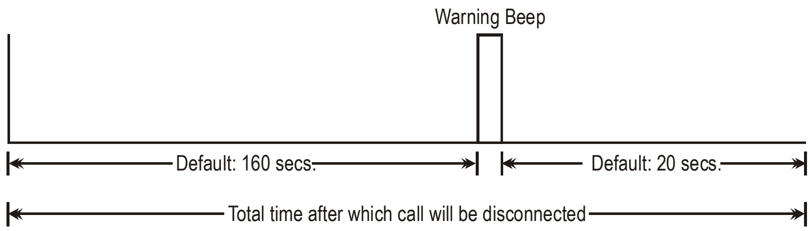
- the **Apply CDC for Incoming Calls** check box is enabled in the Trunk Features Template assigned to the trunk on which the call is received.
  CDC is applied on the call.

- A will get beeps.

Internal Calls

A and B are extension users.

- A calls B.
- The system checks whether the check box, **Apply CDC to Internal Calls**, is enabled in the CDC Table.
- If the check box is enabled, CDC is applied on the call.
- Warning beeps are played to A, who made the call.



In this case the total time after which the call is disconnected will be 160 seconds (default CDC Timer) + 5 seconds (Beep Timer; non-configurable) + 20 seconds (Goodbye Timer; non-configurable).

Feature Interactions:

- **Call Transfer:** In case of transferred call, the CDC timer gets reset and starts again afresh on the transferred extension.
- **Conference, Conversation Recording** and **Call Taping:** CDC is treated as turned OFF.
- **Call Park and Call Hold (Exclusive and Global):** CDC is treated as turned ON.
- **Interrupt Request, Barge-In:** CDC is treated as turned ON.
- **Raid:** CDC is not applicable when you raid an extension as the conversation is converted into a 3 party conference.
- **Emergency Number Dialing:** Emergency calls are not affected by this feature, i.e. CDC will not be applied on the dialing of Emergency Numbers.

How to configure

By default this feature is enabled on the Trunks and Extensions. The feature settings are as follows:

- **Call Duration Control Table 1** is assigned to CDC for Trunks and Extensions.

- In the **Call Duration Control Table 1,** the check boxes **Call Duration Control For IC Calls**, **Call Duration Control For OG Calls**, **Apply CDC to Internal Calls** and **Disconnect Call after CDC Timer** all are enabled.

- In the **Call Duration Control Table 1, List Number 2** is assigned to **Apply CDC for calls matching with numbers** and **List Number 8** is assigned to **Do Not Apply CDC for calls matching with numbers.**

- The **CDC Timer** is set as 300 seconds.

- In the default **Trunk Feature Template 1** assigned to all the Trunks, the check boxes **Call Duration Control For IC Calls** and **Call Duration Control For OG Calls** are enabled.

- In the default **Station Advanced Feature Template 1** assigned to all the extensions, **CDC Table 1** is assigned.

If you wish to change the configurations, follow the steps as mentioned below:

- Configure Call Duration Control Table. You may configure up to 8 different Tables.

- Assign a CDC Table in the Station Advanced Feature Template of those extensions on which Call Duration Control is to be applied.
- Enable CDC for incoming and outgoing calls from the Trunks on which you want to apply this feature.

To configure the Call Duration Control Table,

- decide the types of calls - Outgoing, Incoming and Internal - on which CDC is to be enabled.

- make a list of numbers on which CDC is to be applied, that is, the Apply CDC to Numbers List.

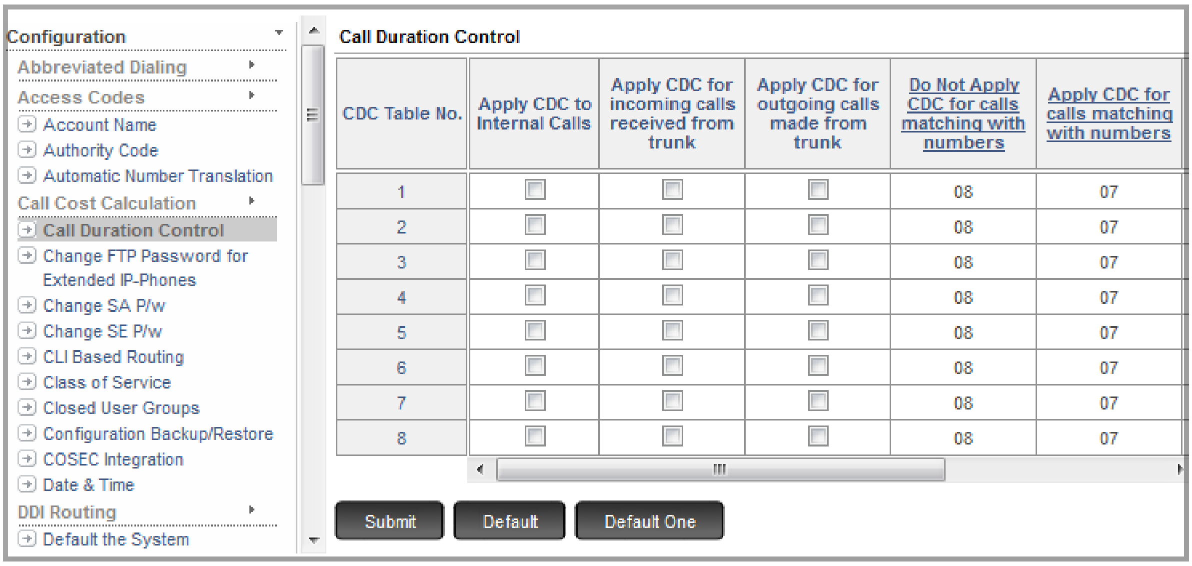
- Make a list of numbers on which CDC is not to be applied, the Do Not Apply CDC to Number List.

Configuring CDC

CDC Table

- Login as System Engineer.

- Under **Configuration,** click **Call Duration Control.**



- The CDC Table will open. There are 8 CDC Tables. By default CDC Table No. 1 is assigned to all extensions of ANANT UCS. If the same CDC is to be assigned to all extensions, configure this table.

  If different CDC is to be applied to different extensions, configure separate CDC tables for these extensions.

- Now configure the following parameters in the table you have selected:

  - **Apply CDC to Internal Calls:** By default, this check box is enabled, that is, CDC will be applied on internal calls. Clear the check box to disable.

  - **Apply CDC for Incoming Calls received from trunk:** By default, this check box is enabled, that is, CDC will be applied to incoming external calls. Clear the check box to disable

  - **Apply CDC for Outgoing Calls made from trunk:** By default, this check box is enabled, that is, CDC will be applied to outgoing external calls. Clear the check box to disable.

  - **Do Not Apply CDC for calls matching with numbers:** This is the list of numbers on which CDC is not to be applied. By default, Number List 08 is assigned to this parameter. You must configure this list with numbers which you want to be exempted from CDC.

To configure the list, click **Do Not Apply CDC for calls matching with numbers**.

Call Duration Control

CDC Table No.	Apply CDC to Internal Calls	Apply CDC for incoming calls received from trunk	Apply CDC for outgoing calls made from trunk	Do Not Apply CDC for calls matching with numbers
1	☐	☐	☐	08
2	☐	☐	☐	08
3	☐	☐	☐	08
4	☐	☐	☐	08
5	☐	☐	☐	08
6	☐	☐	☐	08
7	☐	☐	☐	08
8	☐	☐	☐	08

Submit | Default | Default One

This will lead you to the "Number Lists" page. Click '001-250' of Number list 07-08.

01-02 | 03-04 | 05-06 | 07-08 | 09-10 | 11-12 | 13-14 | 15-16

001-250 | 251-500 | 501-750 | 751-999

Number List

Index	Number List 01	Number List 02
001		00
002		0
003		1
004		2
005		3
006		4
007		5
008		6
009		7
010		8
011		9

Submit | Default | Default One

By default, Number List 08 is assigned to this parameter. You can also program any other Number List you want. Enter the list of numbers on which CDC is not be applied (refer to the list you prepared). Click **Submit.**

Return to the **Call Duration Control** page. If you have prepared a Number List other than the default 08, then enter the number of that list in the **Do Not Apply CDC for calls matching with numbers** column.

- **Apply CDC for calls matching with numbers:** This is the list of numbers on which CDC is to be applied. By default, Number List 02 is assigned to this parameter for Table 1. You must configure this list with numbers on which you want CDC to be applied.

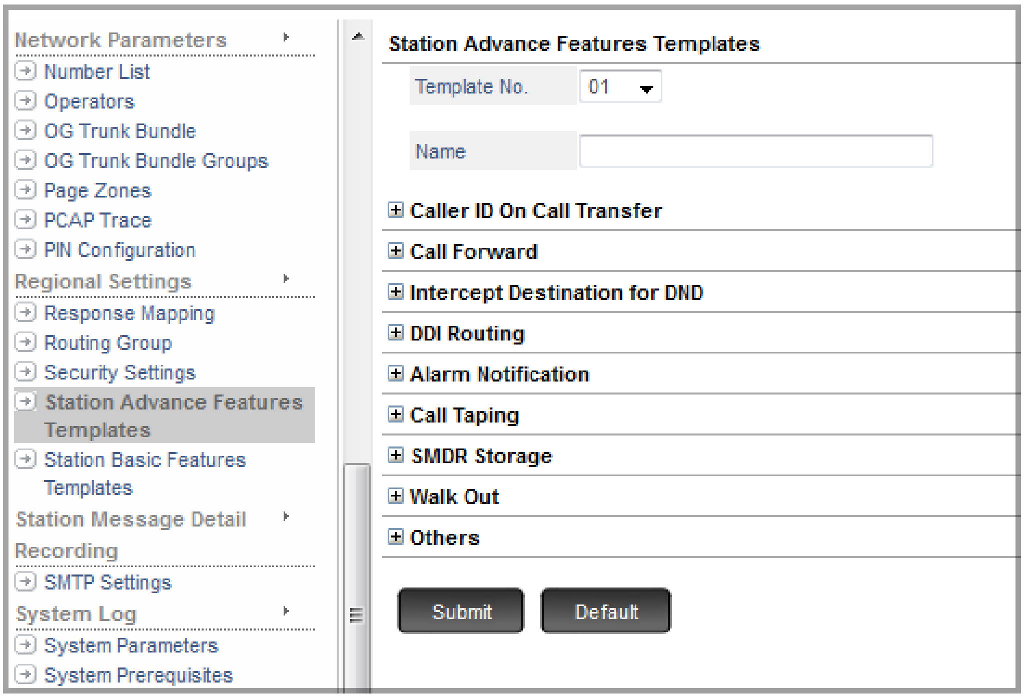
  To configure the list, click **Apply CDC for calls matching with numbers**. The 'Number Lists' page opens.

  Click '001- 250' of Number list 07- 08. Follow the same steps as described above for configuring the Do Not Apply CDC Number List.

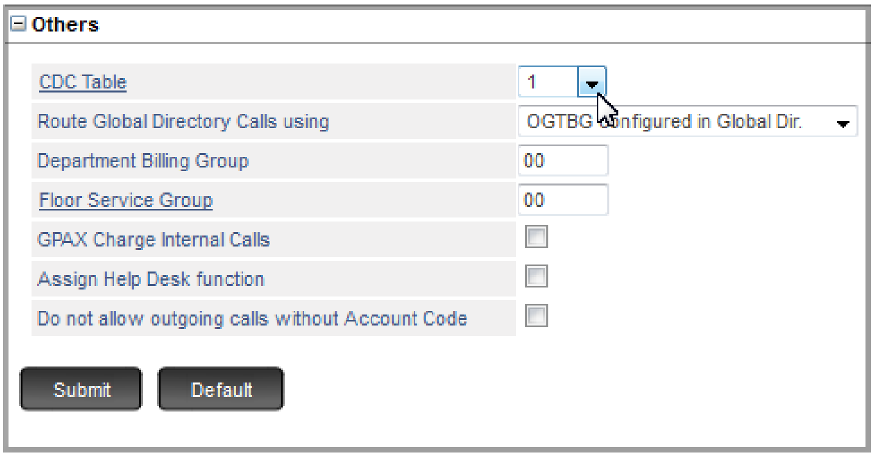
- **CDC Timer:** This is the time for which the warning beeps are to be played before the system disconnects the call. The range of the timer is 0001 to 9999 seconds. By default this Timer is set to 160 seconds. Set the CDC Timer to the desired time limit.

- **Disconnect Call after CDC Timer:** This check box is to be enabled if you want the call to be automatically disconnected on the expiry of the CDC Timer. By default the check box is disabled, which means that calls will not be disconnected on expiry of the CDC Timer. Enable the check box if required.

- Click **Submit.**
- Under **Configuration**, click **Station Advanced Feature Template**.



- Click **Others** to expand. Ensure that the **CDC Table** Number (in this case 01) you have configured is assigned in the Template you want to apply to the extensions.



- Click **Submit.**

- To assign the Station Advanced Feature Template with the CDC Table on SIP Extensions go to the **SIP Extensions Settings** under **VoIP Configuration**.

  Refer “Station Advanced Feature Template” for instructions on customizing the templates and assigning them to extensions.

- If selected extensions are to be allowed CDC or if different CDC parameters are to be allowed to selected extensions (for example, 160 seconds duration timer for a few extensions, 360 duration timer for some other extensions), then follow these steps:

  a. Define a new CDC table.

  b. Configure the different CDC parameters in this table, as required for the extensions.

  c. Apply this CDC table on a separate Station Advanced Feature Template.

  d. Apply the new Station Advanced Feature Template now configured with a different CDC table on the selected extensions which are to be allowed this feature.

CDC on Trunk

- You must enable CDC in the Trunk Feature Template assigned to the trunks. For instructions, see “Configuring Trunks”.

Call Duration Display

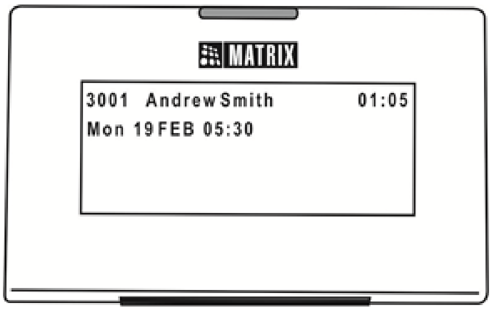
By invoking Call Duration Display, extension users can view the duration of the current call instantly.

The system displays the duration of the current call on the LCD display of the phone.

The current call may be incoming or outgoing, internal or external call.

How it works

- The user goes OFF-Hook
- Dials an external number.
- The remote party answers the call.
- The dialed external number with duration (5-digits in the format of MM:SS) is displayed on the LCD of the phone, when the call is answered.



Call Forward

During a typical workday, it is common for people in an organization to move from one place to another. For instance, a manager might go on the production floor or remain in the conference room for a few hours; a field engineer may spend half of the day on site. So, they need to be able to attend their calls even when they are not present at their desks. The 'Call Forward' feature of ANANT UCS ensures this.

Using this feature, calls landing on an extension can be forwarded to another extension, an external number, Voice Mail, or a Department Group. This way, extension users can ensure that callers can reach them and that they do not miss calls when they are not present at their desks.

You can also set Call Forward for all the extensions from the SA Mode. See "Settings Call Forward for All Extensions".

The Call Forward feature of ANANT UCS offers the following forwarding options:

- **Unconditionally** - calls are forwarded to the destination phone number automatically without waiting for a response from the called party's phone.
- **If Busy** - calls are forwarded to the destination phone number only when the called party's phone is busy.
- **If No Reply** - calls are forwarded to the destination phone number only when the called party does not answer the phone. Each extension can set a different time after which the call should be forwarded, in case of no reply. The default time is 30 seconds for all extensions and can be changed by configuring the Call Forward No-Reply Timer.
- **If Busy or No Reply** - calls are forwarded to the destination phone number when the called party's phone is either busy or does not reply.
- **Dual Ring** - when calls are forwarded to another phone number. Both phones, that is, the source phone (whose calls are forwarded) as well as the destination phone (on which call is forwarded) will ring and the user can answer from either extension.

  Dual Ring is useful to users who may have to be present frequently at two different places. As it is cumbersome to forward the calls from one extension to another and cancel it repeatedly, extensions users can set Dual Ring, so they can attend to their calls at either place they are present.

If you do not want to set/cancel Call Forward manually, you can set Preset Call Forward. Call will be forwarded automatically to the selected destination according to the type of Preset Call Forward set. You can set a different type of forward and destination for each time zone. *To know more, see "Preset Call Forward"*

How it works

A has set Call Forward to extension B unconditionally.

- The system forwards all calls for A to B, without checking for Busy Tone and without waiting for the Call Forward No-Reply Timer to expire.

A has set Call Forward to external number on No-Reply.

- The system waits for the Call Forward No-Reply Timer to expire and then forwards all external incoming calls to the external number.

A has set Call Forward to extension B on Busy.

- The system forwards the call for A to B on detecting Busy signal from A.

B belongs to a Department Group and has set Call Forward-If Busy to C within the Department Group.

- If the system detects Busy signal on B, it forwards the call for B to C in the Department Group.

- However, if the caller has called the Department Group instead of calling B directly, the call will land in the sequence on all Department group extensions. When it is B's turn, the call will not be forwarded to C, B will ring instead.

C belongs to a Department Group and has set Call Forward-No Reply to D within the Department Group.

- The system waits for the Call Forward No-Reply Timer to expire, and forwards the call for C to D in the Department Group.

D has set Call Forward to Voice Mail on Busy or No Reply.

- Whenever there is a call for D, if the system does not detect a busy signal from D, it waits for the Call Forward No-Reply timer to expire.

- The system forwards the call to the Voice Mail System.

E has set Call Forward Dual Ring on extension F.

- When there is a call for E, the system rings on both E and the destination F.

- E can answer the call at E or at F.

- *Call Forward set by member extensions in a routing group will be ignored by the system if, the Ignore call forward set by member extension, when call is routed on Routing/Dept. Group check box is enabled. See* *"System Parameters"* *for more information.*
- *When an incoming call is routed to the Routing Group and if any member has set Call Forward to the Department Group, then only the first extension of that Department Group will ring.*
- *Call Forward when set/canceled from the SA mode, will not depend on the assigned CoS.*

Feature Interaction:

- **Do Not Disturb (DND):**

  When DND or DND with Intercept Destination is set along with Call Forward-Unconditional on an extension, Call Forward is given priority.

  If any other type of Call Forward and DND are set on an extension, DND is given priority. However, DND with Intercept Destination will not work.

  If an extension has set both Call Forward and DND, then Feature Tone will be played to the extension user.

- *You can select the types of calls, that is, internal calls only, or trunk calls, or both, to be forwarded to external numbers. You can configure the system to forward internal calls only, or trunk calls only or both trunk calls and internal calls, to the external number. For this, the parameter 'Allow External Call Forward for' must be configured in the "Station Advanced Feature Template" of the extensions that are allowed Call Forward.*

- *The system supports only single-point Call Forward, which means, if the destination extension is also forwarded, the call will not follow the forwarding path. For example: Calls for extension A are set to be forwarded to extension B. Call Forward is also set on extension B with C as the destination number. Calls for A will land on B only and calls for B will land on C only.*

- *Only one Call Forward Type can be set from an extension. Every new Call Forward Type set overrides the previous one.*

- *When the calls are forwarded the extension user gets the feature tone on lifting the handset to indicate that Call Forward is set on his/her extension.*

How to configure

The functioning of this feature is controlled by three parameters: 'Class of Service' and 'Call Forward No-Reply Timer' and 'Allow External Call Forward for'.

Call Forward must be enabled in the Class of Service (COS) group of the extensions to which this feature is to be allowed.

When Call Forward No-Reply is set, if required the Call Forward No-Reply Timer needs to be configured.

You may select the types of calls, that is, internal, external, both internal and external calls to be forwarded by configuring the 'Allow External Call Forward for' parameter.

You can set Call Forward for All the Extensions from the SA mode only, see "Settings Call Forward for All Extensions".

Call Forward in Class of Service

By default, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS. Station Basic Feature Template 01 is assigned COS group 01 in which Call Forward is enabled. So, all extensions of ANANT UCS can use Call Forward.

If you want to deny Call Forward to certain extensions, follow these steps:

a. Define a CoS group with Call Forward disabled.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".

c. Assign this new Template to the extensions to which Call Forward is to be denied.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for detailed instructions and configuration.

Call Forward No Reply Timer

When using Call Forward -No Reply, each extension can set a different Time after which the incoming call on the extension should get forwarded when there is no reply from the extension. For this, the Call Forward No-Reply Timer must be configured. By default, this timer is set to 30 seconds.

The Call Forward No-Reply Timer is to be configured in the "Station Advanced Feature Template" applied on the extensions which are allowed Call Forward in their CoS.

If you want to set this timer to the same duration for all extensions, simply set the Call Forward No-Reply Timer in the default Station Advanced Feature Template 01 which is assigned to all extensions.

If you want to set different Timer duration for different extensions, then prepare separate Station Advanced Feature Templates with the desired Timer durations and assign different Templates (with different Timer durations) to the extensions as desired.

Allow External Call Forward for

The types of calls to be forwarded to the external number may be selected in the parameter "*Allow External Call Forward for"* in the "Station Advanced Feature Template" applied on the extensions which are allowed Call Forward.

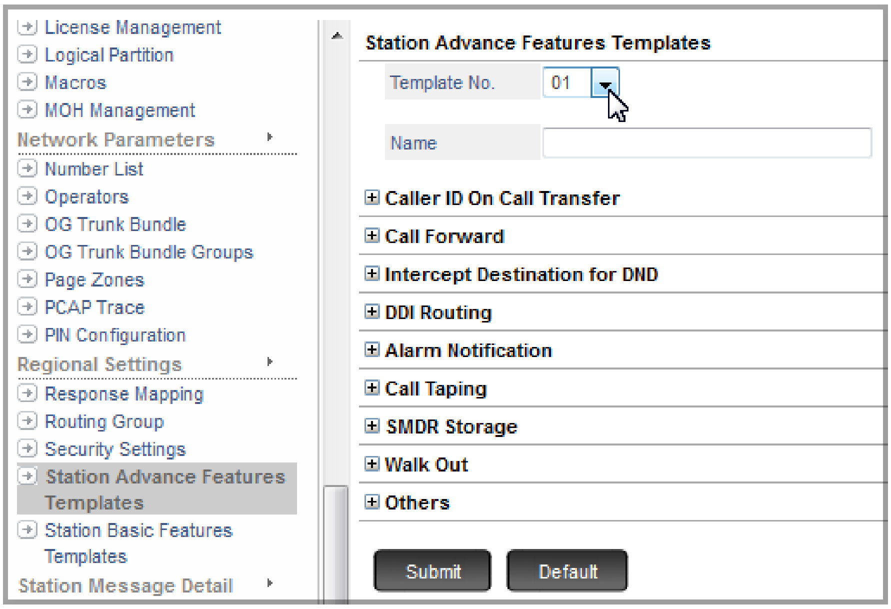
You may select from 'Internal Calls', 'Trunk Calls' and 'Internal + Trunk Calls'. By default, only trunk calls are forwarded to external numbers in the default Station Advanced Feature Template 01 which is assigned to all extensions.

If you want to set different call types for different extensions, then prepare separate Station Advanced Feature Templates with the desired Call Types and assign these different Templates to the extensions as appropriate.

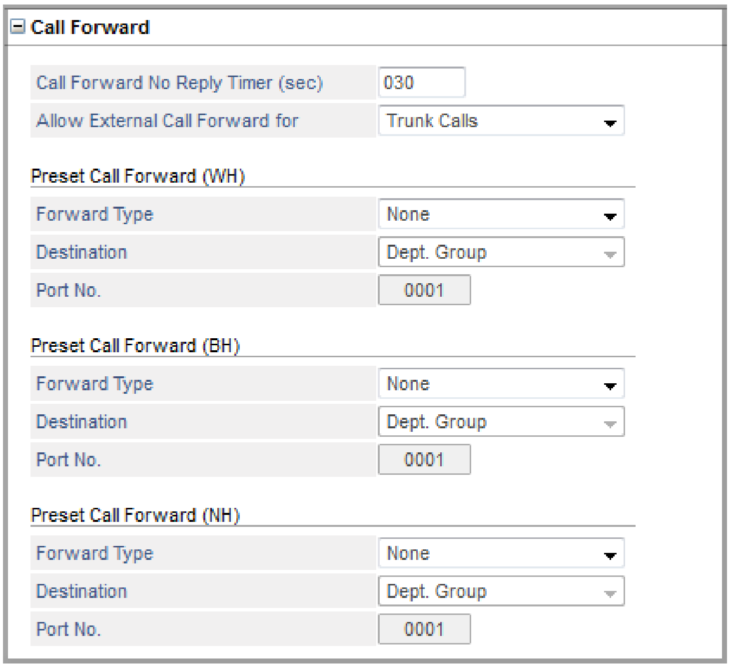
Refer the topic "Station Advanced Feature Template" for instructions on customizing the template and applying the template to extensions using Jeeves.

Changing Call Forward No-Reply Timer

- Login as System Engineer.
- Under **Configuration**, click **Station Advanced Feature Template.**
- Select an Advanced Feature Template from **Template No.** (By default, Template 01 is assigned to all extensions)



- Click **Call Forward** to expand.
- Change **Call Forward No-Reply Timer (Sec)** to the desired value.

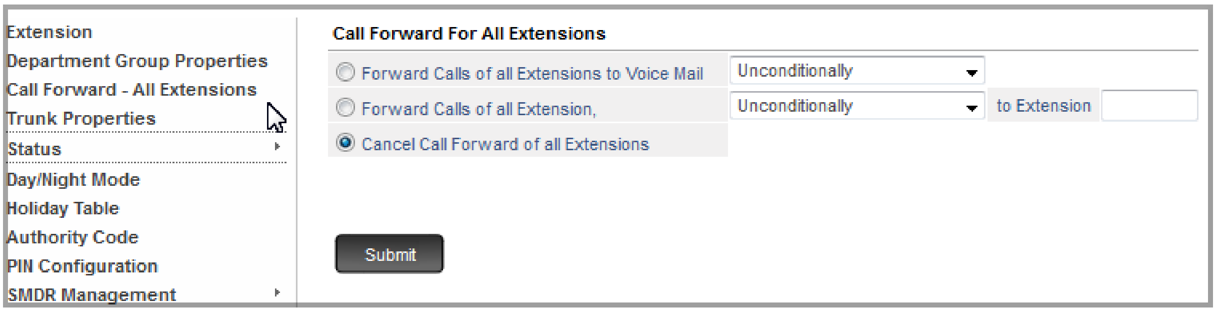


- Click **Submit.**
- Apply the Template now configured with the Call Forward No-Reply Timer to the extensions.

When Call Forward No-Reply is set on a phone that is programmed in a Trunk Landing Group, the calls will be forwarded on expiry of 'Call Forward No-Reply Timer' programmed in the routing group for this member phone. Call Forward No-Reply Timer, programmed in Station Advanced Feature Template will not be applied in this case.

Settings Call Forward for All Extensions

- Login as System Administrator.
- Click **Call Forward- All Extensions**.



  - To set Call Forward of all Extensions to the Voice Mail, select **Forward Calls of all Extensions to Voice Mail**. Select the type of Call Forward—Unconditionally, When Busy, When No Reply, When Busy or No Reply you want to set.
  - To set Call Forward of all Extensions to an Extension, select **Forward Calls of all Extensions to Extension**. Configure the **Extension** Number to which the calls are to be forwarded. Select the type of Call Forward—Unconditionally, When Busy, When No Reply, When Busy or No Reply you want to set.
  - To Cancel Call Forward, select **Cancel Call Forward of all Extensions.**
- Click **Submit.**

How to use

Call Forward can be set/canceled by extension users who are allowed this feature. It can be set/canceled by an extension user for another extension (refer “Call Forward-Remote” to know more).

For Extended IP Phone Users

To set Call Forward,

- Press the 'Forward' key.
  OR
- Dial **13**.
- Scroll to select the desired Call Forward Type.
- Press 'Enter' key.
- Enter destination Phone Number/Voice Mail System/ Department Group Number.
- You get a confirmatory text message and confirmation tone.
- Go Idle or you get dial tone after 3 seconds.

To set Call Forward-Dual Ring,

- First, set the desired Call Forward type.
- Press the 'Forward' key.
  OR
- Dial **13**.
- Scroll to select Dual Ring.

- Press 'Enter' key.
- Select Dual Ring ON
- Press ‘Enter’ key.
- You get a confirmatory text message and confirmation tone.
- Go Idle or you get dial tone after 3 seconds.

- *If the call is to be forwarded to an extension, dial the extension number.*

- *If the call is to be forwarded to an external number, dial Trunk Access Code, then the external phone number and terminate the command with #*.*
  - *For users world wide, Trunk Access Code (TAC) for dialing external numbers are: 0, 5, 61, 62, 63, 64.*
  - *For users in USA, TAC for dialing external numbers are: 9, 5, 81, 82, 83, 84.*

- *If call is to be forwarded on voice mail, dial the Access Code for the Voice Mail System. The default Access Code is 3931. Verify with the System Engineer if the default VMS Access Code has been changed and use the new code to dial the VMS.*

To cancel Call Forward,

- Press 'Forward' Key again.
  OR
- Dial **13**.
- Select 'Cancel'.
- You get a confirmatory text message and confirmation tone.
- Replace Handset on the cradle or you get dial tone after 3 seconds.

Call Forward-Remote

An extension user can set Call Forward for another ('remote') extension from his/her own extension. Thus, Call Forward set for an extension from another extension is called 'Call Forward-Remote'.

This feature can be used by the Operator or the Receptionist to forward the calls for the Managers and other extension users to the destinations where they will be available.

This feature is also useful in Hotels, where the Front Desk can set Call Forward for guests. Refer the *ANANT UCS Hospitality System Manual* to know how this feature can be used in hotels.

Call Forward-Remote is possible only from the System Administration (SA) mode.

How it works

This feature works in the same way as Call Forward. The only difference is that it is set by one extension user for another extension.

For example:

- A and B are extension users.
- A needs to forward calls for B's extension to another extension 'C' or an external number or a Voice Mail System or a Department Number.
- A dials the Call Forward-Remote feature code followed by B's extension number, the destination number where the calls for B should land.
- The system routes all incoming calls for B to the destination number.

How to configure

As Call Forward-Remote can be set only from the SA mode, either the feature 'SA Mode' or 'SA Extension' must be enabled in the Class of Service of extensions that are to be allowed this feature.

It can also be set from Jeeves by the SA.

The feature 'SA Mode' requires a password to be dialed. Users must be provided a password to use this feature from their extensions. The feature 'SA Extension' allows entry into SA mode, without a password.

By default, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS. This Station Basic Feature Template is assigned CoS group 01 by default. The default CoS group 01 has both 'SA Mode' and ‘SA Extension’ disabled.

You may decide which extensions should be allowed Call Forward-Remote feature. In general practice only very few extensions are allowed this feature.

So, to allow this feature to a few extensions only:

a. Define a CoS group with either 'SA Mode' or 'SA Extension' enabled. Recall that the facility 'SA Mode' is password protected, so the extensions allowed access to this feature must also be provided an SA Password.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

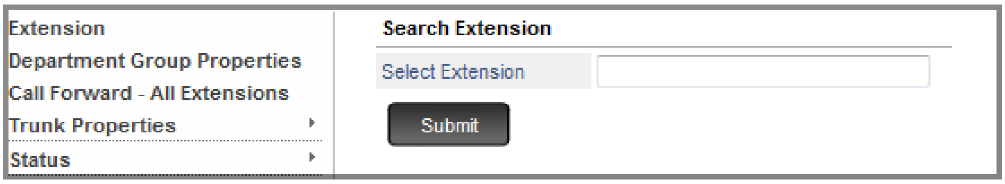
c. Assign this new Template to the extensions to which Call Forward-Remote is to be allowed.

Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for detailed instructions and configuration.

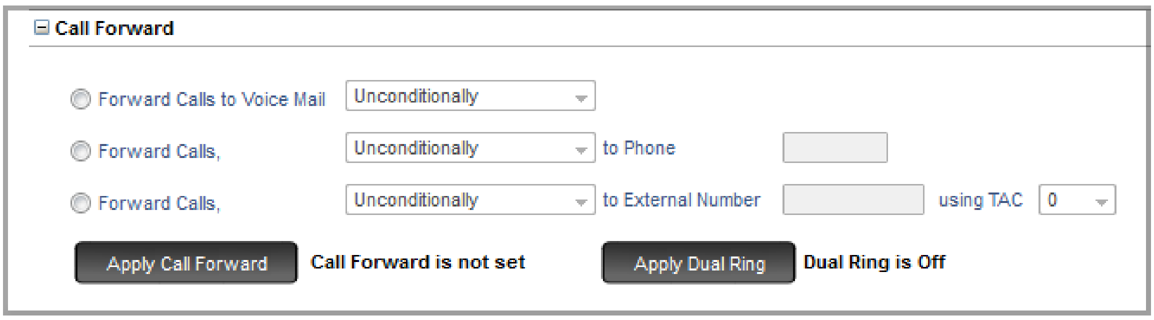
How to use

Settings Call Forward-Remote

- Login as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.
- The searched extension user details appear on your screen.
- Click **Call Forward** to expand.



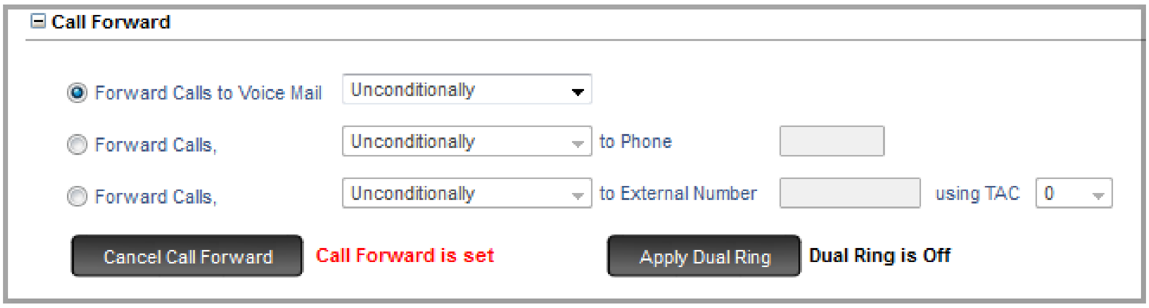
- Select the type of Call Forward you want to set for the extension:
  - To forward calls to voice mail, select **Forward Calls to Voice Mail** and the type of call forward. Default: Unconditionally.
  - To forward calls of this extension to another extension, select **Forward Calls to Phone** and the type of call forward. Default: Unconditionally.
    Enter the extension number to which calls must be forwarded.

- To forward calls of this extension to an external number, select **Forward Calls to External Number** and select the type of call forward. Default: Unconditionally.

  Enter the external number to which calls must be forwarded.

  By default, TAC 0 is assigned for routing external calls. You can change the TAC as per your requirement.

- Click the **Apply Call Forward** button to set Call Forward.



The message “Call Forward is set” appears.

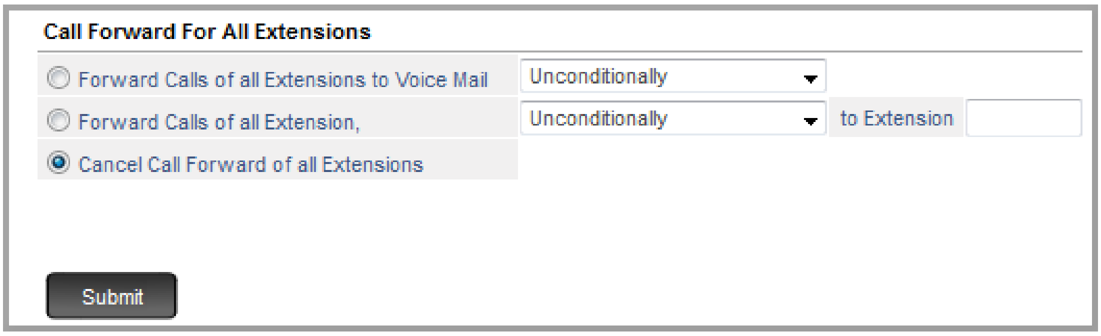
- Click the **Dual Ring** button to set Call-Forward Remote with Dual Ring.

  The message “Dual ring is On” appears.

- To set Call Forward on another extension, follow the same instructions as above.

- You can also forward the calls of all extensions at one go to the same destination. To do this,

- Click **Call Forward-All Extensions**.



- Select the **Forward Calls of all Extensions** and select the Call Forward type.

- In **to Extension**, enter the Call Forward destination number.

- Click **Submit.**

- To cancel, click **Cancel Call Forward of all extensions**.

- Click **Submit**.

For Extended IP Phone Users

To set Call Forward-Remote:

- Press the DSS Key assigned to 'Call Forward-Remote'.
- Enter the Destination Phone Number.
- Scroll to select the desired Call Forward Type:
  - All Calls.
  - If Busy.
  - If No Reply.
  - If Busy or No Reply.
  - Dual Ring.
- Press 'Enter' key.
- Enter Destination Phone Number[81]/Voice Mail System[82]/Department Group.
- You get a confirmation tone and a text message for the Call Forward type set.
- Go Idle or you get dial tone after 3 seconds.

To cancel Call Forward set for an extension:

- Press the DSS Key assigned to 'Call Forward-Remote'.
- Enter Extension Number.
- Scroll to select 'Cancel'.
- Press 'Enter' key.
- You get a confirmation tone and text message for Call Forward canceled.
- Go Idle or you get dial tone after 3 seconds.

81. *If call is to be forwarded to an extension of the ANANT UCS, dial the extension number. If call is to be forwarded on an external number, dial Trunk Access Code, then dial the external phone number and terminate the command with #*.*

    For users world wide, Trunk Access Code (TAC) for dialing external numbers are: 0, 5, 61, 62, 63, 64. For users in USA, TAC for dialing external numbers are: 9, 5, 81, 82, 83, 84.

82. *If call is to be forwarded on voice mail, dial the Access Code for the Voice Mail System. The default Access Code is 3931.*

Call Forward-Scheduled

Extension users may want their calls to be automatically forwarded to a desired destination number during working hours or non-working hours. To cite an example, a Support Technician spends working hours on the field and wants all incoming calls on his extension in the office to be forwarded to his cell phone during working hours. During non-working hours, he wants calls to be forwarded to his voice mail.

Remembering to set and cancel Call Forward and changing the destination number for each Time Zone, that is, working hours, non-working hours, break hours, every day proves to be cumbersome for such extension users.

In addition to “Call Forward”, ANANT UCS supports 'Call Forward - Scheduled', which allows extension users to set call forward for desired Time Zones at one time, and the system automatically forwards the calls to the destination defined for each Time Zone.

How it works

Call Forward-Scheduled supports all the forwarding options of Call Forward: Unconditionally, If Busy, If No Reply, If Busy or No Reply, Dual Ring.

Any of these options can be set for the three Time Zones: working hours, break hours and non-working hours.

The destination for Call Forward-Scheduled can be an internal (extension) number or an external number.

Both 'Call Forward' and Call Forward-Scheduled can be set on the same extension. In this case, priority is given to 'Call Forward' over Call Forward-Scheduled.

The logic for forwarding calls to the destination number remains the same as described in the topic “Call Forward”, illustrated in the following example.

- Extension user A sets Call Forward-Unconditionally to extension B for Non-working hours.
- When there is a call on extension A, the system first checks if there is any 'Call Forward' type (that is, Unconditional, Busy, No Reply, Busy/No Reply, Call Follow Me) set on extension A.
- If 'Call Forward' is set on extension A, the system will follow the logic described in “How it works” under the topic “Call Forward”.
- If no 'Call Forward' is set on extension A, the system will check if Call Forward-Scheduled is set on A.
- Since Call Forward-Scheduled is set on extension A, the system will compare the Time Zone for which the Call Forward is scheduled with the current Time Zone of extension A.
- If the current Time Zone of extension A is the same as the Time Zone set for Call Forward Scheduled, that is, non-working hours, the call will be forwarded to extension B as per the call forward type set.
- As the Call Forward Type set by A is Unconditional, the system will forward the call to B, without checking for the Busy Tone and without waiting for the Call Forward No-Reply Timer to expire.
- If the current Time Zone of extension A is not the same as Time Zone set for Call Forward-Scheduled, the call will not be forwarded. The system will consider that no call forward has been set.

- *Call Forward - Scheduled can be set simultaneously for more than one Time Zone from the same extension. For example, extension A can set Call Forward-Scheduled for working hours, then again set Call Forward-Scheduled for non-working hours, and again for break hours.*
- *A different Call Forward Type can be set for a different Time Zone. For example, extension A can set Call Forward -Unconditional for non-working hours, and Call Forward -Busy for working hours. Also, a different destination number can be set for forwarding calls in each Time Zone. For example, extension A can set Call Forward-Unconditional for non-working hours to a mobile number and set extension B as destination number for working hours.*
- *When more than one Call Forward type is set on the same extension for the same Time Zone, the latest Call Forward type set for the Time Zone will override the previous Call Forward type set for that Time Zone. For example, extension A sets Call Forward -Busy for working hours, then sets Call Forward Busy or No Reply for working hours, the latter will override the former. The system will consider the latest, that is, Busy or No Reply as the Call Forward type for forwarding calls during working hours.*
- *Call Forward-Scheduled can be canceled individually for a desired Time Zone or all at once for all Time Zones.*
- *Call Forward-Scheduled can be set by extension users as well as for extension users from the System Administrator mode.*
- *It is also possible to select the types of calls, that is, internal calls only, or trunk calls, or both, to be forwarded to external numbers. You can program the system to forward internal calls only, or trunk calls only or both trunk calls and internal calls to the external number. For this, the parameter 'Allow External Call Forward for' must be configured in the "Station Advanced Feature Template" of the extensions that want to use Call Forward-Scheduled.*
- *Call Forward-Scheduled when set/canceled from the SA mode, will not depend on the assigned CoS.*

How to configure

The configuring of this feature involves the same parameters as in Call Forward.

Call Forward must be enabled in the Class of Service (CoS) group of the extensions to which this feature is to be allowed. Refer the topic "Call Forward".

If Call Forward No-Reply is to be set, and if required, the Call Forward No-Reply Timer may be programmed in the "Station Advanced Feature Template" applied on the extensions which are to be allowed this feature. Refer the topic "Call Forward".

The types of calls to be forwarded to the external number may be selected in the parameter "Allow External Call Forward for" in the "Station Advanced Feature Template" applied on the extensions which are allowed Call Forward-Scheduled. You may select from 'Internal Calls', 'Trunk Calls' and 'Internal + Trunk Calls'. By default, only trunk calls are forwarded to external numbers.

How to use

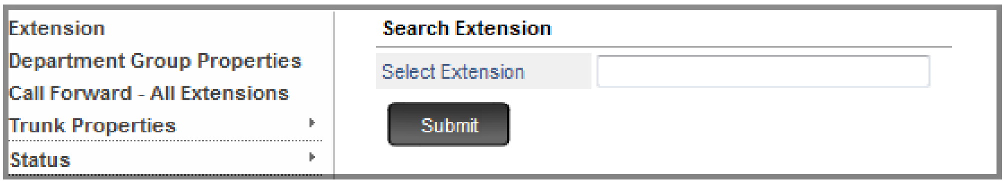
Call Forward-Scheduled can be set/canceled by users for their own extension, or for any other extension from the SA mode.

Setting Call Forward-Scheduled for Extension Users

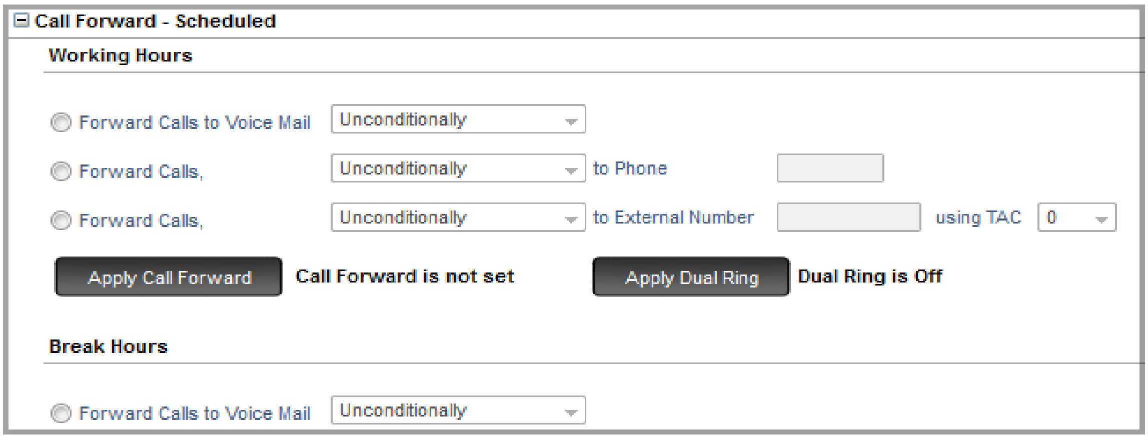
The Operator or any extension user having access to System Administrator mode can set or cancel Call Forward-Scheduled for other extension users using Jeeves.

Setting Call Forward-Scheduled

- Login as System Administrator.
- Click **Extension**.



- Enter the Extension Number or the Extension Name of the extension you want to search in **Select Extension**.
- Click **Submit**.
- The searched extension users details appear on your screen.
- Click **Call Forward - Scheduled** to expand.



- You can set **Call Forward - Scheduled** for Working Hours, Break Hours as well as for Non-working Hours

  To set Call Forward for the extension for the Working Hours, under **Working Hours,** select the type of Call Forward:

  - To forward calls to voice mail, select **Forward Calls to Voice Mail** and the type of call forward. Default: Unconditionally.

- To forward calls of this extension to another extension, select **Forward Calls to Phone** and the type of call forward. Default: Unconditionally.

  Enter the extension number to which calls must be forwarded.

- To forward calls of this extension to an external number, select **Forward Calls to External Number** and select the type of call forward. Default: Unconditionally.

  Enter the external number to which calls must be forwarded.

  By default, TAC 0 is assigned for routing external calls. You can change the TAC as per your requirement.

- Click the **Apply Call Forward** button to set Call Forward -Scheduled.

  The message “Call Forward is set” appears.

- Click the **Dual Ring** button to set Call Forward-Scheduled with Dual Ring.

  The message “Dual ring is On” appears.

- To set call forward for Break Hours and Non-working Hours, follow the same instructions as above.

- To set Call Forward - Scheduled for another extension, follow the same instructions as above.

Setting Call Forward-Scheduled using a Phone

- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number has fewer than 16 digits.*
- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number.*

For Extended IP Phone Users

Call Forward-Scheduled by Extension Users

To set Call Forward-Scheduled:

- Press DSS key assigned to Call Forward-Scheduled.
  OR
- Dial **1175**.
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to the desired Call Forward type for the selected Time Zone.
- Press Enter key to select Call Forward type.
- Enter Destination Number on prompt.
- You get confirmation tone and message showing extension to which Call Forward is set.

To cancel Call Forward-Scheduled for a Time Zone:

- Press DSS key assigned to Call Forward-Scheduled.
  OR
- Dial **1175.**
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to select Cancel.
- Press Enter key.
- You get confirmation tone and message.

To cancel Call Forward-Scheduled in all Time Zones:

- Press DSS key assigned to Call Forward-Scheduled.
  OR
- Dial **1175.**
- Scroll to 'Cancel Call Forward'.
- Press Enter key.
- You get confirmation tone and message.

Call Forward-Scheduled from SA mode

To set Call Forward-Scheduled:

- Press DSS key assigned to Call Forward-Scheduled - Remote.
- Enter extension number (from which calls are to be forwarded)
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to the desired Call Forward type for the selected Time Zone.
- Press Enter key to select Call Forward type.
- Enter Destination Number on prompt.
- You get confirmation tone and message showing extension to which Call Forward is set.

To cancel Call Forward-Scheduled for a Time Zone:

- Press DSS key assigned to Call Forward-Scheduled - Remote.
- Enter extension number (for which it is to be canceled)
- Scroll to the desired Time Zone.
- Press Enter key to select Time Zone.
- Scroll to select Cancel.
- Press Enter key.
- You get confirmation tone and message.

To cancel Call Forward-Scheduled in all Time Zones:

- Press DSS key assigned to Call Forward-Scheduled - Remote.
- Enter extension number (for which it is to be canceled)
- Scroll to 'Cancel Call Forward'.
- Press Enter key.
- You get confirmation tone and message.

Call Forward-When Not Registered

SIP Phones connected as extensions may fail to register with ANANT UCS when the network link is down or when there is power failure. Using the Call Forward-When Not Registered feature, the extension users can have their calls forwarded even when their extension phone is not registered with ANANT UCS.

The destination for ‘Call Forward-When Not Registered’ can be an internal number, an external number or the Voice Mail.

It is also possible to select the types of calls—internal calls only, or trunk calls, or both—to be forwarded to external numbers.

Call Forward-When Not Registered can be set/canceled by,

- the System Administrator mode
- SIP phone users from their phones

Call Forward- When Not Registered can also be set for each Time Zone—Working Hours, Break Hours, Non-working Hours, by setting *Call Forward-When Not Registered - Scheduled*.

Call Forward - When Not Registered-Scheduled can be set for more than one Time Zone at a time on the same SIP phone. It can be canceled individually for a desired Time Zone, or all at once for all Time Zones. A different destination number can be set for forwarding calls in each Time Zone. For example, the destination number for non-working hours can be a mobile number and the destination number for working hours can be another extension number.

When the VARTA AMP100 / ADR100 application is in background, and is a member in a Routing Group or Department Group, then Call Forward functionality will not be achieved.

Feature Interaction:

- If ‘Call Forward-Unconditional’ and ‘Call Forward-When Not Registered’, have been set on the same SIP phone. ‘Call Forward-Unconditional’ will have priority over ‘Call Forward-When Not Registered’.

- If ‘Call Forward-Scheduled-Unconditional’ and ‘Call Forward-When Not Registered-Scheduled’, have been set on the same SIP phone. ‘Call Forward - Scheduled - Unconditional’ will have priority over ‘Call Forward-When Not Registered-Scheduled’.

How to configure

The Call Forward-When Not Registered feature does not require any specific configuration except:

- ensuring that 'Call Forward' in the “Class of Service (CoS)” group in the “Station Basic Feature Template” applied to the SIP phones.

- if required, selecting the types of calls to be forwarded to the external number. By default, only trunk calls are forwarded to external numbers. If you want to select a different type of call, configure the parameter “Allow External Call Forward for” in the *Station Advanced Feature Template* applied to the SIP phones. Refer the sub-topic “Station Advanced Feature Template”, under *Configuring Extensions*.

- If you want to allow Call Forward-When Not Registered to be set only by the System Administrator (SA) for the extension users, the System Engineer (SE) must disable ‘Call Forward’ feature in the Class of Service (CoS) group in the Station Basic Feature Template applied to the SIP phones.

If you disable ‘Call Forward’ in the CoS of a SIP phone, the user will not be able to set any other type of Call Forward.

Setting Call Forward-When Not Registered

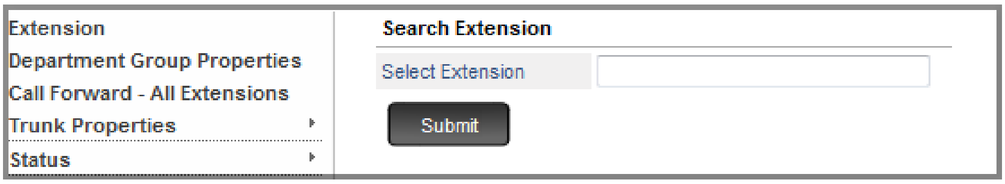
Call Forward-When Not Registered can be set from

- the SA mode from Jeeves.
- the SIP phones connected as extensions.

Call Forward-When Not Registered set by SA

- Login as System Administrator.

- Click **Extension**.

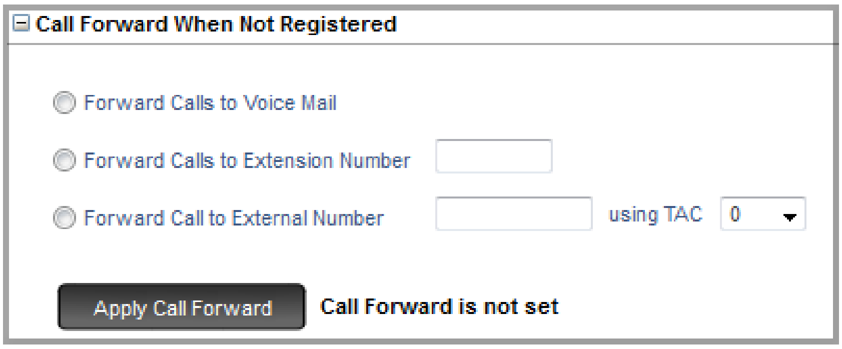


- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.

- Click **Submit**.

- The searched extension user details appear on your screen.

- Click **Call Forward When Not Registered** to expand.



- Select the destination for forwarding calls when the SIP Extension fails to register from the following:

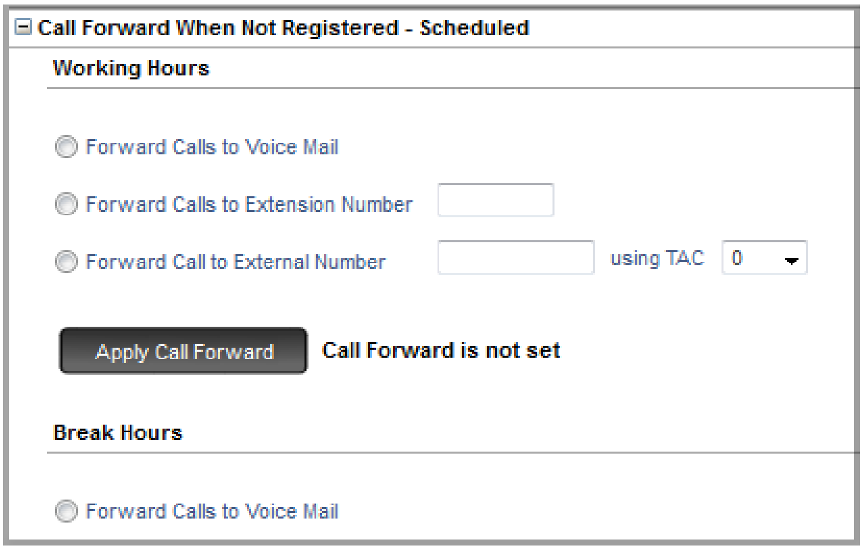
  - **Forward Calls to Voice Mail**.

  - **Forward Calls to Extension Number**. If you select this option, you must enter the desired Extension Number in the corresponding box.

  - **Forward Calls to External Number**. If you select this option, you must enter the desired external number in the corresponding box. Also, assign a trunk to route the call by selecting the Trunk Access Code from the **using TAC** list.

- Click the **Apply Call Forward** button. The message “Call Forward is set” appears.

- To set time-zone based Call Forward - When Not Registered, click **Call Forward When Not Registered-Scheduled** to expand.



- To set Call Forward When Not Registered for working hours, under **Working Hours,** select the desired destination from the following options:
  - **Forward Calls to Voice Mail**.
  - **Forward Calls to Extension Number**. If you select this option, you must enter the desired Extension Number in the corresponding box.
  - **Forward Calls to External Number**. If you select this option, you must enter the desired number in the corresponding box, and assign a trunk to route the call by selecting the Trunk Access Code in the **using TAC** list.
- Click the **Apply Call Forward** button. The message “Call Forward is set” appears.
- To set call forward for Break Hours and Non-working Hours, follow the same instructions as above.
- To set Call Forward When Not Registered - Scheduled for another extension, follow the same instructions as above.

Call Forward-When Not Registered set/canceled by SIP Phone Users

SIP extension users can set/cancel Call Forward-When Not Registered from their SIP phones. The SIP phone may be a Matrix Extended IP Phone or any Standard SIP phone.

Using Matrix Extended IP Phone

- Lift handset.
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Scroll to the desired option.

To set Call Forward - When Not Registered regardless of time-zone,

- Select ‘Always’ and press 'Enter' key.
- Select ‘Set’ and press ‘Enter’ key.

To set Call Forward When Not Registered - Scheduled,

- Select ‘Working Hours’/’Break Hours’/’Non-Working Hours’, and press ‘Enter’ key.
- Select ‘Set’ and press ‘Enter’ key.

- On the prompt, ‘Forward to Number’, enter the Destination Number—Extension Number/External Number/ Voice Mail System.

- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number has fewer than 16 digits.*
- *If the you want to route the calls to the Voice Mail, enter the VMS Access Code as the destination number.*
- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number and #*.*

- You get confirmation tone and message.

To cancel Call Forward - When Not Registered,

- Lift handset.
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Select ‘Always’ and press 'Enter' key.
- Select ‘Cancel’ and press ‘Enter’ key.

To cancel Call Forward When Not Registered - Scheduled for each Time Zone,

- Lift handset.
- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Select the desired time-zone ‘Working Hours’/’Break Hours’/’Non-Working Hours’, and press ‘Enter’ key.
- Select ‘Cancel’ and press ‘Enter’ key.

To cancel All Call Forward When Not Registered,

- Press DSS key assigned to Call Forward-When Not Registered (if programmed).
  OR
- Dial ***13**.
- Select ‘Cancel Call Forward’ and press ‘Enter’ key.

Using Standard IP Phone

- Lift handset.

To set Call Forward - When Not Registered regardless of time-zone,

- Dial ***13-1-1-Destination Number**

- *The destination number for forwarding calls can be a maximum of 16 digits. Terminate the command with #* if destination number has fewer than 16 digits.*
- *If you want to route the calls to the Voice Mail, enter the VMS Access Code as the destination number.*
- *If the destination number is an external number, enter the Trunk Access Code followed by the destination number and #*.*

To set Call Forward - When Not Registered - Scheduled,

- Dial ***13-2-1-Destination Number** for working hours.
- Dial ***13-3-1-Destination Number** for break hours.
- Dial ***13-4-1-Destination Number** for non-working hours.
- Replace handset.

To cancel Call Forward - When Not Registered,

- Lift handset.
- Dial ***13-1-0**.

To cancel Call Forward When Not Registered - Scheduled,

- Dial ***13-2-0** for working hours
- Dial ***13-3-0** for break hours.
- Dial ***13-4-0** for non-working hours.
- Replace handset.

To cancel All Call Forward-When Not Registered,

- Lift handset.
- Dial ***13-0.**
- Replace handset.

Call Hold

Call Hold enables you to put an on-going conversation (with an internal or external number) on hold. ANANT UCS offers three types of Call Hold:

- **Exclusive Hold:** An on-going conversation is put on hold from an Extended IP Phone and is retrieved from the same Extended IP Phone that put it on hold.
- **Global Hold:** An on-going conversation is put on hold from an Extended IP Phone and is retrieved from any Extended IP Phone connected to ANANT UCS.
- **Consultation Hold:** An on-going conversation is put on hold in order to perform any further activity, such as Call Transfer, Conference, Call Toggle.

ANANT UCS supports interoperability with the Polycom IP Phones. When any extension of ANANT UCS puts a SIP Extension on hold (Exclusive, Global or Consultation Hold), ANANT UCS will send Re-Invite message to the SIP Extension put on hold.

How it works

Exclusive Hold using the Hold Feature Key

When a call is put on Exclusive Hold,

- ANANT UCS starts the *Exclusive Hold Retrieval Timer* (configurable; default: 2 minutes).
- The call remains on hold for the duration of this timer.
- The extension user can retrieve the call within this timer.
- If the call is not retrieved before the expiry of this timer, it will return back to the Extended IP Phone that has put the call on hold. The Extended IP Phone rings and the user may answer the call.
- The returned call is disconnected, if the Extended IP Phone is not in the idle state, or if the call is not answered by the IP Phone user.

A call placed on Exclusive Hold can be retrieved in the following ways:

- Pressing the Hold key again (when the Extended IP Phone is idle).
- Pressing the Call Appearance key of the call put on hold (when the Extended IP Phone is busy).
- Pressing DSS key assigned to the Trunk/extension you put on hold.
- Answering the call, when it returns at the end of the Exclusive Hold Retrieval Timer.

To be able to place calls on Exclusive Hold, you must select **Exclusive Hold** as the **Default Call Hold Type** in the System Parameters. See “System Parameters” for instructions.

If multiple calls have been put on Exclusive Hold and you press the Hold key, then the last call that was put on hold will be retrieved.

Exclusive Hold using DSS Key

Extended IP Phone users can configure upto 8 DSS keys for Exclusive Hold, namely Exclusive Hold 1 to 8. Using DSS key assigned to Exclusive Hold calls can be put on Exclusive Hold only. The functioning of the DSS key does not depend on the Default Call Hold Type you select.

For instructions to assign a DSS Key to Exclusive Hold, see "DSS Keys Programming".

When a call is put on Exclusive Hold using DSS key assigned to Exclusive Hold 1,

- ANANT UCS starts the *Exclusive Hold Retrieval Timer* (programmable; default: 2 minutes).
- The call remains on hold for the duration of this timer.
- The extension user can retrieve the call within this timer.
- If the call is not retrieved before the expiry of this timer, it will return back to the IP Phone that has put the call on hold. The Extended IP Phone rings and the user may answer the call. The LCD displays the message 'Held X Recall', where X is the hold position number.
- The returned call is disconnected, if the Extended IP Phone is not in idle state, or if the call is not answered by Extended IP Phone user.

A call placed on Exclusive Hold can be retrieved in the following ways:

- Pressing the DSS key (when the Extended IP Phone is idle).
- Pressing the Call Appearance key of the call put on hold (when the Extended IP Phone is busy).
- Pressing DSS key assigned to the Trunk/extension you put on hold.
- Answering the call, when it returns at the end of the Exclusive Hold Retrieval Timer.

Global Hold

When a call is placed on Global Hold,

- The call remains connected in the system. The call remains on hold for the duration of the *Global Hold Retrieval Timer* (configurable; default: 60 seconds).
- Any Extended IP Phone connected to ANANT UCS can pick up the call put on Global hold by:
  - Pressing DSS key assigned to the Trunk put on Global Hold.
  - Pressing the DSS key assigned to the extension put on Global Hold.
- If this call is not retrieved before the expiry of the Global Hold Retrieval Timer, the call is returned to the Extended IP Phone which put it on hold. The Extended IP Phone rings and the user may answer the call.
- The returned call is disconnected, if the Extended IP Phone is not in idle state, or if the call is not answered by Extended IP Phone.

To be able to place calls on Global Hold, you must select 'Global Hold' as the Default Call Hold Type in the System Parameters of ANANT UCS. The Extended IP Phone (which picks up the call) must have a DSS Key to access the Trunk or the Extension which is put on hold. See “System Parameters” for instructions.

- *ANANT UCS provides the flexibility to use Exclusive Hold and Global Hold at the same time. You can put calls on Exclusive Hold even when Global Hold is enabled in the system using the Hold Feature key only.*
- *External calls cannot be put on global hold.*
- *You must first retrieve the call that is put on Exclusive or Global Hold, if it is to be transferred or included in a Conference.*

Consultation Hold

During an on-going conversation, any Extended IP Phone can place a call on Consultation Hold to perform any of the following:

- “Call Transfer”
- “Call Toggle”
- “Conference-3 Party”, “Conference-Multiparty”, “Conference Dial-In”
- “Call Park”
- “Mute”
- “Call Chaining”
- “Conversation Recording”

The call is released from the held state once the operation has been performed or canceled.

For instructions on using the above mentioned features, refer *How to Use* of the respective feature.

How to configure

For *Exclusive and Global Hold*, you must configure the following parameters:

- **Class of Service:** Call Hold must be enabled in the Class of Service (CoS) of the Extended IP Phone you want to allow this feature.

  By default, Station Basic Feature Template 01 assigned to all extensions of ANANT UCS, Call Hold is included in the 'Basic Features' assigned to all Class of Service groups, including the default CoS group 01. So, all extensions of ANANT UCS can use this feature.

  Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” to know more.

- **Call Hold Type:** Enable the desired option, that is, Exclusive Hold or Global Hold in the “System Parameters”.

- **Send Re-INVITE over SIP Trunk on Hold:** When an external call over a SIP Trunk is put on hold by any extension, and you want ANANT UCS to send Re-INVITE message over SIP Trunk to the remote end, you must enable this flag on the SIP Trunk. See “Configuring SIP Trunks” to know more.

- **DSS Keys:** Program DSS Keys for Trunks and Extensions which are allowed to retrieve calls on Global Hold. Program the DSS Keys for Exclusive Hold, if required. Refer the topic “DSS Keys Programming”, “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in

"Configuring Matrix SPARSH VP210" and "DSS Key Settings" in "Configuring Matrix SPARSH VP510" for instructions.

- **Global Hold Retrieval Timer:** Change the default setting of this timer to the desired duration, if required.
- **Exclusive Hold Retrieval Timer:** Change the default setting of this timer to the desired duration, if required.

  For instructions to change the Timers, see "System Timers and Counts".

For *Consultation Hold* to work, Call Hold must be enabled in the "Class of Service (CoS)" of the Extended IP Phone you want to allow this feature.

How to use

Exclusive Hold

For Extended IP Phone Users

To put a call on Exclusive Hold, when Exclusive Hold is selected as the Default Call Hold Type:

- You are in speech with a Trunk/with an extension.
- Press 'Hold' Key.
- Go idle.
- Call with Trunk/extension is put on 'Exclusive Hold'.

To retrieve the call:

- Press 'Hold' Key again.
- Press Call Appearance key of your Extended IP Phone
- Press DSS Key of the Trunk/extension you put on hold from your Extended IP Phone.

To put a call on Exclusive Hold, when Global Hold is enabled:

- You are in speech with a Trunk/with an extension.
- Press 'Hold' Key twice in quick succession within 2 seconds.
- Go idle.
- Call with Trunk/extension is put on 'Exclusive Hold'.

To retrieve the call:

- Press 'Hold' Key again.
- Press Call Appearance key of your Extended IP Phone.
- Press DSS Key of the Trunk/extension you put on hold from your Extended IP Phone.

To put a call on Exclusive Hold using DSS key assigned to Exclusive Hold:

- You are in speech with on a Trunk/with an extension.
- Press DSS key assigned to Exclusive Hold.
- Go idle.
- Call with Trunk/extension is put on 'Exclusive Hold'.

To retrieve the call put on hold using DSS key:

- Press DSS Key again.
- Press Call Appearance key of your Extended IP Phone.
- Press DSS Key of the Trunk/extension you put on hold from your Extended IP Phone.

Using DSS key assigned to Exclusive Hold calls can be put on Exclusive Hold only. The functioning of the DSS key does not depend on the Default Call Hold Type you select in the System Parameters.

Global Hold

For Extended IP Phone Users

To put a call on Global Hold, when Global Hold is selected as the Default Call Hold Type:

- You are in speech with a Trunk/with an extension.
- Press 'Hold' Key.
- Go idle.
- Call with Trunk/extension is put on 'Global Hold'.

To retrieve a call on Global Hold:

- From any Extended IP Phone, press the DSS Key of the Trunk/extension put on Global Hold.

To put a call on Global Hold, when Exclusive Hold is enabled:

- You are in speech with a Trunk/with an extension.
- Press 'Hold' Key twice in quick succession within 2 seconds.
- Go idle.
- Call with Trunk/extension is put on 'Global Hold'.

To retrieve a call on Global Hold:

- From any Extended IP Phone,
- Press the DSS Key of the Trunk/extension put on Global hold.

Consultation Hold

For Extended IP Phone Users

To put a call on Consultation Hold:

- Press the DSS Key assigned to the feature (if configured).
  OR
- Press Transfer key.

Call Park

Call Park allows you to place a call on Hold, so it can be retrieved from the same or another extension of the system.

A call is 'parked' when the extension user temporarily places the call into a location in the system called 'Orbit'. The user can attend to other calls. The parked call can be retrieved on completion of the current call by dialing the Orbit number.

Call Parking is useful in offices housed in different parts of a building or multi-storied offices. It is useful in situations like:

- the person who picked up the call is not the desired called party or the desired party is at an unknown location. The person who picked up the call can then either go to find the desired called party or call other numbers to find him/her. When found, the desired called party can pick up the call from the same or any extension by dialing the Orbit number.

- the person who picked up the call may have to go to another part of the office to look up a file or consult a colleague. The person can park the call and continue the conversation from the other part of the office.

- the person who picked up the call is an Operator. The Operator needs to handle many calls on a daily basis. It becomes difficult to know the available free orbit or to remember the orbit number after parking the call. In such cases, the Operator can assign a separate DSS key for each General Orbit. The availability of the orbit will be indicated by the LED of the assigned DSS key. Thus, the Operator can now easily park or retrieve the call by pressing the DSS key.

ANANT UCS offers various types of Call Park facility:

- **Call Park-General Orbit:** The calls can be parked either manually or automatically in the General Orbit. The calls parked in the General Orbit can be retrieved from any extension including your own extension. General Orbit number can vary from 2 to 9.

  - **Manual Call Park:** The extension user can park calls in any of the 8 General Orbits, which are like fictional extensions located in the system. The calls parked in the General Orbit can be picked up from any extension by dialing the General Orbit Number. At a time, only one call can be parked in each General Orbit.

    The extension user can assign a separate DSS key for each General Obit, this will help the user to know the status of the Orbit - available, occupied. During an ongoing call, the user can park the call by pressing the DSS key of the available orbit depending on the LED indication. When the call is parked in the orbit, the LED blinks in blue and when it is free, the LED is off.

  - **Auto Call Park:** To park the call automatically in the free General Orbit, ANANT supports General Call Park - Auto. When the DSS key assigned to General Call Park - Auto is pressed, the system searches for a free General Orbit (2 to 9) and automatically parks the call in the free orbit. The Orbit number is then displayed on the phone’s LCD. At a time, only one call can be parked in each General Orbit.

- **Call Park-Personal Orbit:** Each IP Phone connected as an extension has one Personal Orbit. Calls parked in personal orbit can be picked up only from where the call is parked. So, no other person can pick up this call. Multiple calls can be parked in the Personal Orbit at a time.

Extension users can park the call either in the General Orbit or the Personal Orbit by dialing an Orbit Number from 1 to 9, where:

- 1 is the Personal Orbit Number.
- 2 to 9 are General Orbit Numbers.

After parking a call, the extension user can continue to make and answer other calls and use other system features.

For Standard SIP phones, ANANT UCS supports Call Park and Retrieve using REFER Message. For a list of IP phones on which this feature has been tested, see *"ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.*

How it works

A and B are extension users. C, D and E are callers.

Parking Calls in General Orbit

- C calls B.
- A picks up the call.
- As B is not present at his extension, A parks the call in General Orbit Number 2 by dialing the Access Code.
- C is played on-hold music.
- A tries to find B (by either calling several numbers or by going in person or sending someone).
- The parked call remains in orbit for the duration of the Call Park Timer, which is set to 2 minutes by default.
- A finds B.
- B retrieves the call from another extension by dialing the feature access code for retrieving Call Park and Orbit number 2.
- If A cannot locate B or if B cannot attend the call, A can also retrieve the call from his extension.

However,

- If neither A nor B retrieves the parked call within the Call Park Timer, the system will hunt for the extension that parked the call (A) on the expiry of the Call Park Timer.
- Meanwhile, if A is busy, the system again keeps the call parked in orbit number 2 for the period of the Call Park Timer. This process continues for the duration of the Call Park Release Timer, which is set to 3 minutes by default.
- If A is free, the system will ring on A's phone. A gets connected to C again.
- If A does not retrieve the parked call till the end of the Call Park Release Timer, C gets disconnected.

Automatically Parking Calls in General Orbit

- C calls B.
- A picks up the call.
- As B is not present at his extension, A presses the DSS key assigned to General Call Park - Auto.
  OR
  Press the Transfer key and dial access code for Call Park+0.

- System internally checks for the available free General Orbit and parks the held call there.
- Once the call is parked automatically in the free orbit, a confirmation message appears on the LCD of A's phone 'Call Parked in Orbit X' (where value of X varies from 2 to 9).
- C is played music on-hold.
- A tries to find B (by either calling several numbers or by going in person or sending someone).
- The parked call remains in orbit for the duration of the Call Park Timer, which is set to 2 minutes by default.
- A finds B.
- B retrieves the call from another extension by dialing the feature access code for retrieving Call Park and Orbit number X.
- If A cannot locate B or if B cannot attend the call, A can also retrieve the call from his extension.
- If the parked call is not retrieved before the expiry of the Call Park Release Timer, C gets disconnected.

Parking Calls in Personal Orbit

Parking calls in the Personal Orbit works the same way as in General Orbit. The only difference is that A can park multiple calls by dialing the Personal Orbit Number. But calls can be retrieved from A's phone only.

When there are multiple calls to be retrieved from the Personal Orbit, they are retrieved one by one, without following any particular sequence like FIFO or LIFO.

To be able to use 'Call Park', this feature must be enabled in the CoS of the requesting extension. However, for retrieving parked calls, the system does not check CoS. So any extension can retrieve parked calls.

How to configure

To provide this feature to extensions,

- Enable the Call Park in the "Class of Service (CoS)" of the "Station Basic Feature Template" of the extensions. By default, this feature is enabled in the CoS of all extension types for all the time zones.

- Assign a DSS key for 'General Call Park - Auto'.
  OR
  Assign a separate DSS key for each General Orbit (2-9).

  For instructions refer to "DSS Keys Programming".

- If required, you may change the duration of the Call Park Timer and the Call Park Release timer. See "System Timers and Counts" for instructions.

How to use

For Extended IP Phone Users

To park a call:

- You are in speech with extension/external caller.
- Press DSS Key assigned to 'Call Park'.
  OR
- Press Transfer Key and dial **115**
- Enter Orbit Number (1-9)
  (Personal Orbit:1; General: 2-9).

To park a call manually in the available General Orbit (indicated through LED):

- You are in speech with extension/external caller.
- Press DSS Key assigned to the specific General Orbit (2-9).
  Call is parked.

To park a call automatically in the General Orbit:

- You are in speech with extension/external caller.
- Press DSS Key/CSF Key[83] assigned to ‘General Call Park - Auto[84]’
  OR
  Press Transfer Key and dial 115-0.
- The system checks for the available free General Orbit (2-9) and automatically parks the held call in it.
  The General Orbit number is displayed on the phone’s LCD.

To retrieve a parked call from your phone, when your phone is in idle state:

- Press DSS Key assigned to 'Call Park - Retrieve'.
  OR
- Dial **116**
- Enter Orbit Number where you parked the call (1-9)
  (Personal Orbit:1; General: 2-9).
- You are in speech with the extension/external caller.

To retrieve a parked call from your phone, when you are in speech with someone:

- Press Transfer Key.
- Press DSS Key assigned to 'Call Park - Retrieve'.
  OR
- Dial **116**
- Enter Orbit Number where you parked the call (1-9)
  (Personal Orbit:1; General: 2-9).

83. *This key is applicable only for SPARSH VP510.*
84. *This feature is not applicable for SPARSH VP330 and SPARSH VP210.*

Call Logs

ANANT UCS stores the details of 20 each, of the following types of calls:

- **Missed calls:** incoming calls that were not answered by extension users.
- **Answered calls:** incoming calls answered by extension users.
- **Dialed calls:** calls made by extension users.

The call history of each of the above types of calls is stored by Name, Number, and Date-Time of the Call.

If there is no name in the CLI of the above types of calls, the system stores and displays the Number and the Date-Time. In case there is no number in the CLI, the system will display the Port number on/from which the call was received/made.

The Call Logs contain details of both internal as well as external calls made or received by the extension users.

Using call logs you can:

- **view call history:** you can see the calls you missed, answered or dialed.
- **make calls:** you can call any number that you have missed, answered, or dialed.
- **edit the numbers:** you can change or modify the number in the call log. This is useful when the CLI received and stored in the call log is not in the same format that is to be used to make calls.
- **save the numbers:** you can store the external numbers in your call logs in the "Personal Directory" and use them for “Personal Abbreviated Dialing”.

The maximum number of calls that can be stored under each Call Log type is 20. The logs will be cleared automatically using the First-In, First-Out method, that is, the latest call detail will replace the record of the oldest call detail.

Given the limited Call Log capacity, the system also allows you to choose if you want internal calls to be displayed or not in the Missed, Answered and Dialed Call Logs. And accordingly it will store internal calls in the logs.

The system stores each Missed, Answered and Dialed call individually even if the same number is received multiple times.

How to configure

This feature does not require any specific configuration, except:

- Selecting whether internal calls should be logged in the Missed, Answered and Dialed Call Logs. This can be done on the 'System Parameters' page of Jeeves.

- Configuring of a DSS key for the Call Logs feature. For instructions please refer the topic “DSS Keys Programming”, “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP210” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510” for instructions.

Configuring Internal Call Logging

- Login as System Engineer.
- Under **Configuration**, click the **System Parameters.**

- Click **System Parameters** to expand.
- You may enable any or all of the following flags by selecting the respective check box:
    - Store Internal Calls in Missed Call Log
    - Store Internal Calls in Dialed Call Log
    - Store Internal Calls in Answered Call Log
    - Store Internal Calls in Redial Call Log
- Click **Submit.**

How to use

The Call Logs feature allows you to view calls and edit numbers, make calls to any number logged, and store numbers.

For Extended IP Phone Users

Viewing Call Logs

There are two ways to view call logs:

- From the Phone Menu
- Using the Feature Key assigned to Call Logs.

If you are using a DSS Key for the Call Logs feature, whenever there is a missed call, the LED of the DSS key will glow. If you press the Call Logs key, the system will display the last missed call details.

To view Call Logs from Phone Menu:

- Press Enter key when the phone is idle.
- Place your cursor on Call Logs option, press Enter key.
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- To view another call log, scroll with the Back Navigation key to return to the previous option.

- You may exit the Phone Menu by going ON-Hook or pressing the Cancel key.

- *If there is no name in the CLI, the Call Log will only display the number.*
- *If you press the 'Enter' key, the system will dial out the number you just viewed.*

To view Call Logs using DSS Key:

- Press DSS Key programmed for Call Logs, when the phone is idle.
- Scroll to select the desired Call Log: Missed, Answered, Dialed.
- The phone will display the call log details.
- To view another call log, scroll with the Back Navigation key to return to the previous option.
- You may exit the Phone Menu by going ON-Hook or pressing the Cancel key.
  OR
- Press the DSS Key assigned the Call Logs feature, when it glows.
- The Phone will display the Call Logs: Missed, Answered, Dialed.
- Press Enter key to select the Missed Call Log.
- The phone will display the call log details.
- To view another call log, scroll with the Back Navigation key to return to the previous option.
- You may exit the Phone Menu by going ON-Hook or pressing the Cancel key.
- The LED of the Call Logs DSS key will be turned off once you have viewed the missed call.

Editing Numbers in Call Logs

- Go to Call Logs from the Phone Menu or by pressing Call Logs DSS Key. (see instructions given above).
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- Scroll with the Up/Down Navigation Key to reach the desired number.
- To edit the number, move the cursor with the Forward (>) navigation key.
- Place the cursor under the digit you want to delete.
- Press 'Cancel' key to delete a digit.
- To insert a digit, place the cursor where you want to insert the digit, and enter the digit using the dial pad. The digit will be inserted in the number string accordingly.
- Repeat the same to delete/insert another digit.
- After editing the number, you may store it in the Personal Directory or dial the edited number by pressing the Enter key.

The original number (you now changed) will remain unaffected in the Log. However, if you make a call to the new number (you changed), it will be logged in the "Dialed" call log and the Last Number Redial list.

Making calls using Call Logs

- Go to Call Logs from the Phone Menu or by pressing Call Logs DSS Key.
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- Scroll with the Up/Down Navigation Key to reach the desired number.
- Press Enter key.
- The system will dial out the selected number using the Outgoing Trunks assigned for dialing '0'.
- The dialed number will be logged in the "Dialed" call log and the Last Number Redial List.

Storing numbers of Call Logs in the Personal Memory of the Phone

To store any external call record (trunk call) from Received, Missed and Dialed call logs to Personal Directory,

- Go to Call Logs from the Phone Menu or by pressing Call Logs DSS Key.
- Scroll with the Up/Down Navigation Key to reach the desired Call Log: Missed, Answered, Dialed.
- Press Enter key to select the desired Call Log.
- The phone will display the call log details.
- Scroll with the Up/Down Navigation Key to reach the desired number.
- Edit the number (following instructions given above), if required.
- Press 'v' (Down Navigation key).
- You will get the prompt: "Enter Name"[85].
- Enter the name of the contact.
- Press Enter key.
- You will get the confirmation message: "Stored at Memory: <XXX>".

- *When you store the number in the Personal Directory, the system will automatically assign Trunk Access Code "0".*
- *If all 25 Location Index Numbers of the Personal Directory are already configured, the message "Memory Full" will appear on your phone's display and you will get an Error Tone. Refer the topic* *"Abbreviated Dialing" to know more.*

85. *Only if there is a free Memory Index in the Personal Directory.*

Call Pick Up

Call Pick-Up allows extension users to answer calls ringing on other extensions from their own extension; without physically going to the ringing extensions.

Extension users can 'pick-up' both internal and trunk calls ringing on other extensions.

As extension users can answer calls of their colleagues or co-workers without physically going to their extensions, this feature ensures that all incoming calls are answered.

Call Pick-Up Notification will be displayed if you have SPARSH VP510. Make sure you have enabled the **Call Pick-up Notification (Only for SPARSH VP510)** in “Configuring Matrix SPARSH VP510”. For details also refer to the EON510_SPARSH VP510 V2 User Guide.

ANANT UCS offers two types of Call Pick-Up:

- **Call Pick Up-Group** - extensions are assigned to Pick-Up Groups. Any extension in a Pick-Up Group can answer calls ringing on other extensions within the same group only.
- **Call Pick-Up Selective** - calls ringing on any extension of the system can be answered.

- *On SIP extensions, ANANT UCS supports Call Pickup-Selective and Call Pickup-Group using Temporary Subscription. For a list of IP phones on which this feature has been tested, see* *“ANANT UCS Features tested on IP Phones of different Brands” in the Appendix.*
- *ANANT UCS will send only first 3 ringing call's information in NOTIFY message to the SIP Extension, which has requested Group Call Pickup. This feature has been supported in SIP Phones of CISCO and POLYCOM.*
- *SIP Extension which has subscribed for BLF of SIP Extension of ANANT UCS, ANANT UCS will send information for the call present in the first call loop only.*

How it works

Call Pick-Up Group

- Extensions must be assigned to Call Pick-Up Groups.
- As many as 99 such groups may be formed.
- Each group is assigned a number 01 to 99.
- For example, extensions 2007, 2008, 2009, 2010. 2011, 2012, 2013 are assigned to Pick-Up Group number 03.
- When an extension in this group rings, any extension in the group can pick up the call by dialing the feature access code for “Call Pick-Up Group” (default: 4).
- The ringing extension should be in the same Pick-Up Group.

Call Pick-Up Selective

- Extensions need not be in Call Pick-Up Groups.

- Whenever an extension in the system rings, the call can be picked up by any extension of the system by dialing the feature access code and the number of ringing extension.

When more than one extension in a Pick-Up Group is ringing, you can choose which one to answer first, using Call Pick-Up Selective.

Feature Interactions:

- **Call States:** Call Pick-Up will fail if the ringing extension goes into idle state just when you are dialing the pick-up access code.
- **Auto Call Back:** Call Pick-Up will fail if the call ringing on the extension is an Auto Call Back request.
- **Alarms:** Call Pick-Up will fail if the call ringing on the extension is an Alarm Call.

How to configure

For this feature to function, **Call Pick-Up** should be enabled in the **Class of Service** of extension that are to be allowed this feature.

Call Pick-Up Groups

On a sheet of paper, list the extensions that are to be grouped into a Call Pick-Up Group. Make as many Call Pick-Up Groups as required. Assign each group a number.

Call Pick-Up Group Number	SIP Extensions
01	2002, 2003, 2006, 2014
02	
03	2007, 2008, 2009, 2010. 2011, 2012, 2013
:	
99	

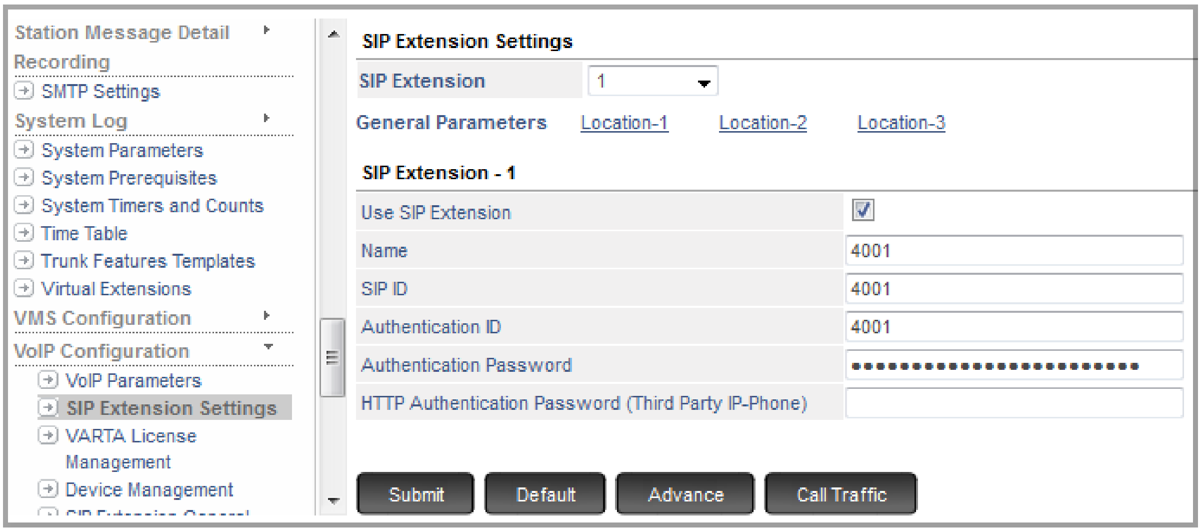
The numbering of Call Pick-Up Groups must start from 01 and end at 99.

Do not assign '00' as Call Pick-Up Group. '00' is the command to de-assign from a Call Pick-Up Group.

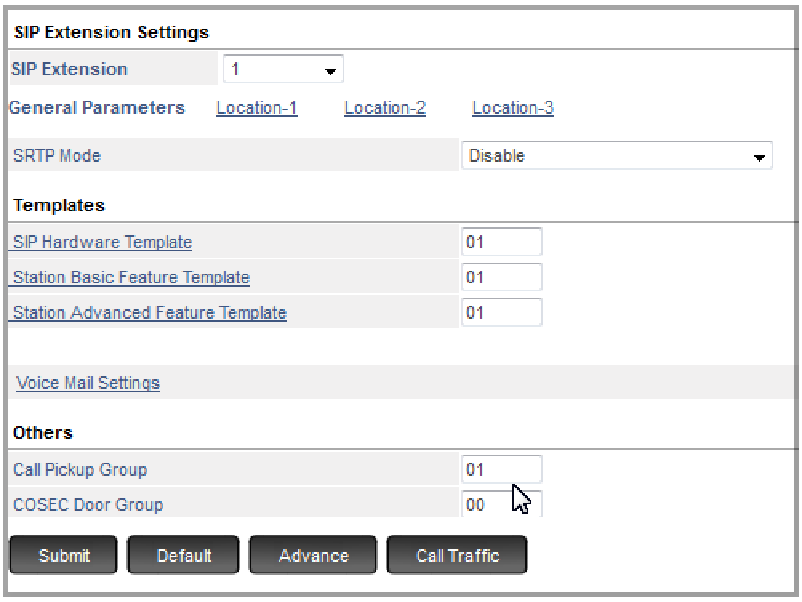
Assigning Extensions to Call Pick-Up Groups

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Extension Settings**.



- Under **Others**, in **Call Pick-up Group**, assign the group number. Refer to the sheet of paper you prepared.



- Click **Submit.**

Call Pick-Up in Class of Service

Decide which extensions are to be assigned to Call Pick-Up groups. Ensure that the feature Call Pick-Up is enabled in the Class of Service (COS) of these extensions.

By default, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS. The Station Basic Feature Template 01 is assigned COS group 01 which has Call Pick-Up feature enabled. Thus, all the extensions of the system can use Call Pick-Up by default.

If you want to deny Call Pick-up feature to all extensions, you can simply disable Call Pick-Up in the default CoS group 01.

However, if Call Pick-Up is to be denied on only selected extensions then:

1. Define a CoS group with Call Pick-Up disabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.
3. Assign this new Template to the extensions to which Call Pick-Up is to be denied.

Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for detailed instructions and configuration.

How to use

For Extended IP Phone Users

To pick up a ringing extension in your Group:

- Press DSS Key assigned to Call Pick-Up Group.
  OR
- Dial **4**.
- Talk.
- Go idle.

To pick up any one of several ringing extensions ringing or the extension that is not in your group:

- Press DSS Key assigned to Call Pick-Up Selective.
  OR
- Dial **12**.
- Dial number of the Extension you want to pick up.
- Talk.
- Go idle.

Call Progress Tones

Call Progress Tones (CPT) are audible tones sent from switching systems such as ITSP or System to calling parties to show the status of phone calls, like dial tone, error tone, ringing error in number dialed, ringing called party, busy line, etc.

Each CPT has a distinctive tone frequency and cadence assigned to it, for which some standards have been established by the International Telecommunication Union (ITU).

On the basis of specific frequency, modulating frequency and cadence, the CPTs generated by ANANT UCS are categorized as:

CPT	Event	Sound	Duration	Timer
Dial Tone 1	Played on lifting the handset.	Toooooooooooo	Played for 7 seconds. After which Error Tone starts	Dial Tone Timer
Dial Tone 2	Played on lifting the handset, when 'Store and Forward Dialing[a]' is done.	Toooooooooooo	Played for 7 seconds. After which Error Tone starts	Dial Tone Timer
Ring Back Tone	Played when the internal number you have dialed is free.	Turroo... Turrroo	Played for 45 seconds	Ring Back Tone Timer
Busy Tone (Engaged Tone)	High pitch beeps with equal ON and OFF periods, played when the dialed extension is busy. Busy tone continues for 7 seconds. This Busy Tone Timer is programmable.	Tooooooo......... Tooooooo	Played for 7 seconds.	Busy Tone Timer
Error Tone (Congestion/ Refusal Tone as per ITU	Fast beeps, played on a wrong operation being performed or a feature invoked without access.	Too…Too…Too …Too	Played for 30 seconds	Error Tone Timer
Internal Call Waiting Tone (Intrusion Tone as per ITU)	Short beep followed by longer OFF duration repeated every second; played to the busy extension when another extension attempts Interrupt Request/ Barge-In	Beep…………. Beep	Played for duration of the Interrupt Request Timer or the Barge-In Timer.	Interrupt Request Timer, Barge-In Timer

CPT	Event	Sound	Duration	Timer
External Call Waiting Tone (Call Waiting Tone as per ITU)	Two ticks followed by a longer OFF time of approx. 3 seconds; played to a busy extension when there is a new incoming Trunk call.	Beep...Beep… ……......Beep... Beep	Played for the duration of the Transfer-On Busy Timer.	Transfer-On Busy Timer.
Confirmation Tone (Acceptance Tone as per ITU)	Continuous, fast beeps, played to confirm successful use of features.	Beep... Beep... Beep	Played for 7 seconds.	Confirmation Tone Timer
Feature Tone	Short beep followed by a longer off duration repeated every second; played when dialing feature access codes	Beep................ Beep	Played until user goes ON-Hook or dials a feature code.	
Programming Tone	Short beep followed by a longer off duration repeated every second; played to prompt entering of fresh commands during programming.	Beep................ Beep	Played until user goes ON-Hook or dials a command.	
Programming Confirmation Tone	Continuous, fast beeps; played to indicate that system has received a valid command and is processing it.	Beep... Beep... Beep	Played for 3 seconds.	Programming Confirmation Timer
Programming Error Tone	Fast beeps, played on a wrong programming command being dialed.	Too…Too…Too …Too	Played for 3 seconds.	Programming Error Tone Timer

a. In Store and Forward dialing, the digits are first stored in a memory location and then these are dialed on the trunk. For example: When Least Cost Routing (LCR) is enabled, the system will store the dialed digits first, check the trunk through which the call is to be routed and then dials the number on the appropriate trunk.

Tone standards vary with the country of application. For example, as per ITU standard, the Dial Tone for India consists of 400Hz modulated by 25Hz, whereas it is 350+440Hz, without modulation, for USA/Canada. Further, many countries use different frequencies and cadences for the same tone. For example, in the US, five different frequency and cadence are used for Dial Tone.

ANANT UCS offers the flexibility of setting the Call Progress Tone Generation (CPTG) type to match the country-specific CPT standards established by ITU.

India being the default 'Region' for ANANT UCS, the CTPG for India is set as default in the system.

How it works

At the time of installation, when you select the **Region** (according to the geographical location of the site where the ANANT UCS is installed), ANANT UCS sets the country-specific CPTG type defined for the selected Region. To see default CPTG types applicable for each region, see “CPTG Region Codes”.

For countries that use different frequencies and cadences for the same tone, e.g. USA, only one frequency/cadence among the group is considered. See “Default CPTG Type” at the end of this topic.

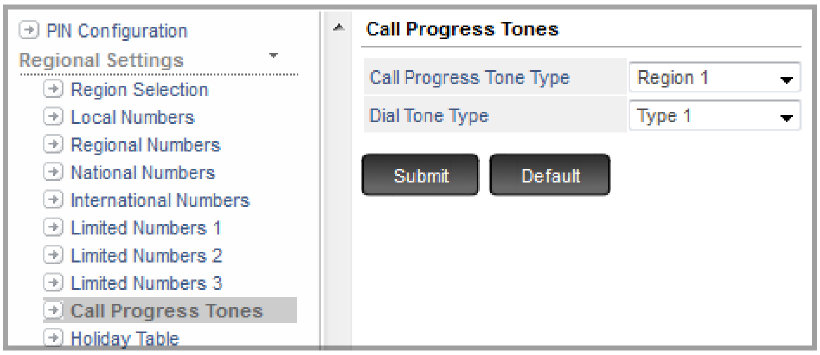
How to configure

Programming of Call Progress Tones involves configuration of three parameters: CPTG Type (Region), CPT related Timers, and Dial Tone Type.

The country-specific CPTG type is set automatically by the system when the 'Region' is selected. However, if required, the System Engineer can change the CTPG type set by the system.

Configuring CPTG Parameters

- Login as System Engineer.
- Under **Configuration**, click **Regional Settings**.
- Click **Call Progress Tones**.

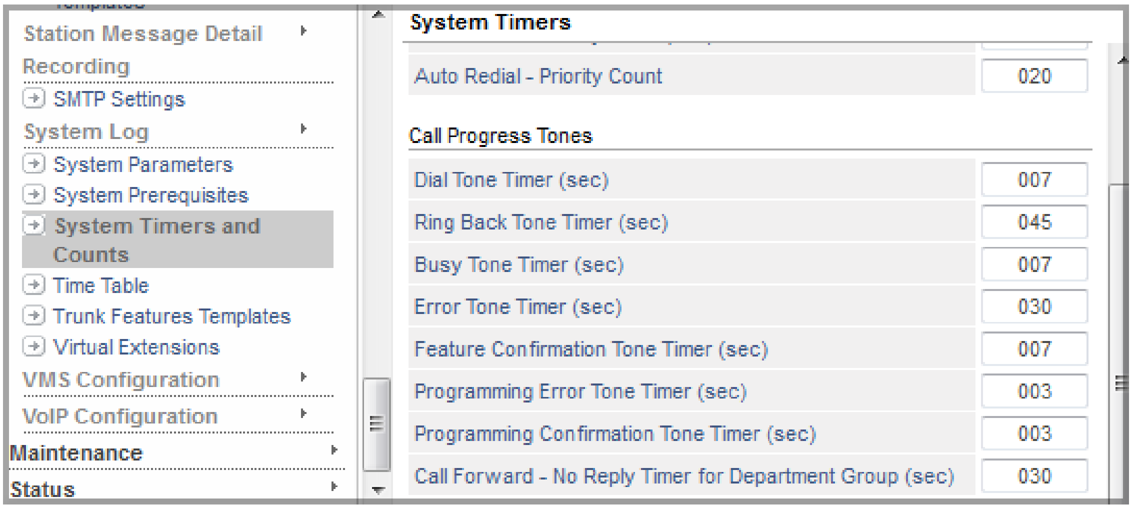


- To set the **Call Progress Tone Type**, select the desired **Region** from the list.
- Select the desired **Dial Tone Type** from Type 1 or Type 2.
- Click **Submit.**

To change CPT-related Timers

- Under **Configuration**, click **System Timers and Counts.**

- Go to **Call Progress Tones**.



- Change the values of the CPT-related Timers as desired.
- Click **Submit**.

How to use

It is important that users of ANANT UCS also get acquainted with the different Call Progress Tones played by the system, so that they understand the meaning of the terms used for various tones. Therefore, ANANT UCS makes it possible for users to listen to the various Call Progress Tones.

CPTG Region Codes

Region Type	Tones	FREQ	CADENCE (ms)					
			ON	OFF	ON	OFF	ON	OFF
Region 1	Ring Back Tone	350+440	400	200	400	2000		
	Busy Tone	440	750	750				
	Error Tone	440	250	250				
	Confirmation Tone	350+440	100	100				
	Feature Tone	350+440	100	900				
	ICWT	440	100	2900				
	CCWT	440	100	100	100	2700		
	Dial Tone 1	440	Continuous					
	Dial Tone 2	350+440	Continuous					

Region 2								
	Ring Back Tone	**400**	**600**	**200**	**200**	**2000**		
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**100**	**100**				
	Feature Tone	**400**	**1500**	**100**				
	ICWT	**400**	**100**	**2900**				
	CCWT	**400**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400**	**Continuous**					
	Dial Tone 2	**400**	**Continuous**					
Region 3								
	Ring Back Tone	**440+480**	**2000**	**4000**				
	Busy Tone	**480+620**	**500**	**500**				
	Error Tone	**440**	**250**	**250**				
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					
Argentina								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**300**	**200**				
	Error Tone	**425**	**300**	**400**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Australia								
	Ring Back Tone	**400*25**	**400**	**200**	**400**	**2000**		
	Busy Tone	**425**	**375**	**375**				
	Error Tone	**425**	**375**	**375**				
	Confirmation Tone	**425*25**	**100**	**100**				
	Feature Tone	**425*25**	**100**	**900**				
	ICWT	**425*25**	**100**	**2900**				
	CCWT	**425*25**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425*25**	**Continuous**					
	Dial Tone 2	**425*25**	**Continuous**					
Brazil								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**250**	**250**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					

Canada								
	Ring Back Tone	**440+480**	**2000**	**4000**				
	Busy Tone	**480+620**	**500**	**500**				
	Error Tone	**480+620**	**250**	**250**				
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					
China								
	Ring Back Tone	**450**	**1000**	**4000**				
	Busy Tone	**450**	**350**	**360**				
	Error Tone	**450**	**700**	**700**				
	Confirmation Tone	**450**	**100**	**100**				
	Feature Tone	**450**	**100**	**900**				
	ICWT	**450**	**100**	**2900**				
	CCWT	**450**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**450**	**Continuous**					
	Dial Tone 2	**450**	**Continuous**					
Egypt								
	Ring Back Tone	**425*50**	**2000**	**1000**				
	Busy Tone	**425*50**	**1000**	**4000**				
	Error Tone	**450**	**500**	**500**				
	Confirmation Tone	**425*50**	**100**	**100**				
	Feature Tone	**425*50**	**100**	**900**				
	ICWT	**425*51**	**100**	**2900**				
	CCWT	**425*50**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425*50**	**Continuous**					
	Dial Tone 2	**425*50**	**Continuous**					
France								
	Ring Back Tone	**440**	**1500**	**3500**				
	Busy Tone	**440**	**500**	**500**				
	Error Tone	**440**	**250**	**250**				
	Confirmation Tone	**440**	**100**	**100**				
	Feature Tone	**440**	**100**	**900**				
	ICWT	**440**	**100**	**2900**				
	CCWT	**440**	**100**	**100**	**2700**			
	Dial Tone 1	**440**	**Continuous**					
	Dial Tone 2	**440**	**Continuous**					
Germany								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**480**	**480**				
	Error Tone	**425**	**240**	**240**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					

Greece								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**300**	**300**				
	Error Tone	**425**	**150**	**150**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**200**	**300**	**700**	**800**		
	Dial Tone 2	**425**	**200**	**300**	**700**	**800**		
India 1								
	Ring Back Tone	**400*25**	**400**	**200**	**400**	**2000**		
	Busy Tone	**400**	**750**	**750**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**1000**	**4000**				
	Feature Tone	**400*25**	**100**	**900**				
	ICWT	**400*25**	**100**	**2900**				
	CCWT	**400*25**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400*25**	**Continuous**					
	Dial Tone 2	**400*25**	**Continuous**					
Indonesia								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Iran								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Iraq								
	Ring Back Tone	**400**	**Continuous**					
	Busy Tone	**400**	**1000**	**1000**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**100**	**100**				
	Feature Tone	**400**	**100**	**900**				
	ICWT	**400**	**100**	**2900**				
	CCWT	**400**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400**	**400**	**200**	**400**	**1500**		
	Dial Tone 2	**400**	**400**	**200**	**400**	**1500**		

Israel								
	Ring Back Tone	**400**	**1000**	**3000**				
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**170**	**140**	**340**	**5000**		
	Feature Tone	**400**	**100**	**900**				
	ICWT	**400**	**100**	**2900**				
	CCWT	**400**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400**	**Continuous**					
	Dial Tone 2	**400**	**Continuous**					
Italy 1								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**425**	**200**	**200**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**200**	**200**	**600**	**1000**		
Japan								
	Ring Back Tone	**400*25**	**1000**	**2000**				
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**100**	**100**				
	Feature Tone	**400**	**100**	**900**				
	ICWT	**400**	**100**	**2900**				
	CCWT	**400**	**100**	**100**	**100**	**2700**		**3450**
	Dial Tone 1	**400**	**Continuous**					
	Dial Tone 2	**400**	**Continuous**					
Kenya								
	Ring Back Tone	**425**	**670**	**3000**	**1500**	**5000**		
	Busy Tone	**425**	**200**	**600**	**200**	**600**		
	Error Tone	**425**	**200**	**600**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Korea								
	Ring Back Tone	**440+480**	**1000**	**2000**				
	Busy Tone	**480+620**	**500**	**500**				
	Error Tone	**480+620**	**300**	**200**				
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					

Malaysia								
	Ring Back Tone	**425**	**400**	**200**	**400**	**2000**		
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**425**	**2500**	**500**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Mexico								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**250**	**250**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
New Zealand								
	Ring Back Tone	**400+450**	**400**	**200**	**400**	**2000**		
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**100**	**100**				
	Feature Tone	**400**	**100**	**900**				
	ICWT	**400**	**100**	**2900**				
	CCWT	**400**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400**	**Continuous**					
	Dial Tone 2	**400**	**Continuous**					
Philippines								
	Ring Back Tone	**425+480**	**1000**	**4000**				
	Busy Tone	**480+620**	**500**	**500**				
	Error Tone	**480+620**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Poland								
	Ring Back Tone	**425**	**1000**	**4000**				
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**425**	**500**	**500**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					

Portugal								
	Ring Back Tone	**425**	**1000**	**5000**				
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**450**	**330**	**1000**				
	Confirmation Tone	**425**	**1000**	**200**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Russia								
	Ring Back Tone	**425**	**800**	**3200**				
	Busy Tone	**425**	**400**	**400**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Saudi Arabia								
	Ring Back Tone	**425**	**1200**	**4600**				
	Busy Tone	**425**	**500**	**500**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Singapore								
	Ring Back Tone	**425*24**	**400**	**200**	**400**	**2000**		
	Busy Tone	**425**	**750**	**750**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**125**	**125**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
South Africa								
	Ring Back Tone	**400*33**	**400**	**200**	**400**	**2000**		
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400*33**	**100**	**100**				
	Feature Tone	**400*33**	**100**	**900**				
	ICWT	**400*33**	**100**	**2900**				
	CCWT	**400*33**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400*33**	**Continuous**					
	Dial Tone 2	**400*33**	**Continuous**					

Spain								
	Ring Back Tone	**425**	**1500**	**3000**				
	Busy Tone	**425**	**200**	**200**				
	Error Tone	**425**	**250**	**250**				
	Confirmation Tone	**425**	**100**	**100**				
	Feature Tone	**425**	**100**	**900**				
	ICWT	**425**	**100**	**2900**				
	CCWT	**425**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**425**	**Continuous**					
	Dial Tone 2	**425**	**Continuous**					
Thailand								
	Ring Back Tone	**400**	**1000**	**4000**				
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**300**	**300**				
	Confirmation Tone	**400*50**	**100**	**100**				
	Feature Tone	**400*50**	**100**	**900**				
	ICWT	**400*50**	**100**	**2900**				
	CCWT	**400*50**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400*50**	**Continuous**					
	Dial Tone 2	**400*50**	**Continuous**					
Turkey								
	Ring Back Tone	**450**	**2000**	**4000**				
	Busy Tone	**450**	**500**	**500**				
	Error Tone	**450**	**200**	**200**	**600**	**200**		
	Confirmation Tone	**450**	**40**	**40**				
	Feature Tone	**450**	**100**	**900**				
	ICWT	**450**	**100**	**2900**				
	CCWT	**450**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**450**	**Continuous**					
	Dial Tone 2	**450**	**Continuous**					
UAE								
	Ring Back Tone	**400+450**	**400**	**200**	**400**	**2000**		
	Busy Tone	**400**	**375**	**375**				
	Error Tone	**400**	**400**	**350**	**225**	**525**		
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					
UK								
	Ring Back Tone	**400+450**	**400**	**200**	**400**	**2000**		
	Busy Tone	**400**	**375**	**375**				
	Error Tone	**400**	**400**	**350**	**225**	**525**		
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					

USA								
	Ring Back Tone	**440+480**	**2000**	**4000**				
	Busy Tone	**480+620**	**500**	**500**				
	Error Tone	**480+620**	**250**	**250**				
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					
Italy 2								
	Ring Back Tone	**400**	**1000**	**2000**				
	Busy Tone	**400**	**500**	**500**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**400**	**100**	**100**				
	Feature Tone	**400**	**1750**	**100**				
	ICWT	**400**	**100**	**2900**				
	CCWT	**400**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**400**	**Continuous**					
	Dial Tone 2	**400**	**Continuous**					
Belgium								
	Ring Back Tone	**350+440**	**1000**	**3000**				
	Busy Tone	**440**	**750**	**750**				
	Error Tone	**440**	**250**	**250**				
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**440**	**Continuous**					
India 2								
	Ring Back Tone	**350+440**	**400**	**200**	**400**	**2000**		
	Busy Tone	**400**	**750**	**750**				
	Error Tone	**400**	**250**	**250**				
	Confirmation Tone	**350+440**	**100**	**100**				
	Feature Tone	**350+440**	**100**	**900**				
	ICWT	**350+440**	**100**	**2900**				
	CCWT	**350+440**	**100**	**100**	**100**	**2700**		
	Dial Tone 1	**350+440**	**Continuous**					
	Dial Tone 2	**350+440**	**Continuous**					

Stuttered Dial Tone

Frequency: Same as Dial Tone 1 (Region wise)

Cadence: 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off, 400 ms On - 100 ms Off (same for all Regions)

The meaning of frequency notation is as follows:

- ***f1*f2:*** *f1 is modulated by f2.*
- ***f1+f2:*** *The juxtaposition of two frequencies f1 and f2 without modulation.*

Default CPTG Type

Region Code	Meaning	CPTG Region Code
001	Afghanistan	
002	Algeria	
003	Antigua and Barbuda	
004	Argentina	04
005	Australia	05
006	Austria	
007	Bahamas	
008	Bahrain	
009	Bangladesh	
010	Belarus	
011	Belgium	
012	Bhutan	
013	Bolivia	
014	Bosnia	
015	Botswana	
016	Brunei	
017	Brazil	06
018	Bulgaria	
019	Cambodia	
020	Cameroon	
021	Canada	03
022	Chile	
023	China	08
024	Colombia	
025	Costa Rica	
026	Croatia	
027	Cuba	
028	Cyprus	
029	Czech Republic	
030	Denmark	
031	Egypt	09
032	Fiji	
033	Finland	
034	France	10
035	Germany	11
036	Greece	12
037	Guyana	
038	Holland	
039	Hong kong	
040	Hungary	
041	India	01
042	Indonesia	14
043	Iran	15
044	Iraq	16
045	Ireland	
046	Israel	17
047	Italy	18
048	Japan	19
049	Jordan	
050	Kazakhstan	
051	Kenya	20
052	Korea-North	21
053	Korea-South	21
054	Kuwait	
055	Kyrgyzstan	
056	Lebanon	
057	Libya	
058	Malaysia	22
059	Maldives	
060	Mauritius	
061	Mexico	03
062	Mongolia	
063	Mozambique	
064	Myanmar	
065	Namibia	03
066	Nepal	
067	Netherlands	
068	New Zealand	24
069	Nigeria	
070	Norway	
071	Oman	
072	Pakistan	
073	Paraguay	
074	Peru	
075	Philippines	25
076	Poland	26
077	Portugal	27
078	Qatar	
079	Romania	
080	Russia	28
081	Singapore	30
082	Slovakia	
083	South Africa	31
084	Spain	32
085	Sri Lanka	
086	Sudan	
087	Sweden	
088	Switzerland	
089	Syria	
090	Taiwan	
091	Tajikistan	
092	Thailand	33
093	Turkey	34
094	Uganda	
095	Ukraine	
096	United Arab Emirates	35
097	United Kingdom	02
098	United States	03
099	Uzbekistan	
100	Venezuela	
101	Vietnam	
102	Yemen	
103	Yugoslavia	
104	Zambia	
105	Zimbabwe	

Call Restriction based on IP Address

When ANANT UCS is connected to a public IP network, it is may be necessary to allow traffic from particular IP address only.

With the feature 'Call Restriction based on IP Address', ANANT UCS makes it possible to entertain requests on its LAN/WAN Ports from predefined IP Addresses only.

How it works

For this feature to work,

- the *Trusted IP Address/es* table must be configured for each SIP Trunk.
- with this table configured, incoming call traffic from all IP Addresses, other than those configured in the Trusted IP Address/es table, will be blocked.

All incoming traffic on the SIP Trunk will be rejected if the Trusted IP Address/es Table of that SIP Trunk is blank.

How to configure

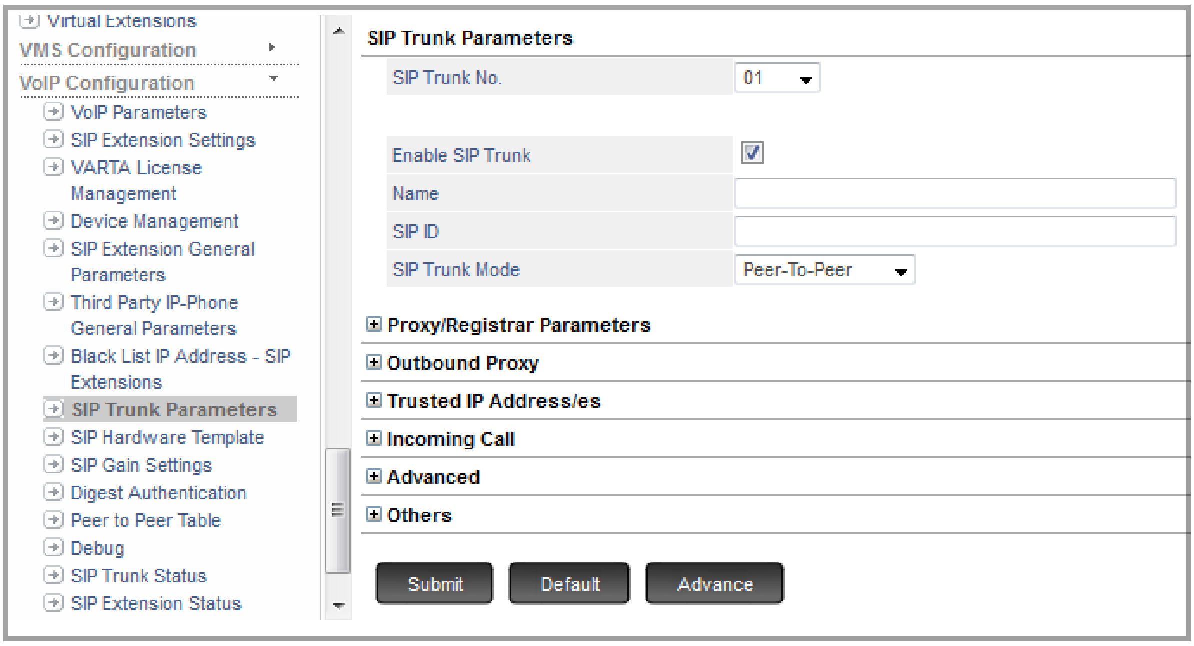
For each SIP Trunk, make a list of IP Addresses:Port from which you want to allow traffic. If you want to allow incoming calls from all ports for a particular IP Address, configure only the IP Address.

You are allowed to configure a maximum of 10 IP Addresses.

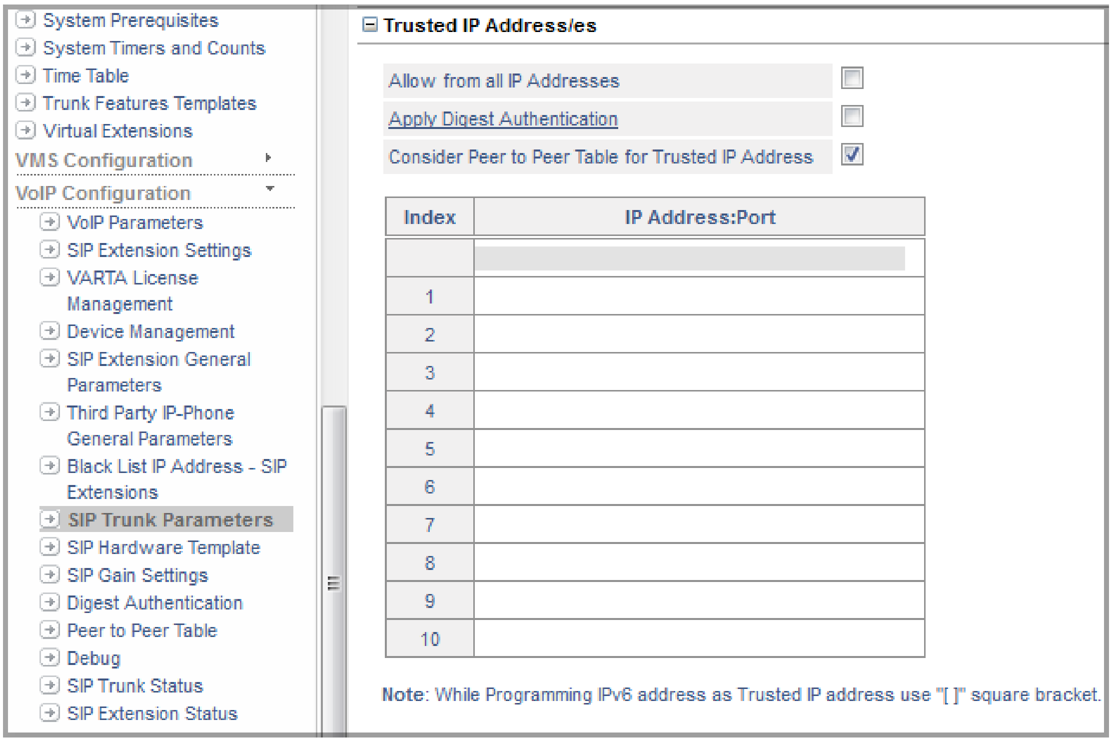
Configuring IP Address Based Call Traffic Restriction

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.

- Click **SIP Trunk Parameters**.



- Click **Trusted IP Address/es** to expand.



The first entry in the table will display the *Proxy/Registrar Server Address:Port* or *Outbound Proxy Address: Port* as configured for the “Configuring SIP Trunks”. For the Index numbers 1 to 10,

- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls.

  Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit.**

Call Taping

Call Taping allows extension users to record the telephone conversations they have with other extensions or external numbers, without the opposite party coming to know about it.

Feature is useful for keeping records of important conversations. This feature will work only if a VMS channel is available.

Call Taping can be done for:

- Incoming and outgoing external calls
- Incoming and outgoing internal calls

ANANT UCS supports tapping of 20 calls simultaneously in a 3-Party Conference. Calls can be taped either in the extension user's personal mailbox or a common mailbox assigned to this feature. The taped calls are stored along with the call details, that is, the time and date of the call, the calling number and the called number. If calls are taped in a common mailbox, only the extension users having access to the common mailbox can retrieve and listen to the recorded conversations.

To be able to record external incoming and outgoing calls,

- a list of phone numbers (both incoming and outgoing) must have been programmed in **Number List-Incoming Calls** and **Number List- Outgoing Calls** respectively, in the Station Advanced Feature Template.

- **Call Taping** must be enabled in the Trunk Feature Template assigned to the trunk used for making/receiving external calls.

Incoming calls without Calling Line Identification (CLI) can also be taped. For this, the check box **Tape calls Incoming without CLI** must be enabled in the Station Advanced Feature Template assigned to the extensions.

To be able to record internal calls, the **Call Taping for Internal Calls** check box must be enabled in the Station Advanced Feature Template assigned to the extension.

- *Use this feature in accordance with the local privacy laws.*
- *Matrix Comsec is not responsible for any misuse/abuse of this feature by users.*

How it works

A and B are extensions. Both have Call Taping parameters configured in the Station Advanced Feature Template assigned to them. Also, the trunks assigned to both the extensions for making outgoing calls have Call Taping enabled.

C and D are external parties. E is the mailbox extension where the tapped calls are recorded.

A calls C

- The system matches the dialed number with the numbers in the **Number List - Outgoing Calls**. The system finds a match.

- When speech is established, the system starts recording the conversation between A and C automatically in E's mailbox.

- Call Taping Beeps will be played to A and C, if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

D calls B

- The system matches the incoming number with the numbers in the **Number List-Incoming Calls**.
- On finding a match, when speech is established, system records the speech between D and B in E's mailbox.
- Call Taping Beeps will be played to D and B only if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.

- *If an incoming call does not have any CLI, the system checks the **Tape calls coming without CLI** check box under the Call Taping in Station Advanced Feature Template assigned to the extension.*
- *If the check box is enabled, all calls without CLI are taped.*

A calls B

- The system checks if **Call Taping for Internal Calls** is enabled under the Call Taping in Station Advanced Feature Template assigned to A.
- If the check box is enabled, the system records the speech between A and B in E's mailbox.
- Call Taping Beeps will be played to A and B only if **Play Beep when Raid/Call Taping/Conversation Recording starts** is enabled in the System Parameters.
- If the check box is disabled, the speech between A and B will not be recorded.
- The same is done when B calls A. The speech will be recorded in E's mailbox.

To listen to the conversation, A and B must have access to the mailbox of E.

During Call Taping, if the user puts the call on Consultation Hold and access the feature — Call Transfer/ Conference/ Call Park/ Call Toggle, then in this case, the conversation after the call is put on consultation hold will not be taped. However, the system will start tapping this call again once it is transferred to the respective user.

For example, consider F and G are in a two-party speech, and during the ongoing conversation, F wants to speak to H. So, when G puts the call of F on consultation hold and speaks with H, the conversation between G and H will not be taped. Call taping will be resumed again, after the call is transfered to H, that is, the conversation between F and H will be taped.

You can save the taped conversation either in Personal Mailbox or in a Common Mailbox. If you select Personal Mailbox, the taped conversations will be saved in each extension user’s Personal Mailbox.

In you select Common Mailbox, you can save the taped conversation either in a single file or in individual files as per your requirement.

If the call is not transferred successfully and returns back to the transferor, then in this case, the system will save the taped conversation in two separate files.

Feature Interaction:

- **Conversation Recording:** If Call Taping and "Conversation Recording" both are enabled for an extension, then priority is given to Call Taping.

How to configure

The functioning of this feature requires the following parameters to be programmed:

- **Save Call Taping Files in**: Specify the location to save Call Taping files. You can select **Common Mailbox** or **Personal Mailbox**.

- **Common Mailbox for Call Taping (Enter Extension Number):** You must program the extension number of the user in whose mailbox the calls are to be taped.

- **Save Call Tapping Files as:** If you select common mailbox, select the type of file you want the system to generate for saving the tapped conversation. You may select — *Individual File before and after call transfe*r or *Single File before and after call transfer* as per your requirement.

- **Call Taping on Trunks:** Call Taping must be enabled in the Trunk Feature Template assigned to the Trunks. Make sure these trunks are used by extension users for making/receiving external calls.

- **Tape calls coming without CLI check box:** This check box must be enabled if you want calls without CLI to be taped.

- **Number Lists for Incoming and Outgoing Calls:** The Call Taping Number List-Incoming Calls and Call Taping Number List-Outgoing Calls are to be programmed in the Station Advanced Feature Template assigned to the extension users so that the system can match the phone numbers of the incoming and outgoing calls and initiate the recording of the speech.

  On a sheet of paper, prepare the Call Taping List Incoming and Call Taping List Outgoing.

  You can add as many as 999 numbers to each list. Each entry on these Lists is stored in a serial order against a 'Location Number'. So, draw three columns and enter the numbers against a location number from 001 to 999.

Location	Number List-Incoming Calls	Number List-Outgoing Calls
001		
002		
:		
999		

  Use this table to program the Number lists. By default Number List 09 is assigned for numbers of incoming calls, and Number List 10 is assigned to numbers of outgoing calls.

- **Call Taping for Internal Calls check box:** This check box is to be enabled in the Station Advanced Feature Template applied on those extensions that are to be allowed Call Taping of internal calls, that is, calls made or received by them to or from other extensions.

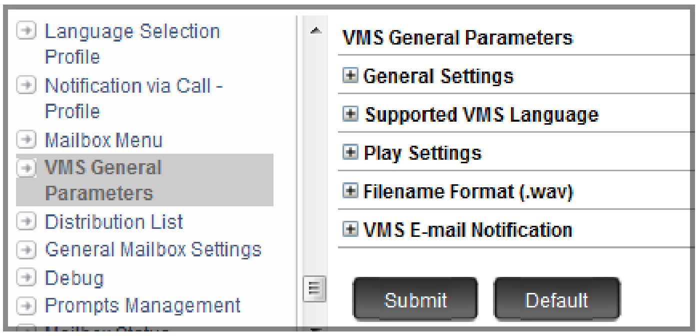
- **Play Beep when Raid/Call Taping/Conversation Recording starts:** This check box is to be enabled if Call Taping Beeps are to be played to the two parties in speech. Enable Call Taping Beeps only when you want indication of speech recording to the two parties in speech. By default, this check box is enabled.

Configuring Call Taping

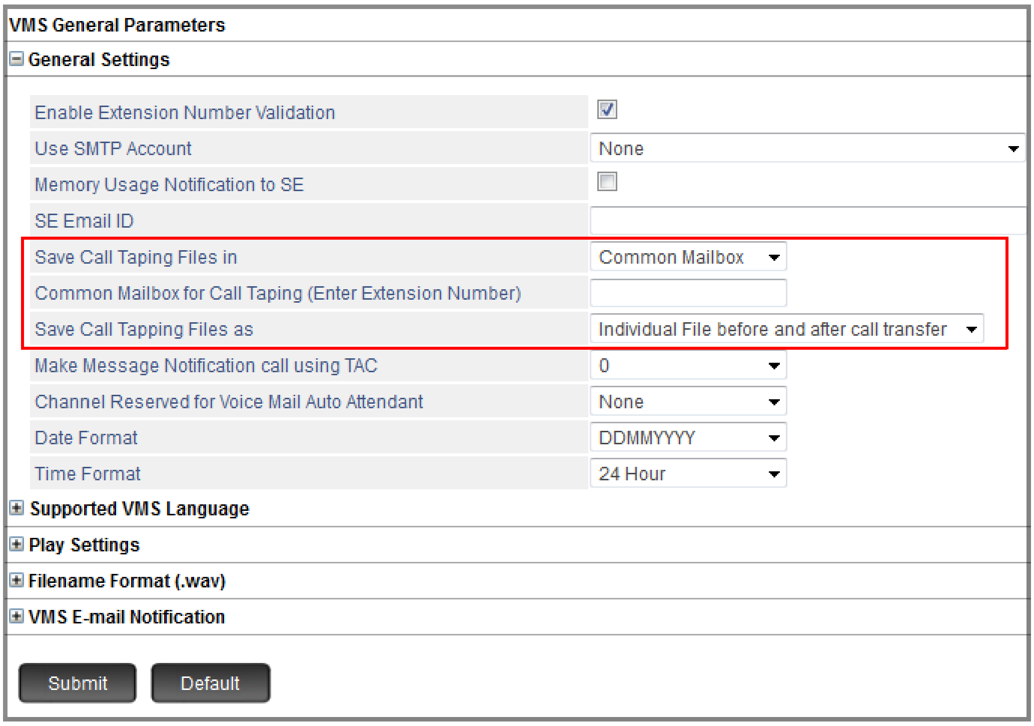
- Login as System Engineer.
- Click **Configuration**.

For Call Taping Mailbox,

- Under **VMS Configuration,** click **VMS General Parameters**.



- Click **General Settings** to expand.



- Specify the location where you want to **Save Call Taping Files in**. You can save the Call Taping files either in the Common Mailbox or in your Personal Mailbox. By default, Call Taping files are saved in the Common Mailbox.
- If you choose to save Call Taping files in the Common Mailbox,

- enter the access code of the SIP Extension, Department Group or General Mailbox, whose mailbox you want to assign for Call Taping in **Common Mailbox for Call Taping (Enter Extension Number)**.

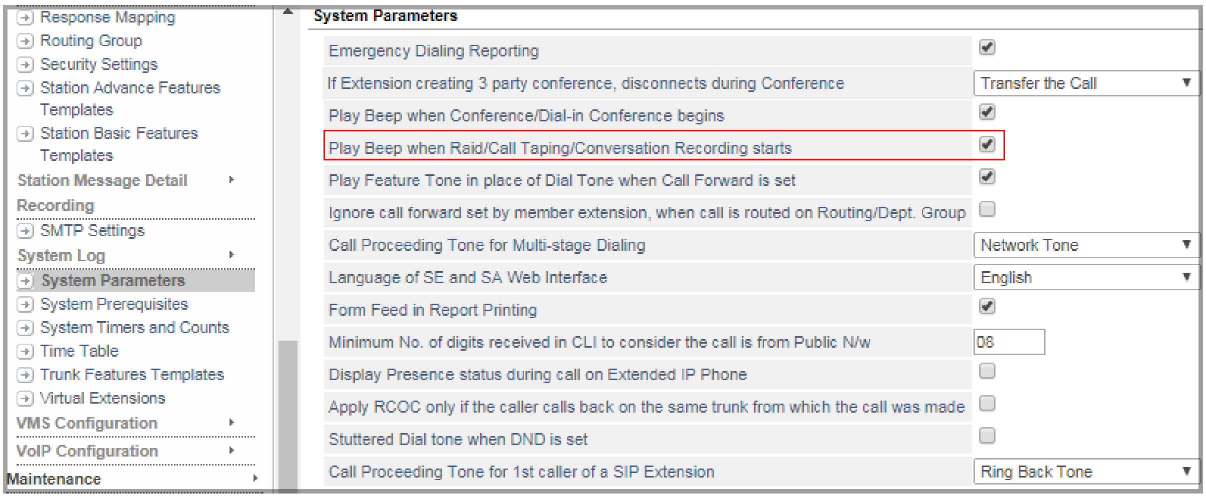
- select the type of file you want the system to generate for saving the tapped conversation in **Save Call Tapping Files as**. You may select — *Individual File before and after call transfe*r or *Single File before and after call transfer* as per your requirement.

  If you select *Individual File before and after call transfe*r, the system will generate two separate files for saving the tapped conversation, that is, one file containing the conversation tapped before the call is transferred and another file containing the conversation tapped after the call is transferred. However, if you select *Single File before and after call transfer,* the system will generate one single file for saving the conversation tapped before and after the call is transferred.

- Click **Submit.**

To enable/disable Call Taping Beeps,

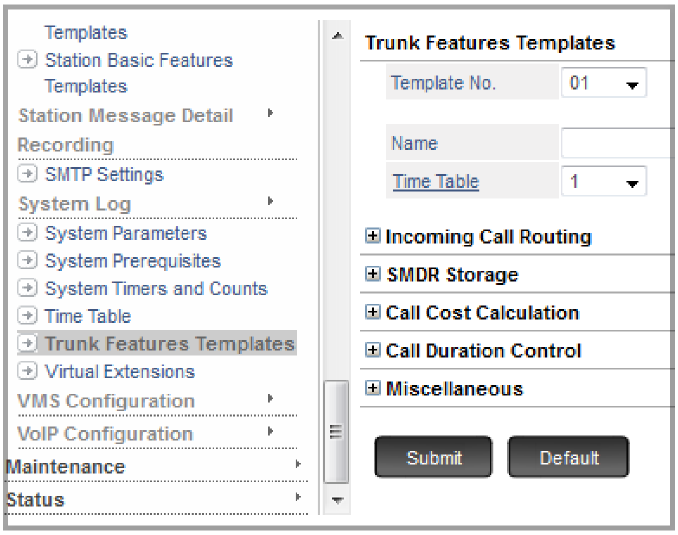
- Under **Configuration**, click **System Parameters**.
- Click **System Parameters** to expand.
- Scroll to **Play Beep when Raid/Call Taping/Conversation Recording starts**, enable/disable beeps by selecting/clearing the check box.



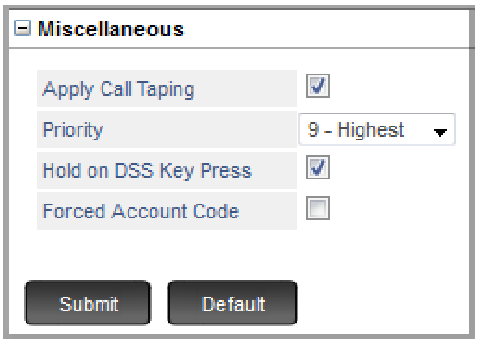
- Click **Submit.**

To enable **Call Taping on Trunks**,

- Under **Configuration**, click **Trunk Feature Template**.



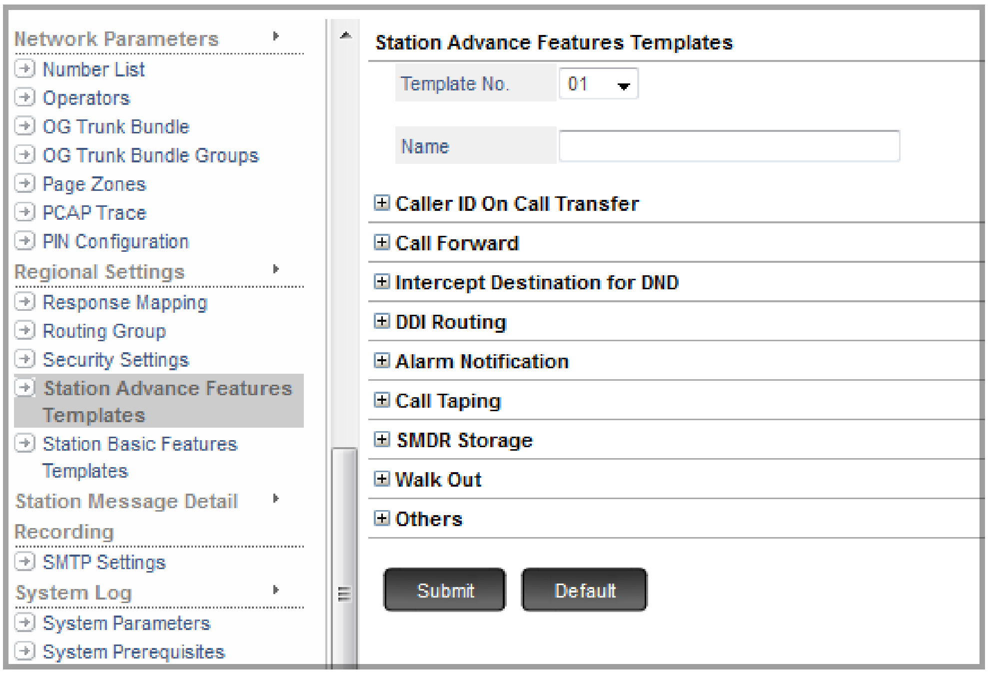
- Click **Miscellaneous**.



- Select the **Apply Call Taping** check box to enable. For detailed instructions, see “Configuring Trunks”.

To configure the Call Taping parameters and Number Lists,

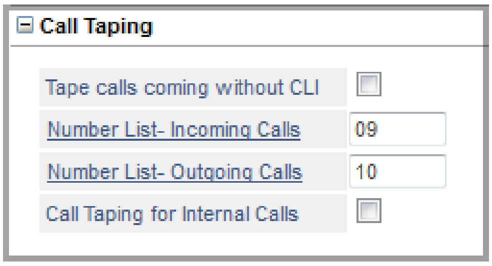
- Under **Configuration,** click **Station Advanced Feature Template**.



By default, Station Advanced Feature Template 01 is assigned to all extensions of the ANANT UCS. If you want to assign Call Taping facility to all extensions, then program the Call Taping related parameters and Number Lists, in Template 01.

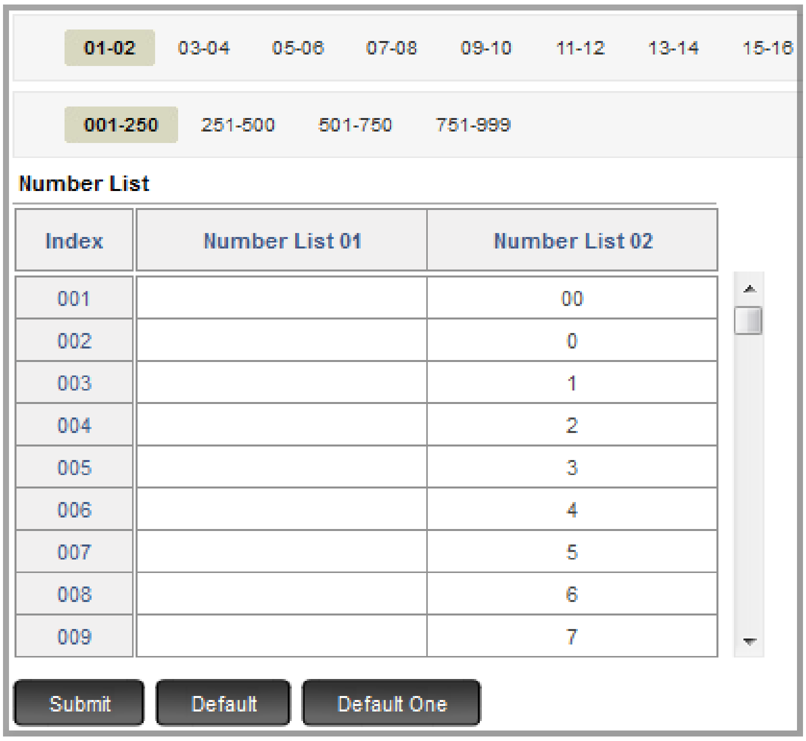
However, if only selected extensions are to be assigned this feature, then:

- Prepare a separate Station Advanced Feature Template.
- Set the Call Taping Parameters in this template.
- Apply this new template to desired extensions that are to be allowed this feature.
- Click the **Call Taping** to expand.



- If you want calls without CLI to be taped, select the **Tape calls coming without CLI** check box.
- To program the list of numbers of incoming calls, click **Number List- Incoming Calls**.

The Number List page will open.



- Click the default number list **9-10** link assigned to Call Taping, then click the **001-250** link of the default.

  The Number Lists 9 and 10 will open.

- *If the same incoming and outgoing numbers are to be programmed for all extensions, you may simply program the default Number lists 09 and 10.*
- *If different incoming and outgoing numbers are to be programmed for different extensions, then prepare different number lists.*

- Enter the List of Incoming Numbers that the system should match in List No. 09.
- Enter the List of Outgoing Numbers that the system should match in List No. 10.

  You can program as many as 999 numbers in each list. Each entry on these Lists is stored in a serial order against a 'Location Index, starting from 001-999'. There are 250 Location Index on each page on your screen. To go to the next set of Location Index, for instance, 251-500, click the link under 09-10.

- Click **Submit.**
- Follow the same steps to program a different Call Taping number list. But ensure that the different List number you programmed is entered in the Station Advanced Feature Template applied to the extensions.
- If you want calls between extensions to be taped, click the **Call Taping for Internal Calls** check box.
- Click **Submit.**
- Now, apply the programmed template to the desired extensions to which you want to provide the Call Taping facility. Refer the topic “Station Advanced Feature Template” for programming instructions.

How to use

This feature works automatically on the extensions which have the Call Taping parameters configured.

Call Taping conversations can be recorded either in a common mailbox or in the extension user's personal mailbox.

- If calls are taped in a common mailbox, only the extension users having access to the common mailbox can retrieve and listen to the recorded conversations.

- If calls are taped in a personal mailbox, the extension user can listen to the recorded conversation by accessing their personal mailbox.

Accessing Personal Mailbox

If you are an Extended IP Phone user

- Press 'Voice Mail' Key.
  OR
- Dial **3931-Your Mailbox Password**[86]

- Follow Voice Mail Prompts to listen to new messages.

86. *Only if the mailbox is password protected, you will be prompted to enter the password.*

Call Toggle

Call Toggle allows you to have two simultaneous telephone conversations, talking to two persons alternately.

Call Toggle is also referred to as Hold-Consult or Call Splitting,

You can toggle between:

- Two internal calls (that is, two extensions).
- An internal Call and an External Call (extension and trunk).
- Two external calls (two trunks).

How it works

- A, B, and C are extensions.
- D and E are trunks.

Toggling between two internal calls

- A is in speech with B and C is on Consultation Hold.
- To talk with C, A dials Flash-1. Speech with C.
- To talk with B, A dials Flash-1. Speech with B.
- A can toggle back to C by dialing Flash-1.

Toggling between internal call and external call

- A is in speech with B and D (external call) is on Consultation Hold.
- To talk with D, A dials Flash-1. Speech with D.
- To talk with B, A dials Flash-1. Speech with B.
- A can toggle back to D by dialing Flash-1.

Toggling between two external calls

- A is in speech with D and E is on Consultation Hold.
- To talk with E, A dials Flash-1. Speech with E.
- To talk with D, A dials Flash-1. Speech with D.
- A can toggle back to E by dialing Flash-1.

- *The party put on Consultation Hold during Call Toggle cannot hear the conversation between the other two parties.*
- *You can also toggle between an incoming internal/external call (indicated by call waiting tone) and an internal/external call you are currently in speech with.*
- *You can also answer an incoming 'Interrupt Request' call and toggle between the interrupting extension and the extension you were in speech with.*
- *You can convert a Call Toggle into a three-party conference by dialing Flash-*3.*
- *You can transfer the call you are currently in speech with to another extension.*
- *You can park the call you are currently in speech with.*

How to configure

Call Toggle is a Class of Service (CoS) dependent feature.

In the default Station Basic Feature Template 01 assigned to all extensions of ANANT UCS, Call Toggle is included in the 'Basic Features' assigned to all CoS groups, including the default CoS group 01. So, all extensions of ANANT UCS can use this feature.

As Call Toggle is a part of the set of 'Basic Features', you cannot disable this feature selectively in the CoS of extensions, without disabling the entire set of features.

No specific configuration is required for this feature, except for programming a DSS key for Call Toggle, if required. Refer the topic "DSS Keys Programming" for instructions.

How to use

For Extended IP Phone Users

Call Toggle between two internal calls:

- Speech with Extension 1.
- Extension 2 on Consultation Hold.
- To talk with Extension 2, press DSS key assigned to Call Toggle.
- Speech with extension 2.
- Press DSS key assigned to Call Toggle again.
- Speech with Extension 1.

Call Toggle between an Internal Call and an External Call:

- Speech with extension.
- External party on Consultation Hold.
- To talk with the external party, press DSS key assigned to Call Toggle.
- Speech with external party.
- Press DSS key assigned to Call Toggle again.
- Speech with Extension.

Call Toggle between two External Calls:

- Speech with external party 1.
- External party 2 on Consultation Hold.
- Press DSS Key assigned to Call Toggle.
- Speech with External party 2.
- Press DSS key assigned to Call Toggle again.
- Speech with External party 1.

Call Traffic

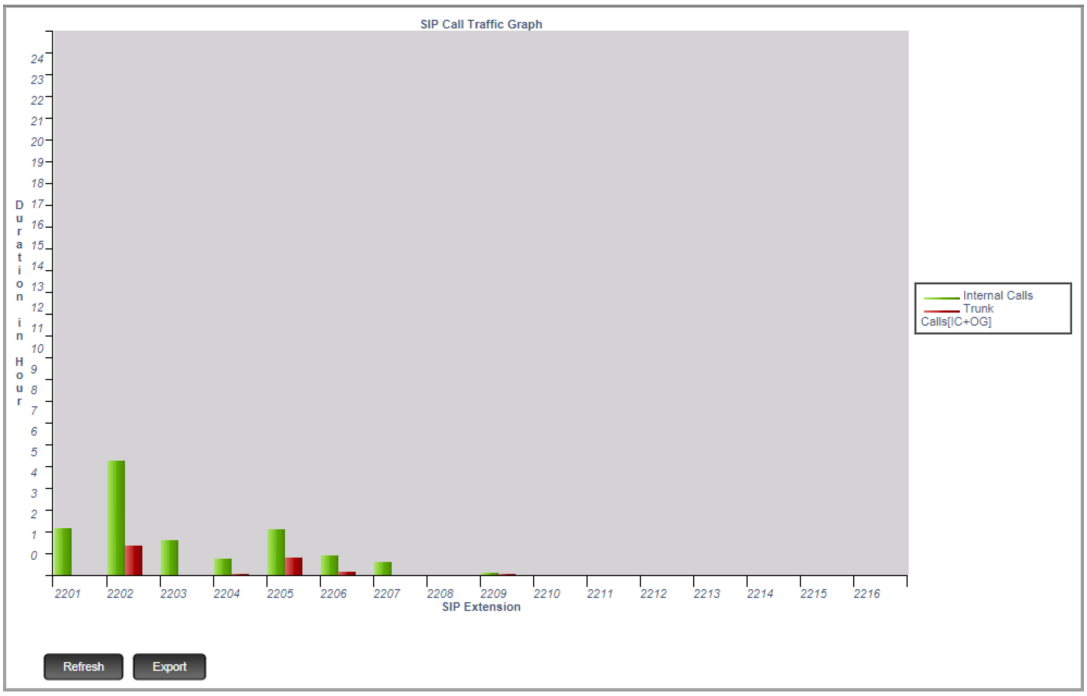
Call traffic measurement feature of ANANT UCS gives a graphical representation of the time duration for which extensions and trunk ports remained Off-hook. The data is represented in a bar graph format.

In the graph, the duration is shown in Hours along the Y-axis while the extension(s) names are shown along the X-axis. The following call traffic measurement data is displayed:

- Time for which each extension remained in speech for making/receiving Internal calls.
- Time for which each extension remained in speech for making/receiving Trunk calls (Incoming + Outgoing).
- Time for which each trunk port remained in speech for Incoming calls.
- Time for which each trunk port remained in speech for Outgoing calls.

This time is measured in terms of the number of hours, and the traffic is measured for last 24 hours.

You can view this traffic information in graphical format on the Jeeves; see the illustration below for Call Traffic information generated for SIP extension.



The two-color bars distinguish Internal calls (green) and Trunk calls (red)

You can also export this information in a database readable format like Microsoft Excel. The call traffic information files can be saved on a local disk.

How to use

To view Call Traffic in graphical format, you need to log into the Jeeves as System Engineer.

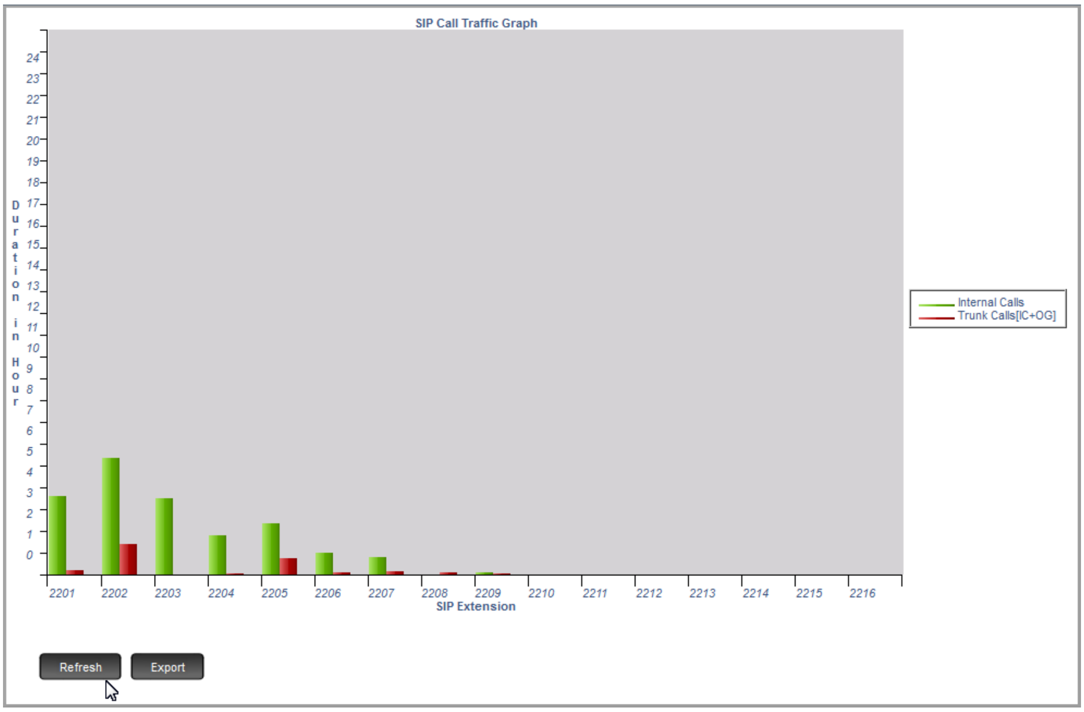
Go to the configuration page of the desired trunk or extension port for which you want to view Call Traffic data.

On the page, click the **Call Traffic** button.

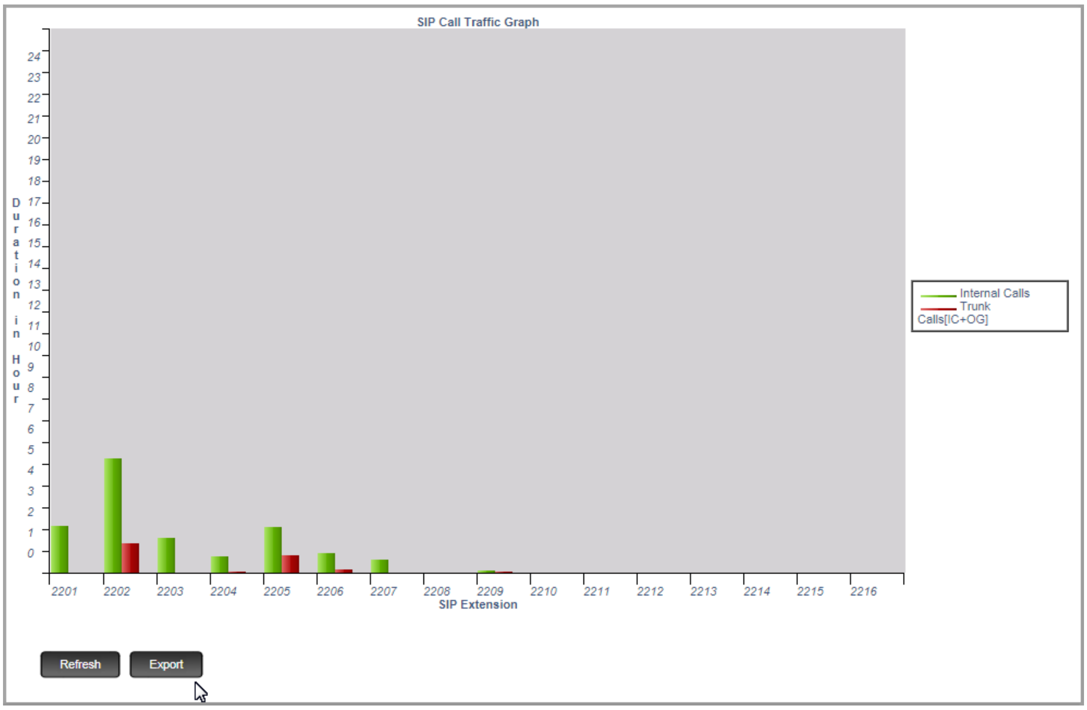
The call traffic will be generated as a graph.

If there is no call traffic usage data is available for the extension/ trunk you are currently viewing, the page will not show any bars.

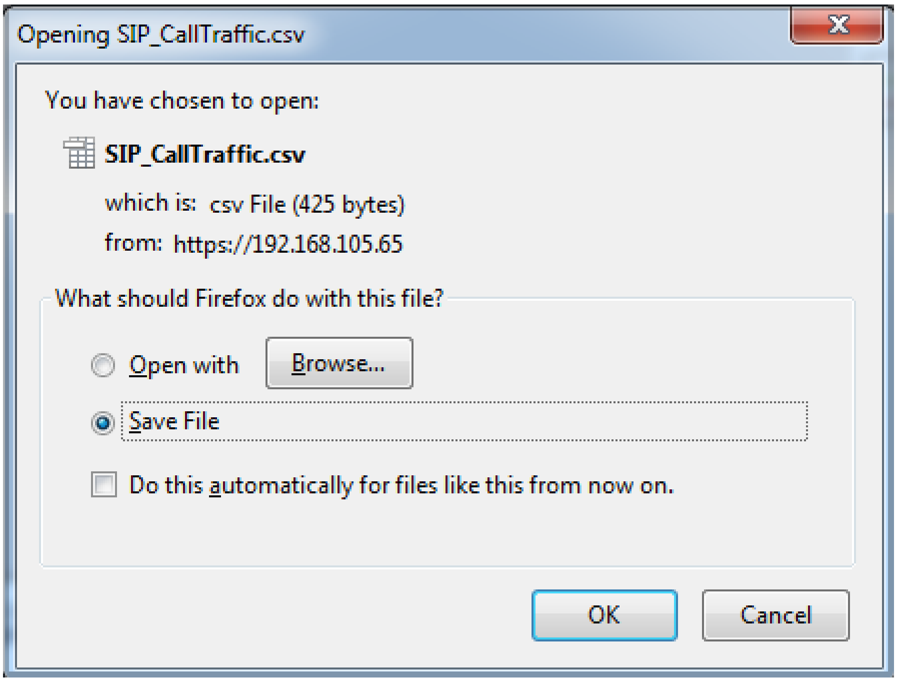
Click the **Refresh** button to get the latest 24hrs call traffic statistics.The graph displays the call data (for last 24hrs) as per the SMDR filters set by you.



- Click the **Export** button to export the data you are currently viewing.

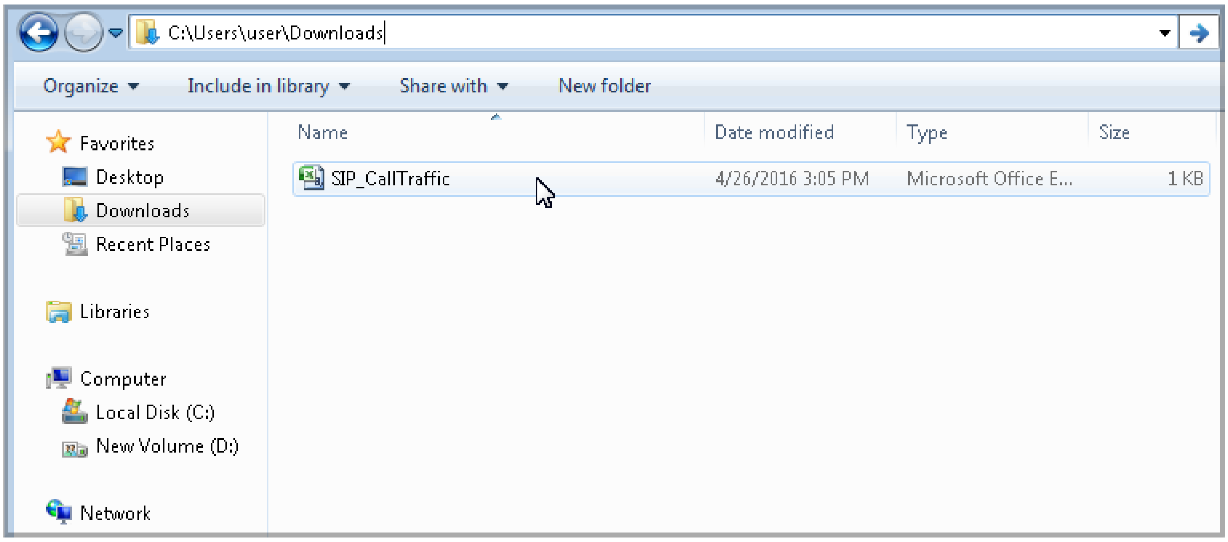


- The **Opening SIP_CallTraffic.csv** window will open.

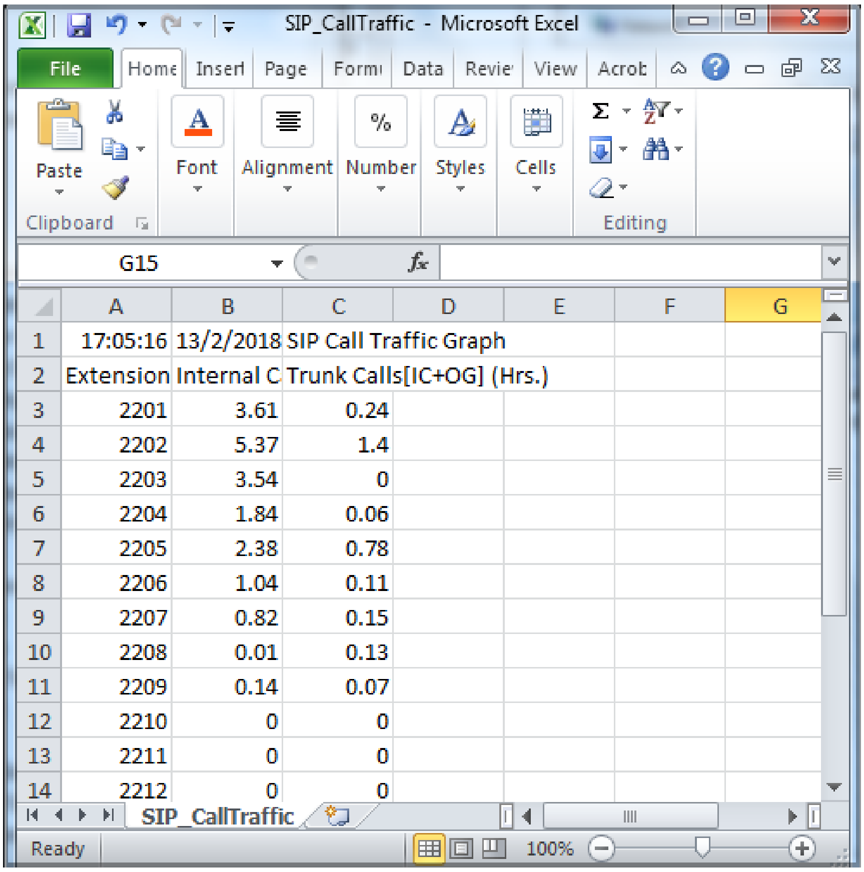


You can choose whether to open this file or save it in.xls format as per your requirement.

- Save the file on the local disk.



- If you open this file in MS Excel then it may look as shown in the following image.



The file contains the date and time when the call traffic data is calculated and the names of the extensions/ trunks for which it is calculated.

- You can save this file on your local disk from Excel window also.

Call Transfer

Call Transfer enables you to relocate an existing call from an extension or trunk to another extension or to an external number. Calls can be transferred after notifying the other extension/external number about the impending transfer or can be transferred directly without notification.

The types of Call Transfer ANANT UCS offers are:

- **Call Transfer - Screened:** The Operator puts the caller on Consultation Hold, dials the desired party's extension, and informs the desired party of the impending transfer. If the desired party chooses to accept the call, the call is transferred to him.

- **Call Transfer - While Ringing:** The Operator puts the caller on Consultation Hold, dials the desired party's number and transfers the call when the desired party's extension starts ringing.

  This feature is used when there are several other calls to be attended and the Operator cannot wait for the desired party to answer.

- **Call Transfer - On Busy:** The Operator puts the caller on Consultation Hold, dials the desired party's number and transfers the call even when the desired party is busy in speech with another person. The busy extension gets intrusion tone and can choose to answer the intruding (transferred) call.

- **Call Transfer - Trunk-to-Trunk:** An external call is transferred on to another trunk line. The Operator puts the external caller on Consultation Hold, dials the desired party's external number, and transfers the call after or without notifying the desired party of the impending transfer.

  Trunk-to-Trunk call transfer may be used to transfer incoming calls for out-of-office extension users to their cell phones, or to connect personnel at remote or distant locations. For instance: an out-of-office executive who does not have long distance dialing permission can call the office and request the operator to connect him to the desired party on a trunk line.

- **Blind Transfer to VMS:** The Operator puts the caller on Consultation Hold, dials the feature access code for Blind Transfer to VMS, dials the desired party's number, and transfers the call. The call is transferred to the mailbox assigned to the desired party. The caller may leave a message in the mailbox.

- *Call Transfer is not exclusively an Operator feature, though it is used mostly by Operators. Calls can be transferred by any extension to another extension or external number, if "Basic Features" (this will allow you to access all types of Call Transfer except Trunk-to-Trunk Transfer) are allowed in Class of Service of the transferring extension.*

- *To access Trunk-to-Trunk Transfer make sure Trunk-to-Trunk Transfer feature is enabled in the Class of Service of the transferring extension.*

- *ANANT UCS enables SIP extensions to resume a transferred call before it has been answered by the transfer target (which may be an extension or an external number). For a list of IP Phones on which this feature has been tested, see "ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.*

- *ANANT UCS allows Semi-attended Transfer and Transfer on Conference Hangup on SIP Trunks. For a list of IP Phones on which this feature has been tested, see "ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.*

How it works

A and B are extension users.
C is an external caller.
D is an external number.

Call Transfer from Trunk/Extension to Extension

Three scenarios are possible:

1. Screened Transfer:
   - C calls a Trunk of ANANT UCS.
   - The Operator answers the call.
   - Operator puts C on Consultation Hold.
   - Operator dials B's extension.
   - When B answers the call, the Operator informs B about the call.
   - If B accepts the call, the Operator transfers the call to B.
   - Now, B and C are in speech.

   - *If B does not accept the call, Operator may dial Flash to retrieve the call and speak to C.*
   - *The Operator can also abort call transfer while B's phone is ringing by dialing Flash. The Operator gets connected to C.*

2. Transfer While Ringing:
   - C calls a Trunk of ANANT UCS.
   - The Operator answers the call.
   - The Operator puts C on Consultation Hold and dials B's extension.
   - The Operator transfers the call when B's phone starts ringing.
   - If B answers the call, B gets connected to C.
   - If B does not answer the call at the end of the Transfer-While Ringing Timer (configurable; default: 30 seconds), the call is returned to the Operator. C gets Ring Back Tone.
   - The Operator answers the call and is in speech with C.
   - However, if the Operator is busy at the time of call return, the system waits for the Operator to become free. When the Operator is free, C gets Ring Back Tone.
   - The Operator answers the call and is in speech with C.

3. Transfer On Busy:
   - C calls a Trunk of ANANT UCS.
   - The Operator answers the call.
   - The Operator puts C on Consultation Hold and dials B's extension.
   - The Operator gets busy tone from B's extension. B is busy with A
   - The Operator transfers the call to B on Busy tone.
   - B gets intrusion beeps. The beeps are played for the duration of the Transfer on Busy Timer (configurable; default: 30 seconds)
   - B may dial Flash to answer the call. A is put on hold.
   - B is now connected to C.

   - *If B does not answer the intrusion beeps at the end of the Transfer on Busy Timer, the call is returned to the Operator. C gets ring back tone.*
   - *If the Operator too is busy at the time of call return, C gets busy tone.*

Call Transfer - Trunk to Trunk

- C calls a Trunk of ANANT UCS.
- The Operator answers the call.
- B is out of office, but is available at external number D.
- The Operator puts C on Consultation Hold.
- The Operator dials trunk access code and calls the external number D.
- The Operator may:
  Wait for D to answer the call, transfer the call if D accepts the call. (screened transfer)
  OR
  Transfer the call as soon as D's phone starts ringing. (transfer while ringing)
- C and D are now in speech.

Call Transfer - Blind Transfer to VMS

- C calls a Trunk of ANANT UCS. C wants to talk to A.
- The Operator attends the call.
- The Operator puts C on Consultation Hold.
- The Operator dials Feature Access Code for Blind Transfer to VMS and A's extension number.
- The system hunts for the mailbox assigned to A's extension.
- When the mailbox is found, the Operator gets confirmation tone. C gets connected to A's mailbox.
- C gets voice prompts. C may follow the prompts to leave a message.

- *If A does not have a mailbox assigned, the Operator will get an error tone while transferring the call.*
- *The Operator may retrieve C's call by pressing Transfer Key/Flash /Call Appearance key.*

Feature Interactions:

- **CLIP and Caller ID Presentation while Transfer:** ANANT UCS provides the flexibility to display either the extension number that is transferring the call or the held party's number, that is, the number of the party that is about to be transferred. Refer “Calling Line Identification and Presentation (CLIP)”.

- **DND:** Call Transfer will not work if the destination extension has set DND.

- **Call Hold:** You must retrieve a held call, to Transfer it.

How to configure

To be able to use Call Transfer, this feature must be enabled in the Class of Service group of the extensions to be allowed this feature. The default values of the related Timers may be changed, if required.

To be able to use Blind Transfer to VMS, the extensions must be assigned a mailbox. For instructions, refer to “Configuring Voice Mail System”.

Call Transfer in Class of Service

In the default Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS, the default CoS group 01 in this Station Basic Feature Template has Call Transfer included in the 'Basic Features'. So, all extensions of ANANT UCS can use this feature.

You cannot disable 'Call Transfer' selectively without disabling the entire set of 'Basic Features'.

Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” to know more.

Call Transfer Related Timers

- **Transfer While Ringing Timer:** This timer is related to Call Transfer - While Ringing. It is the time for which the system rings the extension. By default it is set to 30 seconds. At the end of the timer the call is returned to the transferring extension.

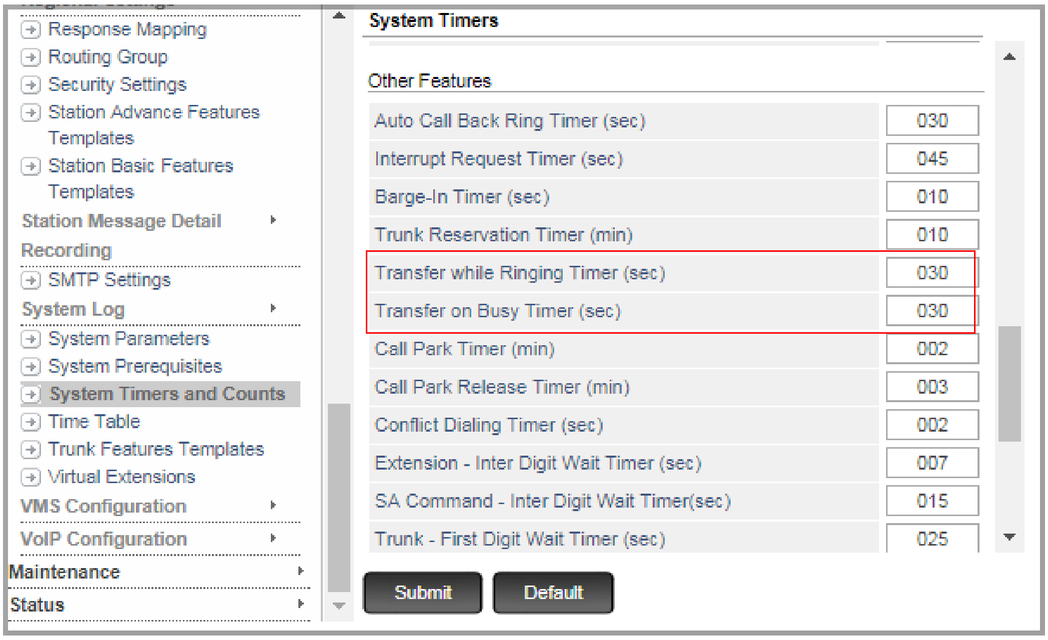
- **Transfer on Busy Timer:** This timer is related to Call-Transfer on Busy. It is the time for which the system waits for the busy extension to respond to the intrusion tone. By default the timer is set to 30 seconds. At the end of the timer the call is returned to the transferring extension.

Changing Call Transfer Related Timers

- Login as System Engineer.

- Under **Configuration**, click **System Timers and Counts**.

- Scroll to **Other Features** and change the values as required for Call Transfer related timers.



- Click **Submit.**

How to use

For Extended IP Phone Users

Extension to Extension:

- Speech with extension.
- Press DSS Key assigned to desired party extension.
- Go ON-Hook or press ‘Transfer’ Key.
  OR
- Speech with extension.
- Press ‘Transfer’ Key. You get feature tone.
- Dial desired party's extension number.
- Go ON-Hook or press ‘Transfer’ Key.

Extension to Trunk:

- Speech with extension.
- Press DSS Key assigned to Trunk.
- Dial External Number (transfer target)
- Go ON-Hook or press ‘Transfer’ Key.
  OR
- Speech with extension.
- Press ‘Transfer’ Key. You get feature tone.
- Dial-Trunk Access Code[87]- External Number.
- Go ON-Hook or press ‘Transfer’ Key.

Trunk to Extension Transfer:

- Speech with Trunk.
- Press DSS Key assigned to desired party extension.
- Go ON-Hook or press ‘Transfer’ Key.
  OR
- Speech with Trunk
- Press ‘Transfer’ Key and dial desired party's extension number
- Go ON-Hook or press ‘Transfer’ Key.

Trunk to Trunk Transfer:

- Speech with Trunk.
- Press DSS Key assigned to Trunk.
- Dial External Number.
- Speech with External Number.
- Go ON-Hook or press ‘Transfer’ Key.
  OR
- Speech with Trunk.
- Press ‘Transfer’ Key and dial Trunk Access Code[88]-External Number.
- Speech with the External Number.
- Go ON-Hook or press ‘Transfer’ Key.

Blind Transfer to VMS:

- Speech with extension.
- Press DSS Key assigned to Blind Transfer to VMS.
- Dial desired party’s extension number.
- Go ON-Hook or press ‘Transfer’ Key.
  OR
- Speech with extension.
- Press Transfer key.
- Dial 1078.
- Dial desired party's extension number.
- Go ON-Hook or press ‘Transfer’ Key.

87. *Trunk Access Code: users worldwide may dial a code from 0, 5, 61, 62, 63, and 64. Users in USA may dial a code from 0, 9, 81, 82, 83, and 84.*
88. *Trunk Access Code: users worldwide may dial a code from 0, 5, 61, 62, 63, and 64. Users in USA may dial a code from 0, 9, 81, 82, 83, and 84.*

Calling Line Identification and Presentation (CLIP)

ANANT UCS provides the facility of detecting the caller's number and presenting it on the display of the called extension phone. This feature is called Calling Line Identification and Presentation (CLIP).

In the case of Call Transfer, the system also provides the option of displaying to the destination extension either the number of the party that is put on hold to be transferred, that is, the Held Party or the number of the Transferring Party, while the call transfer is taking place. This feature is called Caller ID On Call Transfer.

Caller ID On Call Transfer

The Caller ID Presentation while Transfer gives you the choice of displaying to the destination extension, the CLI of either the party put on hold to be transferred, that is, the Held Party or the Transferring Party, that is, the party transferring the call, during a call transfer.

This feature is to be configured in the Station Advanced Feature Template applied on the extension.

In the Station Advanced Feature Template 01 assigned to all extensions by default, Caller ID of the transferring party is selected. But if you want to show the Caller ID of the Held party, select this option in the default Station Advanced Feature Template 01.

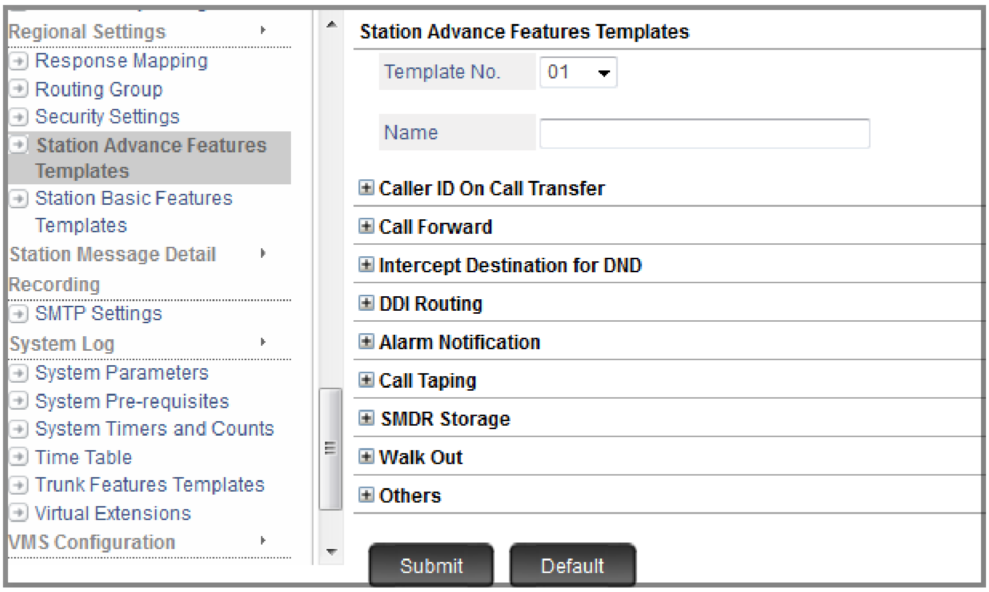
If all certain extensions are to be provided Caller ID of the Held Party and others the Caller ID of the Transferring Party,

1. Create separate Station Advanced Feature Templates.
2. Select 'Held Party' or 'Transferring Party' as desired in each Template.
3. Apply this Template on the extensions.

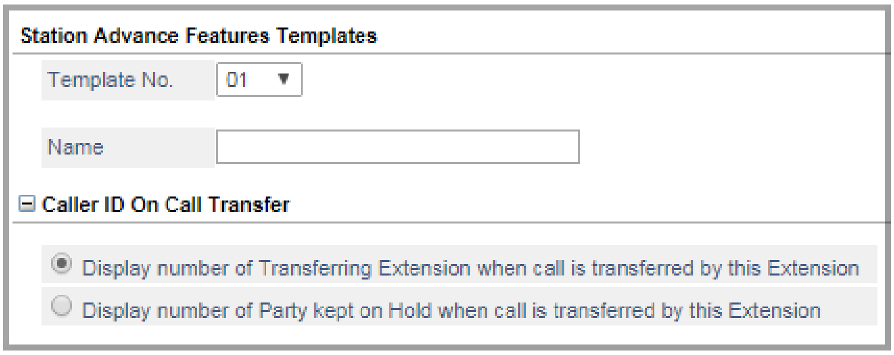
Configuring Caller ID On Call Transfer

- Login as System Engineer.

- Under **Configuration**, click **Station Advanced Feature Templates**.



- Click **Caller ID On Call Transfer** to expand.
- Select the desired option: **Display number of Transferring Extension when call is transferred by this Extension** or **Display number of Party kept on Hold when call is transferred by this Extension**.



- Click **Submit.**

Calling Line Identification Restriction (CLIR)

ANANT UCS allows extension users to suppress their identity, that is, extension number and name, when they call other extensions. This feature is called Calling Line Identification Restriction (CLIR).

Extensions that have 'CLIR Override' facility can view the CLI of those that have suppressed it with CLIR.

This is a feature of the system. It is applicable for internal calls only.

This feature will work only on the Matrix Extended IP Phone, VARTA UC Clients that support Caller Line Identification (CLI).

How it works

- Extension A has CLIR enabled.
- Extension B does not have CLIR Override enabled.
- Extension C has CLIR Override enabled.
- When A calls B, B cannot view the extension name and number of A.
- When A calls C, C can view A's extension name and number.

Now,

- Extension D calls extension E.
- A picks up the call.
- D will be able to view A's name and extension only if it has CLIR Override enabled.

Feature Interactions:

- **CLIP and Caller ID Presentation while Transfer:** Both these features will not work if CLIR is enabled.

How to configure

CLIR and **CLIR Override** are Class-of-Service dependent features. Extensions that are to be allowed these features, must have them enabled in their **Class of Service (CoS)** group.

Decide which extensions should be allowed CLIR and which should be allowed CLIR Override.

By default, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS. Station Basic Feature Template 01 is assigned CoS group 01 in which both CLIR and CLIR Override are disabled. Thus, none of the extensions of the ANANT UCS can suppress their CLI or force any other extension to display its CLI.

If you want to enable both features on all extensions, simply enable CLIR and CLIR Override in the default CoS group 01.

If you want to allow CLIR to all extensions, but not allow CLIR Override to any extension, simply enable CLIR in the default CoS group 01.

If you want to allow CLIR and/or CLIR Override to selected extensions, only, then follow these steps:

- Define a new CoS group with CLIR/CLIR Override enabled.
- Prepare a Station Basic Feature Template with this CoS group applicable in all the “Time Zones”.

- Assign this new template to the extensions to which CLIR/CLIR Override is to be allowed.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for detailed instructions and programming.

How to use

For Extended IP Phone Users

To enable CLIR

- Press DSS Key assigned to CLIR[89].
  OR
- Dial **103-1.**
- You get confirmatory tone and message on the phone's display.
- Go idle or you get dial tone after 3 seconds.

To disable CLIR:

- Press DSS Key assigned to CLIR again[90].
  OR
- Dial **103-0.**
- You get confirmatory tone and message on the phone's display.
- Go idle or you get dial tone after 3 seconds.

---

89. *System Engineer is recommended to assign a DSS Key with LED to this feature. When the assigned DSS key is pressed, it will glow red indicating that CLIR is enabled.*
90. *If a DSS key with LED has been assigned, when you press the key again, the LED will be turned off indicating CLIR is now canceled.*

Cancel All Station Features

For each feature of ANANT UCS that extension users have set/enabled on their extension, the system provides access code for cancellation of the feature.

Extension users can also cancel all features set on their extension by dialing a feature access code. These set of features can also be canceled for any extension user from the SA Jeeves.

How to use

The extension users can cancel all features set on their extension from their own extension or these can be canceled from SA Jeeves for any extension user.

Cancel All Features by Extension Users

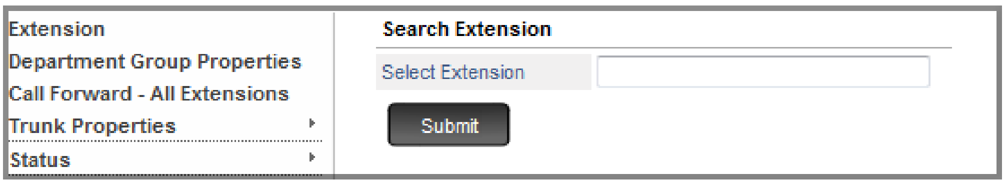
For Extended IP Phone Users

- Press DSS Key assigned to 'Cancel All Features' (if programmed).
  OR
- Dial **1051**.
- You get confirmation tone and confirmatory message on your phone display.
- Go idle or wait for dial tone.

Cancel All Features for Extension Users

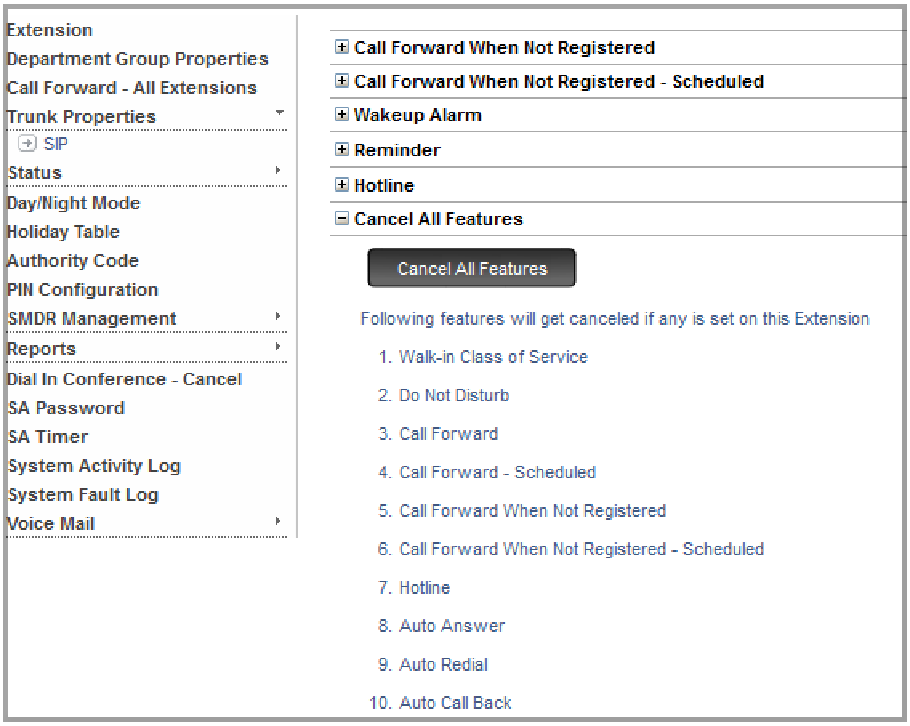
The Operator or any extension user having access to System Administrator mode can cancel all features for extension users using Jeeves.

- Login as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension you want to search.
- Click **Submit**.
- The searched extension users details appear on your screen.

- Click **Cancel All Features** to expand.



- Click the **Cancel All Features** button. The following features will get canceled if any is set on this extension user:
  - Walk-in Class of Service
  - Do Not Disturb
  - Call Forward
  - Call Forward - Scheduled
  - Call Forward When Not Registered
  - Call Forward When Not Registered - Scheduled
  - Hotline
  - Auto Answer
  - Auto Redial
  - Auto Call Back

Class of Service (CoS)

Class of Service (CoS) defines the permission an extension will have on a System. It defines the set features of the system that the extension is to be allowed access to.

Feature requirements vary among users and with time. Certain groups of extension users may have a need for voice mail, while another group may need the ability to forward calls to a cell phone, and still others may have no need to make calls outside the office.

Similarly, certain features that are required during working hours may not be required during break or non-working hours.

ANANT UCS offers the flexibility to allow or deny extension users access to features of the System, on the basis of their requirement and according to time of the day. For users, access to various features from their extensions is their CoS.

How it works

The list of all features allowed to an extension is referred to as 'CoS group'. There are 20 CoS groups numbered from 01 to 20.

Each extension port of the System has an associated CoS group that indicates which features of the System the port is allowed to access.

The CoS group of an extension port is defined in the “Station Basic Feature Template” applied to that extension port. It is defined for each "Time Zone", namely, working hours, break hours, and non-working hours, in the Station Basic Feature Template.

A feature can be allowed or denied to an extension by enabling or disabling it in the CoS group of the Station Basic Feature Template applied to that extension.

The same CoS group uniformly to all extensions ports for all Time Zones. Doing so, all extensions can access the same set of features in all time zones. For example: CoS group 03 is assigned to all extensions for Working, Break and Non-Working hours.

A different CoS group for each Time Zone can be assigned to all extension ports. Doing so, all extensions can access only those features allowed for the particular Time Zone.

For example: All extensions are assigned CoS group 03 for Working, CoS group 04 for Break hours and CoS group 05 for Non-Working Hours.

Different CoS groups can be assigned to different extension ports, for all or for different Time Zones. Doing so, each extension can access a different set of features in each Time Zone.

For example: extensions 3001 to 3010 are allowed CoS group 03 for all Time Zones, while extensions 3011 to 3015 are assigned CoS group 03 during Working Hours, and for the Non-Working and Break Hours, they are assigned CoS group 04 and 05 respectively.

The following features can be set/canceled on extensions from the SA mode, regardless of whether these features are allowed or denied in the CoS assigned to the extensions:

- *Call Forward*
- *DND*
- *Dynamic Lock and Timer*
- *Hotline*

Basic Features

A set of features including Internal Call, Call Hold, Call Toggle, Call Transfer, Department Call, Operator Access, Redial, and Call Mute defined as Basic Features and allowed in all CoS groups.

It is not possible to enable or disable selectively any of the features included in "Basic Features".

How to configure

The table below presents the CoS groups from 01 to 20 with the list of features supported on the extensions.

Default CoS Groups

Feature Name	Class of Service Group																			
	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20
Account Code	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
SA Mode																				
SA Extension																				
Auto Call Back-Busy	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Auto Call Back-No Reply	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Auto Redial																				
Auto Redial Priority																				
Barge-In																				
Call Forward	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Call Park	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Call Pickup	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Change Room Clean Status																				
Global Directory Programming																				
CLIR																				
CLIR Override																				
Conference	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Continued Dialing	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Conversation Recording																				
Decrement Dynamic Lock Timer for Internal Calls																				
DISA																				
Do Not Disturb	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Do Not Disturb - Override																				
Dynamic Lock	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Forced Answer																				
Forced Release																				
Global Directory Part-1																				
Global Directory Part-2																				
Global Directory Part-3																				
Hotline																				

Feature Name	Class of Service Group																			
	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20
Interrupt Request	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Live Call Supervision																				
Message Wait (Set/Cancel)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Mini Bar Details																				
Paging	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Privacy from Interrupt Request, Barge-In and DND-Override																				
Privacy from Raid	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Raid																				
Return Call to Original Caller (RCOC)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Room Monitor																				
Selective Port Access																				
Trunk-Trunk Transfer																				
Basic Features[a]	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
General MailBox	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Dept. Group-Call Forward	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Emergency Conference	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Intercom																				
DSS Call Pickup-Station																				
DSS Call Pickup-Trunk																				
Closed User Group (CUG)																				
PIN Dialing																				

a. Basic Features includes: Internal Call, Call Hold, Call Toggle, Call Transfer, Department Call, Operator Access, Redial, Mute.

CoS group number 01 is assigned for all Time Zones in the default Station Basic Feature Template 01 assigned to all extensions of ANANT UCS.

CoS group number 19 and 20 are assigned when the Hospitality Application of ANANT UCS is used. When this application is being used in CoS group number 20, the SA Mode will be enabled. See ANANT UCS Hospitality System Manual.

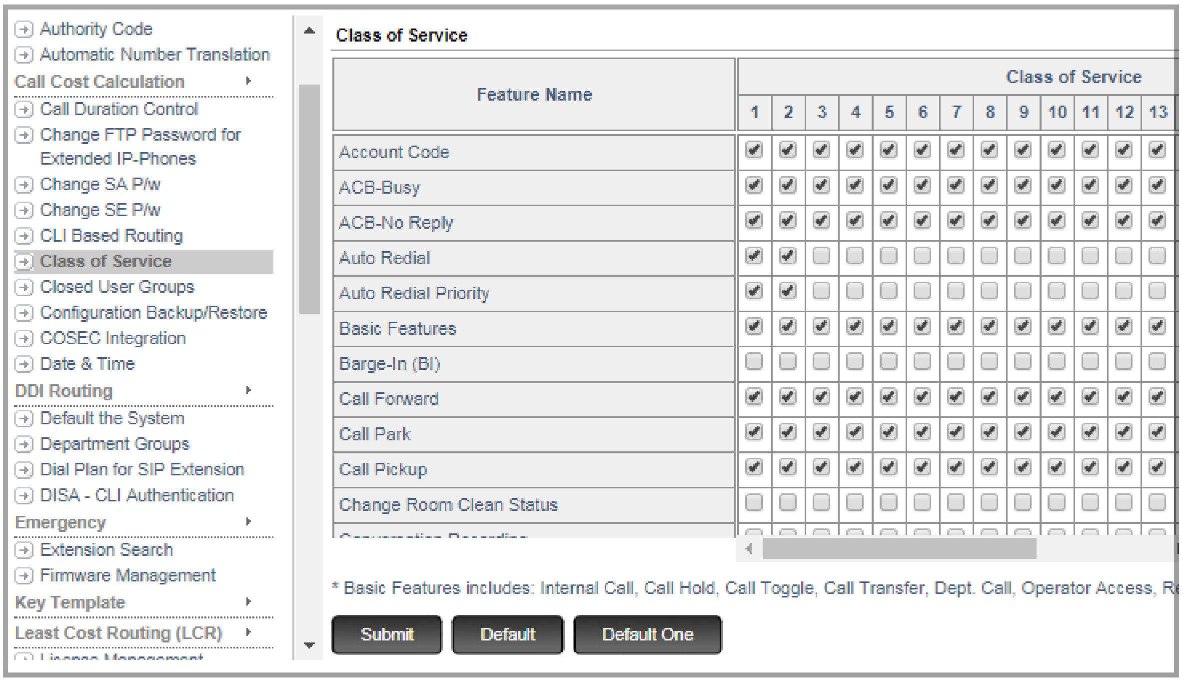
Creating CoS Groups

- Take a pen and a paper pad.
- Prepare a list of extensions.
- Read the list of features supported on the extensions (see above Table 'Default CoS Groups for Features').
- Against each extension name on the list, write the features needed for each Time Zone. You will notice that the features needed by many extensions are identical.
- List the common features to be allowed to and features to be denied to all extensions. Assign a CoS Group Number to this list.
- Are there any other features, in addition to those on the common list, which you want to allow to selected extensions?

- If yes, extend the common list you prepared by adding the features to be allowed to selected extensions. Assign a CoS Group Number to this extended list.
- You can prepare different CoS Groups for different Time Zones and assign a number to each group.
- For example, you may end up creating five different CoS groups. The First group may contain none of the features. The Second group may contain the most common features like Call Forward, Call Transfer, Internal Dialing, etc. The Third group may contain more advanced features, and the Fourth group may contain even more advanced features. The Fifth group may contain all the features.
- When you are finished preparing the CoS groups you need, program the CoS groups using Jeeves.
- Now, the CoS groups to be assigned to extensions must be configured in the Station Basic Feature Template applied to the extensions. This can be done using Jeeves.

Enabling a feature in a CoS Group

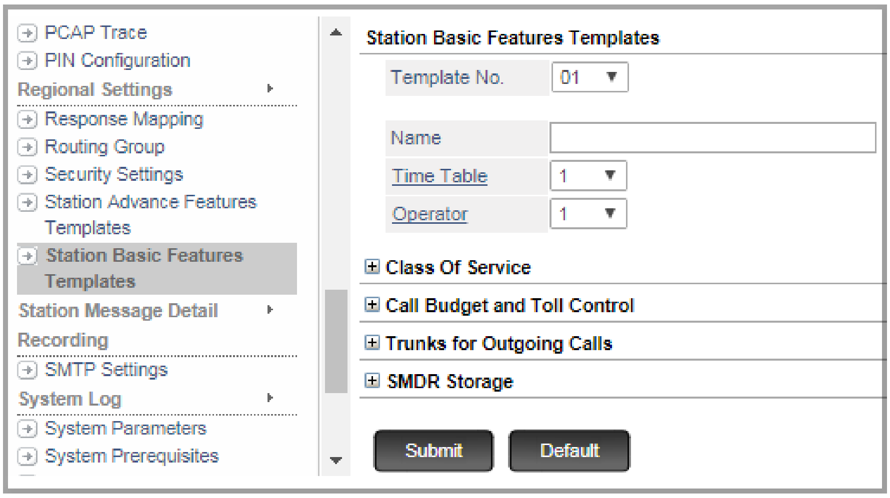
- Login as System Engineer.
- Under **Configuration**, click **Class of Service (CoS)**.



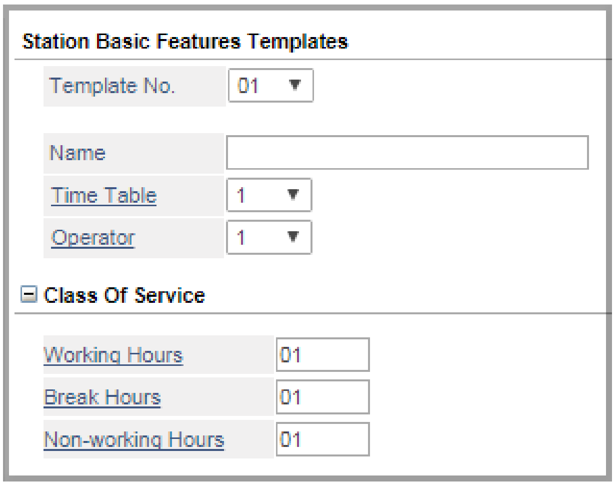
- The default CoS groups from 01 to 20 appear. The check boxes selected under each CoS group column indicate that the feature is enabled in that CoS group. The default CoS groups meet the requirements of most extension users. Check the default CoS groups whether the features you want to allow are enabled and features you want to deny are disabled.
- To enable a feature in a CoS group, select corresponding check box in the CoS group. To disable a feature simply clear the check box. For example: to enable DND-Override in CoS group 01, select the check box against DND-Override in CoS group 01. To disable clear the check box again.
- Click **Submit**.

Assigning a CoS group in a Station Basic Feature Template

- Login as System Engineer.
- Under **Configuration**, click **Station Basic Feature Template**.



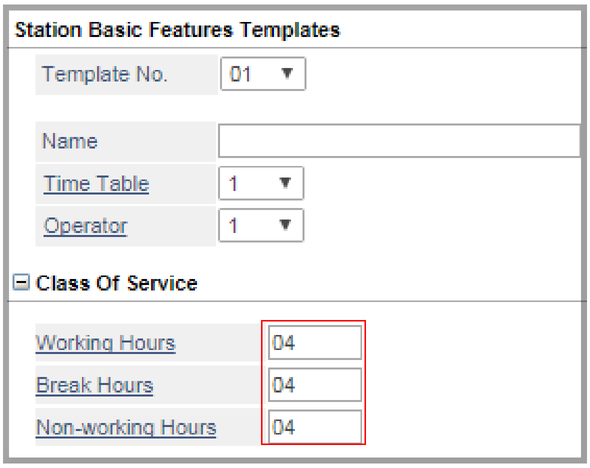
- Select the desired **Template No.**
- Click **Class Of Service** to expand.



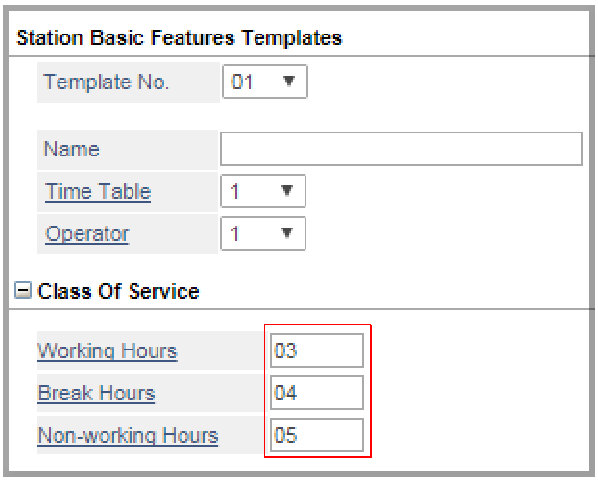
- The default CoS group assigned to each time zone, that is, working hours (WH), break hours (BH) and non-working hours (NH) appears under Class of Service.
- To assign a different CoS group to a Station Basic Feature Template, enter the CoS group number for each time zone under Class of Service.
- Click **Submit.**

  By default, Station Basic Feature Template 01 is applied on all extensions and CoS group 01 is assigned by default to this template in all time zones.

  If all extensions to be allowed the same set of features during working hours, break hours, non-working hours, enter the same CoS group number in all time zones in the Template Number applied to all extensions. For example: to assign CoS group 04 to all time zones in Station Basic Feature Template Number 01; enter 04 under WH, BH and NH.



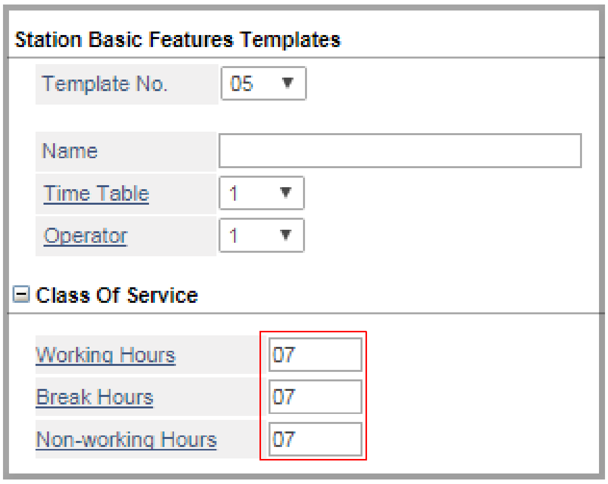
If all extensions are to be allowed a different set of features in each Time Zone, enter the CoS group for each Time Zone. For example: to assign CoS group 03 in Working hours, 04 in Break Hours and 05 in Non-Working hours in Station Basic Feature Template 01, enter 03, 04, 05 under WH, BH and NH respectively.



- If a set of features is to be allowed to selected extensions only, assign the CoS group with these features enabled to a separate Station Basic Feature Template. Apply this template to the selected extensions which are to be allowed this CoS.

  For example: To assign all features to extensions, create a CoS group with all features enabled, CoS group 07. Select a different Station Basic Feature Template, for example 05. Enter CoS Group 07 in all

Time Zones in Station Basic Feature Template 05. Apply Station Basic Feature Template 05 to the software ports of the extensions that are to be assigned all features.



- Click **Submit.**

After you have configured the CoS group in the Station Basic Feature Template, you must assign this template to the stations. Refer the topic "Station Basic Feature Template" for instructions on applying templates on SIP extensions.

CLI Based Routing

ANANT UCS offers the facility to detect the calling party's number and name. This is known as Calling Line Identification.

On the basis of CLI, it is possible to land calls from a particular number on a particular extension or group of extensions. This is known as CLI Based Routing.

How it works

A, B, C are extensions. D and E are external callers.

Calls made by D are to be landed on A.

Calls made by E are to be landed on B and C.

The CLI of D and E and their corresponding landing destinations should be entered in the CLI Based Routing Table.

CLI Based Routing should be enabled on the desired trunks for each Time Zone (working hours, break hours and non-working hours).

The system can match the incoming call CLI with the numbers configured in the CLI table in two ways, that is, Match from last digit of CLI or Match from first digit of CLI.

- If you select Match from last digit of CLI, this is how the call will be routed:
  - D calls on a trunk of ANANT UCS.
  - The system checks if CLI Based Routing is enabled on the trunk for the current time zone.
  - If CLI Based Routing is enabled on the trunk, the system checks the numbers stored in the CLI Based Routing table.
  - D's number is found in the CLI Based Routing Table.
  - The system checks the destination number stored against D's CLI.
  - A's number is found as the destination extension.
  - The system lands the call on A.
- If you select Match from first digit of CLI, this is how the call will be routed:
  - D calls on a trunk of ANANT UCS.
  - The system checks if CLI Based Routing is enabled on the trunk for the current time zone.
  - Then the system checks if the parameter **Replace '+' from CLI**[91], is enabled.
  - If enabled, the system will replace **+** sign received in the incoming CLI with the number string configured in the **Replace '+' from CLI with the number string**. To know more, see "System Parameters".
  - Now the system checks if **Incoming CLI Modification**[92] is enabled.
  - If enabled the system modifies the incoming number according to the parameters configured in the Incoming CLI Modification. To know more, see "Incoming CLI Modification" in "System Parameters".
  - The system matches the modified number string with the numbers stored in the CLI Based Routing table.

91. *To comply with the Indian Government Laws and Regulation, this parameter is not provided for India Region.*
92. *To comply with the Indian Government Laws and Regulation, this parameter is not provided for India Region.*

- D's number is found in the CLI Based Routing Table.
- The system checks the destination number stored against D's CLI.
- A's number is found as the destination extension.
- The system lands the call on A.

If D's number does not exist in the CLI Based Routing Table, the call will be routed according to the incoming call management logic.

How to configure

For this feature to work, you must do the following:

- enter the numbers of the calling parties and the numbers of the corresponding destination extensions in the CLI Based Routing Table. You can store up to 2000 numbers in the CLI Routing Table.

- enable CLI Based Routing on the desired trunks according to time zones in their “Trunk Feature Template”.

Creating CLI Routing Table

To apply this feature,

- On a sheet of paper, create a 5-column table, as illustrated below. Each calling party number in the CLI table is stored a location index in the system. Enter the telephone numbers and names of the calling parties and the corresponding landing destinations, that is, the Port Type and Port Number. The Port Type may be SIP Extension, Routing Group, Virtual Extension or Voice Mail Auto Attendant.

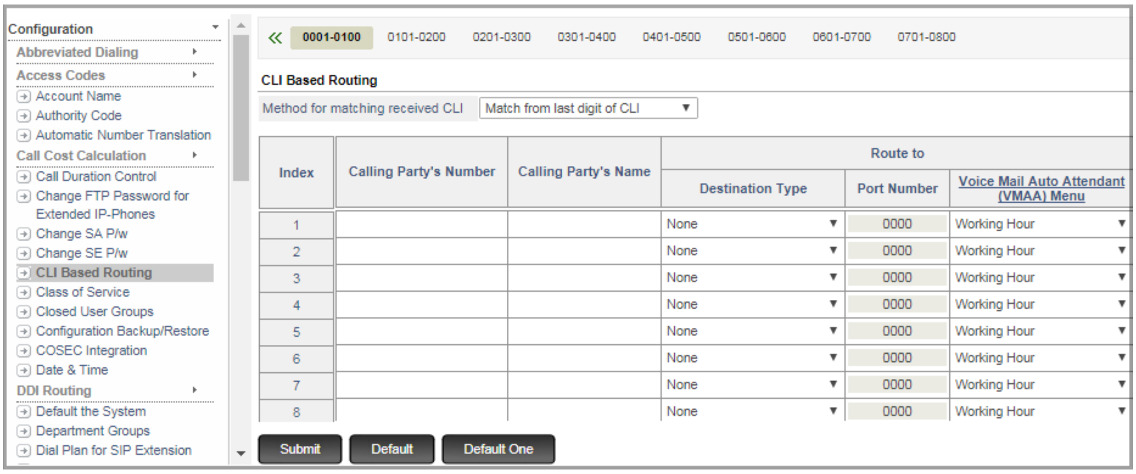
- The 'Name' field is for identifying the entry. When placing a call on the destination extensions, both the number and the 'Name' are presented in the CLI.

- Determine the method which the system should use to match the incoming CLI with the numbers in the table, that is, Match from last digit of CLI or Match from first digit of CLI Based Routing Table.

Index	Phone Number	Name	Port Type	Port Number
1	2640459	MidasBiz	Routing Group	0002
2	022281110001	Jet Set	SIP Extension	0004
:				:
10	2640075	Bacchus	Routing Group	0003

Configuring CLI Based Routing

- Login as System Engineer.
- Under **Configuration**, click **CLI Based Routing**.



- In **Method for matching received CLI**, select the method according to which you want the system to match the received CLI with the numbers stored in the CLI table. You can select:
  - Match from last digit of CLI
  - Match from first digit of the CLI

  The method you select will be applicable to all the numbers configured in the CLI Based Routing Table.

- In the CLI table each number is to be stored at a Location Index numbered from 0001 to 2000.

  There are 100 entries on each page. To go to the next 100 Index numbers, click the tabs 0101-0200, 0201-0300, 0301-0400 , 0301-0400...............................1901-2000.

- At each Location Index, enter the following parameters:

  - **Calling Party's Number:** enter the number of the calling party, not exceeding 16 digits. You can also enter '+' in the number string.

  - **Calling Party's Name:** enter the name of the calling party. You can enter a maximum of 8 characters in this field.

  - **Destination Type:** select the landing destination extension. It may be SIP extension, Routing Group, Virtual Extension or Voice Mail Auto Attendant.

  - **Port Number:** enter the port number (software port or routing group number) to which the landing destination is connected. This is applicable for Destination Types — SIP Extension or Routing Group.

    If you have selected *Virtual Extension* as the Destination Type, enter the software port of landing destination of the Virtual Extension.

    If you have selected *Routing Group* as the Destination Type, enter the number of the Routing Group (01 to 96) in this field.

- **Voice Mail Auto Attendant (VMAA) Menu:** if you have selected the *Voice Mail Auto Attendant* as the Destination Type, select the VMAA Menu to be assigned to the respective calling party.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- Click **Submit**.
- Enable CLI Based Routing for the desired Trunks in their “Trunk Feature Template”. Refer the topic “Customizing Trunk Feature Templates” for instructions.

Closed User Group (CUG)

A Closed User Group is a network of systems to provide seamless connectivity. To have private networks, few systems can be connected to each other over the IP Network. These systems connected in the network behave as a single unit. Extension users of one system can reach the extension users of the other system without dialing any access code, as if they were dialing extension numbers of their own system. Users will not know whether they are dialing an extension number of their own system or of the other system.

There can be two different application scenarios in this case:

- A network in which you have unique extension number in all the systems
- A network in which you have same extension number in all the systems

How it works

Scenario 1

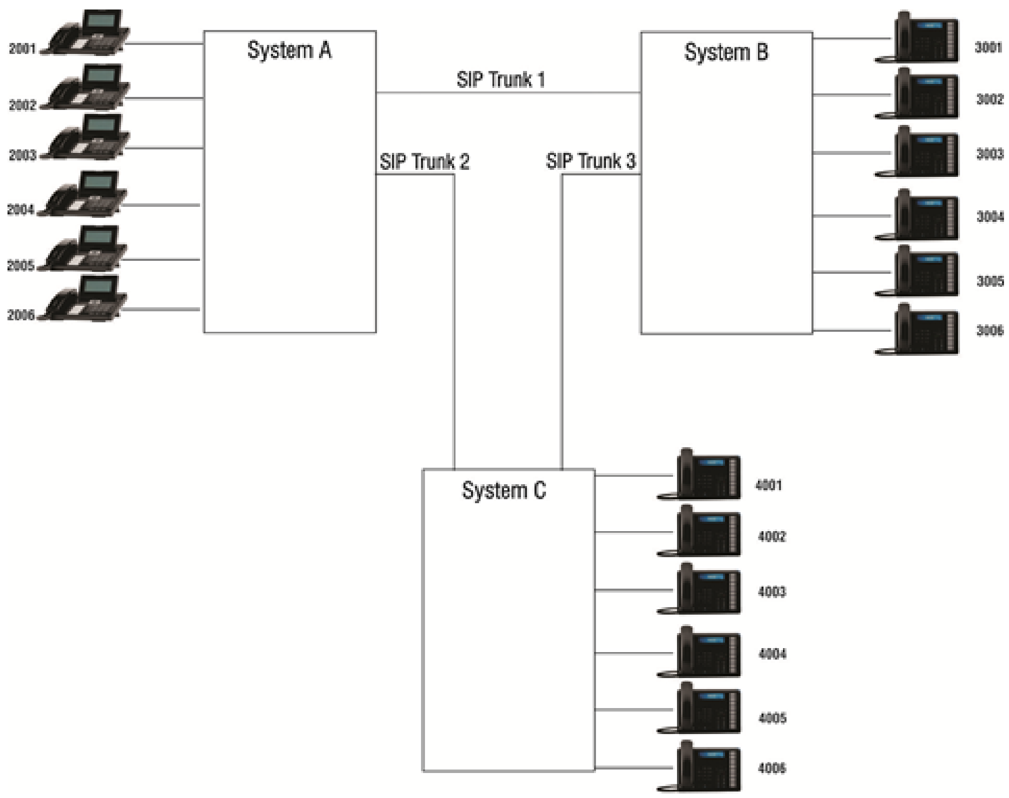
A network in which you:

- have unique extension number in all the systems, that is, one cannot have extension number 2001 in System-A as well as in System-B or System-C.

- S1 to Sn are extensions.

- SIP Trunk 1 to SIP Trunk 3 are SIP Trunks connecting the three Systems.

- enable Closed User Group (CUG) in the Class of Service assigned to the extension users.



You need to make sure the following parameters of the CUG table are configured as mentioned below:

- **Index:** Maximum 250 different routes (001-250) can be configured.
- **Route Code:** Generally route code will be a truncated number of the extension numbers. For example in the figure given above, route code for System-B can be defined as ‘3’ and that for System-C can be defined as ‘4’.

  Route code could be of maximum sixteen digits. Valid Digits: 0 to 9, * # A B C D F P, where P is Pause, F is Flash, A to D is DTMF Digits.

  If System-B were having extension numbers from 3100 to 3199 and System-C were having extension numbers from 3200 to 3299 then route code for System-B can be defined as ‘31’ and that for System-C can be defined as ‘32’. If System-B were having extension numbers from 301 to 399 and 401 to 499 then two route codes can be defined for System-B viz. ‘3’ and ‘4’. Likewise for System-C.
- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of Rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the group is selected. This helps to select an alternate route. Whereas if Rotation is ON then the trunks in the OGTBG are selected in round robin fashion.
- **Strip Digit Count:** For this application, configure the parameter as 0. This parameter is relevant when you have the same extension numbers in each system.
- **Self-Route:** For this application make sure it is disabled. This parameter is relevant when you have the same extension numbers in each system.
- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be programmed here. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.
- **Apply Toll Control**: When Self Route check box is disabled, system will check this parameter.

  By default, the Apply Toll Control check box is enabled. The system will apply toll control to all the outgoing calls. Disable this check box, if you do not want to apply toll control to the CUG numbers dialed by you. For detailed information, see “Toll Control”.

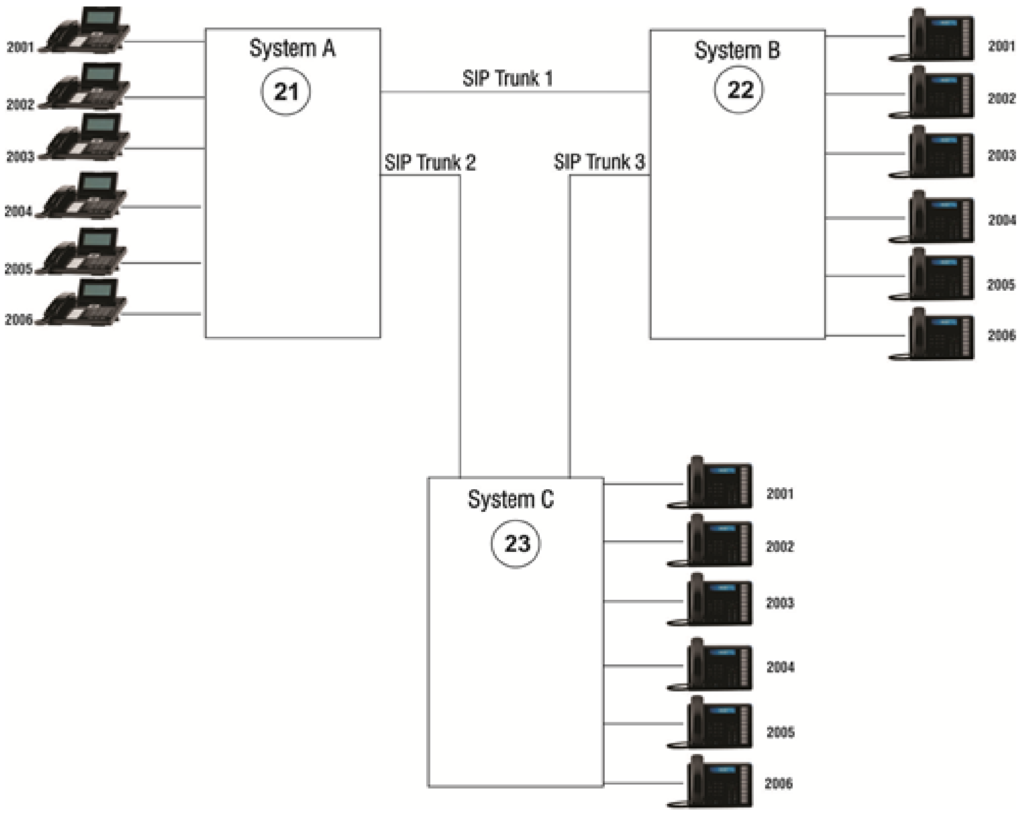
  Apply Call Cost: By default, this check box is enabled and the system will calculate the cost of each call.

  For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries. You can also set the filter **Calls with units more than** to generate a report according to the Call Cost. For details, see “Station Message Detail Recording-Report”.

Scenario 2

A network in which you:

- have same extension number in all the systems, that is, extension number 2001 in System-A as well as in System-B or System-C.
- S1 to Sn are extensions.
- SIP Trunk 1 to SIP Trunk 3 are SIP Trunks connecting the three Systems.
- enable Closed User Group (CUG) in the Class of Service assigned to the extension users.



You need to make sure the following parameters of the CUG table are configured as mentioned below:

- **Index:** Maximum 250 different routes (001-250) can be programmed.
- **Route Code:** Route code could be of maximum sixteen digits. Valid Digits: 0 to 9, * # A B C D F P, where P is Pause, F is Flash, A to D is DTMF Digits. Generally, route code will be a unique number. The route code should not clash with any of the extension numbers of same System. For example in the figure given above, route code for System-A can be defined as ‘21’, route code for System-B can be defined as ‘22’ and that for System-C can be defined as ‘23’. This means that no extension in System-A can start with ‘22’ or ‘23’. Similarly, no extension in System-B can start with ‘21’ or ‘23’ and no extension in System-C start with ‘21’ and ‘22’.
- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the

group is selected. This helps to select an alternate route. Whereas if rotation is ON then the trunks in the OGTBG are selected in round robin fashion.

- **Strip Digit Count:** This count signifies the number of digits to be stripped off while dialing a number. To elaborate as per the requirement if extension 2001 of System-B dials 212002 and if SIP Trunk 1 is busy then the call should reach extension 2002 of System-A through alternate route. In this case the strip digit count of System-A should be programmed as 2 and that of System-B and System-C should be programmed as 0. Doing so, when extension 2001 of System-B dials 212002 and if SIP Trunk 1 is busy then the call is routed through System-C. In this case, System-B dials 212002 on SIP Trunk 3, System-C receive this code and dials out the same code, that is, 212002 on SIP Trunk 2 without striping of any digit. On receiving 212002, System-A strips of two digits as per the programming and routes the call to extension 2002.

- **Self-Route:** This check box signifies that the digits being dialed are for the same System and are not to be dialed on the SIP Trunk.

- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be configured in this field. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.

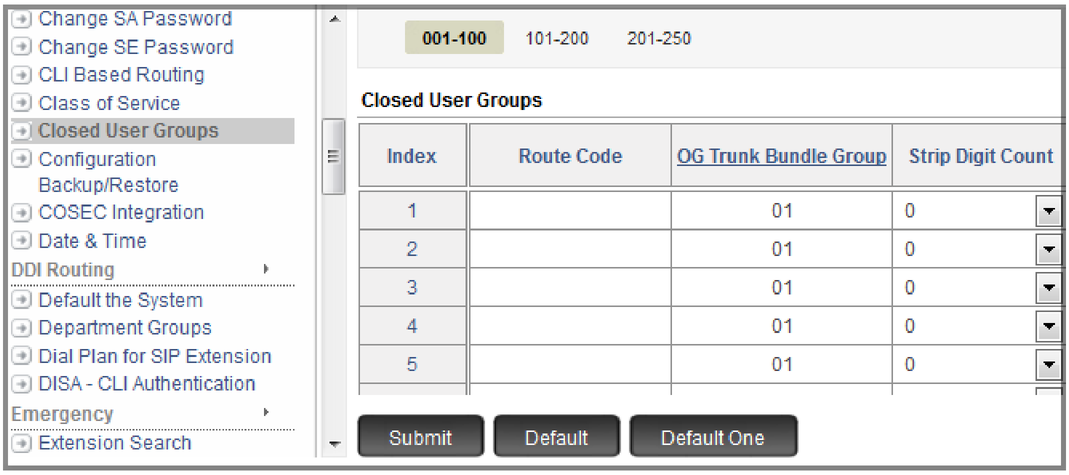
- **Apply Toll Control:** This parameter is not relevant as Self Route check box is enabled.

- **Apply Call Cost:** By default, this check box is enabled and the system will calculate the cost of each call.

  For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries. You can also set the filter **Calls with units more than** to generate a report according to the Call Cost. For details, see “Station Message Detail Recording-Report”.

Configuring Closed User Groups

- Login as System Engineer.
- Under **Configuration**, click **Closed User Groups**.



Against each Index configure the following parameters:

- **Route Code:** Route code could be of maximum sixteen digits. Valid Digits: 0 - 9, *, #, +, . (Dot). Generally route code will be a truncated number of the extension numbers. For example in the figure given above, route code for System-B can be defined as ‘3’ and that for System-C can be defined as ‘4’.
- **OG Trunk Bundle Group:** An OG Trunk Bundle Group (OGTBG) is assigned to each route code. Whenever a call is to be made on that route, a free trunk from the OGTBG is selected and the extension number is dialed on it. The same logic of Rotation On/Off for trunk selection from the OGTBG is used. If rotation is OFF then always the first trunk in the OGTBG is selected. If it is busy then the next trunk in the group is selected. This helps to select an alternate route. Whereas if Rotation is ON then the trunks in the OGTBG are selected in round robin fashion.
- **Strip Digit Count:** Configure this as per your requirement. Refer to the examples above.
- **Self-Route:** Configure this as per your requirement. Refer to the examples above.
- **Dialed Digit Count:** When digits are dialed on the trunk, the system waits for inter digit timer after the last digit is dialed. In order to avoid this timer and number of digits dialed to be routed without further delay, count for the number of digits to be programmed in this field. If the number of digits received are equal to the parameters programmed then the number is dialed out immediately without waiting for the inter digit timer. If the number of digits dialed by the user are not equal to the digits programmed, the number is dialed after inter digit timer.
- **Apply Toll Control**: By default, this check box is enabled. The system will apply toll control to all the outgoing calls. Disable this check box, if you do not want to apply toll control to the CUG numbers dialed by you.
- **Apply Call Cost:** By default, this check box is enabled and the system will calculate the cost of each call.

  For certain calls (internal calls) you do not require the call cost calculation, clear the check box corresponding to these entries.

- Click **Submit.**

Conference-3 Party

ANANT UCS offers three types of conference calls: Conference-3 Party, *"Conference Dial-In"*, and *"Conference-Multiparty"*.

Conference-3 Party (also referred to as Three-Way Calling) is a call, in which the calling party can talk to two other persons simultaneously.

A 3-Party Conference is initiated by dialing the number of the first person one wishes to talk to. The first person is informed about the conference and put on Consultation Hold. The number of the second person one wishes to talk to is dialed. When the second person answers, s/he is informed about the conference. Three-way speech is established by pressing Flash-*3.

An already connected two-way speech can be converted into a conference by adding a second person, without disconnecting the call with the first person.

Thus, a 3-Party Conference may be planned or conducted on the spur of the moment.

A 3-Party Conference can be conducted with extensions of ANANT UCS and between extensions and external numbers.

The maximum number of 3-Party conferences supported by ANANT UCS is 512.

Conference is a licensed feature. Decide the number of conferences you want to conduct and purchase the license accordingly. Refer the topic "Licenses Supported in ANANT UCS" to know more.

How it works

A, B, C are extensions.
D and E are external numbers.

3-Party Conference between extensions

- A is in speech with B.
- A and B want to include C in their conversation.
- A presses the 'Conference' Key. B is put on Consultation Hold.
- A gets feature tone. B gets on-hold music.
- A dials C's extension number. A gets ring back tone.
- A is in speech with C. B cannot hear their conversation.
- A presses the 'Conference' key a three-way speech is established.
- A, B, and C are now in speech.

- Any of them can disconnect to withdraw from the conference.
- If C disconnects, A and B will be in two-way speech.
- A and B can carry on the conversation or can have a conference with another trunk (external number) or with another extension.

3-Party Conference between two extensions and a trunk

- A is in speech with B.
- A and B want to include D in their conversation.
- A presses the ‘Conference’ Key. B is put on Consultation Hold.
- A gets feature tone. B gets on-hold music.
- A grabs a Trunk and dials D's extension number. A gets ring back tone.
- A is in speech with D. B cannot hear their conversation.
- A presses the ‘Conference’ Key to enable three-way speech.
- A, B, and D are now in speech.

- Any of them can disconnect to withdraw from the conference.
- If B disconnects, A and D will be in two-way speech.
- A can now conduct a conference with another extension or trunk.

3-Party Conference between an extension and two trunks

- A is in speech with D.
- A and D want to include E in their conversation.
- A presses the ‘Conference’ Key. D is put on Consultation Hold.
- A gets feature tone. D gets on-hold music.
- A grabs a Trunk and dials E's number.
- A is in speech with E. D cannot hear their conversation.
- A presses the ‘Conference’ Key to enable three-way speech.
- A, D, and E are now in speech.

- Any of them can disconnect to withdraw from the conference.
- If A disconnects, D and E are now in two-way speech.

- *A, B and C are in speech. When A disconnects, either B and C are also disconnected or speech is established between them depending on the option you select in* ***If the Extension creating 3 party conference, disconnects during Conference*** *in the System Parameters.*

- *The Conference can be broken only by the master Extended IP Phone that has initiated the Conference.*

- *If a call put on hold is to be included in a Conference, it must be retrieved first.*

- *If all the parties to the conference are SIP Extensions/Trunks and if the initiator of the Conference goes on-Hook during the conference, the other parties will still remain in conversation. This is known as Transfer on Conference Hangup.*

How to configure

For this feature to work, the feature 'Conference' must be enabled in the Class of Service group of the extensions that are to be allowed this feature.

If extension users at remote locations are to be allowed to initiate the 3-party conference, “Direct Inward System Access (DISA)” or “Auto Attendant” must be enabled on the trunk on which their call lands.

3-Party Conference in Class of Service

By default, Station Basic Feature Template Number 01 is assigned to all the extensions of the ANANT UCS. The Station Basic Feature Template 01 is assigned CoS group 01 which has 'Conference' enabled. So, all the extensions of the ANANT UCS can make Conference calls.

If you want to deny 3-Party Conference to selected extensions, follow these steps:

- Define a CoS group with 'Conference' disabled.
- Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
- Assign this new Template to the extensions to which Conference is to be denied.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for detailed instructions and configuration.

The feature 'Conference' in the Class of Service also includes Dial-In and Multi-party Conference. Extensions that are denied 'Conference' in their Class of Service will not be allowed any of these conference type - 3-Party, Dial-In and Multi-party Conference.

How to use

For Extended IP Phone Users

- Speech with Party 1 on trunk/extension.
- Press 'Conference' Key. Party 1 put on Consultation Hold.
- Dial the number of Party 2.

If Party 2 is a trunk,

- Dial Trunk Access Code, to grab a trunk.
- You get Trunk dial tone.
- Dial telephone number of Party 2. You get ring back tone.
- Speech with Party 2.
- Press the 'Conference' Key.
- Three-way speech is established.

When the Conference is established, the Conference Key LED will glow continuously and you will get a message "Conference" on your phone display.

Conference-Multiparty

Like the Dial-In Conference, a Multi-party conference allows speech between more than three participants.

The key difference between Dial-In and Multi-party conference is that in a Dial-In conference participants can include themselves in the conference by dialing into it without assistance, whereas in a Multi-party Conference the party initiating the conference must include the participants by dialing their numbers.

In a 3-party Conference, when you add the fourth participant, a Multiparty Conference is initiated.

A Multiparty conference may be

- between extensions
- between extensions and trunks, that is, external numbers.

Any participant in a Multiparty Conference can Include a party, Remove a party, Leave a conference temporarily or can Cancel a conference. When any participant is included in the conference, the system plays a beep to indicate the inclusion.

External callers can initiate multiparty conference using “Direct Inward System Access (DISA)”.

Refer the table below for details regarding the multiparty conferences supported by the system.

Maximum conference participants	Maximum simultaneous conferences (If all the conferences involve 3 parties)	Maximum parties included in a conference (If all the parties are in a single conference)
1536	512	64

Conference is a licensed feature. Decide the number of conferences you want to conduct and purchase the license accordingly. Refer the topic “Licenses Supported in ANANT UCS” to know more.

How it works

A, B, C, and D are extension users.
E and F are external numbers.
A decides to hold a teleconference with B, C, D, E and F.

Initiating a Multiparty Conference

- A has initiated a 3-party conference with B and C.
- If A dials the D’s number followed by the 3-party conference code. D is included in the conference.

  Now, the 3-party Conference is converted into a Multiparty Conference.

After the Conference has been initiated, conference participants can:

- **Include a Party in an on-going Multiparty Conference**.

Internal as well as external callers can be included in an ongoing conference by the any of the participants. To include external callers, in this case, E and F. A must dial the Trunk Access Code followed by E's number and the 3-party conference code, Similarly F can be included.

- **Remove a Participant from an on-going Multiparty Conference**.

  Participants can remove another participant from the Multiparty Conference. The IP Phone displays the numbers of all the participants, select the number of the participant to be removed from this list.

- **Temporarily Leave a Multiparty Conference**.

  Any participant in a conference can dial the Temporarily Leave Conference code to leave the conference for a short time period.

If all participants, Temporarily Leave the conference one-by-one, the system will start the 'Release Conference if idle for more than (Minutes) Timer'. This Timer is programmable, and by default it is set to 002 Minutes. If no participant rejoins the conference before the expiry of this Timer, the system will free the resource occupied by the conference on the conferencing circuit.

- **Rejoin a Multiparty Conference**.

  The participants who leave temporarily can rejoin the conference at a later stage by going Off-Hook and dialing the Rejoin Conference conference code.

- **Permanently Leave from a Multiparty Conference**.

  Participants in a conference can exclude themselves by going ON-Hook. Once any participant goes ON-Hook, he/she cannot rejoin the conference.

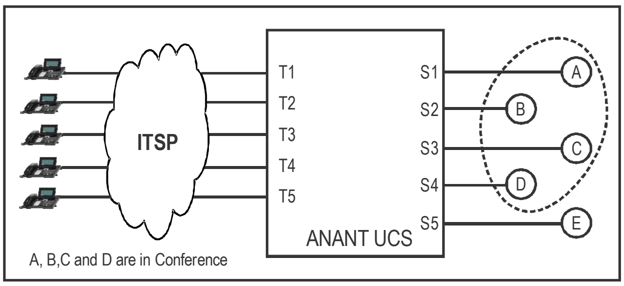
- **Cancel a Multiparty Conference**.

  Any participant in a conference can dial the Cancel conference code to end the conference.

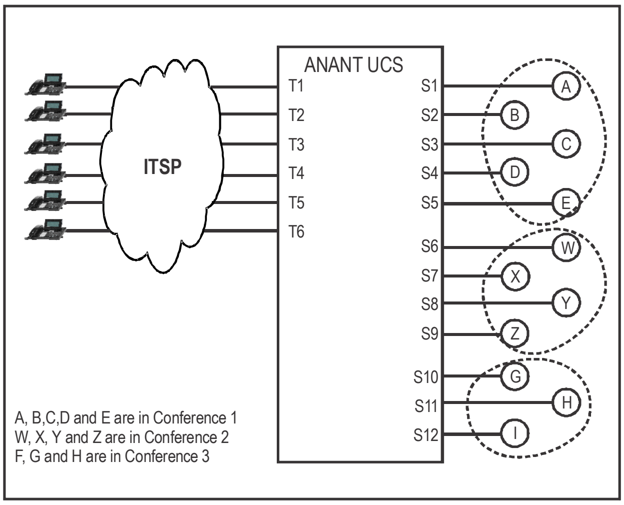
  All participants will get Error Tone and the system resource occupied by the conference will be freed.

Examples of Multiparty Conferences:

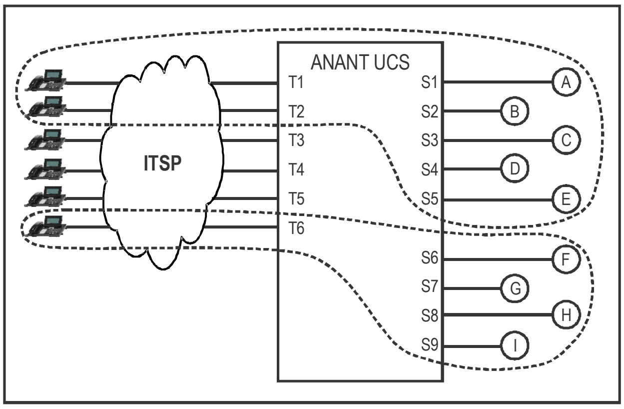
A Multiparty conference between extensions.



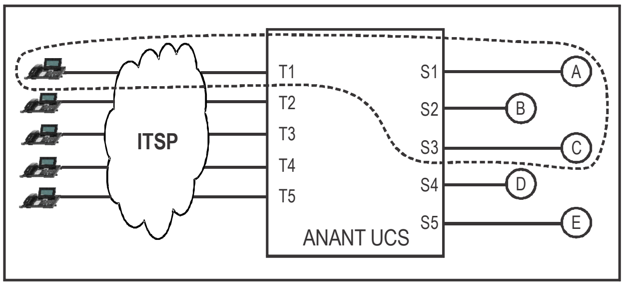
Simultaneous Multiparty conferences between extensions.



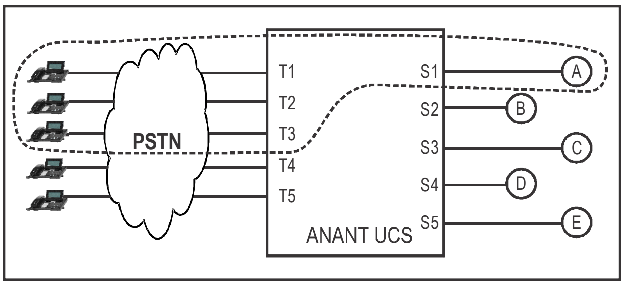
Multiple Multiparty conferences between trunks and extensions.



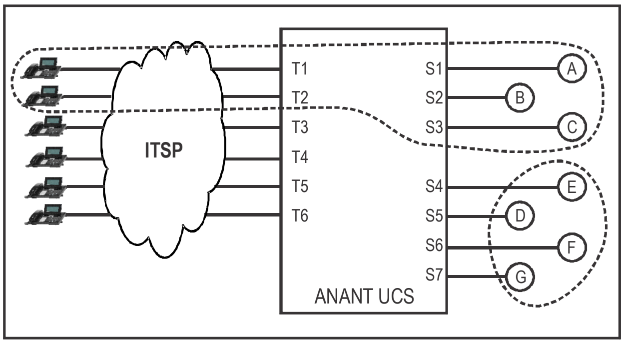
Multiparty conference between one trunks and few extensions.



Multiparty Conference between a few trunks and one extension.



Simultaneous Multiparty Conference between a few trunks and extensions and between extensions.



How to configure

To provide this feature to extensions,

- You must enable the feature 'Conference' in the "Class of Service (CoS)" of the extensions in their "Station Basic Feature Template". By default, this feature is enabled on all extensions, so all extensions can use this feature.

The feature 'Conference' in the Class of Service also includes 3-Party and Multi-party Conference. Extensions that are denied 'Conference' in their Class of Service will not be allowed Dial-In as well as 3-Party and Multi-party Conference.

For confidential conferences, you must:

- If desired, you may also change default value of the ***Release Conference if Idle for more than (min.) Timer***. See "System Timers and Counts".

- If external parties are to be allowed to initiate or join the Conference, "Direct Inward System Access (DISA)" must be enabled on the trunk on which they call.

- You can program a DSS key for Terminating a Conference, Temporarily Leave/Rejoining a Conference, if required. Refer the topic "DSS Keys Programming" for instructions.

How to use

For Extended IP Phone Users

To initiate multiparty conference:

- Dial the number of party 1.
- When you are in speech with party 1, press ‘Conference’ Key.
- Party 1 is put on hold and gets on-hold music.
- You get feature tone. Dial number of party 2.
- When you are in speech with party 2, press the ‘Conference’ Key.
- Party 2 is included in the conference. A 3-way speech is established.
- Press the ‘Conference’ Key. Dial the number of party 3.
- When in speech with party 3, press the ‘Conference’ Key.
- A Multiparty Conference is initiated. Press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option Include Party and dial the number.
- When you are in speech with the party, press the ‘Conference’ Key.
- Repeat the above steps to add new participants in the conference.

To include a party in a multiparty conference:

- After the Multiparty Conference has been initiated. Press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option Include Party and dial the number.
- When you are in speech with the party, press the ‘Conference’ Key.
- Repeat the above steps to add new participants in the conference.

To temporarily leave multiparty conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Leave Temporary’ and press the Enter Key.

  OR

- Press the DSS key assigned to Temporary Leave Conference/Rejoin Conference.

To rejoin multiparty conference:

- Press the ‘Conference’ Key.

  OR

- Press the DSS key assigned to ‘Temporary Leave’ Conference/Rejoin Conference.

To remove a party from multiparty conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Remove Party from Conf’
- The LCD displays the numbers of all the participants.
- Select the number of the participant you want to remove and press the Enter Key.

- The selected participant is disconnected.

To permanently leave from the multiparty conference:

- While in Conference, go ON-Hook.

To cancel multiparty conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Terminate Conference’ and press the Enter Key.

  OR

- Press the DSS key assigned to Terminate Conference.
- All the participants will get an Error Tone and the system resource occupied by the conference will be freed.

To use Multiparty Conference from DISA mode, you may see the instructions “Dial-In Conference using DISA” under “Conference Dial-In”.

Conference Dial-In

Dial-In Conference is a multi-party conference held at a pre-defined time. Extension users can schedule a Dial-In Conference and inform other participants to join in the conference at the scheduled time. Dial-In Conference can be used to conduct client meetings or sales presentations, project meetings and updates, regular team meetings, and to communicate with coworkers who operate in different locations. Thus, this feature helps to increase productivity by saving time and cost of travel for out-of-office meetings.

Refer the table below for details regarding the Dial-In conferences supported by the system.

Maximum conference participants	Maximum simultaneous conferences (If all the conferences involve 3 parties)	Maximum parties included in a conference (If all the parties are in a single conference)
1536	512	64

Conference is a licensed feature. Decide the number of conferences you want to conduct and purchase the license accordingly. Refer the topic *"Licenses Supported in ANANT UCS"* *to know more.*

How it works

- A Dial-In Conference can be scheduled by dialing the access code for Dial-In conference followed by the conference number and a password.
  - The **Conference Number** can be: **01** to **64**[93].

    The Conference Number must correspond with the number of simultaneous Dial-In Conferences supported by the ANANT UCS. If a user dials a conference number other than this, system will play an Error Tone.
  - The **Conference Password** is a four digit number string. The default conference password is **1111** and must be changed before using this feature.

    To avoid unauthorized access, make sure the password is strong and is provided to the participants only.
- All the other participants must be informed about the conference.

  You can also have external callers join the conference by providing them the DISA login.

Let us understand how Dial-In Conference works with the following example.

- Extension user A wants to schedule a Dial-In Conference at 4:30 p.m. with B, C, D, E and F.
- B and C are extension users. C is an extension user who has been provided a DISA login to access an extension of ANANT UCS.
- D, E and F are external parties.
- Any extension user can initiate the conference, in this case A initiates the conference.

93. *For SPARSH VP330 and VARTA Clients, the conference number is 01 to 15.*

Scheduling a Dial-In Conference

- A schedules a Dial-In Conference for 4.30 pm.
- A informs B, C, D, E and F about the conference and provides them conference number, for example, '1' and the password, '4040'.

If C wants to schedule a conference, C must log into his extension from DISA mode.

Initiating a Dial-In Conference

- Any extension user can initiate the Dial-In conference by dialing the feature access code for Dial-In conference followed by the two-digit conference number and the password, for instance: '01' and '4040'. In this case, A initiates the conference, the system will play beep.

After the Conference has been initiated, conference participants can:

- **Join a Dial-In Conference**.

  In this, example, B can join the conference by dialing the feature access code to join the conference followed by the two-digit conference number and password.

  C can join the conference from the DISA mode.

  When a new party joins the conference, the system plays beeps to the existing participants, to inform them of the new inclusion. Beeps are configurable (default: enabled).

- **Include a Party in an on-going Dial-In Conference**.

  Internal as well as external callers can be included in an ongoing conference by the any of the participants.

  To include external callers, in this case D, E and F, A must dial the Trunk Access Code followed by the their numbers.

- **Remove a Participant from an on-going Dial-In Conference**.

  Participants can remove another participant from the Dial-In conference. The IP Phone displays the numbers of all the participants, select the number of the participant to be removed from this list.

- **Temporarily Leave a Dial-In Conference**.

  Any participant in a conference can dial the Temporarily Leave Conference code to leave the conference for a short time period.

If all participants, Temporarily Leave the conference one-by-one, the system will start the 'Release Conference if idle for more than (Minutes) Timer'. This Timer is programmable, and by default it is set to 002 Minutes. If no participant rejoins the conference before the expiry of this Timer, the system will free the resource occupied by the conference on the conferencing circuit.

- **Rejoin a Dial-In Conference**.

  The participants who leave temporarily can rejoin the conference at a later stage by going Off-Hook and dialing the Rejoin Conference conference code.

- **Permanently Leave from a Dial-In Conference**.

  Participants in a conference can exclude themselves by going ON-Hook. Once any participant goes ON-Hook, he/she cannot rejoin the conference.

- **Cancel a Dial-In Conference**.

  Any participant in a conference can dial the Cancel conference code to end the conference.

  All participants will get Error Tone and the system resource occupied by the conference will be freed.

  The conference can also be canceled by logging into the SA mode.

How to configure

To provide this feature to extensions,

- You must enable the feature 'Conference' in the "Class of Service (CoS)" of the extensions in their "Station Basic Feature Template". By default, this feature is enabled on all extensions, so all extensions can use this feature.

The feature 'Conference' in the Class of Service also includes 3-Party and Multi-party Conference. Extensions that are denied 'Conference' in their Class of Service will not be allowed Dial-In as well as 3-Party and Multi-party Conference.

- If you want beeps to be played when any one joins the conference, enable ***Play Beep when Conference/ Dial-in Conference begins***. See "System Parameters", for instructions.

.

- If desired, you may also change default value of the ***Release Conference if Idle for more than (min.) Timer***. See "System Timers and Counts".

- If external parties are to be allowed to initiate or join the Conference, "Direct Inward System Access (DISA)" must be enabled on the trunk on which they call.

- You can program a DSS key for Dial-In Conference, Terminating a Conference, Temporarily Leave/ Rejoining a Conference, if required. Refer the topic "DSS Keys Programming" for instructions.

How to use

For Extended IP Phone Users

To Schedule a Dial-In Conference:

- Go OFF-Hook.
- Press the DSS key assigned for Dial-In Conference.
  OR
- Dial ***19**.
- The Dial-In Conference menu appears on the LCD.
- Select the option 'Schedule a Conf' and press the Enter Key.
- Enter Conference Number on the prompt.
- Enter Conference Password on the prompt.
- You get confirmation tone and the message 'Conf <number> Scheduled' on your phone's display.

- Go ON-Hook.
- Call all participants and inform them of the time of the Dial-In conference, the Conference Number and Password.

To join the Dial-In Conference:

- Go OFF-Hook.
- Press the DSS key assigned for Dial-In Conference.
  OR
- Dial ***19**.
- The Dial-In Conference menu appears on the LCD.
- Select the option ‘Include in Schd Conf’ and press the Enter Key.
- Enter the Conference Number on the prompt.
- Enter the Conference Password on the prompt. If you enter the wrong password, you get the message 'Check Conf P/w' on your phone's display.
- Conference is initiated.
- You will hear speech if any other participant has joined it. You will hear silence if no other participant has joined it.
- If you hear silence, wait for others to join in.
  OR
- Include the other participants in the conference, if you initiated the conference.

To include a party in the Dial-In conference:

- After initiating the conference and while you are all in speech, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Include Party in Conf’ and dial the desired number to be included in the conference. You get Ring Back Tone.
- When you are in speech with the party, press the ‘Conference’ Key The party is included in the conference.

To remove a participant from the Dial-In conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Remove Party from Conf’.
- The numbers of the participants appear on your phone's display.
- Scroll to select the participant you want to remove.
- Press 'Enter' key.
- You get confirmation tone. The extension user is now excluded from the conference.

To temporarily leave Dial-In conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Leave Temporary’ and press the Enter Key.
  OR
- Press the DSS key assigned to Temporary Leave Conference/Rejoin Conference.

To rejoin Dial-In conference:

- Press the ‘Conference’ Key.
  OR
- Press the DSS key assigned to ‘Temporary Leave /Rejoin Conference’.

If you have configured a DSS key for Temporary Leave and you leave the Conference by pressing the DSS key, to Rejoin the Conference press the DSS key again.

.

To permanently leave from the Dial-In conference:

While in Conference, go ON-Hook.

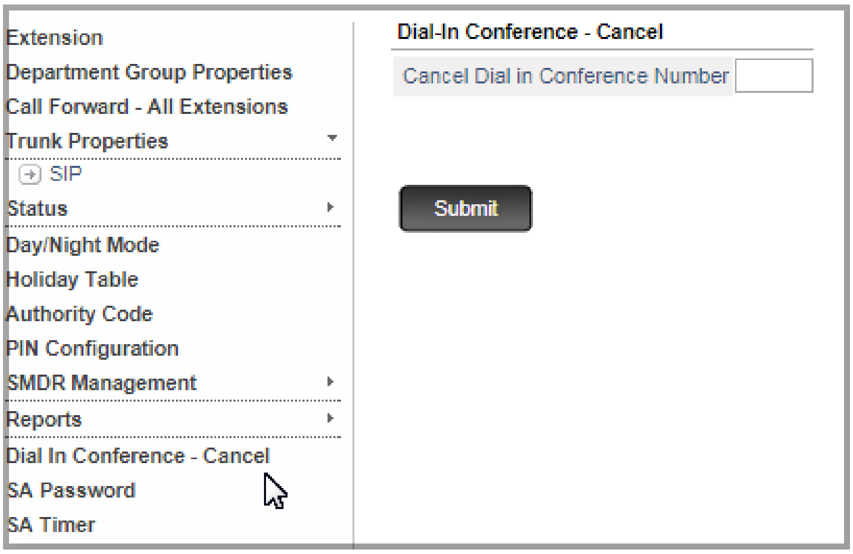
To cancel Dial-In conference:

- While in Conference, press the ‘Conference’ Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option ‘Terminate Conference’ and press the Enter Key.
  OR
- Press the DSS key assigned to Terminate Conference.
- All the participants will get an Error Tone and the system resource occupied by the conference will be freed.

You can cancel a Dial-In Conference from System Administrator (SA) Mode using Jeeves. To do this,

- Login as System Administrator.

- Click **Dial-In Conference - Cancel**.



- Enter the two-digit conference number (**01** to **64**[94]**)** which you want to cancel in the **Cancel Dial-In Conference Number** field.

- Click **Submit**.

94. *For SPARSH VP330 and VARTA Clients, the conference number is 01 to 15.*

Dial-In Conference using DISA

To schedule a Dial-In Conference:

- Dial a DISA enabled Trunk
- Dial DISA Login Code 1079-Extension Number-User Password, if PIN Authentication is required.

- After DISA Login beeps, dial *191-Conference Number-Conference Password.

To initiate or join a Dial-In Conference:

- After DISA Login beeps, dial *192-Conference Number-Conference Password.

To include a party midway of the Conference:

- Dial #2.
- You get dial tone.
- Dial the extension number to be included in the conference. You get Ring Back Tone.
- When you are in speech with the extension user, dial #2.
- You get feature tone, and the extension user is played music on hold.
- Dial *3. Both of you are now included in the conference.

To temporarily leave the Conference:

- Dial Flash-191.

To rejoin the Conference

- Go Off-Hook Dial 191.

To permanently leave the Conference:

- Dial #0 to go ON-Hook.
- Dial #0#9 to end DISA session.

- *When you enter DISA mode, you get beeps, dial digits before the DISA Inactivity Timer elapses.*
- *Never dial 'Flash' when in DISA mode, you will get disconnected.*
- *Keep dialing any digit to continue the conference.*
- *See "Direct Inward System Access (DISA)" to know more.*

Join Dial-In Conference using VMAA

Users can also Join a Dial-In Conference using the trunks on which the Voice Mail Auto Attendant is enabled. To know more refer to "Join Conference Dial-In using VMS". After joining the conference users can Temporary Leave, Rejoin or Permanently Leave the Conference also.

Conflict Dialing

You may recall that "Access Codes" are dialed at different call phases. No two Access Codes must be the same in the same call phase.

For example, the same access code cannot be used for two different features like Call Forward and Redial, since both these features are invoked in the 'Dial' phase. Similarly, Station (Extension) and Logical Group Codes too must be unique and should not match with any of the features invoked in the 'Dial' phase.

However, ANANT UCS allows overlaps within Feature Codes and "Flexible Numbers" (Station Codes). One Feature Access Code can be a part of (subset) another code, for example, 4, 41, 412; Flexible Numbers of extensions can be 201, 2011 etc.

So, when such overlapping access codes are dialed, the system matches the first digit. On finding more than one Access code starting with the same digit, the system will not know how to interpret the instruction and act accordingly.

Conflict Dialing feature resolves this confusion. When an access code that is a subset of any other access code is dialed, the system waits for some time for the extension user to dial the next digit. If the user does not dial any digit within that time, the system interprets it as the smaller Access Code, and invokes the associated feature.

The time for which the system waits for the next digit to be dialed before resolving the Access Codes is called "Conflict Dialing Timer". This timer is set to 2 seconds and is programmable.

Refer the topics "Access Codes" to know more.

How it works

You may set,

- The Access code of Call Pick Up as '4'.
- The Access code for Alarms as '41'.
- The Access code for Department Group 01 as '412'.

- Extension user A dials '4'.
- The system finds three access codes starting with '4' (4, 41, 412).
- So, it waits for 2 seconds, which is the default duration of the Conflict Dialing Timer, for the next digit to be dialed.
- If A does not dial any other digit before the Timer expires, the system interprets the code as '4' and invokes Call Pick-Up.

- If A dials '1' before the Timer elapses,
  - The system interprets it as '41'.
  - The system detects another access code starting with '41'.
  - So it waits for 2 seconds again for the next digit to be dialed.

- If A does not dial any other digit before the Timer elapses, the system interprets the code as '41' and invokes the Alarm feature.

- If A dials '2' before the Timer elapses,

- The system interprets the code as '412' and invokes Department Call to Group 01, provided there are no other access codes like 4121, 4123, etc.

- If such access codes exist, the system again waits for the duration of the Conflict Dialing Timer for another digit to be dialed.
- Thus, only when the conflict in the access codes is resolved will the system respond accordingly.

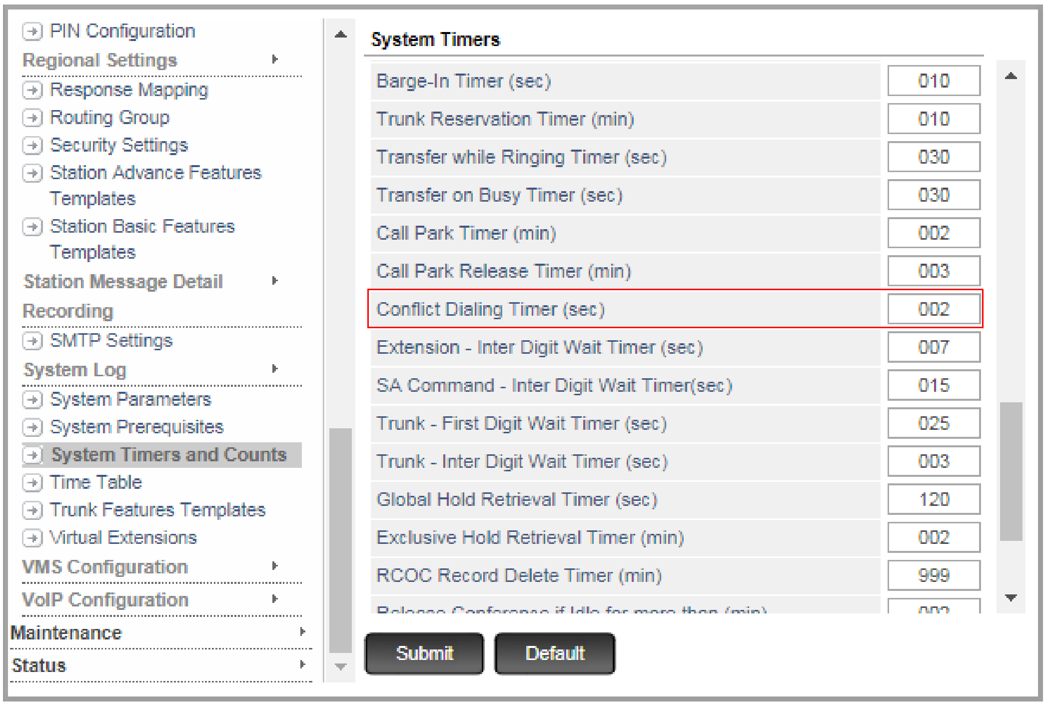
How to configure

The working of this feature is controlled by the Conflict Dialing Timer, which is set by default to 2 seconds and can be changed as desired.

If the duration of the Conflict Dialing Timer is long, it may cause delay in the system's response to the feature. If the duration is less, the system may misinterpret the access codes. Ensure that the value of the Timer is programmed optimally (that is, at least the default value).

Changing Conflict Dialing Wait Timer

- Login as System Engineer.
- Under **Configuration**, click **System Timers and Counts**.
- Scroll to **Other Features** and set the time in seconds as desired for the **Conflict Dialing Timer**.



- Click **Submit.**

Conversation Recording

Conversation Recording allows extension users to record their talk with other extension users or external parties, after or without informing the opposite party.

This feature can be used to record verbal agreements, important discussions, instructions, interviews, client requirements, take or place orders, etc.

Extensions must have a mailbox assigned to them for recording conversations. This feature will work only if a VMS channel is available.

ANANT UCS supports recording of 64 calls simultaneously.

- *Use this feature in accordance with the local laws.*
- *Matrix Comsec is not responsible for any mis-/abuse of this feature by users.*

- *On SIP extensions, ANANT UCS supports Conversation Recording using INFO Message. For a list of IP Phones on which this feature has been tested, see* *"ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.*

- *It is not possible to pause the Conversation Recording in SIP Extension. So, when SIP Extension puts the call on hold and then retrieves the call then for the hold duration, silence will be recorded in the recorded file.*

How it works

A and B are extensions. Both are assigned a mailbox each.
C and D are external parties.

- A calls C.
- C answers the call.
- A presses the Transfer Key.
- C is put on Consultation Hold.
- A dials the command for Conversation Recording.
- The system sends a string of digits to the Voice Mail System to initiate Conversation Recording.
- A and C are in speech again.
- The conversation recording starts in A's mailbox. The system plays beeps, if Conversation Recording Beeps are enabled.
- A or C disconnects the call.
- Conversation recording ends.
- A can listen to the recorded conversation by invoking the Voice mail feature.

The same is repeated when B calls A. As both have mailboxes assigned, both can record the conversation.

How to configure

The functioning of this feature is controlled by three parameters: 'Class of Service', 'Mailbox', and 'Conversation Recording Beeps'. These parameters can be configured using Jeeves.

Conversation Recording in Class of Service

Conversation Recording must be enabled in the CoS group of the extensions to which this feature is to be allowed.

In the default, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS, CoS group 01 is assigned as default. Conversation Recording is disabled in CoS group 01. So, none of the extensions of the ANANT UCS can record conversations.

Decide which extensions should be allowed Conversation Recording.

If you want to allow this feature to all extensions, simply enable Conversation Recording in the default CoS group 01.

If you want to allow Conversation Recording to selected extensions,

1. Define a CoS group with Conversation Recording enabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".
3. Assign this new Template to the extensions to which Conversation Recording is to be allowed.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for detailed instructions and configuration.

Mailbox

Extensions that are to be allowed Conversation Recording must also have a mailbox. Refer, "Viewing SIP Extension Status" for more information and configuring instructions.

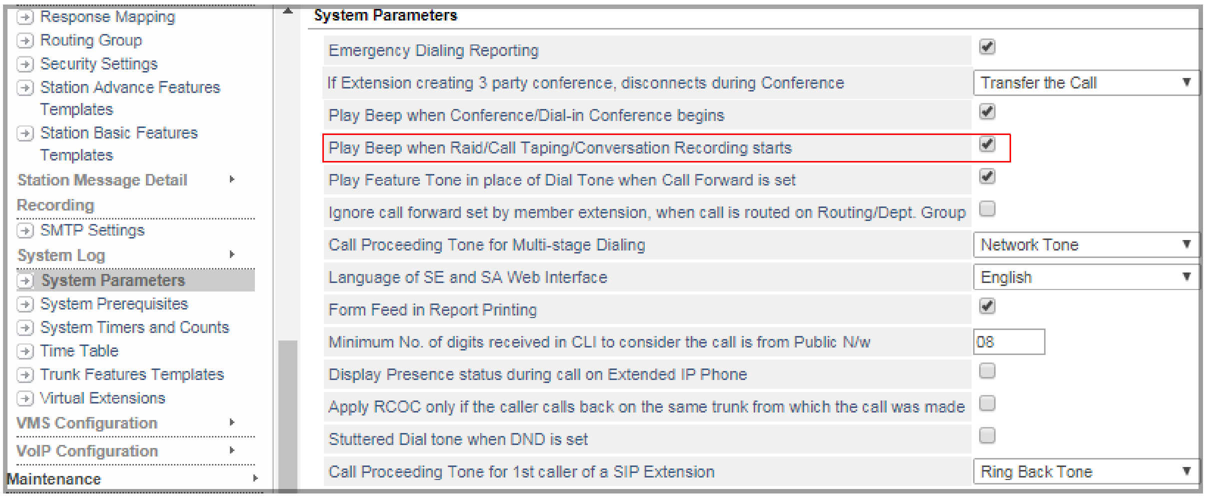
Conversation Recording Beeps

Decide whether you want Beeps to be played during Conversation Recording. Follow the instructions given below.

Configuring Conversation Recording Beeps

- Login as System Engineer.

- Under **Configuration**, click **System Parameters.**



- Go to **Play Beep when Raid/Call Taping/Conversation Recording starts**. Click the check box to enable or clear the check box to disable this feature.

- Click **Submit.**

How to use

For Extended IP Phone Users

To record a conversation:

- You are in speech with another extension/external number.
- Press DSS Key assigned to Conversation Recording (if programmed).
  OR
  Press Transfer Key and Dial **1095.**

- You get beeps (if enabled).
- Speech with party re-established.
- Recording starts.
- Go ON-Hook, after conversation ends.

To listen to a recorded conversation:

- Press 'Voice Mail' Key
  OR
  Dial **3931**[95]
- Follow Voice Mail Prompts.
- Go Idle or you get dial tone after 3 seconds.

Conversations are recorded as New Messages. So, follow the voice mail prompts for listening to new messages.

95. *This is the default Voice Mail Feature Access Code. Verify with your System Engineer if this has been changed and use the new code.*

COSEC Integration

ANANT UCS supports integration with COSEC to support Matrix COSEC Door Controllers. However, COSEC Integration can also be used in VIP Apartments and Villa's, wherein ANANT is installed.

With this integration, the users of ANANT UCS can unlock the COSEC Door Controller using their Extended IP Phone.

Matrix COSEC is an enterprise-grade people mobility management solution for organizations covering Time-Attendance, Access Control, Visitor Management, Employee Self Service Portal, Roster Management, Contract Workers Management and Cafeteria Management. To know more about Matrix COSEC, refer to COSEC documentation or visit our website: www.MatrixComSec.com.

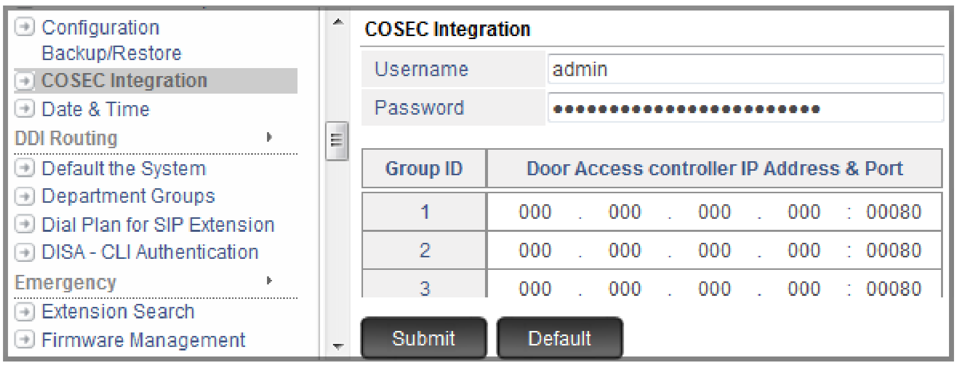
How it works

- Group together extensions of ANANT UCS that need to access the same COSEC Door Controller. Assign a Group ID to each group. You can create 50 groups. Each group must have a unique Group ID.
- In ANANT UCS, for integration a User Name and Password is configured. Make sure the same User Name and Password is configured in each COSEC Door Controller you have installed. You can integrate 50 such Door Controllers with ANANT UCS.
- Each COSEC Door Controller has a unique IP Address and port.
- Each Group ID is mapped to the IP Address and Port of the COSEC Door Controller. The extensions in the same Group ID can only unlock the COSEC Door mapped with their Group ID.
- The extension users can open the door by dialing *7 (access code to unlock the COSEC Door, programmable, see "Access Codes")or by pressing the DSS key assigned to COSEC Door Open.

*Standard SIP Phone users can put a call on hold by dialing #2. These users can also unlock the COSEC Door by dialing *7.*

How to configure

- Login as System Engineer.
- Under **Configuration**, click **COSEC Integration**.



- For COSEC Integration, enter the **User Name** and **Password**. Default User Name: admin, Password: 1234.

  The User Name can be a maximum of 24 characters. Valid characters: 0 - 9, a - z, A - Z.

  The Password can be a maximum of 24 characters.Valid characters are 0 - 9, a - z, A - Z, !,@,*,(,),-,.+,/ and comma

  Make sure the same User Name and Password is configured in the COSEC Door Controllers you have installed.

- Against each **Group ID**, enter the **COSEC Door Controller IP Address** and **Port**.

  The IP Address can be a maximum of 15 characters (only IPv4 Addresses are supported).

  The Port can be a maximum of 5 digits. Valid range:1025 - 65535 or 80.

The same COSEC Door Controller IP Address and Port cannot be assigned to different Group ID's.

- Click **Submit.**

To assign Group IDs to SIP users, see parameter COSEC Door Group in *"Configuring Matrix SPARSH VP248"*, *"Configuring Matrix SPARSH VP310"*, *"Configuring Matrix SPARSH VP330"*, *"Configuring Matrix VARTA ADR100/ AMP100 UC Clients"*, *"Configuring Matrix SPARSH VP510"*, *"Configuring Matrix SPARSH VP210"*, *"Configuring Matrix Extended SPARSH VP710"*, *"Configuring Matrix VARTA WIN200 UC Client"* and *"Configuring Standard SIP Phones"*.

To assign a DSS key to COSEC Door Open, see *"DSS Keys Programming"*.

Customer Name

Customer Name is the name of the organization/enterprise that has deployed ANANT UCS. As an user, you can enter the name of your company/organization in the system.

When Customer Name is assigned in the system, this name will appear as header on the various System Reports generated and printed by ANANT UCS like SMDR Incoming, Outgoing and Internal Call Reports, Alarm Status reports, etc.

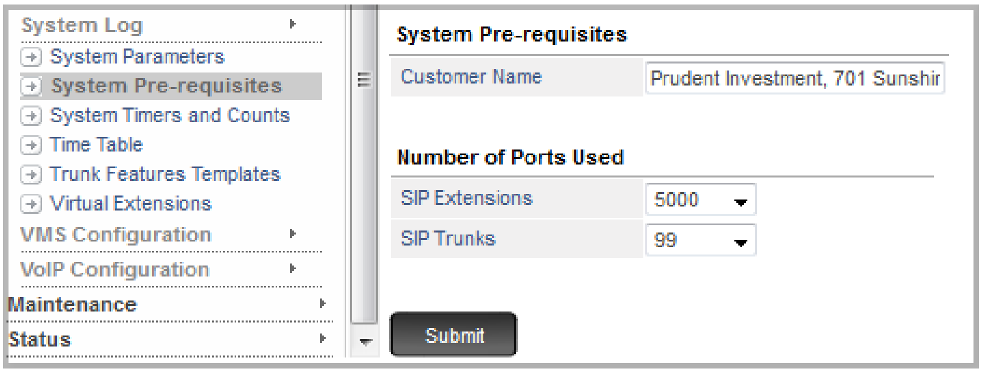
The Customer Name may consist of a maximum of 80 alphanumeric characters, including punctuation marks. So, you can enter the organization's address along with the Customer Name.

How to configure

Customer Name can be configured using Jeeves at the time of installation, or any time thereafter. It can also be corrected or changed any time.

Configuring Customer Name

- Login as System Engineer.
- Under **Configuration**, click **System Prerequisites.**



- In **Customer Name,** enter the name (and address, if desired) of the organization/enterprise. For example: Prudent Investment, 701 Sunshine Boulevard, Bannerghatta, Bangalore.
- Click **Submit.**

Day Night Mode

Certain features of ANANT UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Bundle Access Groups, Trunk Landing Group, Auto Attendant, Direct Inward System Access (DISA) etc require extensions and trunks to behave differently according to the working hours, non-working hours and break hours, which are referred to as Time Zones.

These Time Zone-dependent features and facilities are operated automatically according to the Time Tables configured in the system. In a Time Table, the Time Zones - Working Hours, Non-Working Hours, Break Hours - are defined for the entire week. Time Table is assigned to trunks, extensions and other time-zone dependent features. The system executes the Time-Zone dependent features and facilities automatically according to the Time Table.

To know more refer the topic “Time Tables”.

Day/Night Mode allows you to change the Time Zone of the system at any point in time, by using Jeeves or by pressing the DSS key on the phone. For example, the office is to be closed on account of an unplanned holiday or emergency. So, the Time Zones of all extensions and trunks must be set to Non-working hours to route outgoing calls and land incoming calls from/to the appropriate destination. You can set ANANT UCS to Night Mode until the office remains closed and set it back to operate as per the Time Table, when work is resumed.

To cite another example, the office must work for extended hours. You can set ANANT UCS to Day Mode and set it back to operate as per the Time Table.

When you set the system in Day/Night Mode, the system overrides the Time Tables assigned to Trunks, Extensions and Operator. According to the mode you selected, it applies Working Hours/Non-Working Hours to run all the Time-Zone dependent features of the system.

When the system is set to Day Mode, it applies Working Hours as the Time Zone for all extensions, trunks and time zone dependent features and facilities. When the system is set to Night Mode, it applies Non-Working Hours as the Time Zone on Time-Zone dependent features of the system.

Thus, Day/Night Mode forces the system to work in a particular Time Zone, until it is changed again, manually.

Day/Night mode can be set by the System Engineer (SE mode) as well as by the System Administrator (SA Mode).

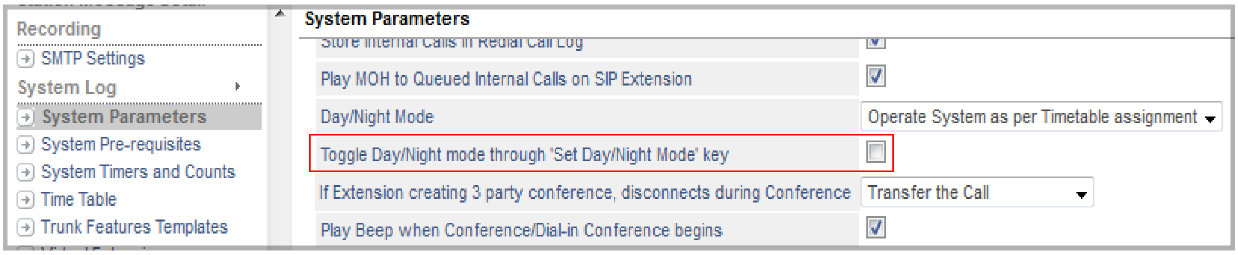
How to configure

Setting Day/Night Mode from SE mode

- Login as System Engineer.
- Under **Configuration**, click **System Parameters**.

- Go to **Day/ Night Mode**. Select the desired option:

- Set System in Day Mode (Working Hrs)
- Set System in Night Mode (Non-Working Hrs)
- Set System in Break Mode
- Operate System as per Timetable assignment (default).



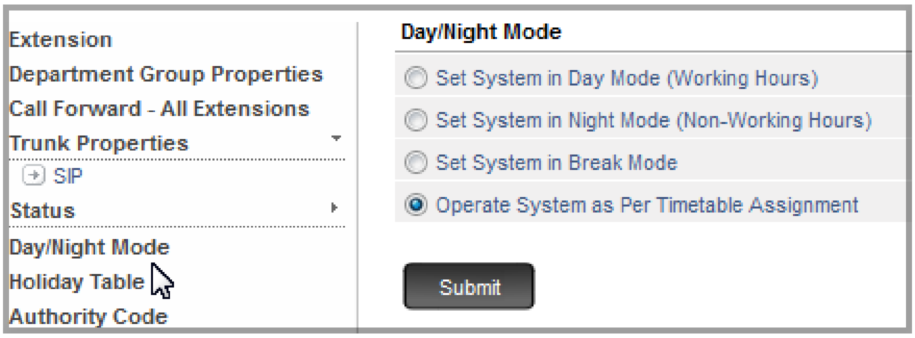
- Select the **Toggle Day/Night mode through ‘Set Day/Night Mode’ key** check box to switch to Day Mode (Working Hours) or Night Mode (Non-Working Hours) on pressing the DSS Key. Default: Disabled.

- Click **Submit.**

Setting Day/Night Mode from SA mode

- Login as System Administrator.

- Click **Day/Night Mode.**



- Select the desired option:

  - Set System in Day Mode (Working Hours)
  - Set System in Night Mode (Non-Working Hours)
  - Set System in Break Mode
  - Operate System as Per Timetable Assignment (default)

- Click **Submit.**

Setting Day/Night Mode using DSS key

You cannot switch to Break Hours or As per Timetable assignment using the DSS key. This can be achieved only by logging in through Jeeves.

How to configure

- Login as System Engineer.
- Assign a DSS Key for **Day/Night Mode** to the required extension. To know more about assigning a DSS Key for a specific feature, refer to “DSS Keys Programming”.
- To enable the toggle functionality of the DSS Key, select the **Toggle Day/Night mode through ‘Set Day/ Night Mode Key’** check box. To know more, refer to “Setting Day/Night Mode from SE mode”.

How to use

- Press DSS Key assigned to Day/Night Mode.

  When you press the DSS Key of extension, the system overrides the Time Table assigned to that extension. According to the current Day/Night Mode, it switches to Working Hours/Non-Working Hours.

  The following table displays the LED indication of DSS key with Toggle functionality disabled.

Model	**Event**	**Color**	**Cadence**
SPARSH VP248/ SPARSH VP310/ SPARSH VP510	Day Mode Set	Blue	Continuous ON
	Night Mode Set	Red	Continuous ON
	Break Hours Mode	Violet	Continuous ON
	System set to work as per Time Table	--	OFF

  The following table displays the LED indication of DSS key with Toggle functionality enabled.

Model	**Present Mode**	**Next mode on pressing Day/Night Mode DSS Key**	**LED Color after toggle**	**Cadence**
SPARSH VP248/ SPARSH VP310/ SPARSH VP510	As per Time Table (Day Mode)	Night Mode	Red	Continuous ON
	As per Time Table (Night Mode)	Day Mode	Blue	Continuous ON
	As per Time Table (Break Hour Mode)	Day Mode	Blue	Continuous ON
	Day Mode	Night Mode	Red	Continuous ON
	Night Mode	Day Mode	Blue	Continuous ON
	Break Hour Mode	Day Mode	Blue	Continuous ON

Daylight Saving Time (DST)

Daylight Saving Time (DST) is the practice of advancing clocks so that afternoons have more daylight and mornings have less. Typically clocks are adjusted forward one hour near the start of spring and are adjusted backward in autumn.

Many countries of the world use DST, though the start and end dates of DST vary with location and year. Even within countries, uniform DST may not be observed. For example the states of Arizona and Hawaii do not observe DST. Certain countries may observe DST in certain years, for instance Guatemala, while in most countries of Asia and Africa, and in certain countries of South America, DST is not observed at all.

When ANANT UCS is installed in a country/region where DST is used, it is necessary to synchronize the Real Time Clock of ANANT UCS with the local time.

So, if you are installing ANANT UCS in a country where DST is used, find out the DST convention currently in use in that country, and adjust DST accordingly.

How it works

The forward and backward adjustment of clocks can be Scheduled or Manual.

- **Scheduled DST Adjustment:** The Real Time Clock of ANANT UCS is advanced and set backward automatically according to the DST convention of the country/region where ANANT UCS is installed.

  Scheduled DST Adjustment is useful in countries/regions where DST Time is fixed, such as in Europe, USA and Canada, without yearly variations.

  The table below describes the DST conventions followed in the different countries for which ANANT UCS will automatically adjust DST.

  ANANT UCS supports 18 DST Types for Scheduled DST Adjustment.

DST Type	DST Timings		Applicable in Countries
	Start Time	**End Time**	
01	Last Sun MAR From 01:59 to 03:00	Last Sun OCT From 02:59 to 02:00	Austria, Poland, Russia, Spain
02	Last Sun OCT From 01:59 to 03:00	Last Sun MAR From 02:59 to 02:00	Australia, Australia-Tasmania, Belgium, France, Germany, Greece, Hungary, Italy, Sweden, Switzerland
03	Second Sunday MAR From 01:59 to 03:00	First Sunday NOV From 01:59 to 01:00	Bahrain, Mexico, Turkey, United States
04	First Sun NOV From 23:59 to 01:00	Third Sun FEB From 23:59 to 23:00	Brazil
05	Second Sunday MAR From 01:59 to 03:00	First Sunday NOV From 01:59 to 01:00	Canada
06	Second Sat OCT From 23:59 to 01:00	Second Sat MAR From 23:59 to 23:00	Chile

DST Type	DST Timings		Applicable in Countries
	Start Time	**End Time**	
07	Last Sun MAR 00:59 02:00	Last Sun OCT 01:59 01:00	Denmark, Ireland, Portugal, United Kingdom
08	Last Sun MAR 02:59 04:00	Last Sun OCT 03:59 03:00	Finland
09	First APR 02:59 04:00	First OCT 03:59 03:00	Iraq
10	Last Sun MAR 02:29 03:30	Last Sun OCT 02:29 01:30	Kyrgyzstan
11	Last Fri APRIL 23:59 01:00	Last Thu SEP 23:59 23:00	Egypt
12	Last Sun MAR 23:59 01:00	Last Sun OCT 23:59 23:00	Lebanon
13	First Sun SEP 01:59 03:00	First Sun APRIL 01:59 01:00	Namibia
14	Last Sun SEP 01:59 03:00	First Sun APR 02:59 02:00	New Zealand
15	Last Sun MAR 01:59 03:00	Last Sun OCT 02:59 02:00	Norway
16	First Sun OCT 23:59 01:00	First Sun APRIL 23:59 23:00	Paraguay
17	First APRIL 23:59 01:00	First OCT 23:59 23:00	Syria
18	First APRIL 23:59 01:00	Last Sun OCT 23:59 23:00	Cuba

The DST Type is to be selected according to the country/region where the system is installed.

When DST Mode is set to 'Scheduled' and the DST Type is selected, the system will automatically adjust DST at the preset dates and time for the country/region where the system is installed.

For example, if ANANT UCS is installed Spain, the DST Type 01 applicable to this country should be programmed as Scheduled DST. The system will automatically advance the clock on the last Sunday of March at 01.59 to 03:00 am every year (the start date of DST) and set the clock backward on the last Sunday of October at 02.59 to 02:00 am of the same year.

- **Manual DST Adjustment:** The Real Time Clock of ANANT UCS is advanced manually and set backward automatically according to the DST convention of the country/region where ANANT UCS is installed.

  Manual DST Adjustment is to be used in regions/countries that have no fixed DST Convention and where yearly variations in DST practices are likely.

  When DST Mode is set as 'Manual', you must set the start and the end time, that is, the time at which the clock is to be advanced and the time at which the clock is to be delayed.

There are two ways to adjust DST manually:

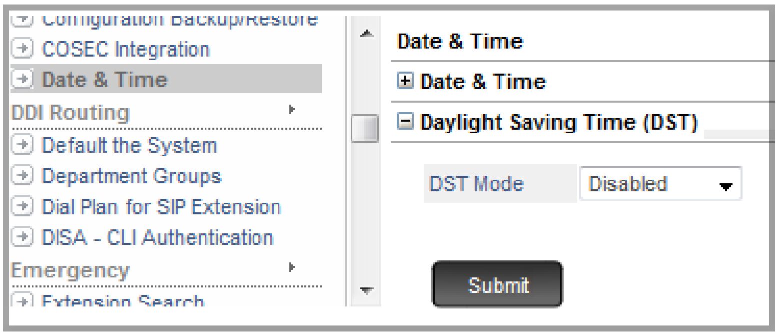
1. The 'Day of Month' method, which specifies a day of the month DST will start or end. For example: starting on the 2nd Sunday of March and ending on 1st Sunday of November.
2. The 'Date and Month' method, which specifies a date of the month that DST will start or end. For example: starting on March 11 and ending on November 4.

DST is not applicable in certain regions/countries, like Asia and South America. In such cases, the DST Mode is to be 'Disabled'.

How to configure

Adjusting DST

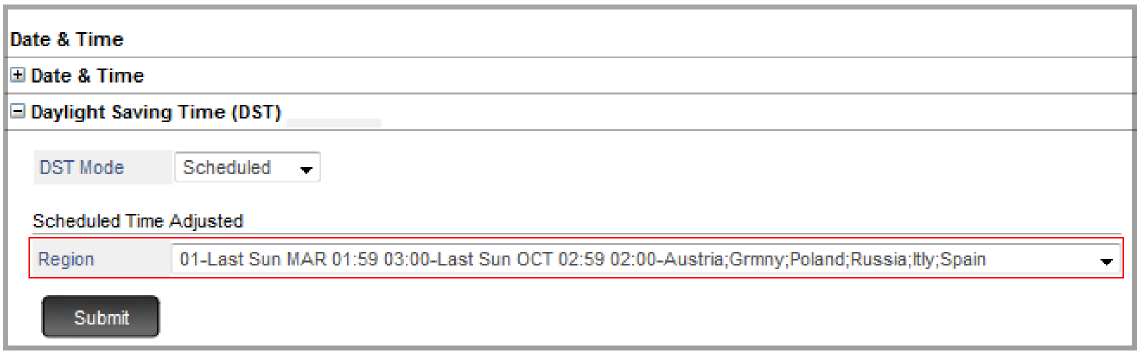
- Login as System Engineer.
- Under **Configuration**, click **Date and Time**.
- Click **Daylight Saving Time (DST)** to expand.



- Set the **DST Mode** to **Manual** or **Scheduled** as per your requirement.

Scheduled DST Adjustment

- If you have selected **Scheduled** as DST mode, in the **Region** list, select the name of the country/region where your system is installed.



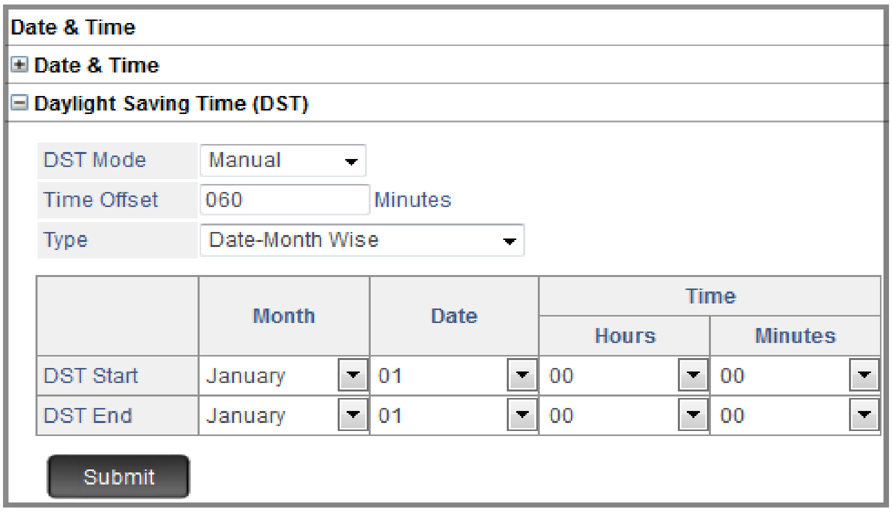
- Click **Submit.**
- If you do not find your region on this list, you are recommended to set DST Mode to 'Manual' and adjust DST manually.

Manual DST Adjustment

- If you have selected **Manual** as DST Mode.
- In **Time Offset,** enter the time you wish to forward the DST start time with.
- In **Type,** select the desired option:
  - **Date-Month Wise** to specify the date of the month DST will start.
    OR
  - **Day-Month Wise** to specify the day of the month DST will start.

Date-Month Wise

- If 'Date-Month Wise' is selected in **Type**, you should now select the desired options in each of the following to specify the **DST Start** details,
  - **Month:** Select the month when DST begins (January-December).
  - **Date:** The date on which DST begins (1-31).
  - **Time:** Select the time when DST will begin to change.The time mode is 24 hours, with options from 00 to 23 hours and 00 to 59 minutes.



  - Similarly, in the **DST End** configure the desired DST End Time details.
- Click **Submit.**

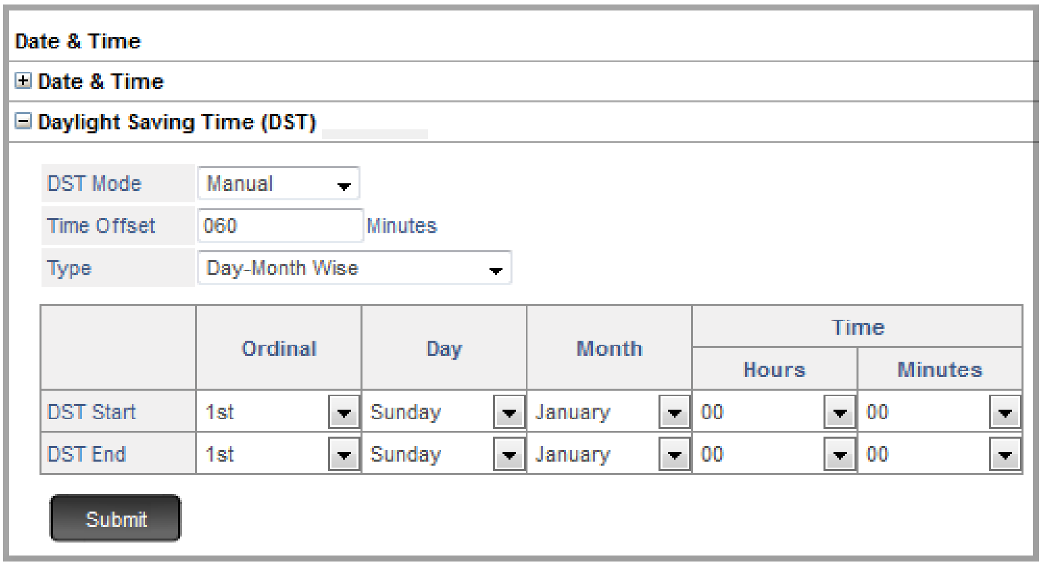
Day-Month Wise

- If 'Day-Month Wise' is selected in **Type**, you should now select the desired options in each of the following to specify the **DST Start** details,
  - **Ordinal number:** Select the Ordinal number of the day of the month, that is, the 1st, 2nd, 3rd, 4th, 5th day, when DST begins in the DST Start.

- **Day:** Select the day of the month - Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday - when DST begins.

- **Month:** Select the month when DST beings (January-December).

- **Time:** Select the time when DST will begin to change.The time mode is 24 hours, with options from 00 to 23 hours and 00 to 59 minutes.



Similarly, in **DST End** configure the DST End Time details.

Once the DST Ends, the time of the system is set back to the Standard time automatically.

- *When the DST of a particular country starts or ends on the Last Sunday or any other day, for example, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.*

- *Wherever time adjustments are made at 00:00 hours, use the previous date and set DST start time (that is, "from" time) at 23:59 hrs.*

- *If you synchronize the RTC with the SNTP Server and the Date and Time changes, the DST will be applicable as per the new Date and Time.*

Department Call

Department Call enables you to group together extensions of a particular department so that callers can reach anyone in the department by dialing a common access code assigned to the department.

Calls made to such groups of extensions are called Department Calls and the access code used to make department calls is called Department Number.

This feature is useful in situations where any member of a department may interact with callers, as for instance in a information counter, a customer care cell, a technical support team, etc.

Callers can also reach individual extensions in a Department group by dialing the extension number.

ANANT UCS supports the formation of 32 department groups. The *member* extensions of a department group may be SIP Extensions, Virtual Extensions, Outgoing Trunk Bundle Group, Voice Mail Auto Attendant.

Each Department Group can also be assigned a mailbox for voice mail, which any member extension can access.

Each Department Group can forward its calls to an extension or to its voice mail, or to another Department Group.

How it works

Extensions A, B, C, D are grouped as a Department with the access code 3901.

Internal Calls

- Extension E dials 3901 to call the Department.
- The system checks E's Class of Service for the Department Call feature.
- The feature is enabled. The system checks if Rotation is enabled in the routing group assigned to the Department.
- The Rotation check box is enabled. The system lands the call on the extension which is set to ring first.
- Extension A, configured as the first landing destination rings for the duration of the Ring Timer (configurable; default:15 seconds).
- A answers the call. Speech is established between A and E.

- If A does not answer, the system hunts for the next extension in the group to land the call, say B.
- B starts ringing for the duration of the Ring Timer.
- If Continuous Ring is enabled on A, A will continue to ring even as B is ringing.
- If B does not answer the call at the end of the timer, the system hunts for the next extension, C.
- If B has Continuous Ring enabled, B will continue to ring even as C is ringing.
- If the call is not answered even after hunting the last extension, the system will loop back and start from the first extension once again.

External Calls

Department Calls can be made using Voice Mail Auto Attendant. For example, a company may use the Voice Mail Auto Attendant to have callers who want information only to dial the Information Department instead of waiting for the Operator.

- An external caller places a call to Department 3901 using the Voice Mail Auto Attendant.
- The system checks if Rotation is enabled in the routing group assigned to the Department.
- As the Rotation check box is enabled, and the first call landed on A, the system lands this call on the next extension B.
- Extension B rings for the duration of the Ring Timer (configurable; default: 15 seconds). If the Continuous Ring check box is enabled for B, it will continue to ring, even as the system hunts for another extension in the group to land the call.

- A third call internal/external made to Department 3901.
- The same process as described above will be repeated.
- But the system will land the call on extension C first, because Rotation check box is enabled on this routing group.
- The subsequent incoming calls will land on the extension which is next to the one that received the last call. So the next call to the Department will land on extension D, the one thereafter on A, and so forth.

Thus for each call, the system will hunt for a landing extension as per the Rotation set for the routing group. The extensions will ring for the duration of the Ring Timer, either continuously or one-by-one (as per the Continuous check box configured), and according to the sequence in which the extensions in the group are arranged.

Rotation ensures equal distribution of call traffic. If Rotation is disabled, the fresh call will always land on first extension of the Department group.

Voice Mail

A Department Group can be assigned a common mailbox for Voice Mail, called the *Department Group Mailbox*. This common mailbox for the group is called Department Group Mailbox. You can assign Department Group Mailbox to selected extensions or to *all* extensions in the Department.

To take the example of Extensions A, B, C, D with the Department Access Code 3901 further,

- Extensions A, B, C and D are all members of Department Group 1 with the Access Code 3901.

- Department Group Mailbox is assigned to all the four extensions.

- When there is a new message in the Group Mailbox, all four extensions - A, B, C, D - will get the Message Wait Notification.

- The message wait indication may be a Stuttered Dial Tone or Silence when the extension user goes OFF-Hook[96].

- To the first extension that answers the notification call, for example, Extension A, the Voice Mail System informs about the new message(s) waiting in the Department Group Mailbox and in the Personal Mailbox. *"You have <x> new Messages in your Personal Mail Box. You have <y> new Messages in your Department Group Mail Box".*

96. *This will depend on the type of Message Wait Indication configured for the Extension's Voice Mail Settings.*

- If there is no new message in both mailboxes, the VMS will play the message: *"You have Zero new Message"*

- If there is a new message in the Department Group Mailbox, but none in the Personal Mailbox, the VMS will play the message *"You have <x> new Message in your Department Group Mailbox"*

- If there is no new message in the Department Group Mailbox, but new message in the Personal Mailbox, the VMS will play the message: *"You have <x> new Message in your Personal Mailbox"*

- The VMS prompts Extension A to access the Group mailbox: *"To go to Personal Mailbox, press 1. To go to Department Group Mailbox, press 2."*

- The user of Extension A presses 2, and is taken to the Department Group Mailbox,

- VMS prompts A: "Enter your mailbox password". Enter your department group mailbox password.

- The VMS checks the utilized mailbox memory,

  - if 80% of the mailbox memory has been consumed, the VMS prompts the caller: "Your Mailbox is 80% Full. Please Delete few messages.".

  - if 100% of the mailbox memory has been consumed, the VMS prompts the caller."Your Mailbox is Full. Please Delete few messages."

- VMS prompts "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."

- Extension A presses 1.

- VMS plays the new messages.

- After playing the new messages, the VMS cancels Message Wait Notification set for extensions B, C and D.

Call Forward

Just as calls can be forwarded to a Department Group, a Department Group can also forward its calls to:
- an extension
- its own Department Group Mailbox
- another Department Group

For Department Groups, ANANT UCS does not support Call Forward to an external destination number.

You can set Call Forward for Department Group from the SA Mode only.

ANANT UCS supports the following Call Forward options for Department Groups:

- **Call Forward - unconditionally:** calls are forwarded to the destination number, without checking the status or waiting for a response from the Department Group.

- **Call Forward- if Busy:** calls are placed on the Department Group as per the Rotation configured for it and are forwarded to the set destination, only when all the member extensions of the Department Group are found to be busy.

- **Call Forward- if No Reply:** when a call is made to the Department group, ANANT UCS will place the call as per the Rotation configured for the Department Group for the duration of the '*Call Forward No Reply Timer for Department*' (default: 30sec). If none of the member extensions answers the call before the expiry of this timer, the call is forwarded to the destination.

  If you select this option, you may set the *Call Forward No Reply Timer for Department* to the desired value. This Timer is commonly applied on all Department Groups which set Call Forward No Reply.

- **Call Forward - if Busy/No Reply:** calls made to the Department Group will be routed to the destination, if all members of the Department Group are busy or when none of the member extensions answered the call within the Call Forward No Reply Timer for the Department.

- *Call Forward - Dual Ring is not supported for Department Groups.*

- *Member extensions of a Department Group can set Call Forward on their extensions. However, Call Forward set for the Department Group will have precedence over Call Forward set by individual member extensions.*

- *Call Forward set by member extensions in a routing group will be ignored by the system if, the Ignore call forward set by member extension, when call is routed on Routing/Dept. Group check box is enabled.* *See "System Parameters" for more information.*

- *Call Forward for a Department Group can also be set from SA Mode.* *See "Setting Call Forward for Department Group".*

Again, taking the above example of Department Group 1 further, here's how call forward will work:

- Extensions A, B, C, D of Department Group 1 are allowed Call Forward Department Group in their Class of Service.

- Any of them can set Call Forward for Department Group 1. Extensions A, B, C and D can also set Call Forward on their own extensions.

- When any extension or an external caller (also using Auto Attendant or Direct Inward System Access) dials the Access Code 3901 to call Department Group1, ANANT UCS will check the Call Forward option set for the Department Group and route the call accordingly.

- If Call Forward - unconditionally is set, the call will be routed to the destination number, if the *Ignore call forward set by member extension, when call is routed on Routing/Dept. Group is enabled*.

- If Call Forward - Busy is set, and the first extension in the Department Group is busy, the system will hunt for the next free extension in the group. It will continue to hunt for a free extension. If all extensions in the group are busy, the call will be forwarded to the destination number.

  Call Forward unconditional, busy, or busy/No reply set by any member extension will not work.

- If Call Forward - No Reply is set, the system will start the Call Forward No Reply Timer Department Group and place the call as per the Rotation set for the Department Group. If the call is not answered by any of the extensions before the timer expires, the call will be forwarded to the destination number.

  If a member extension that is offered the call has set Call Forward-Unconditional, and the Call Forward No Reply Timer Department Group has not expired, the call forward set by the extension will be applied. If the timer expires, the Call Forward No Reply set for the Department Group will be applied.

If a member extension that is offered the call has set Call Forward-No Reply, or No-Reply/Busy, the Call Forward No Reply Timer (for individual extension) will start simultaneously with the Call Forward No-Reply Timer Department Group. If the No Reply Timer for the extension expires first, the call will be forwarded to the destination set for the extension. If the No Reply Timer of the Department Group expires first, before the call is answered, the call will be forwarded to the destination set for the Department Group.

Call Forward-No Reply Timer can be set from the SE Mode only.

How to configure

The functioning of this feature requires you to do the following:
- create Department Groups.
- configure "Routing Group" (each routing group consisting of extensions related to a Department) and assign the routing groups and appropriate access codes to the department groups.

If you want to provide voice mail facility to the Department Group, you must:
- assign a Mailbox to the Department Group.
- allow member extensions access to the Department Group Mailbox.

If you want to enable Call Forward to the Department Group, you must:
- enable 'Department Group Call Forward' in the Class of Service (CoS) of the member extensions.
- change, if required the default value of the *Call Forward No Reply Timer for Department Group*.

Creating Department Groups

- On a sheet of paper, draw a table.
- Decide the number of department groups you want to create, e.g.: 4 groups.
- Group all the extensions you want to put in each department group. You cannot group more than 32 extensions in a single department group.
- Decide in what sequence the extensions in each group should ring, that is, which extensions should ring first, second, third, and so forth.
- Decide the access code you want to assign to each department group.
- You may also assign a name to the department group.
- Your table may look like this:

Department Group Index	Access Code to be assigned	Name	Extensions to be included as members
			SIP Extension
1	3901		3301, 3302
2	3902		3305
:			
4	3904		3315, 3319, 3320

The access codes for the department groups and extensions in this table are default access codes.

Now, with this information ready, you may configure the department groups using Jeeves.

Enabling Call Forward for Department Call in Class of Service

To be able to set Call Forward for the Department Group, member extensions must have this feature enabled in their CoS.

By default, Station Basic Feature Template Number 01 is assigned to all the extensions of ANANT UCS, the default CoS group 01 in the Station Basic Feature Template has ‘Department Group Call Forward’ enabled. So, all extensions of ANANT UCS can set Call Forward Department Group.

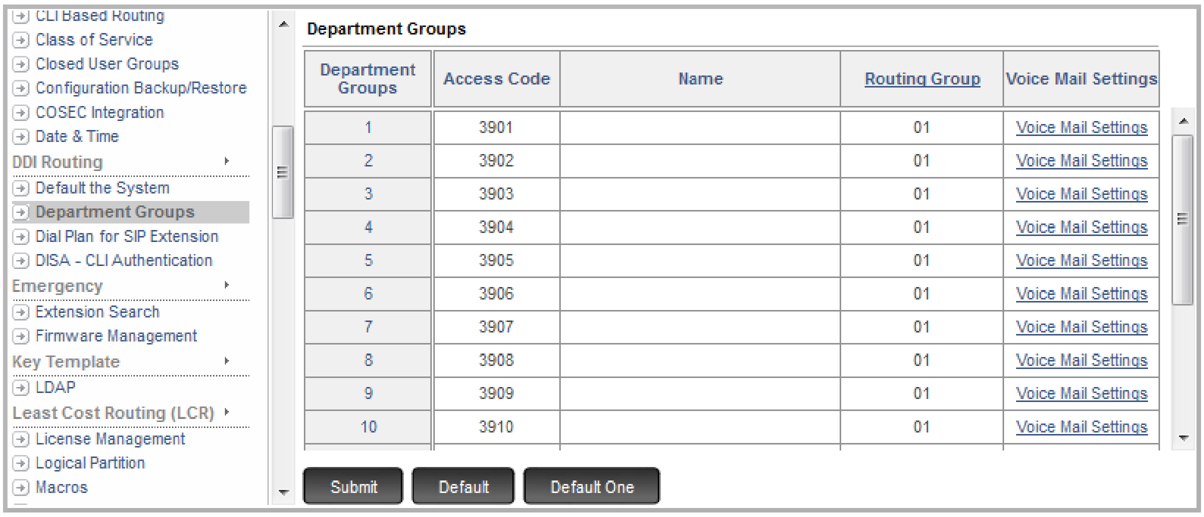
If you wish to allow this feature to member extensions only, retain this feature ‘Department Group Call Forward’ in the CoS group of member extensions, and disable Call Forward feature in the CoS of all the extensions. For this, you may create separate Station Basic Feature Templates for member extensions and other extensions.

Refer the topic “Class of Service (CoS)” and “Station Basic Feature Template” for instructions.

Configuring Department Groups

- Login as System Engineer.

- Under **Configuration**, click **Department Groups**.



Creating Department Groups

- To create a Department group, assign an **Access Code** for the department group against the Index Number.

  By default, the Access Codes assigned to Department groups for Index Numbers 1 to 16 are 3901 to 3916 and for Index Numbers 17 to 32 is blank.

  If you decide not to use the default access codes, ensure that the access code you assign to each department group is unique and does not match with any extension access code or any feature access code of the Dial Phase. Refer the topic “Access Codes” to know more.

To assign Station Access Codes according to your preference and requirement to a range of Department Groups, see “Assigning Access Codes to a Range of Extensions”.

- You may also assign a **Name** to the department group to facilitate identification. This name will appear in the Dial by Name directory along with the department group number. The Name can be a maximum of 18 characters.

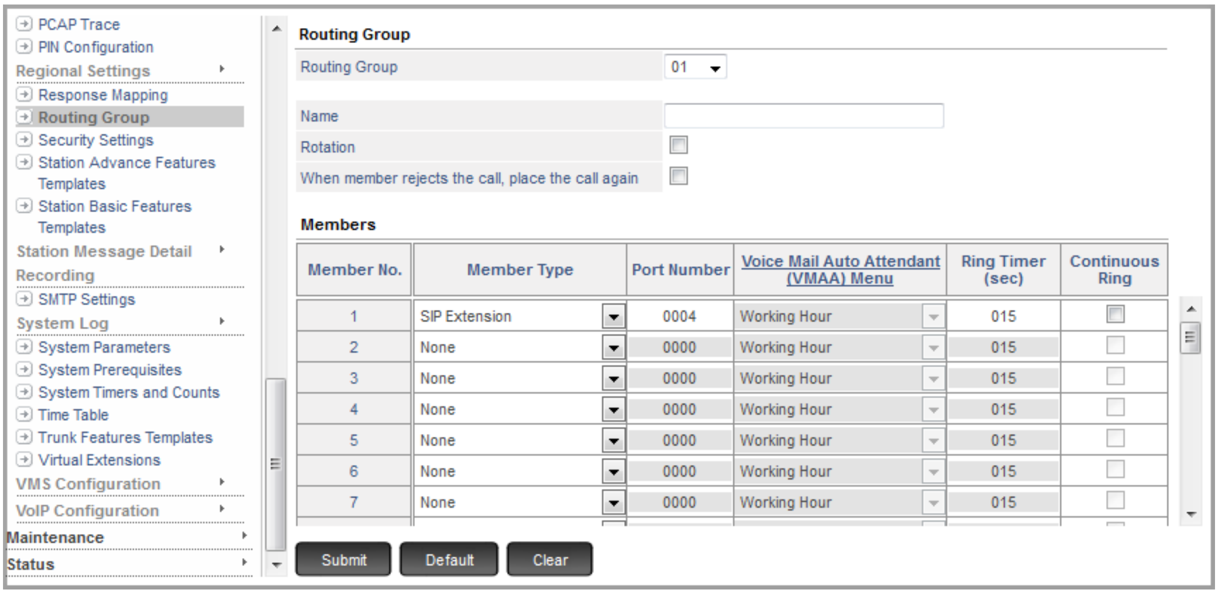
- Now, enter the **Routing Group**, that is, the number of the group you created for this department. Where multiple departments exist, you must create separate routing groups for each department group.

  This can be done in two ways:

  - create the routing groups first and simply enter the relevant routing group number against the Department Group Index (to which you have assigned the access code).

    OR

  - in Routing Group, click the **Routing Group** link. The Routing Group page opens. Now create the Routing Group as per your requirement. For details, see “Routing Group”.



  - In **Voice Mail Settings**, click the Voice Mail Settings link. The respective Extension Voice Mail Settings window will open. You may edit the parameters. For details, see “Extension Voice Mail Settings”.

The Voice Mail Settings link will be visible only if you have configured the respective access codes.

  Click **Close.**

- Click **Submit.**

For Example:

The Customer Care Department of a company has four extensions: 201, 202, 203 and 204 (on software ports 001, 002, 003, and 004 respectively), which needs to be grouped for Department Calls.

Requirement: The company wants the following:

1. Call traffic should be distributed equally on all four extensions.
2. Ring sequence should be 201, 202, 203, and 204 (first to last).
3. First 201 should ring for 20 seconds.
4. If no reply, 201 should continue to ring and 202 should ring for 10 seconds.
5. If still no reply, 201 should continue to ring and 203 should ring for 15 seconds.
6. If still no reply, 201 should continue to ring and 204 should ring for 20 seconds.
7. 51 should be the access code for this department group.

Solution: Select a routing group e.g. 03, and configure as follows:

1. 201 as member 01, with member type SIP Extension, and Port number 0001.
2. 202 as member 02, with member type SIP Extension, and Port number 0002.
3. 203 as member 03, with member type SIP Extension, and Port number 0003
4. 204 as member 04, with member type SIP Extension, and Port number 0004.

   Set 'member type' of members 5 to 32 in Routing Group 3 to 'None'.

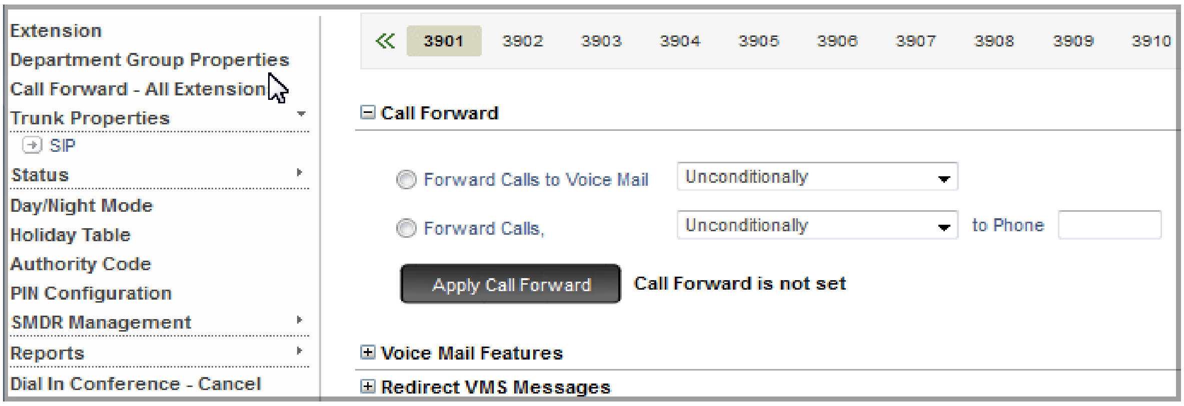
1. Select the 'Rotation check box' on routing group number 03 to distribute call traffic.
2. Select the 'Continuous Ring Check box' for member 01 (201) and set the 'Ring Timer' to '20 seconds.
3. Set the Ring Timer of member 02 (202) to 10 seconds. Disable the 'Continuous Ring' check box.
4. Retain the Ring Timer of member 03 (203) as default 15 seconds. Disable 'Continuous Ring' check box.
5. Set the Ring Timer of member 04 (204) to 20 seconds. Disable 'Continuous Ring' check box.
6. Assign 51 as access code and routing group 03 to the Customer Care Department.

Click **Submit**.

Setting Call Forward for Department Group

You can set Call Forward for Department Group from SA Mode only. To set Call Forward,

- Login as System Administrator.
- Click **Department Group Properties**.



- Click the desired Department Group Number tab for which you want to set Call Forward.

- Click **Call Forward** to expand.
- You have two options for Call Forward:
  - To forward all department calls to Voice Mail, select the **Forward Calls to Voice Mail** and select the Call Forward type.
  - To forward all department calls to a specific number, select **Forward Calls-to Phone** and enter an extension number where the call is to be forwarded.
- Click the **Apply Call Forward** button.

  The color of the text indicating that Call Forward is set will change to red.

To cancel Call Forward,

- Click the **Cancel Call Forward** button.

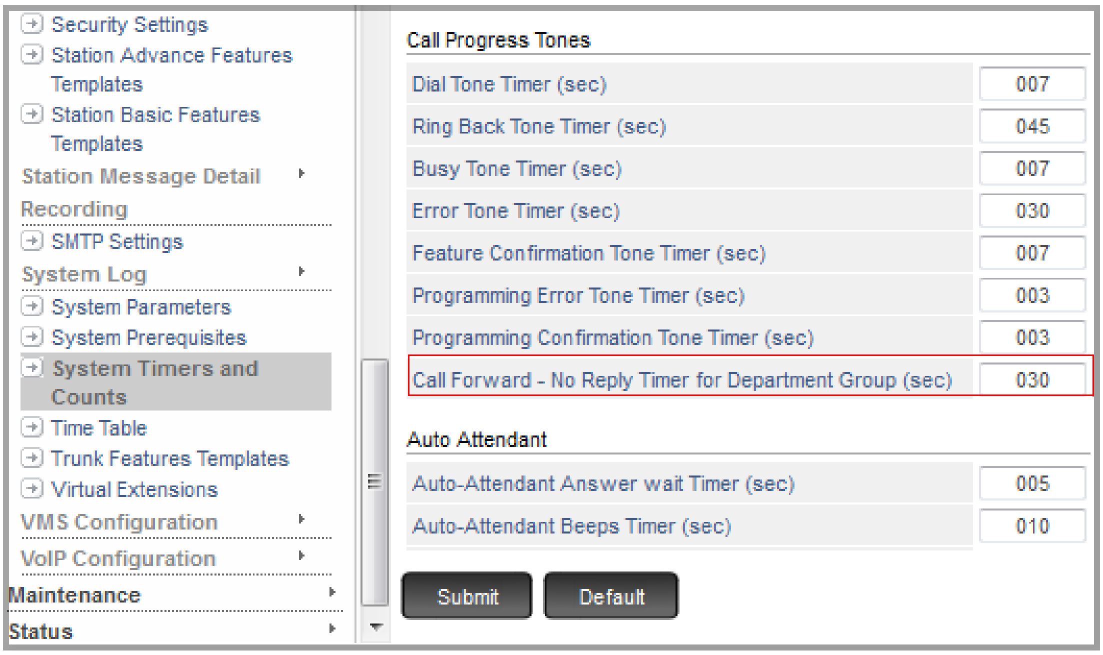
  The color of the text indicating that Call Forward is not set will change to black.

Setting Call Forward No Reply Timer for Department Group

If have enabled Call Forward Department Group, you may change, if required, the Call Forward No Reply Timer for Department Group. You can change this timer only from SE Mode. By default the Timer is set to 30 seconds.

To change the Call Forward No Reply Timer,

- Login as System Engineer.
- Under **Configuration**, click **System Timers and Counts**.



- Scroll with the vertical bar to **Call Forward No-Reply Timer for Department Group (sec)**.
- Set the timer to the desired value. The range of this timer is 1 to 255 seconds.
- Click **Submit.**

How to use

Making a Department Call

Making a department call is the same as calling another extension.

For Extended IP Phone Users

- Press DSS Key assigned to the desired Department Group.
  OR
- Dial the desired Department Group Number.
- You get Ring Back Tone as the call lands on an extension within the department group.
- Talk when the call is answered.
- Go Idle or you get dial tone after 3 seconds.

Accessing Department Group Voice Mail

For Extended IP Phone Users

When LED of Voice Mail Key is turned on to indicate new message,

- Press Voice Mail key.
- VMS informs you about new message(s) in your Department Group mailbox and Personal Mailbox.
- Follow Voice Mail prompts to access Department Group mailbox.
- Press 2 to go to Department Group mailbox.
- Press 1 to listen to new messages.
- Go ON-Hook or follow voice prompts for the desired option.

Setting Call Forward for Department Group

For Extended IP Phone Users

- Press DSS Key assigned to the Call Forward Department Group.
  OR
- Dial **1179** (users worldwide). Users in the Philippines, dial **1108**.
- Enter the Department Group Number whose calls are to be forwarded.
- Scroll to select Call Forward Type from the following options:
  - Cancel Call Forward
  - Forward Unconditionally
  - Forward when Busy
  - Forward when No Reply
  - Forward when Busy/No Reply
- Press Enter key.
- Enter Destination number (extension number or voice mail)
- You get confirmation tone and message on your phone’s LCD.
- Go Idle.

Dial By Name

Dial By Name enables extension users to call another extension or an external party by dialing the name of the person, instead of dialing their number.

With Dial By Name users need not remember the desired party's number or short codes, that is, “Abbreviated Dialing” codes.

For each extension, the database for names used in Dial by Name is drawn from:

- **Personal Directory**, which is assigned to each extension, wherein up to 25 external party numbers along with their names may stored. The system uses the Personal Directory to dial external parties by their names. See “Abbreviated Dialing” to know more.

- **Global Directory**, which is assigned to the extension in its “Class of Service (CoS)”. The Global Directory is a system-wide list of external party numbers and names. Up to 2999 numbers can be stored in this directory, and parts of the Global Directory (Part 1, 2, 3) can be assigned to each extension in its Class of Service. See “Abbreviated Dialing” to know more.

- **Names of Extensions**, which are names of users/departments groups. Their names are assigned to the SIP extensions to identify the extension users. Names of Extensions are necessary for making internal calls using the Dial By Name feature.

How it works

- Extension user presses the DSS Key assigned to 'Dial By Name' feature.
- On SPARSH VP248, press the 'Names' key. On SPARSH VP310, press the ‘Contacts’ key.
- The prompt <Name: > appears on the phone display.
- User enters the name of the desired party[97].
- For example, user wants to call Midas Biz, and enters the letter 'M' using the keypad.
- The system displays in alphabetical order, all names starting with 'M'. These numbers are drawn from the Personal and Global Directories assigned to the extension and the Extension Names configured in the system.
- User scrolls the list using the Up/Down navigation keys to reach the desired contact's name.
  OR
  Instead of scrolling the entire list, the user enters more than one initial letter of the contact's name. The search is narrowed down to more accurate matches. The phone displays the matching entries in the directory.

- The user must select the desired name by pressing 'Enter' Key.
- The system dials out the number stored under the selected name. The name and number are displayed on the user's phone.

97. *The process of entering the names is the same as when writing text messages (SMS) from a cell phone. The keys must be pressed multiple times in quick succession to enter the desired alphabet.*

How to configure

For this feature to work, the following must be configured:

1. **DSS Key:** A direct station selection (DSS) key must be configured for the Dial by Name feature. Without the DSS Key this feature will not be accessible.

   The factory-default key map of SPARSH VP248, phones have the DSS Key labeled as 'Names'. SPARSH VP310 has a fixed feature key ‘Contacts'.

2. **Global Directory:** The names of the external parties must be configured against their respective numbers in the directory. Refer the topic “Abbreviated Dialing” for instructions on configuring the Global Directories.

3. **Personal Directory:** The names of the external parties must be configured against their respective numbers in the Personal Directory. Refer the topic “Abbreviated Dialing” for instructions on configuring the Personal Directories.

4. **Extension Names:** Extensions must be SIP phones. Refer the topics related to the configuration of the extensions[98].

5. **Class of Service:** Dial By Name is allowed to all SIP Extension users. However, the use of this feature is related to the following features, which must be enabled in the Class of Service of extension users:

   - Internal Calls - This is a part of the Basic Features. By default these are enabled.
   - Global Directory Part 1
   - Global Directory Part 2
   - Global Directory Part 3.

   Global Directory Part 1 is assigned to the default CoS group 01 assigned to all extensions in the default Station Basic Feature Template 01.

   If you want the names to be drawn from Global Directory Part 2 and Part 3, provided these are programmed, you must enable these two directories in the CoS of the SIP extensions.

   Refer “Abbreviated Dialing”, for instructions on configuring the Global Directory.

   Refer “Class of Service (CoS)” and “Station Basic Feature Template” for configuration instructions on how to enable a feature in the CoS and how to apply it on extensions.

The system will display the names exactly as they have been configured in the Personal and Global Directories.

98. *You may also refer the instructions provided under the topic Configuring Extensions: “Configuring Matrix SPARSH VP330”, “Configuring Matrix SPARSH VP248”, “Configuring Matrix SPARSH VP310” “Configuring Matrix SPARSH VP510”, “Configuring Matrix SPARSH VP210”, “Configuring Matrix Extended SPARSH VP710”, “Configuring Matrix SPARSH VP210”.*

How to use

For Extended IP Phone Users

- Press 'Names' key.
- You get the prompt: 'Name'.
- Enter the initial letters for the contact's name.
- The number of matching entries that will appear at a time on your phone's display will vary according to your phone's LCD display capacity.
- Scroll with the Up/Down navigation keys to reach the desired contact's name on the list.
- Press 'Enter' key to select the name.
- The system displays the name and number being dialed out.
- You get Ring Back Tone or Busy Tone.

Entered the wrong alphabet?

- Go ON-Hook.
- Go OFF-Hook.
- Press the DSS Key labeled 'Names' again.
- Enter the name/initial letters of the contact's name.

Dialed Number Directory

Dialed Number Directory feature is available only to the users of the Extended IP Phones.

It is the list of numbers dialed out from the phone, similar to the call history of recently dialed calls on a cell phone.

ANANT UCS retains up to 16 numbers dialed out from a phone in a directory.

These numbers may have been dialed out using features like Abbreviated Dialing, Quick Dial, Redial, Walk-In Class of Service, or may be a simple outgoing call made by directly dialing the external number.

How it works

- When a IP Phone extension user makes an outgoing call, the number is stored in the Redial Number List.

*By default the system stores only the external numbers in the Last Number Redial List. If you want the system to store internal calls in this list, make sure you enable the **Store Internal Calls in Redial Call Log** check box in the "System Parameters".*

- The list has a capacity of storing a maximum of 16 recently dialed numbers.
- The list is updated using the First-In First-Out logic, whereby the earliest dialed number is replaced with the most recently dialed number.
- To use this feature, the phone user must invoke the "Last Number Redial" feature.
- Doing so, the Redial Number List will appear on the phone display.
- The user may now navigate the list, select the number to be dialed out.
- The system will dial out the selected number using the same Outgoing Trunk Bundle Group used to place this call earlier.
- If the number had been dialed earlier using Abbreviated Dialing, the system will check for Toll Control when dialing out the number again from the dialed number directory[99].

How to configure

No specific programming required.

How to use

For Extended IP Phone Users Only

- Press the 'Redial' Key.
  OR
  Dial **7.**

99. *Recall that the system does not check for Toll Control when Abbreviated Dialing is used.*

- The list of last dialed calls appear on your phone's display.
- Scroll with up/down navigation key to reach the desired number.
- Press 'Enter' key.
- The desired number is dialed out and appears on your phone's display.
- You get ring back tone.
- Talk when speech is established.
- Go ON-Hook after the conversation has ended.

Dial Plan for SIP Extension

ANANT UCS supports 8 Dial Plans with total 32 entries in each table. The Dial Plan contains a series of digits and/ or wild card characters.

When a user dials a number, it is compared with the Rule configured in the Dial Plan. If a match is found, the IP Phone routes the call immediately without waiting for End of Dialing and if a match is not found, the IP Phone will wait for the End of Dialing and then route the call.

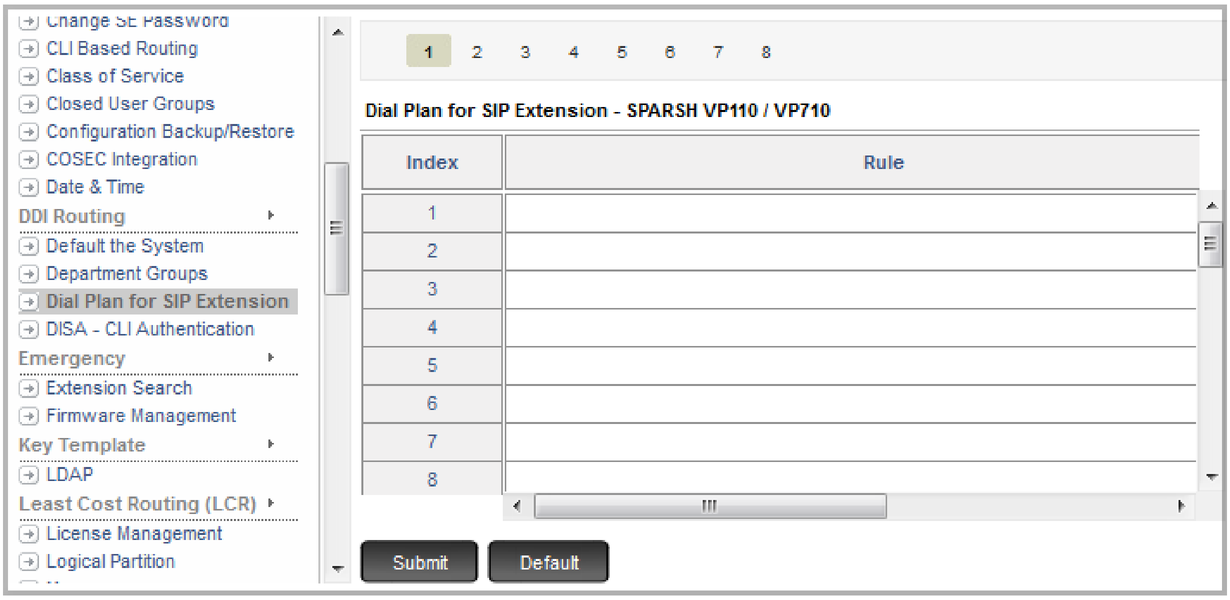
How to configure

Configuring the Dial Plan involves the following steps:

- Selecting a Dial Plan and configuring the rules in the Dial Plan.
- Assigning the Dial Plan to the desired SIP Extensions.

Configuring Dial Plan

- Login as System Engineer.
- Under **Configuration**, click **Dial Plan for SIP Extension**.



- Click the desired Dial Plan number. The **Dial Plan for SIP Extension - VP110/VP710** page opens.
- Against each **Index** configure the **Rule** according to which you want the system to process the call. You can configure a maximum of 32 Rules in each Dial Plan.

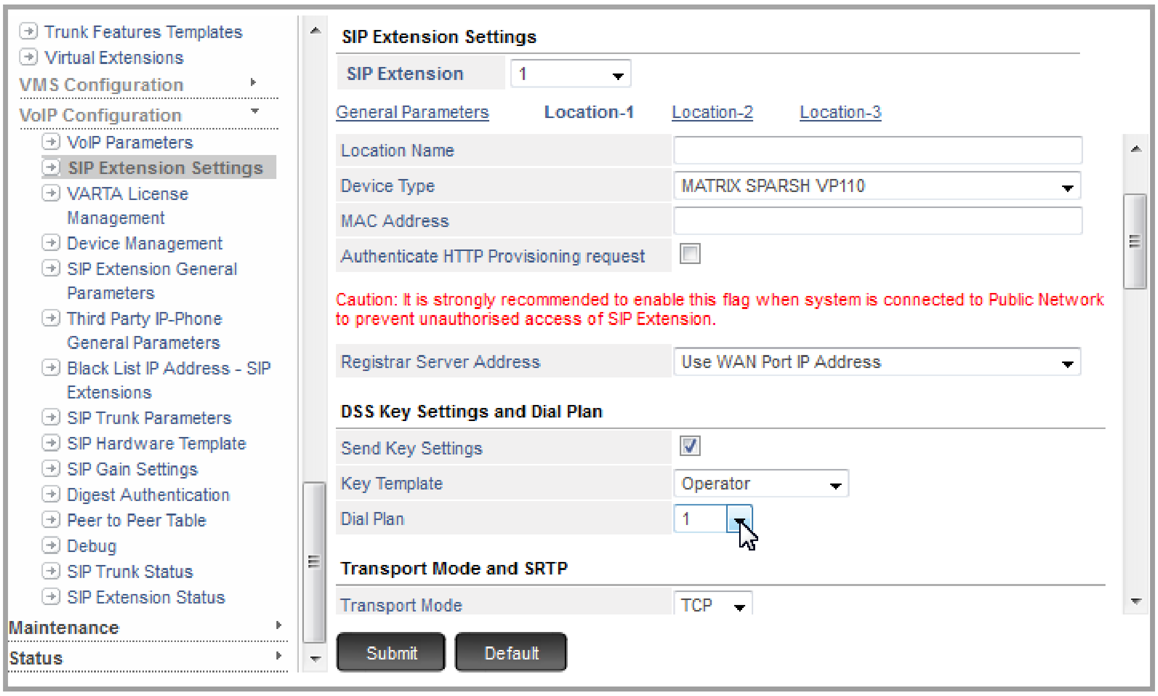
  For example, if you want that users should be able to dial extension numbers from 3000 to 3999 without any delay, configure the Rule as 3XXX.

  For more details to configure the rules, refer to the topic *Dial Plan* in the *SPARSH VP110 User Guide*.
- Click **Submit**.

Assigning Dial Plan to SIP Extensions

To assign the Dial Plan you configured for VP110,

- Under **VoIP Configuration**, click **SIP Extension Settings**.
- Click the location at which you have registered SPARSH VP110, for example, **Location 1**.



- Scroll to **DSS Key Settings and Dial Plan.**
- Select the **Dial Plan** number you configured.
- Click **Submit.**

To assign the Dial Plan you configured for VP710,

- Under **VoIP Configuration**, click **SIP Extension Settings**.
- Click the location at which you have registered SPARSH VP710, for example, **Location 1**.
- Scroll to **Dial Plan.**
- Select the **Dial Plan** number you configured.
- Click **Submit.**

Digest Authentication

Digest Authentication is a challenge-based authentication service of SIP to authenticate the identity of the originator of SIP request in the INVITE message. The recipient of the request can ascertain whether or not the originator of the request is authorised to make the request. When the digest credentials of the originator—User Name and Password—in the INVITE message are authenticated and accepted by the recipient, the originator and the recipient are connected.

ANANT UCS supports Digest Authentication. You may use Digest Authentication to

- restrict access to ANANT UCS to specific callers.
- prevent unwanted or malicious calls.

How it works

The Digest Authentication feature works on the basis of the Digest Authentication Table, in which the credentials, namely the User Name and Passwords of trusted/authorised calling party SIP devices are stored. You must configure this table. The Digest Authentication Table is common for all SIP trunks on which this feature is enabled.

When you enable this feature on a SIP trunk, for all incoming calls (SIP requests):

- ANANT UCS will challenge the identity of the calling party (the SIP device initiating the request) to send its digest credentials.
- When the calling party sends its credentials, ANANT UCS authenticates the credentials by matching it with its Digest Authentication Table.
- If a match is found, the calling party will be authenticated and the call will be allowed on the SIP trunk.
- If no match is found, ANANT UCS will consider it as invalid authentication information and rejects the call.

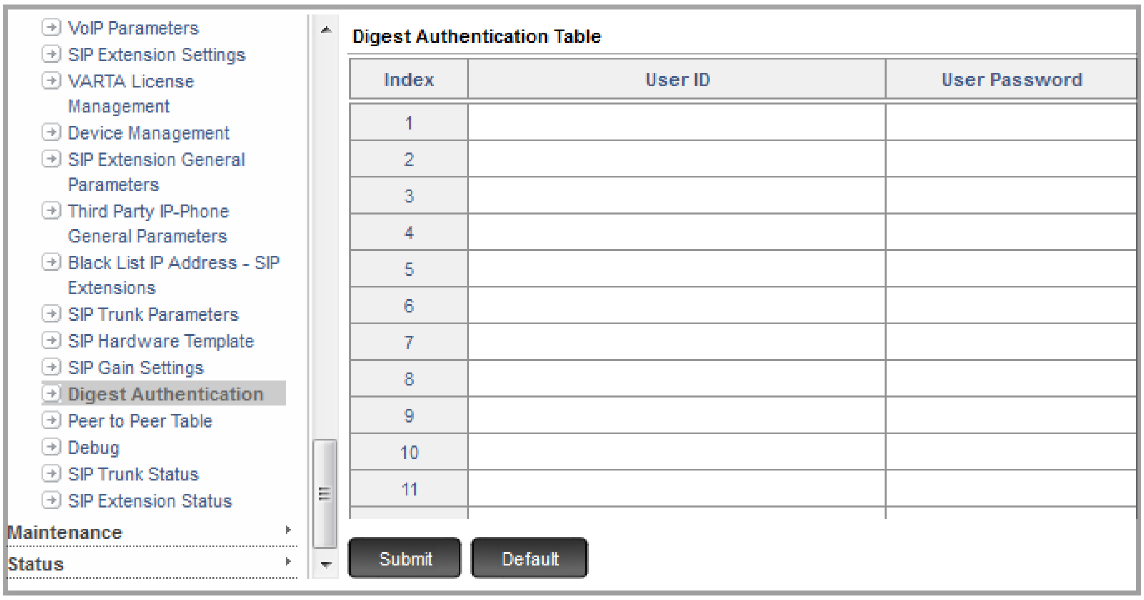
How to configure

To use this feature on SIP Trunks, you must do the following:

- Enable Digest Authentication on the SIP trunks you want to use this feature.
- Configure the Digest Authentication Table.

Configuring Digest Authentication

- Login as System Engineer.
- Under **Configuration**, click the **VoIP Configuration**.
- Click **Digest Authentication**.



The Digest Authentication table opens.

- In the **User ID** field, enter the User ID to be authenticated. The User ID must be within 40 characters.

Make sure the Authentication ID configured in SIP Extension Setting does not conflict with the User ID configured above.

- In the **User Password** field, enter the corresponding Password. The Password must be within 16 characters.

  To avoid unauthorized access, we recommend you to change the Password regularly. Make sure it is strong and is kept confidential.

- Click **Submit.**

- Now, enable Digest Authentication on the desired SIP trunks. For instructions, see “Configuring SIP Trunks”.

Direct Dialing-In (DDI)

DDI is the service feature provided to the customers by the ITSPs over SIP Trunks. DDIs are Direct Dial-In numbers. These are additional numbers that can be associated with the main number assigned to the SIP Trunk. They are normally used with the system to provide separate numbers to extension users.

When there is an incoming call on the a specific DDI number it is routed through to the required extension without operator intervention.

Before we understand how DDI routing works, we must understand the following terms:

- **Pilot Number/ISDN Number:** It is a single number assigned by the service provider to a single SIP trunk. This is the combination of MSN Number and the first DDI Number. The Pilot Number is of maximum 16 digits. This is also known as ISDN Installation Number/ISDN Number. The MSN is a fixed series and is given by the service provider whereas the DDI Numbers can be selected by the user and these series vary. For example, if the Pilot Number is 2630555, the MSN is 2630 and the DDI Number is 555. However the number of digits to be used as DDI Numbers is informed by the service provider as they may differ for each service provider.

- **Multiple Subscriber Number (MSN):** In India, the MSN are the first four digits of the Pilot Number, which remain fixed and are provided by the service provider. The number of fixed digits may vary depending on the service provider.

- **DDI Numbers:** These are additional numbers, which can be selected by you but are allotted by the service provider. These are sequential numbers. In the DDI number only the last 3 or 4 digits vary. These are known as DDI numbers. However the number of DDI digits which vary are provided by the service provider. These numbers are normally used with a PBX and are assigned to the extension user.

How it works

Let us understand this with the help of an example:

- A SIP Trunk has been connected to the PBX.

- 50 DDI numbers have been given by the service provider.

- Incoming calls on the DDI numbers can be routed to different landing destinations as extension users, department groups or other trunks within the organization.

- To assign DDI numbers to the extension users and to route calls as per the DDI numbers, you need to configure the following:
    - Incoming Reference Table
    - Outgoing Reference Table
    - DDI Routing Table

Handling Incoming Calls

Pre-requisites to route incoming calls as per the DDI logic,

- Assign an Incoming (IC) Reference ID to the SIP Trunk. This Incoming (IC) Reference ID acts as a link between the port settings and the Incoming Reference Table.

- In the Incoming Reference Table against the same Incoming (IC) Reference ID, configure the DDI Routing Ref. ID, Route on First Destination, Ring Timer (sec), When No Reply, When Busy, Trunk Features Template as per your requirement.
- To route and place the call on the final destination, the system refers to the DDI Routing Ref. ID configured in the Incoming Reference Table. As per this ID the system checks the DDI Routing Table. The DDI Routing Ref. ID acts as a link between the Incoming Reference Table and the DDI Routing Table. The system places the call on the final destination as configured against the DDI Routing Reference ID in the DDI Routing Table.

When there is an incoming call on the SIP Trunk,

- As per the time zone, the system checks the Incoming (IC) Reference ID assigned to the SIP Trunk
- If the Incoming (IC) Reference ID has been configured, the system then hunts for the same ID in the Incoming Reference Table.

  If no channel match is found in the Incoming Reference Table, the call is routed as per the Trunk Feature Template assigned to the SIP Trunk.

  If a match is found, the system verifies the routing logic is to be applied.
- For the matching entry, the system checks the corresponding DDI Routing Reference ID. The DDI Routing Reference ID acts as a link between the Incoming Reference Table and the DDI Routing Table.
- The system hunts for the same DDI Routing Reference ID in the DDI Routing Table. In the DDI Routing Table, a range of DDI numbers and their respective landing destinations are configured. You can configure as many as 224 DDI Reference ID’s with different DDI numbers and their corresponding landing destinations. From this table the system fetches the extension number configured as the landing destination.
- When a match is found in the DDI Routing Table for the incoming call on the respective DDI number, the system places the call on the corresponding landing destination configured by you.

  When the call is not answered by the landing destination or if the landing destination is busy, the system again checks the Incoming Reference Table for the options you have selected for the parameters, When No Reply and When busy. You may select from the following options - Disconnect, Route to a TLG, or Route to VMS.
- If no match is found for the DDI number in the DDI Routing Table, the call is routed as per the Trunk Feature Template assigned in the Incoming Reference Table.

Handling Outgoing Calls

Reverse DDI is when you want the DDI numbers assigned to the extension users to be displayed as the CLI to the called party. For this you must configure the DDI Routing Table and the Outgoing Reference Table.

When an outgoing call is made by the extension user, the DDI numbers will be sent as CLI to the called parties only if, the service provider supports this facility. If this facility is not supported by the service provider and if ANANT UCS sends the DDI Number, this number will be swapped by with the Pilot number by the service provider and then the call will be routed further.

Pre-requisites to route outgoing calls as per the DDI logic,

- Assign an Outgoing Reference ID to the SIP Trunk. This Outgoing Reference ID acts as a link between the port settings and the Outgoing Reference Table.

- In the Outgoing Reference Table against the same OG Reference ID, configure the DDI Routing Ref. ID and the Pilot Number as per your requirement.

- To fetch the DDI number the system refers to the DDI Routing Ref. ID configured in the Outgoing Reference Table. As per this ID the system checks the DDI Routing Table. The DDI Routing Ref. ID acts as a link between the Outgoing Reference Table and the DDI Routing Table.

When an outgoing call is made using the SIP Trunk,

- The extensions user dials an external number. The system provided the SIP Trunk to out-dial the number.

- As per the time zone, the system checks for the Outgoing (OG) Reference ID assigned to the SIP Trunk.

- If the OG Reference ID has been configured, the system then hunts for the same ID in the Outgoing Reference Table.

  If a match is found, the system verifies the trunks assigned to the user for making an outgoing call.

- The system the checks the corresponding DDI Routing Reference ID. The DDI Routing Reference ID acts as a link between the Outgoing Reference Table and the DDI Routing Table.

- The system hunts for the same DDI Routing Reference ID in the DDI Routing Table. From this table the system fetches the DDI number assigned to the extension user. You can configure as many as 224 DDI Reference ID’s with different DDI numbers.

- When a match is found in the DDI Routing Table, the system sends the MSN Number + DDI number of the extension in the calling party field to the called party.

- If no match is found in the DDI table, the ISDN Number, that is the MSN Number + the first DDI number is sent in the calling party field to called party.

Configuring using Jeeves

To route incoming calls as per the DDI logic you must configure the following:
- Assign the Incoming Reference ID on the respective trunk.
- Incoming Reference Table
- DDI Routing Table

To configure the Incoming Reference Table,

- Determine the SIP Trunk on which you want to apply the DDI logic.

- On the SIP Trunk, assign an Incoming Reference ID number. To do this,

  - Login as System Engineer.

  - Under **Configuration**, click **SIP Trunk Parameters**.

  - Click Incoming Call to expand. Scroll to **Incoming (IC) Reference ID**.

- Assign an **Incoming (IC) Reference ID** number for each time zone.

- Under **Configuration**, click **DDI Routing**.

- Click **Incoming Reference Table** and configure the following parameters:

  - **IC Reference ID**: You must configure the parameters to apply DDI Routing against the same ID as assigned on the SIP Trunk. This ID is a link between the port settings and the Incoming Reference Table.

  - **DDI Routing Reference ID**: Enter the DDI Routing Reference ID number. As per the this reference number the system checks the DDI Routing Table and determines the final landing destination for placing the call.

    This ID is a link between the Incoming Reference Table and the DDI Routing Table.

  - **Route on First Destination**: When the final destination extension is identified as per the DDI number table, the system checks if this check box is enabled. If it is enabled, every incoming call will always be placed on the first extension configured in the DDI Table. If the check box is clear, the call is routed to the respective extension number.

  - **Ring Timer**: This timer signifies the time for which the extension on which the incoming call is placed must ring. On expiry of this timer if the call is not answered the call is routed as per the option you configure for the parameter **When No Reply**.

  - **When No Reply**: When the call is not answered by the DDI extension, the system checks the option you have selected for this parameter and processes the call further. You may select any option from the following:
    - Disconnect the call
    - Route the call to Trunk Landing Group
    - Route the call to Voice Mail, that is to the Mail box assigned to the DDI extension. For this, a mail box must be assigned to the extensions. If the extension is not assigned mail box, the caller will hear the welcome message of the VMS, but will not be able to access the mail box.

  - **When Busy**: If the DDI extension is busy, the system checks the option you have selected for this parameter and processes the call further. You may select any option from the following:
    - Disconnect the call.
    - Route the call to Trunk Landing Group.
    - Route the call to Voice Mail, that is to the Mail box of the DDI extension. For this, a mail box must be assigned to the extensions. If the extension is not assigned mail box, the caller will hear the welcome message of the VMS, but will not be able to access the mail box.

  - **Trunk Feature Template**: A Trunk feature template is assigned to each Incoming Reference ID. You can enable different features in the Trunk Feature Templates. A different Trunk Feature Template can be assigned to each channel or a range of channels, such as set Auto Answer Time Zone wise, allow DID according to the time zone. Hence incoming calls can be answered and routed in different ways. For more details refer the topic “Trunk Feature Template”.

  - Click **Submit.**

- After the Incoming Reference Table has been configured you must configure the DDI Routing Table.

DDI numbers can be assigned to extensions users, Department Groups, Routing Groups depending on the organizations requirement.

You can configure as many as 224 DDI Reference ID's with different DDI numbers and landing destinations. To configure the DDI Routing Table,

- Under **Configuration**, click **DDI Routing**.
- Click **DDI Routing Table** and configure the following parameters:
  - **DDI Routing Reference ID**: You must configure the parameters to apply DDI Routing against the same ID as assigned in the Incoming Reference Table. This ID is a link between the Incoming Reference Table and the DDI Routing Table.

    DDI Routing Ref. ID's in the DDI Routing Table act as identifiers. This Reference ID is assigned in the Incoming Reference Table.

    When there is an incoming call the system checks the Incoming Reference ID assigned to the trunk. The system tracks this ID from the Incoming Reference Table.
  - **Start DDI Number**: This is the first DDI Number from where the DDI range starts. You must only enter the DDI number. For example, if the Pilot Number is 2630555. If 10 DDI numbers have been taken, then the DDI Number range is from 555 to 564. The Start DDI Number in this case is 555.
  - **Total DDI Numbers**: The total number of DDI numbers for you want to set the same landing destination. For example, if the Start DDI Number is 555, Total DDI Number is 1, then the landing destination selected will only be for the DDI Number 555. If Start DDI number is 556 and the Total DDI Numbers is configured as 10, then the landing destination will be for the entire range of numbers, that is from 556 to 565.
  - **DDI Number of Digit**: The number of digits in a DDI number. Suppose 100 DDI numbers are supported on a SIP Trunk, then the Number of Digits for that Trunk should be configured as 3. Suppose 10 DDI numbers are supported on another SIP Trunk, then the Number of Digits for that Trunk should be configured as 2.
  - **Route to Destination**: The incoming calls on the DDI numbers can be placed on extensions users, Department Groups, Routing Groups depending on the organizations requirement.

    Configure the following for the parameters for the landing destination:
    - **Port Type**: Select the port type to place the DDI call from the following:
      - Department Group
      - Quick Dial
      - Routing Group
      - SIP Trunk
      - Flexible Number
      - Virtual Extension
      - Voice Mail Auto Attendant

      For example, to place incoming calls on the extensions, select Flexible Number as the Port Type.
    - **Port Number**: For the Port Type you have selected, enter the Port Number/Group Number. The Port Number range depends on the Port Type you have selected.

- **Start DDI Flexible Number**: If you have selected the Port Type as Flexible Number, enter the first extension number to which you want to assign the first DDI Number. As per the Total DDI numbers configured, the system automatically maps the DDI numbers with the extension numbers. Both the DDI numbers and the extensions numbers are mapped sequentially. For example, if the Start DDI number is 2630555, the Total DDI Numbers are 10 and Start DDI Flexible Number 2001, the system maps the DDI numbers with the extensions numbers as follows:

DDI Number	Start Flexible Number
555	2001
556 564	2002 2010

  Hence the call for the respective DDI Number is placed on the respective extension number.

- **Voice Mail Auto Attendant (VMAA) Menu:** If you have selected the *Voice Mail Auto Attendant* as the Port Type, select the VMAA Menu to be assigned to the respective DDI Routing Table entry.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- Click **Submit**.

If you want to apply Reverse DDI, that is, if you want DDI numbers to be displayed as the CLI to the called party, you must configure the Outgoing Reference Table. The system uses the same DDI Routing Table to trace the extension users and their respective DDI numbers when an outgoing call is made.

To assign the Outgoing Reference ID and to configure the Outgoing Reference Table you must,

- Determine the SIP Trunk on which you want to apply the Reverse DDI logic.
- On the SIP Trunk, assign an Outgoing Reference ID number. To do this,
  - Login as System Engineer.
  - Under **Configuration**, click **SIP Trunk Parameters**.
  - Click **Outgoing Call** to expand.
  - Scroll to **Outgoing (OG) Reference ID** and assign the number.
- Under **Configuration**, click **DDI Routing**.
- Click **Outgoing Reference Table** and configure the following parameters:
  - **Outgoing Reference ID**: To apply DDI Routing configure the parameters given below against the same ID as assigned on the SIP Trunk. This ID is a link between the port settings and the Outgoing Reference Table.

- **Pilot Number**: Each SIP Trunk is given a Pilot Number by the Service Provider. This is the combination of MSN Number and the first DDI Number. The Number is of maximum 16 digits. This is also known as ISDN Installation Number/ISDN Number. The MSN number is given by the service provider whereas the DDI Numbers can be selected by the user. However the number of digits to be used for the DDI Number is informed by the service provider.

- **DDI Routing Reference ID**: Enter the DDI Routing Reference ID number. As per the this reference number the system checks the DDI Routing Table and determines the DDI number assigned to the extension user. The system then swaps this DDI number with the ISDN number when an outgoing call is made. This DDI number is displayed as the CLI to the called party.

The DDI numbers will be displayed as CLI to the called party only if the service provider supports this facility over the SIP Trunk.

- Click **Submit.**

Direct Inward System Access (DISA)

With Direct Inward System Access (DISA) remote users can access and use the system's features and facilities using Trunks, on which this feature is enabled.

Using DISA, remote users can:

- call any extension.
- make external calls.
- use features and facilities of the system.
- configure features and facilities of the system and administer the system.

All these can be done as if being done from a local extension of ANANT UCS.

DISA Variants

ANANT UCS offers three types of DISA, each with a different method of authentication and level of access:

- PIN Authentication-Multiple Calls
- DISA with CLI Authentication-Multiple Calls
- DISA with CLI Authentication-Single Calls

PIN Authentication - Multiple Calls

Callers can access an extension of ANANT UCS by dialing the DISA Login Code that consists of:

- the DISA Feature Access Code.
- the extension number they want to access.
- the User Password of that extension.

Callers are authenticated and allowed to use the extension on which they are logged in.

The callers must dial codes to go On-hook, Off-hook. They are allowed to make as many trunk calls and internal calls for as long as they remain logged into the DISA mode.

To end the DISA login session, callers must dial the Termination code or disconnect from the remote end.

Callers can access an extension to use DISA PIN Authentication-Multiple Calls only if the extension has DISA feature enabled in its Class of Service.

DISA with CLI Authentication - Multiple Calls

The system authenticates the caller by matching the caller's CLI with the entries of the *DISA-CLI Authentication Table* and logs the caller into the extension configured as 'Auto Login' extension for the CLI.

Callers are not required to dial any DISA Login Code or any password.

When a caller is authenticated on the basis of CLI, the system plays the ('internal' system) Dial Tone to the caller.

The callers must dial codes to go On-hook, Off-hook. They are allowed to make as many trunk calls and internal calls for as long as they remain logged into the DISA mode.

To end the DISA login session, callers must dial the Termination code or disconnect from the remote end.
For this type of DISA, the DISA CLI Authentication Table must be configured first.

DISA with CLI Authentication - One Call

This type of DISA is similar to the previous one. The system authenticates callers by matching the callers' CLI with the entries of the DISA-CLI Authentication Table and logs the callers into the extension designated as 'Auto Login' extension for the CLI.

When the caller is authenticated on the basis of CLI, the system gives the caller direct access to the Outgoing Trunks selected for TAC-1 for the current time zone (working hours, break hours, non-working hours) in the "Station Basic Feature Template" assigned to the Auto Login extension. It plays the dial tone.

Callers are allowed to make a single external call. The system ends the DISA session on the completion of the call by the caller or by the other remote party.

For this type of DISA, the DISA CLI Authentication Table must be configured first.

To make another call, the caller must enter the DISA mode again, by calling the ANANT UCS from the remote location.

DISA with CLI Authentication - One Call is generally used when ANANT UCS is deployed in the Gateway Mode, where ANANT UCS is configured to send an answer signal to the caller/calling device, receive the DTMF digits dialed by the caller/calling device and dial out the digits dialed by the caller/calling device.

For this feature to work, you must enable the desired DISA variant on the desired SIP trunk.

- A call lands on a DISA enabled Trunk.
- The system checks if a DISA variant is enabled on the trunk for the current time zone, that is, working hours, break-hours and non-working hours.
- If a DISA variant is enabled on the trunk, the system processes the call according to the DISA variant enabled on the trunk.
- If **DISA with PIN Authentication - Multiple Calls** is enabled,
  - The system plays Welcome Greeting message to the caller.
  - The caller must dial the DISA Login Code consisting of:
    - the DISA Feature Access Code.
    - the number of the extension the caller wants to access.
    - the user password of the extension.
  - On successful login, the system starts the *DISA Idle State Timer* (configurable; default: 20 seconds). The system waits for the caller to go Off-hook[100].
  - When the caller goes Off-hook by dialing the Off-hook code #1, the system plays the internal dial tone and waits for the caller to dial digits.
  - The system collects the digits dialed by the caller and then routes the call.
  - The caller can make as many trunk calls and internal calls as the caller wants.

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100. If the caller does not go Off-hook within this timer, the system releases the call.

- The caller can terminate the DISA login session either by disconnecting from the remote end or by dialing the Termination Code *#9*.

- If **DISA with CLI Authentication - Multiple Calls** is enabled,

  - The system compares the CLI of the caller with the *Calling Party Numbers* configured in the CLI Authentication Table.

  - If the CLI matches with any of the Calling Party Numbers in the Table, the system provides access to the extension configured as *Auto Login* extension for this Calling Party Number in the Table[101].

  - The caller gets logged into the Auto Login extension and gets the dial tone of ANANT UCS.

  - At the end of the call, the caller dials the On-hook code *#0* to go On-hook. To make another call, the caller dials Off-hook code #1 and dials the desired number. Thus the caller dials the On-hook and Off-hook codes to make as many trunk and internal calls as desired.

- If **DISA with CLI Authentication - One Call** is enabled,

  - The system compares the CLI of the caller with the *Calling Party Numbers* configured in the CLI Authentication Table.

  - If the CLI matches with any of the Calling Party Numbers in the Table, the system provides access to the extension configured as *Auto Login* extension for this Calling Party Number in the Table[102].

  - The caller gets logged into the Auto Login extension and gets dial tone of the outgoing trunks selected for TAC-1 for the current Time Zone (working hours, break hours, non-working hours).

  - The caller can now call the desired external number.

  - After the call is completed, that is, the caller disconnects from the remote end or the other remote called party has disconnected, the caller is logged out.

  - To make another external call, the caller must call the DISA enabled trunk of ANANT UCS again.

In all the variants of DISA, the caller can use all the features allowed in the "Class of Service (CoS)" of the extension the caller is logged into (using PIN Authentication or CLI Authentication).

- *DISA calls in the SMDR report are marked as "O" in the remarks column. See "Station Message Detail Recording-Report".*

- *If DISA is disabled, ANANT UCS will route the call by Auto Attendant logic, if Auto Attendant is enabled.*

- *If DISA and Auto Attendant both are disabled, the incoming call will be routed as per the incoming call routing configured. To know more, see "Auto Attendant".*

---

101. *If no match is found for the CLI of the caller in the Table, the call will be routed as per the Incoming Call Routing configured in ANANT UCS.*
102. *If no match is found for the CLI of the caller in the Table, the call will be routed as per the Incoming Call Routing configured in ANANT UCS.*

WARNING! *This feature allows access to system resources to remote users, and therefore has serious implications for your system's security. There is a risk of fraudulent calls being made from your system, if a third party comes to know the authentication PIN or the User Password of an extension number. The cost of such fraudulent calls will have to be borne by the owner of ANANT UCS.*

To avoid unauthorized access, we recommend you to change the PIN regularly. Make sure it is strong and is provided to users who need to access the system using DISA only.

Feature Interaction:

- If DISA are enabled on the trunk, ANANT UCS supports all types of DISA.
- If both, VMS Auto Attendant and DISA are enabled on the trunk, ANANT UCS supports only PIN Authentication-Multiple Calls. To know how the VMS handles a DISA call, see "VMS DISA Login".

How to configure

To provide DISA to remote users you need to do the following configuration:

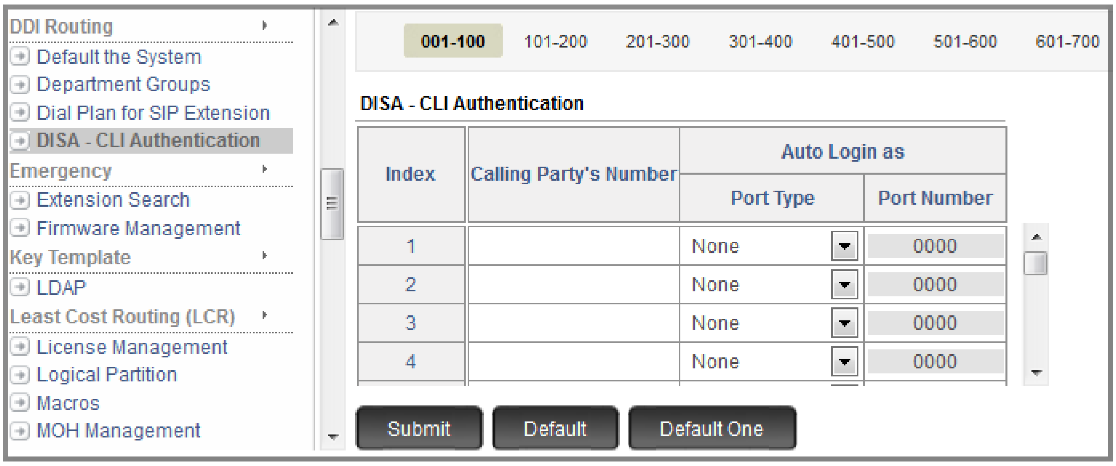
- Select the DISA variant for the Trunks on which you want to apply this feature in their "Trunk Feature Template".
- Enable DISA in the "Class of Service (CoS)" of the extensions which you want to allow callers to access using DISA. This is applicable for DISA PIN Authentication-Multiple Calls only. DISA CLI Authentication-Single Call/Multiple Calls is not dependent on CoS.
- Change the User Password of the DISA extensions, if you selected DISA PIN Authentication-Multiple Calls. If you selected *DISA PIN Authentication-Multiple Calls* on a trunk, the default User Password (1111) will not work. See "User Password" and "System Security" more information and instructions.
- Configure the *DISA Idle State Timer,* if required. See "System Timers and Counts" for instructions.
- If you have selected the *DISA CLI Authentication-Multiple Calls* or *CLI Authentication-One Call* on a trunk, you must configure the **CLI Authentication Table**.

Configuring DISA CLI Authentication Table

To configure the DISA CLI Authentication Table,

- Make a list of remote users and their numbers whom you want to allow DISA.
- For each remote user's number on your list, write the Extension number of the ANANT UCS you want to allow this extension user to log in.
- Login as System Engineer.

- Under **Configuration**, click **DISA - CLI Authentication**.



- You can configure as many as 999 numbers in this table, by clicking the tabs of the index on the top of the table.

- Refer to the list of remote user numbers and the corresponding ANANT UCS extension numbers you made.

- In the **Calling Party's Number**, enter the number of the remote users whom you want to allow access to DISA using CLI Authentication. The system will match the CLI of the callers with the numbers you store here.

- For each Calling Party Number, in the **Auto Login as** field, select the extension **Port Type** (SIP Extension, or Virtual Extension) and **Port Number** you want to allow access to after the Calling Party Number is authenticated.

- Click **Submit.**

How to use

If you are a Remote user, to be able to use DISA, you must know:

- the number of the Trunk on which DISA is enabled and the variant of DISA enabled on this trunk.

- the number of the extension and the user password which you want to access, if using DISA with PIN Authentication.

- the duration of *DISA Idle State Timer* so that you may dial digits accordingly, without delay.

- the special digits to be dialed during a DISA login session.

Dialing Special Digits

After successful login, you will be required to go on-hook, go off-hook, use 'flash', use 'pause' or dial characters like A, B, C, D, from your remote device to use the features and facilities of ANANT UCS.

ANANT UCS will not be able to understand the conventional way of dialing 'flash' key, therefore, the server supports specific codes for specific activities. If these codes are received during a DISA session, ANANT UCS interprets it and performs the associated activity.

When you are in DISA mode, use the following codes to indicate an activity:

Special Digit/activity	Code to be dialed
on-hook	#0
off-hook	#1
Flash	#2
Pause	#3
A	#4
B	#5
C	#6
D	#7
+	#8
To Terminate the DISA	#9
#	##
End of String	#*

To use DISA,

- Dial the number of the Trunk on which DISA is enabled for the current time zone, Working, Break, Non-working hours.
- ANANT UCS answers the call. You will get music.

If **DISA PIN Authentication-Multiple Calls** is enabled,

- You will get beeps at the end of music.
- Dial DISA Login Code 1079 during the beeps.
- Dial the extension number. You will get beeps.
- Dial the User Password for the extension during the beeps.
- You get feature tone on successful login.
- Dial the special digits **#0** to go On-hook and then **#1** to go Off-hook, you get the dial tone. Dial the number to make your calls like any local extension. For example, making an external call, dial Trunk Access Code '0' to grab a trunk and dial the number.
- To terminate DISA session dial **#9** or disconnect the call from your (remote) end.

If **DISA CLI Authentication-Multiple Calls** is enabled,

- You will get system dial tone.
- Dial the special digits **#1** to go Off-hook, **#0** to go On-hook, and make your calls like any local extension.
- To terminate DISA session dial **#9** or disconnect the call from your (remote) end.

If **DISA CLI Authentication - One Call** is enabled,

- You will get Trunk dial tone.
- Dial the external number, without the Trunk Access Code.
- The DISA session will be terminated when you or the remote called party disconnects.
- To make another call, you must dial the number of the DISA enabled trunk again.

To configure DISA Idle State Timer, refer to "System Timers and Counts".

The features listed below are not supported in the DISA mode.

- *Auto Call Back*
- *Auto Redial*
- *Call Park*
- *Call Chaining*
- *Walk-In Class of Service*
- *Self Ring Test*
- *Live Call Supervision*

Direct Station Selection Console

The Direct Station Selection (DSS) Console is an add-on module with tri-color LEDs for SPARSH VP510. It provides you quick access to Extensions, Trunks, Features/Functions of ANANT UCS; also making calling operations easy.

While the DSS Console is generally used by the Operator/receptionist in an organization, it is also meant to be used by anyone who needs to access various features of ANANT UCS at a touch of a single key.

The DSS Console **DSS532** is offered by Matrix for SPARSH VP510.

Maximum 32 DSS Consoles (DSS532) are supported by ANANT UCS.

DSS532

A maximum of four DSS532 can be attached to SPARSH VP510.
With each DSS532, 32 additional keys are at your disposal to be used as DSS keys. When all four DSS532 are connected, you get 128 additional DSS keys at your disposal to be used.

For instructions:

- to install the DSS532 with SPARSH VP510, see “Installing DSS532 with SPARSH VP510”.
- to configure the DSS keys of the Console, see “Programming DSS Console Keys”.

DSS Keys

You can assign Extension numbers or features/functions to the keys on the DSS Console, so that they can be accessed easily simply by pressing a single key.

LEDs

Each DSS Key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

Status of Extensions and Trunks

The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the phone.

Thus, the status of user's own Extension, status of other Extensions and status of the trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the phone.

The following table shows the relationship between the color of the LED and various events:

LED Color	**LED Mode**		
	Continuously ON	Slow Blink	Fast Blink
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

- **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

- **Red** indicates the state of other Extension/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

- **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1, the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

Status of Features

The LED of a DSS key is activated when the feature assigned to this key is used.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Distinctive Rings

Distinctive Rings are ringing patterns used for distinguishing between different types of call events.

ANANT UCS supports the following types of call events:

1. Internal Call
2. Trunk Call
3. Auto Call Back Call
4. Auto Redial Call
5. Alarm
6. Emergency
7. Operator Alarm
8. Message Wait
9. Ring Test
10. Priority
11. Emergency Conference
12. Conference

With Distinctive Rings, it is possible to use ring cadence of user's choice for each of these call events. For instance, Triple ring can be set for 'Priority Internal Calls' and long rings can be set for 'Alarm Calls'.

Distinctive Rings on SIP Extensions

On SIP Extensions, ANANT UCS supports Distinctive Rings using Alert-INFO field in the INVITE message. To indicate the different call events, ANANT UCS sends the Ring Text for the respective Ring Type.

The default Distinctive Ring Types and their corresponding Ring Texts are given below.

Call Events	Ring Type - T1	Ring Text
Internal Call	Short, Very Slow	internal
Trunk Call	Double	external
Auto Call Back (ACB)	Short, Slow	acb
Auto Redial (AR)	Long, Very Slow	autord
Self Alarm	Long, Fast	selfalarm
Emergency	Long, Fast	emergency
Operator Alarm	Long, Fast	opratoralarm
Message Wait	Short, Fast	msgwait
Ring Test	Short, Slow	test
Priority	Triple	priority
Emergency Conference	Triple	emergencyconf
Conference	Triple	conf

The Ring Text is sent in the Alert-INFO field of the INVITE message and the corresponding Ring Type is played on terminal registered as the SIP Extension, if the terminal supports Distinctive Rings.

The Ring Text is configurable. You can change the Ring Text, if required.

How to configure

At the time of installation, when you select the “Configuring Region” (as per the geographical location of the system), and set the system to default, ANANT UCS loads the country-specific Distinctive Ring Type defined for the selected Region.

Refer the topic “Default Settings” for the default Distinctive Ring Type applied to your country/region.

However, if required, you can change the default Ring Pattern and the Ring Text (for SIP Extensions) loaded by the system.

When you change the Ring Texts for the Ring Types in ANANT UCS, you must configure the same Ring Text in the SIP phones.

Configuring Distinctive Rings

- Login as System Engineer.
- Under **Configuration**, click **System Parameters**.

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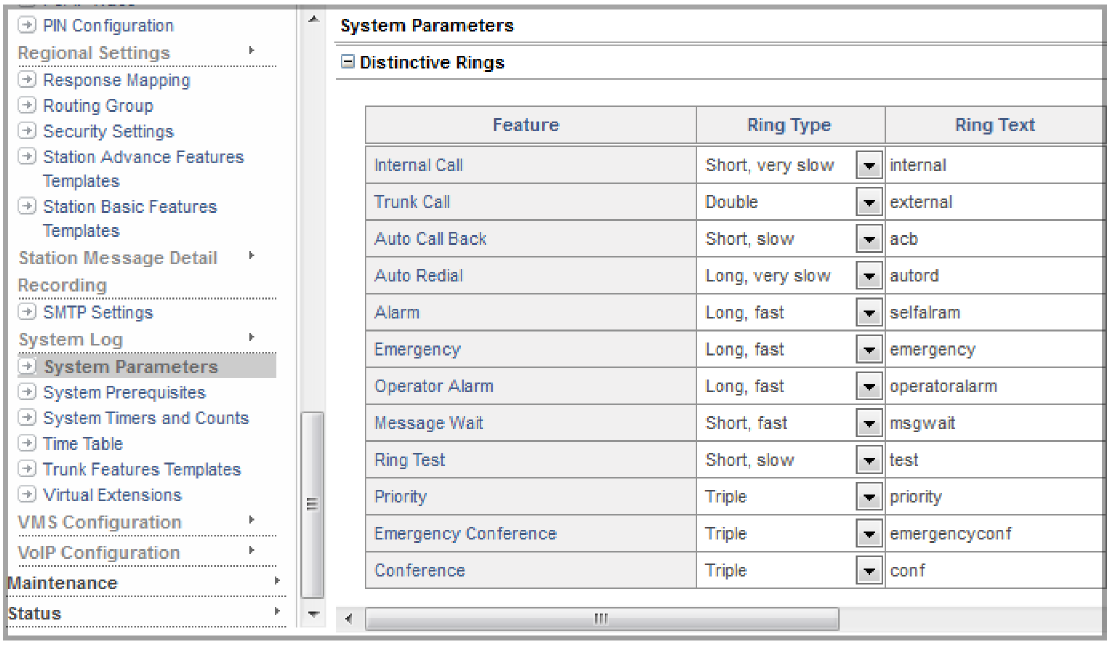
- Click **Distinctive Rings** to expand.



- Select the desired Ring Pattern (Ring Type) for each call event that you want to customize.
- You can also customize the Ring Text of each call event on SIP extensions. The text can be a maximum of 20 alphanumerical characters.
- Click **Submit**.

Do Not Disturb (DND)

Extension users may restrict calls to their extensions in order to work uninterrupted by frequent phone calls. The feature, Do Not Disturb, enables users accomplish this.This feature is useful to extension users who are in the middle of a meeting or any important work that requires their undivided attention.

Using DND users can restrict—all calls, internal calls or external calls. However even if DND is set, users can route their incoming calls to an Intercept Destination. This destination can be the extension users own mailbox or another extension. In this way, extension users can ensure that they do not miss any important calls.

If required, when DND is set, a Stuttered Dial Tone can be played to the user for notification.

DND can be set and canceled by

- Extension users
- Operator for extension users, referred to as DND-Remote.

Doing so, calls — all, internal or external— will be barred. However, the extension user would continue to receive:

- Alarm calls
- Reminder calls
- Auto Call Back calls (Auto Callback as well as Auto Redial)
- Emergency Reporting Calls

Also, the extension user can:

- use all the features of the System
- make Outgoing calls
- make Internal calls to other extensions.

DND has two supplementary features— DND-Override and Privacy from DND-Override.

The 'Do Not Disturb' feature bars calls to the phone on which DND is set. The 'DND-Override' feature breaks this bar and allowing the calls to land on the phone on which DND is set. Protection is also given to the phone on which DND-Override is attempted. If the phone on which DND-Override is attempted has 'Privacy from DND-Override' enabled, the calling phone shall not be able to Override the DND.

When a caller calls a phone on which DND is set, he/she gets Routing tone (Feature tone). The caller can dial DND-Override code. On dialing DND-Override code the call is placed on the called phone and the called phone starts ringing.

The DND-override feature works only if the calling phone has 'DND-Override' feature enabled in its CoS group.

DND-Override will not work if the called phone has ‘Privacy from DND-Override’ enabled in its Class of Service or if the called extension has opted for intercept routing.

So, using DND-Override feature, the users can be reached in case of some emergency despite the DND set on the phone.

- *DND when set/canceled from the SA mode, will not depend on the assigned CoS.*
- *The system supports only single-point DND with Intercept Destination, which means, if the destination extension has also set DND with Intercept Destination, the call will not follow the forwarding path.*

How it works

For this feature to work,

- you must select the "DND Call Type".
- you may select the "Intercept Destination for DND".
- you may select the "DND Text Message" as per your requirement.

DND Call Type

The extension user/Operator can select the type of calls to be restricted while enabling DND. They can select either All, Internal or External Calls.

Intercept Destination for DND

If the user wants that the calls are attended to even if DND is set, System Engineer must configure the Intercept Destination for the user. Incoming calls landing on the extension that has set DND will be routed to the Intercept Destination. This destination can be the users own mailbox or another extension.

DND Text Message

A DND Message is a short Text Message such as 'In a Meeting', 'In a Conference', 'On Vacation'.

When setting DND (also DND-Remote), the extension user/Operator can select an appropriate text message to be displayed to the calling extension.

This DND text message is displayed on the calling extension, only if the calling extension is an Extended IP Phone.

The ANANT UCS supports 9 different DND Text Messages, out of which 8 messages can be changed as per user requirement by the System Engineer. User can select and set on their phones any of the DND messages programmed by the System Engineer.

Let us understand this feature with an example:

A, B and C are extension users.
B has DND-Override in his Class of Service, C does not have this feature.
DND Text messages have been programmed by the System Engineer.

- A has set DND on his extension with the DND Text message 'In Meeting'.
- B calls A.
- As B has DND-Override, the DND message 'In Meeting' set by A appears on B's phone display. B gets routing Beeps.
- To exercise DND-Override, B must dial '4' the feature access code for 'DND-Override' during the routing Beeps.
- B gets Ring Back Tone, if A's extension is free.
- B gets Busy Tone, if A's extension is busy.
- However, if A has Privacy from DND Override, B will get error tone and the DND message set by A appears on B's phone.

If B fails to dial the DND-Override code before the end of the routing beeps, error tone will be played to him.

Now, to take another example,

- C calls A.
- An error tone is played to C and the DND message 'In Meeting' set by A appears on C's phone display.

In the above examples, if A has set E's extension as the Intercept Destination, then calls from B and C will be routed to E's extension.

Feature Interaction:

Call Forward: When DND and Call Forward-Unconditional are set on an extension, Call Forward is given priority. If any other type of Call Forward and DND are set on an extension, DND is given priority. However, if DND with Intercept Destination is set, it will not work. If an extension has set both Call Forward and DND, then Feature Tone will be played to the extension user.

If an extension A has set DND with Intercept Destination as E, incoming calls on A's extension will be routed to E's extension.

If E sets Call Forward or DND with Intercept Destination as B, incoming calls from A's extension will be still routed to E's extension. Only incoming calls on E's extension will be routed to the B's extension.

How to configure

For this feature to work, 'DND' must be enabled in the Class of Service of the group of the extensions which is to be allowed this feature.

Also, 'DND-Override' and 'Privacy from DND-Override' can be enabled in the Class of Service of the extensions to whom this features are to be provided.

Besides these, the System Engineer may program the DND Text Message, Stuttered Dial Tone when DND is set, Intercept Destination for DND as per user requirements.

The user can then select the type of calls for which he wants to set DND.

Configuring DND in Class of Service

In the default Station Basic Feature Template 01 assigned to all extensions of the ANANT UCS, the default CoS group 01 has DND enabled. So, all extensions of ANANT UCS can set and cancel DND. DND-Override and Privacy from DND-Override are disabled in the default CoS group 01. So, none of the extensions can use DND-Override, or be exempt from DND.

While it makes sense to offer all extensions DND, providing DND-Override and Privacy from DND also to all extensions will not serve the purpose of DND.

Decide which extensions are to be allowed 'DND', which are to be allowed 'DND-Override', and which are to be allowed 'Privacy from DND-Override'. Generally, DND-Override is allowed to the Operator extension. It may be allowed to extensions of persons in senior positions in the organization. Similarly, Privacy from DND-Override may be allowed to persons in senior positions in the organization.

If you want to allow DND to all extensions, retain the default CoS group 01 in Station Basic Feature Template 01. However, if you want to allow DND only to selected extensions, disable this feature in the default CoS group 1.

Now, to assign DND to selected extensions, follow these steps:

- Define a CoS group with DND enabled.
- Prepare a Station Basic Template with this CoS group applicable in all the time zones.
- Assign this newly prepared Template to the extensions on which 'DND' is to be enabled.

Similarly, if 'DND-Override' is to be to be allowed to the Operator and a few other extensions, follow these steps:

- Define a CoS group with DND-Override enabled. If DND is also to be allowed, enable both DND-Override and DND in this CoS group.
- Prepare a Station Basic Template with this CoS group applicable in all the time zones.
- Assign this newly prepared Template to the Operator extension on which 'DND-Override' is to be enabled.

Repeat the same steps to allow 'Privacy from DND-Override' to selected extensions. For extensions that are to be allowed 'DND' as well as 'Privacy from DND-Override', enable both features in the CoS group in the Station Basic Feature Template applied on these extensions.

Similarly, for extensions that are to be allowed 'DND', 'DND-Override' and 'Privacy from DND-Override', enable all three features in the CoS group that you prepare for these extensions.

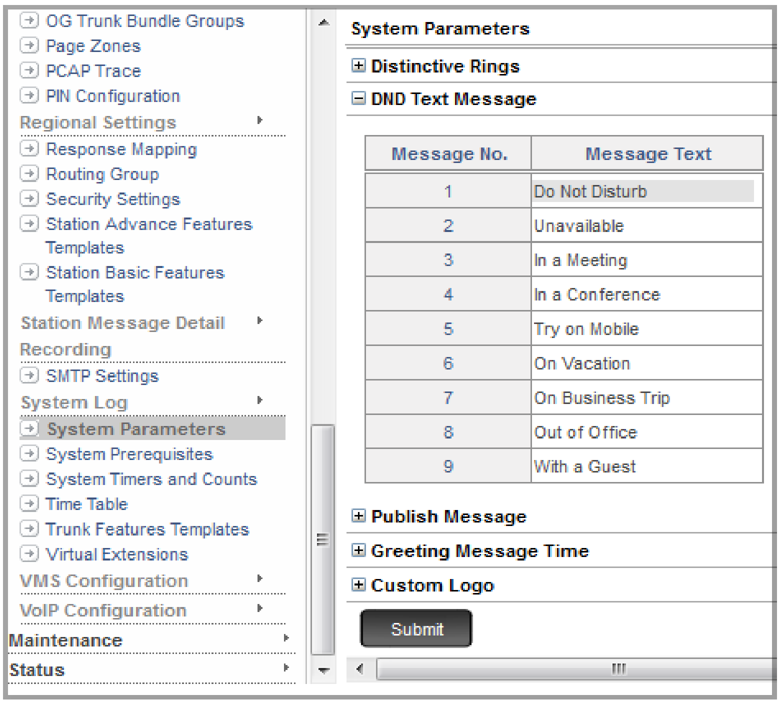
Refer the topics “Class of Service (CoS)” and “Station Basic Feature Template” for detailed configuration and instructions on how to prepare a CoS in the Station Basic Feature Template and how to apply this template on extensions.

Configuring DND Messages

- Login as System Engineer.

- Under **Configuration**, click **System Parameters**.

- Click **DND Text Message** to expand.



- All the default text messages appear in the DND message field. You may change the DND text messages 2 to 9 as per your requirement. Click the field and enter your custom DND text message.

- Click **Submit.**

Configuring Stuttered Dial Tone

If required by users, then the system will play Stuttered Dial Tone on the extension when DND is set.For instructions, see “System Parameters”.

Configuring Intercept Destination for DND

By default, the Station Advanced Feature Template 01 is assigned to all extensions of the ANANT UCS and in this template the Intercept Destination is configured as None.

You can either configure the Intercept Destination in this template or you may select another template and customize it as per your requirement. For instructions see “Station Advanced Feature Template”.

How to use

DND set/canceled by Extension Users

For Extended IP Phone Users

To set DND:

- Press the 'DND' Key.
  OR
- Dial **18**
- Scroll to select the type of call:
  - All calls
  - Internal calls
  - External calls

- Press 'Enter' key.
- You get a text message 'DND Set' on the phone's display and confirmation tone.

To select a DND Message

- Press the 'DND' Key.
  OR
- Dial **18**
- Scroll to select the Set DND Message option.
- The list of DND messages appear on the phone's display:

  - Do Not Disturb
  - Unavailable
  - In a Meeting
  - In a Conference
  - Try on Mobile
  - On Vacation
  - On Business Trip

    - Out of Office
    - With a Guest
- Scroll to the desired option and press 'Enter' key.
- You get a text message 'DND Set' on the phone's display and confirmation tone.
- Go Idle or you get dial tone after the confirmation tone.

To cancel DND:

- Press the DND Key again.
- The following options appear on the phone's display
    - All calls
    - Internal calls
    - External calls
    - Cancel DND

- Select Cancel DND and press 'Enter' key.
  OR
- Dial 1**8-0**
- You get a text message 'DND Canceled' on the phone's display and confirmation tone.

DND-Remote

For Extended IP Phone Users

To set DND for an extension user:

- Press the DSS Key assigned to DND-Remote.
- Enter the extension number.
- Scroll to select the type of call:
    - All calls
    - Internal calls
    - External calls

- Press 'Enter' key.
- You get a text message 'DND Set on <Extension Number>' and confirmation tone.
- Go Idle or you get dial tone after confirmation tone.

To select a DND Message:

- Press the DSS Key assigned to DND-Remote.
- Enter the extension number.
- Scroll to select the option Set DND Message
- The list of DND messages appear on the phone's display:
    - Do Not Disturb
    - Unavailable
    - In a Meeting
    - In a Conference
    - Try on Mobile
    - On Vacation
    - On Business Trip
    - Out of Office
    - With a Guest

- Scroll to the desired option and press 'Enter' key.
- You get a text message 'DND Set' on the phone's display and confirmation tone.
- Go Idle or you get dial tone after the confirmation tone.

To cancel DND-Remote:

- Press the key assigned Remote-DND function.
- Enter the Extension Number.
- Scroll to select the message 'Cancel DND'.
- Press 'Enter' key.
- You get a text message 'DND Canceled on <Extension number>' and confirmation tone.
- Go Idle or you get dial tone after confirmation tone.

DND-Override

For Extended IP Phone Users

To select a DND-override:

- Dial an extension number.
- You get routing beeps and a DND Text message (if using Extended IP Phone)
- Dial '4', the DND-Override Code, during the message or the routing beeps.
- The called extension will start ringing.
- You will get Ring Back tone.
- If the dialed phone is busy, you will get busy tone.

DSS Call Pick-Up

DSS Call Pick-Up allows Extended IP Phone users to answer calls ringing on other extensions or incoming calls on trunks by pressing the DSS Keys assigned to those extensions/trunks on their phones.

There are two types of DSS Call-Up:

- **DSS Call Pick-Up-Station** - Internal or external calls ringing on any extension, can be picked-up by pressing the DSS Key assigned to that extension on your phone.

- **DSS Call Pick-Up-Trunk** - Incoming calls on any trunk for any extension can be picked-up by pressing the key assigned to that trunk on your phone.

For SIP Trunks, DSS keys can be assigned for each Call Appearance.

- *If you have assigned a DSS key to All the Call Appearances for SIP Trunks, you will only be able to grab the trunk to make outgoing calls. You will not be able to pick up incoming calls on these trunks.*

How it works

For this feature to work, you must:

- enable the desired Call Pick-up in the CoS of the extension user.
- assign DSS Keys with LED to the desired extensions/trunks on their Extended IP Phone.

This is how DSS Call Pick-Up works:

- Extension user A has configured a DSS Keys for extension 2007 and SIP trunk 1 on his/her Extended IP Phone.

- When a call lands on extension 2007 and it rings, the DSS Key assigned to 2007 blinks fast in **Blue** color to indicate that the extension is ringing. A presses the DSS Key to pick-up the call ringing on extension 2007.

  If DSS Call Pick-Up-Station is not enabled in the CoS assigned to extension user A, the DSS Key blinks fast in **Red** color to indicate that the extension is ringing. However, A will not be able to pick-up the call ringing on extension 2007.

- Similarly when there is an incoming call on SIP trunk 1, the DSS Key assigned to SIP trunk 1 blinks fast in **Violet** color to indicate that there is an incoming call on the trunk. A presses the DSS Key to answer the call on the trunk.

  If DSS Call Pick-Up-Trunk is not enabled in the CoS assigned to extension user A, the DSS Key blinks fast in **Red** color to indicate that the there is an incoming ringing call. However, A will not be able to pick-up the incoming ringing call on that trunk.

Feature Interactions:

- **Call States:** DSS Call Pick-Up-Station and DSS Call Pick-Up-Trunk are possible only when the calls are in ringing state.

- **Priority**: If multiple calls are ringing on an extension, when you press the DSS Key assigned to that extension, you will be connected to the ringing call with the highest priority. To know more, see "Priority".

How to configure

To provide this feature to extension users, you must

- enable these features in their Class of Service to be assigned to the users. For instructions, see "Class of Service (CoS)" and "Station Basic Feature Template".

- configure the DSS Keys for the desired extensions and trunks on their Extended IP Phone. To know more about assigning DSS Keys, see "DSS Keys Programming".

How to use

For Extended IP Phone Users

To use DSS Call Pick-Up-Station:

- When the DSS key assigned to the station blinks fast in blue color to indicate that the station is ringing, press the DSS Key.
- You are in speech with the calling party.
- You may talk.

To use DSS Call Pick-Up-Trunk:

- When the DSS key assigned to the trunk blinks fast in violet color to indicate that there is an incoming call on the trunk, press the DSS Key.
- You are in speech with the calling party.
- You may talk.

Dynamic Lock

Dynamic Lock allows extension users to change the Toll Control Levels (Calling Permissions) of their extensions on their own by dialing a code.

The System Administrator/Operator can also change the Toll Control Levels of extensions using Dynamic Lock.

With this feature, extension users can prevent misuse of outgoing call facility from their extensions, especially in their absence.

Dynamic Lock also forms the basis of 'Call Privilege', which is feature of the Hotel Application of ANANT UCS. Refer the *ANANT UCS Hospitality System Manual* to know more.

There are four types of Toll Control Levels, starting from Level 0 to Level 3 that can be set for extension phones.

For each Toll Control Level from 0 to 3, a 'Call Privilege'[103] is to be assigned and corresponding numbers strings to be allowed and number strings to be denied for each Call Privilege are to be programmed.

- **Toll Control - Level 0** is Time Zone based, wherein the Call Privilege Type must be defined for each Time Zone, that is, Working Hours, Break Hours and Non-Working Hours. For instance, you may define 'All Calls' as Call Privilege for Working Hours, 'Local Calls' as Call Privilege for Break Hours and 'No Calls' as Call Privilege for 'Non-Working' Hours.

  By default, Call Privilege 'No Calls' is selected for all three Time Zones.

- **Toll Control - Level 1** is not based on Time Zones. By default, the Call Privilege Type for this level is 'No Calls'.

- **Toll Control - Level 2** is not based on Time Zones. By default, the Call Privilege type set for this level is 'No Calls'.

- **Toll Control - Level 3** is not based on Time Zones. By default, Call Privilege 'No Calls' is selected for this level.

  The Call Privilege for each of the above Toll Control Levels can be redefined according to user requirements. For example, Toll Control Level 3 can be programmed for allowing all types of calls by selecting 'All Calls' as Call Privilege Type and Level 0 can be programmed to allow only Local Calls, by programming the strings of 'Local Numbers'.

  Refer the feature description for “Toll Control” to know more.

  Extension users who are allowed the Dynamic Lock feature in their Class of Service, can set the Toll Control Level in two ways:

  - **Manually:** the extension user changes the Toll Control Level of the extension whenever s/he wants by dialing the feature access code.

    For example, an extension user having Toll Control Level 2 (National calls) can restrict long distance dialing on his/her extension by setting the Toll Control Level to 1 (Local calls) before leaving the workplace. On return, the user can restore the previous Toll Control Level, by setting it back to Level 2.

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103. The Call Privilege types are: No Calls, Local Calls, Regional Calls, National Calls, International Calls, All Calls and Limited Calls.

Thus the extension user sets Dynamic Lock, s/he manually selects the desired Toll Control Level for his/her extension and restores the original Toll Control Level assigned to the extension.

- **Automatically:** the extension user changes the Toll Control Level of the extension using the Dynamic Lock Timer. The user sets the Timer to the desired number of minutes. On the expiry of this Timer, the system restores the original Toll Control Level assigned to the extension.

  For example, an organization has defined Toll Control Level 0 as Local Calls, and Level 3 as All Calls. An extension user of this organization is assigned Level 0. When this extension user wants to make international calls, he sets the Dynamic Lock Timer and selects Toll Control Level 3. At the end of the timer, Level 3 gets locked and Toll Control Level 0 is reapplied on the extension phone.

- *The changing of Toll Control level requires the user to dial the 4-digit User Password. The system will not accept the default User Password (1111). The extension user must first change the default User Password.*
- *The Dynamic Lock Timer can be set to a maximum of 99 minutes.*
- *The Dynamic Lock Timer must be set to '00' when using Manual Dynamic Lock.*
- *Dynamic Lock when set/canceled from the SA mode, will not depend on the assigned CoS.*

How it works

The Pre-requisites

- The Toll Control Levels 0 to 3 are programmed in the Station Basic Feature Template applied on the extension.

- Dynamic Lock is allowed to the extension in its Class of Service.

The Process

For Dynamic Lock - Manual

- The user of extension A sets the Dynamic Lock manually by entering the User Password and selecting the desired Toll Control Level.

  OR

- The Operator sets Dynamic Lock manually for an extension by entering the extension number and selecting the Toll Control Level.

For Dynamic Lock - Automatic

- The user of extension A sets the Dynamic Lock by entering the User Password, setting the Dynamic Lock Timer, and selecting the desired Toll Control Level.

  OR

- The Operator sets Dynamic Lock for an extension by entering the extension number, setting the Dynamic Lock Timer, and selecting the Toll Control Level.

- Now, whenever a call is made from extension A, the system checks for Toll Control Level.

- The system then checks the associated Lists of allowed and denied numbers.
    - If the Toll Control Level is 0, then Toll control is time zone based, that is, working hours, break hours and non-working hours. The outgoing call is allowed/denied as per the Call Privilege and the corresponding Allowed and Denied Number List programmed for that time of the day by the System Engineer.
    - If the Toll Control Level is 1, 2, 3 the outgoing call is allowed/denied as per the Call Privilege and the corresponding number list programmed for each level.
- If Dynamic Lock - Automatic has been set by user/Operator, the system waits for the duration of the Dynamic Lock Timer set for the extension. At the end of each outgoing call made during the period of this Timer, the system will restart the Timer again. The system will change the Toll Control back to the previous Level when no outgoing call is made till the expiry of this Timer.
- If Dynamic Lock - Automatic has been set by user/Operator, and an internal call is made during the period of the Dynamic Lock Timer, the system will check for the 'Decrement Dynamic Lock Timer for Internal Calls' feature in the Class of Service of allowed to the extension. If this feature is enabled, the system will start the decrement of the Dynamic Lock Timer. The system will change the Toll Control back to the previous level on the expiry of this Timer. However, if the 'Decrement Dynamic Lock Timer' feature is disabled in the Class of Service, the system will reset the Toll Control as described in the previous step.
- If Dynamic Lock - Manual has been set, the extension user/Operator must set the Toll Control Level back to the previous Level.

Feature Interactions

- **Redial and Auto Redial:** The system will check for Toll Control Level when an extension on which Dynamic Lock is set, attempts Redial. The system will not check the same for Auto Redial.
- **Emergency Number Dialing:** All extensions will be able to dial Emergency numbers always, regardless of the Toll Control set on them.

  ANANT UCS provides for separate programming of Emergency Numbers, which remain unaffected by Dynamic Lock set on the phones. Refer the topic “Emergency Dialing” to know more about this feature.

How to configure

For this feature to work, it must be enabled in the Class of Service of the extensions; Toll Control Level must be programmed in the Station Basic Feature Template of the extensions. The user must change the default User Password.

Dynamic Lock in Class of Service

In the default Station Basic Feature Template 01 assigned to all extensions of the ANANT UCS, the default CoS group 01 has Dynamic Lock enabled. So, all extensions can set Dynamic Lock.

In the default CoS group 01, 'Decrement Dynamic Lock Timer for Internal Calls' is disabled.

Retain the default template, if you want to allow this feature to all extensions and keep the Decrement Timer disabled.

If you want to deny Dynamic Lock to all extension, simply disable this feature in the default CoS group 01 of Station Basic Feature Template 01.

If you want to allow Dynamic Lock and/or the Decrement Dynamic Lock Timer for Internal Calls only to selected extensions, then follow these steps:

a. Define a new CoS group with Dynamic Lock and the Decrement Dynamic Lock Timer for Internal Calls enabled.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the time zones.

c. Assign this newly prepared Template to the extension on which 'Dynamic Lock' and 'Decrement Dynamic Lock Timer' for Internal Call is to be allowed.

Similarly, if you want to deny Dynamic Lock/Decrement Dynamic Lock Timer to selected extension, prepare a new Template with this feature disabled in the CoS group. Assign this feature to those extensions which are to be denied this feature.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for programming instructions.

Toll Control Levels in Station Basic Feature Template

Program the Toll Control Levels in the Station Basic Feature Template that are assigned to the extensions which are to be allowed the Dynamic Lock feature. Refer the topic "Toll Control" for instructions.

How to use

Dynamic Lock, Manual and Automatic, can be set by extension users as well as from their own extensions or from the SA mode by the Operator.

The extension user/Operator must first set the Dynamic Lock Timer and then change the Dynamic Lock Level.

To set Dynamic Lock-Manual, the extension user/Operation must set the Dynamic Lock Timer to ***00.***

Recall that

- *When the Dynamic Lock-Manual is set (Timer set to 00), the extension user/Operator must dial the feature access code to restore the previous Toll Control Level.*

- *When Dynamic Lock-Automatic is set (Timer set to desired number of minutes), the system will restore the previous Toll Control Level at the end of the Timer.*

- *The extension user must change the default User Password to be able to set the Dynamic Lock on his/her extension. Refer the topic "User Password" for instructions on changing the password.*

Changing Dynamic Lock by Extension Users

For Extended IP Phone Users

To set Dynamic Lock-Manual:

- Press DSS Key assigned to Dynamic Lock.
  OR
- Dial **142**.
  OR
- Enter Phone Menu by pressing 'Enter' key.
- You get the prompt <Enter User Password>
- Enter your User Password.
- You get the prompt: <Lock Timer = XX Minutes>.
- Enter 00
- You get the Text message <Lock Timer = 00 Minutes> and confirmation tone.

To set Dynamic Lock-Automatic:

- Press DSS Key assigned to Dynamic Lock.
  OR
- Dial **142**.
  OR
- Enter Phone Menu by pressing 'Enter' key.
- You get the prompt <Enter User Password>
- Enter your User Password.
- You get the prompt: <Lock Timer = XX Minutes>.
- Enter the desired number of minutes (max. 99 minutes).
- You get the Text message <Lock Timer = xx Minutes> and confirmation tone.

To set Dynamic Lock Level:

- Press DSS Key assigned to Dynamic Lock.
  OR
- Dial **141**.
  OR
- Enter Phone Menu by pressing 'Enter' key.
- You get the prompt: <Enter User Password>.
- Enter your User Password.
- Select the desired Toll Control Level:

  - Toll Control Level 0
  - Toll Control Level 1
  - Toll Control Level 2
  - Toll Control Level 3

- Press Enter key.
- You get the text message 'OK!' and a confirmation tone.

If you are still working with the default User Password, the system will prompt you to 'Change User Password' when you attempt to set Dynamic Lock. Change your User Password first, before you use this feature.

Changing Toll Control Level for an Extension (SA mode)

For Extended IP Phone Users

To set Dynamic Lock-Manual:

- Press DSS key assigned to Dynamic Lock.
  OR
- Enter Phone Menu.
- You get a text message 'Enter Room/Phone Number'.
- Enter the extension number for which you want to set Dynamic Lock.
- Select the option 'Change Lock Timer'.
- You get the following prompt: 'XX Minutes'.
- Enter 00
- You get the text message <Lock Timer = 00 Minutes> and confirmation tone.

To set Dynamic Lock-Automatic:

- Press DSS key assigned to Dynamic Lock.
  OR
- Enter Phone Menu.
- You get a text message 'Enter Room/Phone Number'.
- Enter the extension number for which you want to set Dynamic Lock.
- Select the option 'Change Lock Timer'.
- You get the following prompt: 'XX Minutes'.
- Enter the desired number of minutes (max. 99)
- You get the text message <Lock Timer = xx Minutes> and confirmation tone.

To set Dynamic Lock Level:

- Press DSS key assigned to Dynamic Lock.
  OR
- Enter Phone Menu.
- You get a text message 'Enter Room/Phone Number'.
- Enter the extension number for which you want to set Dynamic Lock.
- Select the option 'Change Toll Ctrl. Level'
- Press Enter key.
- Scroll to select the desired Toll Control Level:
  - Toll Control Level 0
  - Toll Control Level 1
  - Toll Control Level 2
  - Toll Control Level 3

- Press 'Enter' key.
- You get the confirmatory text message 'OK!' and confirmation tone.

Emergency Calls (911) - Reporting to PSAP

Make sure:

- *you have selected the **Region** as USA or Canada.*
- *In the Emergency Table for the Number 911, you have assigned the outgoing trunk bundle group with only SIP Trunks. For instructions, see "Configuring Emergency Number Dialing".*

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 Users license. For details, refer "Licenses Supported in ANANT UCS".

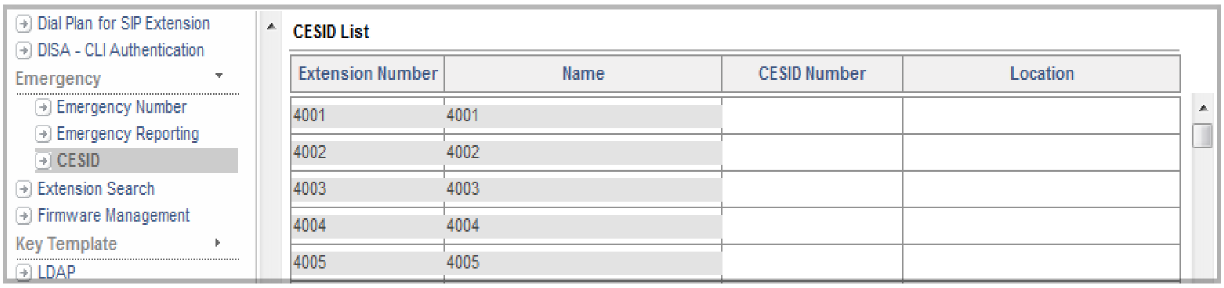
ANANT UCS allows you to assign Customer Emergency Services Identification (CESID) number to each extension user. When the user dials the Emergency number 911, the system sends the caller's CESID number to the network and the call is routed to the Public Safety Answering Point (PSAP).

The PSAP stores information, such as name, location, and CESID number of the users in an Automatic Line Information (ALI) database. The PSAP uses the CESID number that you have assigned to an extension to locate the user's information in the ALI database. Hence, the PSAP can provide users the desired service without delay.

The CESID numbers are provided by the Service Provider. In case these are not provided, you can assign DDI numbers to extension users. See "Direct Dialing-In (DDI)" to know more.

Configuring CESID List

- Login as System Engineer.
- After you have selected the **Region** and configured the **Emergency Number** table.
- Under **Configuration**, click **Emergency**.
- Click **CESID.**



- The list of **Extension Number** and Extension **Name** of the users appear as configured by you.
- For each Extension Number assign the **CESID Number** and enter his/her **Location**.

  The CESID Number can a maximum of upto 12 digits and the Location can be a maximum of upto 18 alphanumeric characters.
- Click **Submit.**

Emergency Conference

Emergency Conference enables you to establish a conference between a pre-defined group of extensions using a feature access code.

This feature can be used to call and consult with a group of people in emergency situations.

The number of parties that can be included in an Emergency Conference group depends on the Multiparty Conference capacity of the ANANT UCS. For details, see “Conference-Multiparty”.

How it works

For this feature to work, you must do the following:

- First decide the key persons in the organization who should be parties to the Emergency Conference.
- Form a Department Group with the extensions of these key persons as members. A single Department Group can have up to 32 extensions. For more information on forming Department Groups, see the topic “Department Call”.

For example, you have formed a Department Group for Emergency Conference, with the extensions A to G as members. The Access Code assigned to the Department Group is 3901. H is the initiator of the conference as Emergency Conference is enabled in the Class of Service assigned to his/her extension.

Another Emergency Conference is formed with Department Group 3902, having extensions J to P as members and J as the initiator of the Conference as Emergency Conference is enabled in the Class of Service assigned to his/her extension.

Now, extension H wants to initiate an Emergency Conference.

This is how the feature will work:

Initiating an Emergency Conference

- H dials the feature access code for Emergency Conference, followed by access code of the Department Group (3901).

- All extensions in the Department Group (extensions A to G) which are free will start ringing. The system will play *Emergency Conference* ring (default: Triple Ring) on the Extensions. Extensions that are busy will not be included in the call.

  If there are Extended IP Phone extensions in the group, and these phones have a Call Appearance free, the system will ring these extensions on the free Call Appearance, but will not wait for the extensions to become free.

When the Emergency Conference is initiated, the system will ring on the extensions that are not occupied in any other single multi-party or 3-party conference.

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- Extension A goes Off-Hook to answer the call first. A gets connected to the initiator of the conference, extension H.

- Two-way speech is established with extension A and H. All other extensions continue to ring.

- When another extension, B goes Off-Hook to answer the call, A and H get a beep, and three-way speech is established between A, B, H.
- Thus, whenever a new member joins the conference, all other extensions already in conference will get a beep, if the check box *Play Beep when Conference/Dial-In Conference begins* is enabled in the "System Parameters".
- If the conference initiator, extension H, goes idle, all other extensions in the conference will still be in conversation.

Merging Emergency Conferences

- Only the initiator of the conference, extension H, can merge the conference using the phone menu.
  - While in Conference, H presses the 'Conference' Key.
  - From the Multiparty Conference Menu, H selects Merge Conference to merge with another ongoing Emergency Conference.
  - H selects 3902 from the list of ongoing Emergency Conferences. Both the Emergency Conferences (3901 and 3902) are merged.

Canceling an Emergency Conference

- Extension H, can cancel the conference. The initiator of the conference can cancel the conference at two stages:
  - When speech is established with one or more member extensions of the Emergency Conference department group.
    OR
  - During Ring Back Tone, as the system rings on the extensions of the group, after the initiator of the conference has dialed the feature access code.
- To cancel the Emergency Conference, extension H must dial the feature access code for Cancel Conference, 190 (default).

How to configure

To provide this feature to extensions,

- You must enable the feature **Emergency Conference** in the "Class of Service (CoS)" of the extensions in their "Station Basic Feature Template". By default, this feature is enabled on all extensions, so all extensions can use this feature.
- If the extension you are providing this feature is an Extended IP phone, you may configure a DSS key on the phone with this feature.
- You must also create a Department Group as Emergency Conference group. For instructions, see "Department Call".
- By default, the system plays a beep when the Emergency Conference starts. If you do not want the beep to be played, you must disable the check box *Play Beep when Conference/Dial-In Conference begins*. For instructions, see "System Parameters".

This check box is common for other features like "Conference-Multiparty", "Conference Dial-In".

- By default, the system plays *Triple Ring* as Ring Type for Emergency Conference. If necessary, you may configure a different ring type. For more information and for instructions, see "Distinctive Rings".

How to use

For Extended IP Phone Users

To initiate an Emergency Conference:

- Press DSS Key assigned to Emergency Conference.
  Or
- Dial **1177**
- Dial Department Group Number.
- All free extensions in the group will ring.
- You get connected to the extensions that answer.

- *If no resources are free, you will get the 'Conf. Resource full' message on your LCD.*
- *You can also initiate an Emergency Conference using "Direct Inward System Access (DISA)".*

To merge Emergency Conferences:

- While in Conference, press the 'Conference' Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option 'Merge Conferences'.
- The LCD displays list of Department Number and Group Name of ongoing Emergency Conferences.
- Select the desired Department Group number and press the Enter Key.
- Both the conferences will be merged.

Emergency Conferences can be merged by an Extended IP Phone users only through the Phone Menu.

To cancel an Emergency Conference while in speech with one or more extensions:

- While in Conference, press the 'Conference' Key.
- The Multiparty Conference menu appears on the LCD.
- Select the option 'Terminate Conference' and press the Enter Key.
  OR
- Press the DSS key assigned to Terminate Conference.
- All the participants will get an Error Tone and the system resource occupied by the conference will be freed.

To cancel an Emergency Conference while the extensions are ringing:

- Go ON-Hook during Ring Back Tone.
- All extensions in the conference group will stop ringing.

- *The initiator or participants can cancel an emergency conference.*
- *Conference is a licensed feature. Decide the number of conferences you want to conduct and purchase the license accordingly. Refer the topic "Licenses Supported in ANANT UCS" to know more.*

Emergency Detection and Reporting

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 Users license. For details, refer “Licenses Supported in ANANT UCS”.

When an emergency call is made from an extension, the system dials out the number using any of the free trunks selected for routing Emergency Numbers. Since the number is dialed out by the System, the Emergency Service that attends to the call will be able to locate the System, but not the extension that made the call.

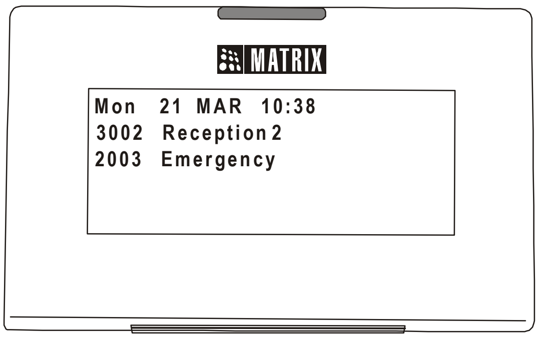
Similarly, the Operator too has no way of knowing which extension made the call, thus making it difficult to quickly reach and provide help to the extension that made the emergency call.

With the Emergency Detection and Reporting feature, the Operator can know from which extension the emergency call is being made. Whenever an Emergency call is made by an extension user, the system detects and reports it to the Operator extension.

How it works

When an extension of ANANT UCS makes an emergency call by dialing an Emergency Number:

- The system hunts for a free trunk in the OGTBG selected for routing the emergency number, and dials out the number from a free trunk.
- Simultaneously, the system informs the Operator by ringing on the Operator extensions for the duration of the *Emergency Reporting Call - Ring Timer* (configurable; default: 10 minutes).
- If Operator has an Extended IP Phone, it will ring continuously, and an emergency message will be displayed on the LCD.



The emergency message shows the number of the extension which has made the emergency call, in this case, extension 2003.

- To acknowledge the Emergency call the operator must press the enter key. The acknowledged Emergency calls are logged into the System Activity Log.
- If the Emergency Call is not acknowledged by the operator, the emergency call is logged into the Emergency Alarms Log. To know more about the Emergency Alarms Log, see “Emergency Alarms Log” at the end of the topic.

Also see the topics “Configuring Emergency Number Dialing” and “Emergency Dialing”.

How to configure

Configuring Emergency Reporting

- Login as System Engineer.
- Under **Configuration**, click **Emergency**.
- Click **Emergency Reporting**.

- **Emergency Dialing Reporting:** By default this check box is enabled. The system will detect the extension that makes an emergency call.
- **Time Table:** A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week.

  You can define and select the Time Table for the Emergency Reporting Group as per you requirement.

  There are 8 different Time Table templates to select from. By default, the Time Table 1 is selected. In Time Table 1, six days of the week - Monday to Saturday -have working hours from 9:00-18:00, break hours from 13:00-14:00 hours and non-working hours from 18:00 to 09:00. Sunday is a holiday, with all three Time Zones set to 00:00 hours.

  You may also customize the default Time Table 1 OR customize and assign a different Time Table. Refer to “Time Tables” for more details.
- **Emergency Reporting Group:** By default **Operator assigned to extension** is selected as the landing destination group for placing emergency reporting calls, for all the time zones.

  You can either customize the default group OR customize and assign a different group.

  You can select a different group for each time zone, that is Working Hours (WH), Break Hours (BH) and Non-working Hours (NH).
- Click **Submit**.

If required, you can change the Emergency Reporting Call-Ring Timer, see “System Timers and Counts”.

Emergency Alarms Log

Emergency Alarms Log is the log of unacknowledged Emergency Calls.

When an Emergency call is made by any extension user and is not acknowledged, that call is logged into the Emergency Alarms Log.

The log can be viewed by an Extended IP Phone user, using the DSS Key assigned to Emergency Alarms Log only. When an Emergency call is made by any extension user, the LED of the DSS Key glows continuous RED.

To view the log from an Extended IP Phone user,

- Press the DSS Key assigned to Emergency Alarms Log.
- A list of the last 20 unacknowledged Emergency calls appears with the following details:
  - Extension number from which the Emergency call was made.
  - Date and Time when the Emergency call was initiated from that Extension.
- Press the enter key to acknowledge the Emergency Call. The message "Emergency Acknowledged" appears on the screen.
- The system plays the Confirmation Tone followed by the Dial Tone.
- The acknowledged Emergency call is removed from the Emergency Alarms Log and is logged into the System Activity Log with the details of the extension that acknowledged the call.

Emergency Dialing

If you are using the system in the Hospitality mode, to dial the Emergency Number 911, you must purchase the E911 Users license. For details, refer "Licenses Supported in ANANT UCS".

ANANT UCS supports dialing of Emergency number immediately without any blocking.

When an extension user dials an Emergency number, the system will hunt for a free trunk from the outgoing trunk bundle group selected for the emergency number. See *"Configuring Emergency Number Dialing"*.

The system will not apply any of the following on the extension dialing the Emergency number:

- Toll Control (Allowed Denied Numbers, Dynamic Lock)
- Call Budget (even when call budget is consumed)
- Call Duration Control
- Automatic Number Translation

The system will allow the extension to dial the Emergency number even in the following conditions:

- the extension is in Off-Hook state.
- the extension is in Standby Mode.
- the extension has grabbed the trunk line using Trunk access code.
- the call state is in any state: Ringing, Busy, Error, Confirmation.
- the keypad of the extension phone is locked.

Emergency Numbers will always be out dialed through the OGTBG you have selected for the numbers.

If the SIP trunk through which the number is to be routed is disabled, Emergency Number will not be out dialed.

Emergency dialing will not work if Mains Power to ANANT UCS fails.

How to configure

The Emergency numbers are fixed as per the Region where ANANT UCS is installed, you can add emergency numbers, as required. For instructions, see "Configuring Emergency Number Dialing".

How to use

To dial an Emergency number,

- Go Off-Hook
- Dial the Emergency Number
  OR
- Dial Trunk Access Code-Emergency Number
  For example: Dial **0-112**

- *Wherever the Trunk Access Code conflicts with the Emergency Number, the emergency number should be dialed after dialing the Trunk Access Code.*
- *Let us take the example of Australia, where the emergency number is 000 and the trunk access code is 0. Now, when an extension user of ANANT UCS located in Australia dials ‘0’ of the emergency number, the system will consider it as trunk access code and will apply the trunk access code logic. Therefore, in such cases, the extension user must first dial the Trunk Access Code and then the Emergency Number. In this case, the extension user must dial 0-000 for emergency number dialing, so that the system will not wait for the Conflict Timer to apply the Trunk access code logic.*

Extended IP Phone/VARTA UC Client - Operation

Matrix offers the following proprietary Extended IP Phones/Mobile UC Clients:

- SPARSH VP248, The High-Definition Edge to your IP Communication. For detailed description, see *"Matrix SPARSH VP248"*.
- SPARSH VP310, The Executive IP Phone. For detailed description, see *"Matrix SPARSH VP310"*.
- SPARSH VP330, Intuitive Touchscreen IP Phone. For detailed description, see *"Matrix SPARSH VP330"*.
- SPARSH VP510, The Premium IP Phone. For detailed description, see *"Matrix SPARSH VP510"*.
- SPARSH VP210, the Entry Level IP Phone. For detailed description, see *"Matrix SPARSH VP210"*
- Extended SPARSH VP710, the Smart Video IP Phone. For detailed description, see *"Matrix Extended SPARSH VP710"*.
- Mobile UC Clients - VARTA ADR100, Mobile UC Client for Android Smart Phones and VARTA AMP100, Mobile UC Client for iPhones. See *"Matrix VARTA ADR100 UC Client"* and *"Matrix VARTA AMP100 UC Client"*
- MATRIX VARTA WIN200, UC Client for Windows. For detailed description, see *"MATRIX VARTA WIN200"*.

To know the list of featured supported, refer to *"ANANT UCS Features Supported in Terminals"*.

Matrix SPARSH VP248

The Matrix SPARSH VP248, the proprietary Extended IP Phone is a feature-rich, VoIP (Voice over Internet Protocol) phone, providing voice communication over IP network. It looks and works like any normal phone, having all the traditional phone features such as redial, speed dial, call transfer, call hold, call forward, conference, and so forth. It allows you to make and receive calls using the handset, the headset (if connected) and the speaker.

The models of SPARSH VP248 are:

- **SPARSH VP248S** - the standard model, with a 2-line x 24-character LCD display.
- **SPARSH VP248P** - the premium model, with a 6-line x 24-character LCD display.

It is a powerful extension, supporting a host of phone and ANANT UCS features, as listed below.

IP Phone Features

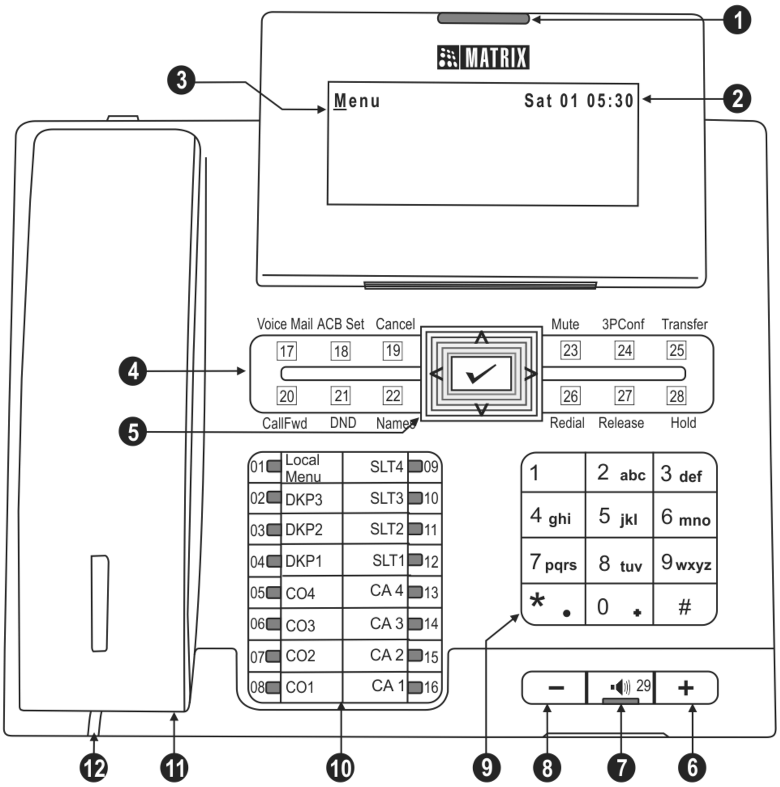
- Status of other ports (Tri-color LED indication)
- Programmable Direct Station Selection (DSS) Keys and Feature keys
- LCD notification messages
- Ringer Tune selection
- Adjustable Speech level
- Adjustable Ringer Volume
- Adjustable Backlight and Contrast levels
- Hands-free operation - Speaker key and headset connectivity.
- Call Logs - last 20 Missed, Answered and Dialed Calls.

- Message Paging
- Menu based operation of ANANT UCS features
- Multiple Language support.

The SPARSH VP248 supports ANANT UCS features. A few of these are listed below:

- Abbreviated Dialing
- Auto Answer
- Call Chaining
- Call Cost Display
- Call Duration Display
- Call Mute
- Dialed Number Directory
- Directory Dialing by Name
- Dynamic Lock
- Forced Answer
- Keypad Lock
- Message Paging
- Off-Hook Alert
- Room Monitor
- User Status (Presence)

SPARSH VP248, Front View



1. Ringer LED
2. Date and Time
3. Cursor
4. Touch sense feature keys
5. Touch sense navigation keys
6. Volume increase key
7. Speaker key with LED
8. Volume decrease key
9. Dial Pad
10. Programmable feature keys
11. Handset
12. 4P4C Spring Cord

Models of SPARSH VP248 at a Glance

Feature	**Model**	
	SPARSH VP248S	**SPARSH VP248P**
Total number of keys	48	48
Number of programmable keys	29	29
Capsense keys	Yes	Yes
LCD display capacity	2 lines x 24 characters	6 lines x 24 characters
LCD with backlight	Yes	Yes
Headset Interface	Yes	Yes
Ringer Lamp (LED)	Yes	Yes
Speaker Phone	Full duplex	Full duplex

SPARSH VP248S

2 lines and 24 characters LCD display, full duplex, capsense feature keys

SPARSH VP248P

6 lines and 24 characters LCD display, full duplex, capsense feature keys.

LCD Display

The LCD display of SPARSH VP248 is backlit and can be tilted at a convenient angle for a clear view of the text/ characters displayed.

The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the "*Phone Settings*" of the SPARSH VP248 Phone Menu.

Ringer LED

The Ringer LED indicates incoming internal and external calls. The LED Cadence will match with the Ring Cadence of the incoming internal/external call.

The Ringer LED cadence changes according to the type of call, as described in the table below.

Type of Call	Cadence
Internal Call	Short, very slow (750ms ON, 2250ms OFF)
External Call	Double (400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Alarm	Long, fast (1500ms ON, 500ms OFF)
Auto Redial Call	Long, very slow (2000ms ON, 4000ms OFF)
Auto Call Back Call	Short, slow (750ms ON, 2250ms OFF)
Priority	Triple (400ms ON, 200ms OFF, 400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Programming mode	Continuous

Navigation Keys

The phone has 5 touch sense navigation keys to be used to move the cursor and scroll through Menu options.

✔ is the Enter key, used to make a selection or to complete an action.

^ is the Up key, used to scroll upwards while navigating the 'Menu'.

v is the Down key, used to scroll downwards while navigating the 'Menu'.

> is the Forward key, used to move the cursor.

< is the Back key, used to move the cursor, return from the Sub-menu to the Main Menu.

Feature Keys

These are 12 capsense keys assigned to important or frequently accessed features of ANANT UCS. Refer to the table given below:

Sr.No.	Description	LED
1.	Voice Mail	Single Color - Blue
2.	Call Back	Single Color - Blue
3.	Cancel	No
4.	Mute	Single Color - Blue
5.	Conference	No
6.	Transfer	No
7.	Forward	Single Color - Blue
8.	DND	Single Color - Blue
9.	Names	No
10.	Redial	No
11.	Release	No
12.	Hold	No

These keys are programmable. However, as you cannot change the labels, avoid programming these keys.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 16 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of ANANT UCS.

For instructions on programming these keys, see "DSS Key Settings" under "Configuring Matrix SPARSH VP248".

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the Extended IP Phone.

  Thus, the status of the Extended IP Phone user's own Extension as well as that of the other Extensions and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the Extended IP Phone.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	**Slow Blink**	**Fast Blink**
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/ Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

  - **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.
  - **Red** indicates the state of other Extensions/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.
  - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters * and #. The dial pad is used for dialing numbers of extensions, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation. The Speaker key is programmable, you can program any other feature/function on this key.

Speaker Key LED

The Speaker Key on SPARSH VP248 is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

SPARSH VP248 provides two Headset interfaces: a 2.5mm Audio Jack and an RJ9 connector at the bottom of the phone body.

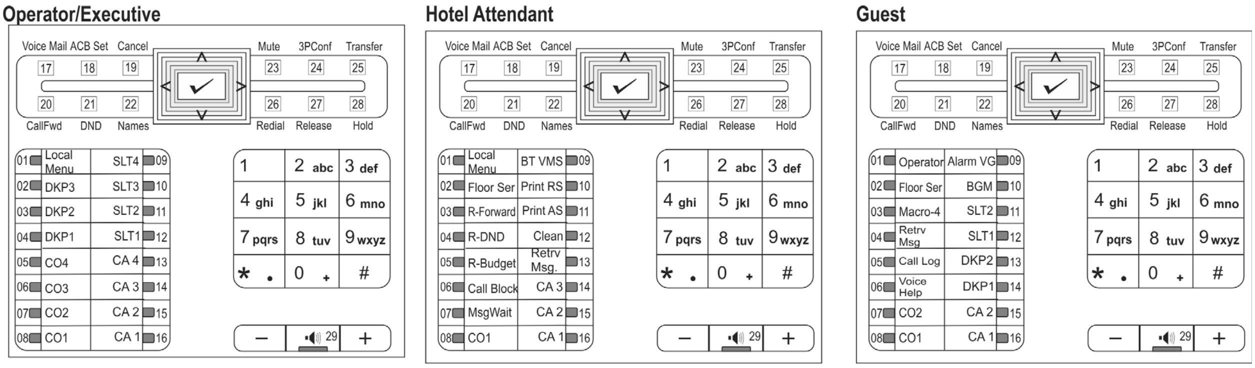
So you can use any stereo headset of standard make with a 2.5 mm single connector or a headset with an RJ9 connector.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

Key Maps

As SPARSH VP248 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates”. You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” under “Configuring Matrix SPARSH VP248”.

Matrix Extended IP Phone, SPARSH VP248 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3 are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

ANANT UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, ANANT UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Phone Menu

You can access the following ANANT UCS and phone features from the Menu of SPARSH VP248:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, i.e. block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To set/clear Transfer Number.

Menu option	Description
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

Navigating the Phone Menu

To navigate the menu,

- Tap on Enter key when the phone is idle.
- Scroll by tapping the Up/Down Navigation Key to reach the desired Menu option.
- Tap on Enter key to select the desired Menu option.
- Scroll by tapping on the Up/Down Navigation Key to reach the desired sub-menu option.
- Tap on Enter key to select the desired sub-menu option.

To exit menu,

- Press Cancel key.
  Or
- Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** option you select in General Parameters under SIP Extensions. See "Configuring Matrix SPARSH VP248" for instructions.

Connecting SPARSH VP248

For detailed instructions to connect SPARSH VP248, see "Connecting SPARSH VP248 as Extended SIP Extension" in SIP Extensions.

Configuring SPARSH VP248

For detailed instructions on how to configure SPARSH VP248, see "Configuring Matrix SPARSH VP248".

Operating SPARSH VP248

Please refer the EON48_310_SPARSH VP248_310 User Guide for instructions on operating the features of ANANT UCS.

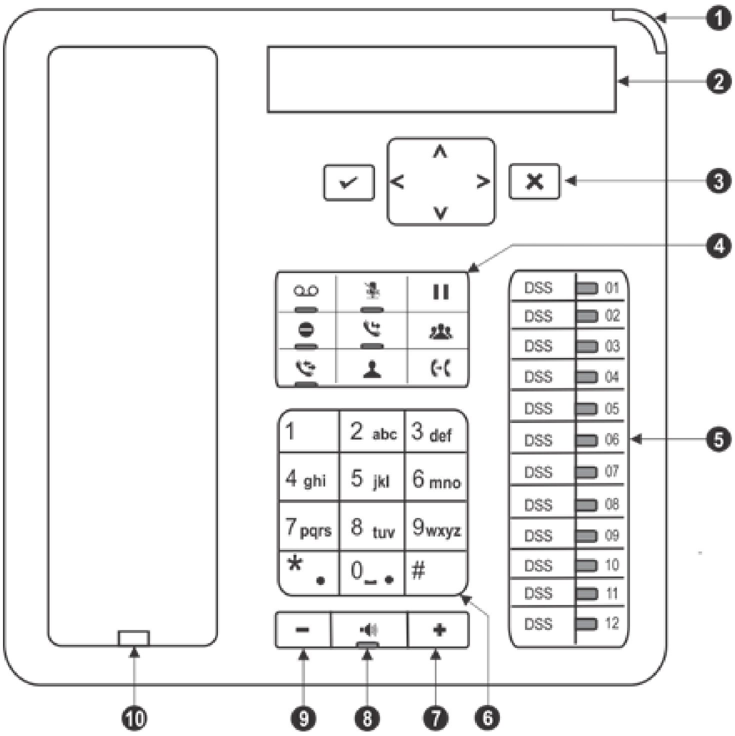
Matrix SPARSH VP310

SPARSH VP310, The Executive IP Phone is engineered to offer a contemporary design with crystal-clear audio and feature-rich capabilities at economical price. Elegant design, built-in programmable DSS Keys and plug-n-play connectivity makes SPARSH VP310 an easy to use phone for executives. SPARSH VP310 works in tight integration with ANANT UCS for speed of operations and better workforce collaboration.

Key Features

- 2 Line LCD with Backlit
- Fixed Function Keys (With LED) - Voice Mail, Mute, Do Not Disturb, Forward, Logs, Speaker
- Fixed Function Keys (Without LED) - Hold, Conference, Contacts, Transfer
- Superior Voice Quality with HD Audio
- Full Duplex Speaker Phone
- PC and LAN Ethernet Ports
- Power over Ethernet (IEEE 802.3af)
- 12 DSS/BLF Keys for Feature, Line, Extension
- Message Wait and Ringer Lamp
- 3.5mm Headset Connectivity
- Adjustable Desk Stand
- Plug & Play

SPARSH VP310, Front View



1	Ringer LED
2	LCD Screen
3	Navigation Keys
4	Fixed Function Keys
5	DSS (Direct Station Selection) Keys
6	Digit Keys/ Dial Pad Keys
7	Volume Increase Key/ "+" Key
8	Speaker Key
9	Volume Decrease Key/ "-" Key
10	Handset

LCD Display

The LCD display of SPARSH VP310 is backlit. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the *Phone Settings* of the SPARSH VP310 Phone Menu.

Ringer LED

The Ringer LED indicates incoming internal and external calls. The LED Cadence will match with the Ring Cadence of the incoming internal/external call.

The Ringer LED changes according to the type of call, as described in the table below.

Type of Call	Cadence
Internal Call	Short, very slow (750ms ON, 2250ms OFF)
External Call	Double (400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Alarm	Long, fast (1500ms ON, 500ms OFF)
Auto Redial Call	Long, very slow (2000ms ON, 4000ms OFF)
Auto Call Back Call	Short, slow (750ms ON, 2250ms OFF)
Priority	Triple (400ms ON, 200ms OFF, 400ms ON, 200ms OFF, 400ms ON, 2000ms OFF)
Programming mode	Continuous

Navigation Keys

The phone has the following navigation keys which are used to move the cursor and scroll through Menu options.

✓ is the Enter key, used to make a selection or to complete an action.

∧ is the Up key, used to scroll upwards while navigating the Menu.

∨ is the Down key, used to scroll downwards while navigating the Menu.

> is the Forward key, used to move the cursor.

< is the Back key, used to move the cursor, return from the Sub-menu to the Main Menu.

✕ is the Cancel key, used to exit a menu.

Feature Keys

There are 9 Feature keys assigned to important or frequently accessed features of ANANT UCS.

Sr.No.	Feature icon	Description	LED
1.		Voice Mail	Single Color - Blue
2.		Mute	Single Color - Blue
3.		Hold	No
4.		DND	Single Color - Blue
5.		Call Forward	Single Color - Blue

Sr.No.	Feature icon	Description	LED
6.		Conference	No
7.		Call Logs	Single Color - Blue
8.		Names	No
9.		Transfer	No

These keys are programmable. However, as you cannot change the labels avoid programming these keys.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP310”.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 12 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of ANANT UCS.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP310”.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the Extended IP Phone.

  Thus, the status of the Extended IP Phone user's own Extension as well as that of the other Extensions and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the Extended IP Phone.

The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	**Slow Blink**	**Fast Blink**
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/ Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

- **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.
- **Red** indicates the state of other Extensions/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.
- **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

  Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0-9, and the characters * and #. The dial pad is used for dialing numbers of extensions, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Speaker Key LED

The Speaker Key on SPARSH VP310 is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.
- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The SPARSH VP310 provides two Headset interfaces: a 3.5mm Audio Jack and an RJ9 connector on the left side panel of the phone.

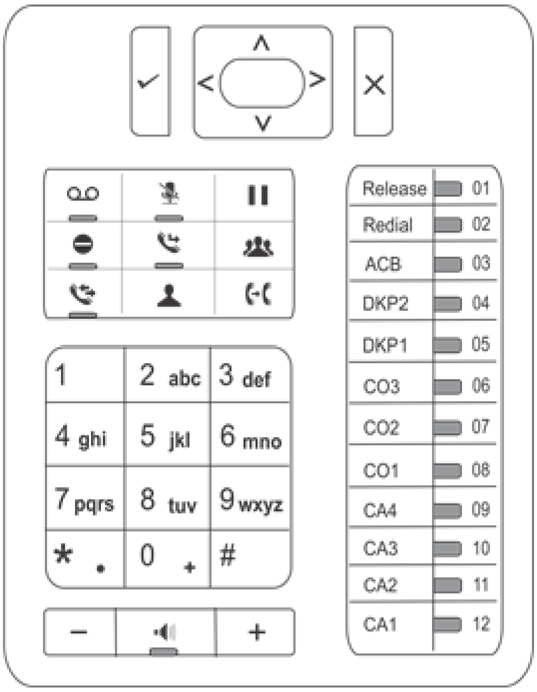
So you can use any stereo headset of standard make with a 3.5 mm single connector or a headset with an RJ9 connector.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see "DSS Key Settings" under "Configuring Matrix SPARSH VP310".

Key Maps

As SPARSH VP310 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see "Customizing Extended IP Phone Templates". You can also personalize the key maps for each location, for instructions, see "DSS Key Settings" under "Configuring Matrix SPARSH VP310".

Matrix Extended IP Phone, SPARSH VP310 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3 are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

ANANT UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, ANANT UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Phone Menu

Press the Enter key ✓ to access the Phone Menu.

You can access the following ANANT UCS and phone features from the Menu of SPARSH VP310.

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Call Forward	To set and cancel Call Forward-Busy, Call-Forward No Reply, Call-Forward-Unconditional, and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent.
Keypad Lock	To lock the keypad of the phone. (when the keypad is locked, the features Call Log, Contact, Call Forward, Dynamic Lock, User Status, DND, Call Cost Display, Hotline, Alarm, Change User Password will not be accessible.)

Menu option	Description
Do Not Disturb	To set/cancel Do Not Disturb on the phone, i.e. block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To set/clear Transfer Number.
Phone Settings	To customize settings of the phone such as Speech and Ringer Controls, LCD Display settings (Brightness and Contrast, Backlight ON/OFF), Headset Connectivity, Call Answering Mode (manual/auto answer).

When the phone is in idle state,

- Press the Down key V to access the Network Settings.
- Press the Up key Λ , if you wish to change the Ringtone and Play Key Tone.

Navigating the Phone Menu

To navigate the menu,

- Press on Enter key when the phone is idle.
- Scroll by tapping the Up/Down Navigation Key to reach the desired Menu option.
- Press on Enter key to select the desired Menu option.
- Scroll by tapping on the Up/Down Navigation Key to reach the desired sub-menu option.
- Press on Enter key to select the desired sub-menu option.

To exit menu,

- Press Cancel key.
  Or
- Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** option you select in General Parameters under SIP Extensions. See “Configuring Matrix SPARSH VP310” for instructions.

Connecting SPARSH VP310

For detailed instructions to connect SPARSH VP310, see “Connecting SPARSH VP310 as Extended SIP Extension” in SIP Extensions.

Configuring SPARSH VP310

For detailed instructions on how to configure SPARSH VP310, see “Configuring Matrix SPARSH VP310”.

Operating SPARSH VP310

Please refer the EON48_310_SPARSH VP248_310 User Guide for instructions on operating the features of ANANT UCS.

Matrix SPARSH VP330

SPARSH VP330, the next generation feature rich SIP phone of Matrix with an intuitive GUI (Graphical User Interface) based touch-screen. It provides you an easy way of managing your communication needs to meet the day to day business requirements. Designed to change the way you communicate and collaborate, SPARSH VP330 delivers a seamless communication experience that is convenient and ready to use in real time, so that you can focus on the task in hand. Once it is registered with the ANANT UCS, you can start operating the phone.

Key Features

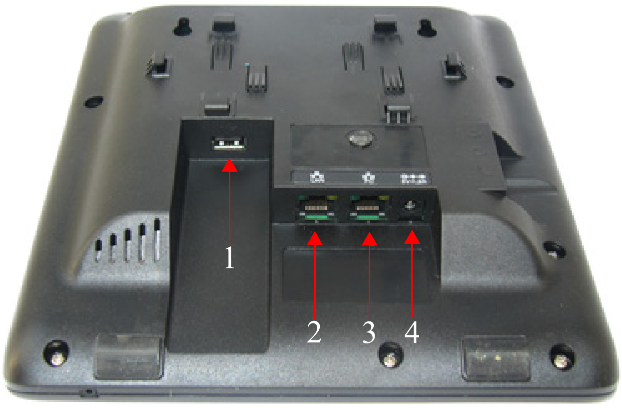
- **Enhanced Call Management:** Dedicated one-touch feature keys and intuitive user interface provides quick access to full range of System call management features including Call Hold, Call Park, Call Transfer, Conference and Voicemail.
- **Access to Corporate Directory:** Easy integration with the enterprise’s Corporate Directory (Global Directory) which allows you to easily locate and dial corporate contacts at one click.
- **Presence:** Provides intuitive Presence status display and supports changing your Presence status which is viewable to other extension users. You can also view the Presence Status of remote users.

- **Busy Lamp Field (BLF) Indication:** You can monitor the status of the extensions and trunks who are assigned DSS Soft keys. You can also pickup ringing extensions/trunks using DSS Soft keys.
- **Plug & Play:** Integrated Plug & Play feature that enables to power up the phone and start using it. On the other hand, it helps in mass deployment of the phones in your organization without requiring a lot of manual intervention to configure each of the phones separately.
- **Multiple Language Support:** The phone can be operated in six different languages including English, French, German, Spanish, Portuguese and Italian.
- **Capacitive Touch Screen:** 4.3 inch Capacitive Touch Screen LCD (Liquid Crystal Display) that delivers easy access to advanced features and a unique experience beyond traditional desk phones. Supports adjustable Brightness controls from the touch screen to suit your customized LCD requirement.
- **Easy-to-Use Hard Keys:** Supports following hard keys and LEDs:
  - 6 Fixed Feature Keys
  - 12 Alphanumeric Digit Keys (Dial pad Keys)
  - 1 Speaker Key.
  - 12 DSS (Direct Station Selection) keys.
  - 1 Ringer LED.
- **Improved Audio:** High Definition (HD) Audio output that delivers crystal clear voice and life like conversations over HD handset and hands-free speaker.
- **High Speed Ethernet connectivity:** Dual switched 10/100 Base-T auto-sensing Ethernet LAN connectivity allows unconstrained bandwidth from the network to the phone. LAN port is used to connect the phone to a Switch or a Hub or a Router or an xDSL Modem. You can connect your PC to the PC port of the phone.
- **Power over Ethernet (PoE):** Integrated IEEE 802.3f Power over Ethernet allows easy deployment with centralized powering without a need for external power adapter.
- **Wi-Fi Support:** Supports Wi-Fi (WLAN) connectivity using which the phone provides seamless connectivity to the corporate Wi-Fi network and offers flexibility to work from anywhere in the office. If your installation setup does not meet the requirements of suitable wired Ethernet connectivity due to any reason, then you can connect the compatible Wi-Fi Adapter supplied by Matrix to the USB port to register and use the phone.
- **Improved Audio:** High Definition (HD) Audio output that delivers crystal clear voice and life like conversations over HD handset and hands-free speaker.
- **Full-duplex Hands-free Speaker:** High quality speaker with acoustic echo cancellation to deliver natural and clear speech even for hands free operation without any distortion.

Front View

1	Ringer LED
2	LCD Screen
3	Fixed Function Keys
4	DSS (Direct Station Selection) Keys
5	Digit Keys/ Dial pad Keys
6	Volume Up Key/ "+" Key
7	Speaker Key
8	Volume Down Key/ "-" Key
9	Handset

Bottom View

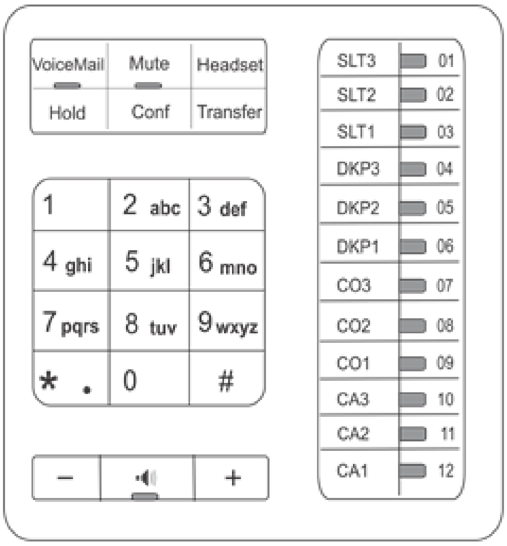


1	USB Port for Wi-Fi Adapter
2	LAN Port for a LAN Switch / Router
3	PC Port for a PC / Computer
4	Power Port for the Power Adapter

Left Side View

1	RJ11 Headset Port
2	Headset Port
3	Handset Port

SPARSH VP330 Key Template (default)



All the key maps of the SPARSH VP330 are the same.

Feature Keys and DSS Keys

Feature Keys

There are 6 Feature Keys. Each Feature Key is accompanied by a feature icon that describes its function. Default features assigned to these keys are as follows.

SR. No.	Feature icon	Assigned Feature	LED
1		Voicemail	Single Color - Blue
2		Mute	Single Color - Blue
3		Headset	Single Color - Blue
4		Hold	No
5		Conference	No
6		Transfer	No

These Feature keys are programmable. You cannot change the labels of these keys and therefore it is recommended that you avoid programming these keys. However, if you still decide to reprogram features on these keys, keep a note of the changes you have made.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

DSS (Direct Station Selection) Keys

There are 12 DSS Keys which can be used for quick access to different features and facilities. For example, to dial an extension number at one key press, just press the DSS key assigned to that extension and the call will be placed automatically. These Keys have dual color LEDs - blue and red. Features/facilities assigned to these keys can be changed by the System Engineer.

As SPARSH VP330 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates”. You can also personalize the key maps for each location, for instructions, see, “DSS Key Settings” under “Configuring Matrix SPARSH VP330”.

For detailed information about SPARSH VP330, refer to the *SPARSH VP330 User Guide*.

Connecting SPARSH VP330

For detailed instructions to connect SPARSH VP330, see “Connecting SPARSH VP330 as Extended SIP Extension” in SIP Extensions.

Configuring SPARSH VP330

For detailed instructions on how to configure SPARSH VP330, see “Configuring Matrix SPARSH VP330”.

Operating SPARSH VP330

Refer the *SPARSH VP330 User Guide* for instructions on operating the features of ANANT UCS.

Matrix SPARSH VP510

SPARSH VP510, the Premium IP Phone sets the benchmark for quality performance with elegant design and crystal-clear voice. SPARSH VP510 features a Vivid LCD Graphical Display, Direct Station Selection (DSS) Keys, 3.5mm Headset connectivity, High Quality speakerphone and high definition audio quality.

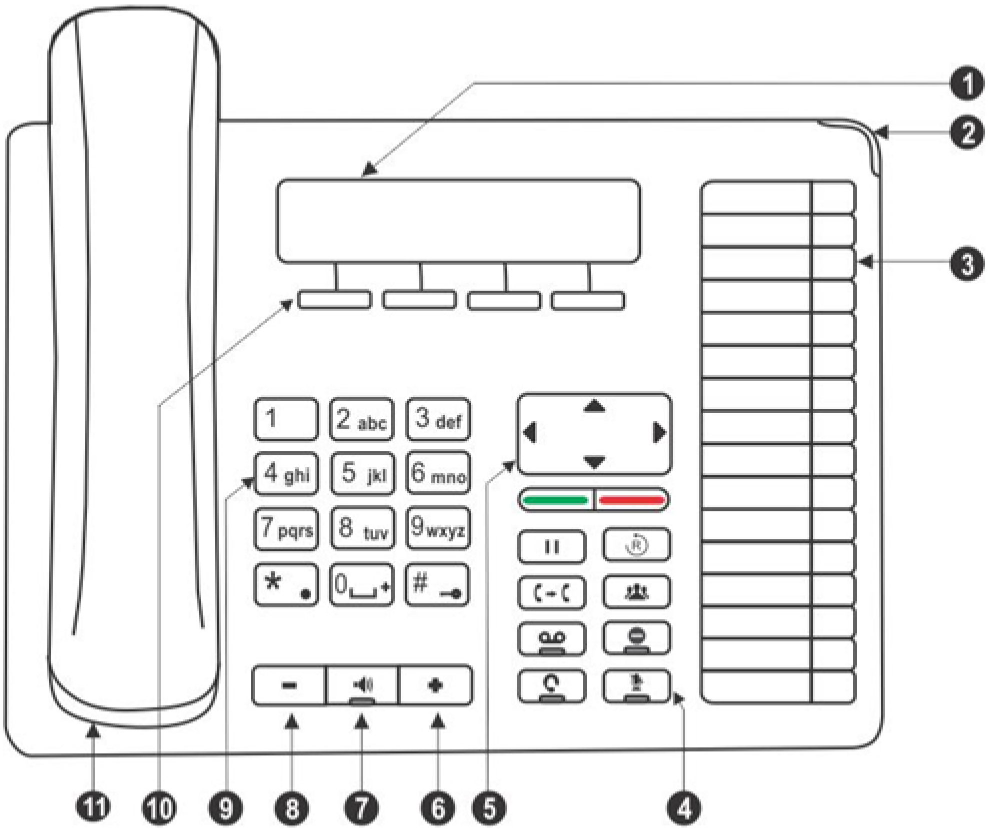
Engineered to deliver full feature access of ANANT UCS, SPARSH VP510 acts as face of your communication system covering wide array of business environments.

The State-of-the-art Deskphone is best suited to deployment for Reception, Supervisors, Managers, Executives, Call Center Agents and Office professionals. These phones offer flexibility to streamline communications and attain higher return over investment.

Key Features

- Minimum 240*64 pixel and above graphical LCD with Back light
- Message Wait and Ringer Lamp
- Built-in 16 DSS Keys for Feature, Line, Extension
- Add -on DSS module facility
- HD Voice,HD Handset, HD Speaker
- 4 Context Sensitive Keys
- 5 Features Keys (With LED) -Voice mail, Headset, Mute, DND, Speaker
- Fixed Function Keys (Without LED) - Hold, Conference, Redial, Transfer
- Tight integration with Server over SIP protocol /Proprietary Protocol
- HTTP Auto Provisioning
- Dual Color LED illuminated LED for line status
- One Touch Transfer
- Call logs
- Ringtone selection
- Wideband Codec: G722
- Narrowband Codec: G.711,G.723,G.729ab,GSM
- VAD,CNG,AEC,AJB,AGC
- Full Duplex speaker phone with AEC,VAD,CNG
- IP Assignment: Static /DHCP
- TCP/DNS-SRV
- AEC encryption for config file
- IEEE802.1x
- 3.5 mm / RJ9 headset port
- Dual port 10/100 Mbps Ethernet
- Stand with 2 adjustable angles
- PoE (IEEE 802.af) class2

SPARSH VP510, Front View



1	LCD Screen
2	Ringer LED
3	DSS (Direct Station Selection) Keys
4	Fixed Function Keys
5	Navigation Keys
6	Volume Increase Key
7	Speaker Key
8	Volume Decrease Key
9	Dial Pad/ Key Pad Keys
10	Context Sensitive Keys
11	Handset

LCD Display

The LCD display of the phone is backlit. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the *Display Settings* of the Phone Menu.

Ringer LED

The Ringer LED will glow in Blue (1 sec ON – 500 msec OFF) to indicate incoming internal and external calls.

Navigation Keys

The functions of each are described briefly below.

- **Up Key:** To scroll upwards while navigating the Menu/sub-menu or to access Phone Settings and set the Ringtone (when phone is in the idle state).
- **Down Key:** To scroll downwards while navigating the Menu/sub-menu.
- **Forward Key:** To move forward when dialing a number or scroll to view the Context Sensitive Key options.
- **Back Key:** To move backwards when dialing a number, to go back one level in the Menu or scroll backwards to view the Context Key options.
- **Menu or Select / OK Key :** To enter the Menu; when the phone is in the idle state (without any incoming or outgoing call being made).

  Menu Key functions as the **Select / OK** Key to make a selection from the Menu/sub-menu options or to complete an action. When there is an incoming call it functions as the **Answer** Key.
- **Cancel Key :** To Cancel all features set by you or exit the Menu/sub-menu.

Feature Keys

There are 8 Feature keys assigned to important or frequently accessed features of ANANT UCS.

Feature icon	Assigned Feature	LED	Programmable
	Hold	No	Yes
	Redial	No	Yes
	Transfer	No	Yes
	Conference	No	Yes
	Voicemail	Single Color - Blue	Yes
	Do Not Disturb	Single Color - Blue	Yes
	Headset	Single Color - Blue	No
	Mute	Single Color - Blue	No

These keys are programmable. However, as you cannot change the labels avoid programming these keys.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

A few of these Feature keys are equipped with an LED to indicate the status of the feature assigned to it. You may re-assign other features to these keys. We recommend you to assign those features that require LED indication to the Feature keys equipped with an LED.

Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a feature key, the LED of the key remains inactive, when Call Pick-Up is accessed.

Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the Feature key to which the Auto Redial feature has been assigned will glow Blue, when Auto-Redial is set, and the LED is turned off when the feature is canceled.

Direct Station Selection (DSS) Keys

These are 16 programmable keys that can be assigned to Stations and Trunks and important or frequently accessed features of ANANT UCS.

For instructions on programming these keys, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

DSS Key LEDs

Each DSS key is equipped with an LED which glows to indicate the status of the Trunk/Extension or Feature assigned to it.

- **Status of Extensions and Trunks:** The LED of DSS keys assigned to Extensions/Trunks glow in three colors to indicate status of the call event on the Extensions/Trunks and on the Extended IP Phone. Thus, the status of the Extended IP Phone user's own Extension as well as that of the other Extensions and the status of Trunk lines are indicated by the LED of the DSS keys assigned to those Extensions and Trunks on the Extended IP Phone.

  The following table shows the relationship between the color of the LED and various events:

LED Color	LED Mode		
	Continuously ON	**Slow Blink**	**Fast Blink**
Blue	The key assigned to the Extension you are in speech with.	The key assigned to the Extension you have kept on hold.	The key assigned to the Extension you are calling or from which you are being called.
Red	The key assigned to the Extension that is now busy with another Extension/ Trunk.	The key assigned to the Extension which has put another Extension/Trunk on hold.	The key assigned to the Extension/Trunk that is called or being called by another.
Violet	You are talking on a Trunk (external call)	You have held a Trunk (external call)	You have an incoming call on the Trunk (external call)

- **Blue** indicates the state of the extension/trunk you access. For example, when you make a call to another Extension 203, the LED of the DSS key assigned to Extension 203 blinks Blue to indicate ringing at the Extension. If you have successfully established speech with Extension 203 the LED glows Blue continuously.

    - **Red** indicates the state of other Extensions/Trunks. For example, if the LED of the DSS key assigned to Extension 201 is glowing Red continuously, it means Extension 201 is busy with another Extension or Trunk.

    - **Violet** indicates the state of the trunk you are in speech with. For example, when you are in speech on an outgoing call on Trunk 1 the LED of the DSS Key assigned to Trunk 1 will be continuously ON. When you put the call on hold, the LED will blink slowly.

- **Status of Features:** The LED of a DSS key is activated when the feature assigned to this key is used.

  Not all features require LED indication. For example, the feature Call Pick-Up does not require an LED. So when you assign this feature to a DSS key, the LED of the key remains inactive, when Call Pick-Up is accessed.

  Feature like Auto Redial requires an LED to show that it has been set or canceled. So, the LED of the DSS key to which the Auto Redial feature has been assigned will glow Red, when Auto Redial is set, and the LED is turned off when the feature is canceled.

The LEDs of the Call Appearance (CA) Keys will function in the same manner as the DSS Key LEDs.

Dial Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters Star (*), Hash (#), Lock (), Plus (+) and Dot. The dial pad is used for dialing numbers of stations, external parties, and for dialing the programming and feature access codes.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Speaker Key LED

The Speaker Key on the phone is equipped with a single color LED which glows Blue when pressed for the speaker mode and is turned off, when you exit the speaker mode.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.
- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The phone provides two Headset interfaces: A 3.5mm Audio Jack and an RJ9 connector at the bottom of the phone body.

So you can use any stereo headset of standard make with a 3.5 mm single connector or a headset with an RJ9 connector.

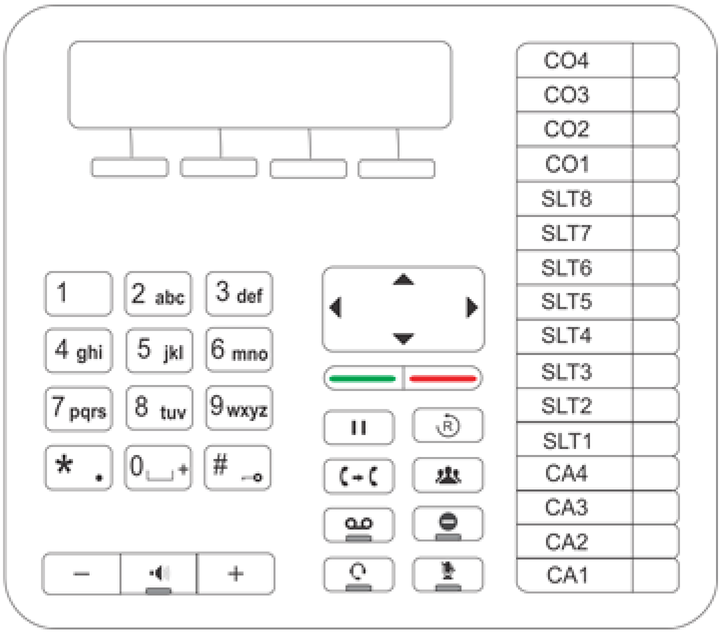
To use the Headset, a Headset Key is assigned on the phone. The Headset Key on the phone is equipped with a single color LED which glows Blue when pressed to indicate that the Headset mode is turned on and is turned off, when you press it again to indicate that you have exit the headset mode.

You can also program any of the DSS keys to function as the Headset key. For instructions on programming the key, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

Key Maps

As SPARSH VP510 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates”. You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” under “Configuring Matrix SPARSH VP510”.

Matrix Extended IP Phone, SPARSH VP510 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3 are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

ANANT UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, ANANT UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Phone Menu

To access the Phone Menu, press the Enter Key. You can access the following System and phone features from the phone:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls. You can also edit numbers in the call logs and store them in the Personal Directory.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.

Menu option	Description
Call Forward	To set and cancel Call Forward-Unconditional, Call Forward-Busy, Call Forward No Reply, Forward On Busy/No Reply and Follow Me.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or User Absent and Presence Status.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Call Cost Display	To view the cost of calls made from the phone.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm	To set/cancel Personalized and Automated Alarms.
Change User Password	To change User Password.
One Touch Transfer	To set/clear the fixed destination number for One Touch Transfer.
Phone Settings	To customize settings of the phone.

When the phone is in idle state,

- Press the Down key V to access the Network Settings.
- Press the Up key Λ , if you wish to change the Ringtone and Play Key Tone.

Navigating the Phone Menu

To navigate the menu,

- Press the Menu Key when the phone is idle.
- Scroll by pressing the Up/Down Navigation Key to reach the desired Menu option.
- Press the Select / OK Key to select the desired Menu option.
- Scroll by pressing the Up/Down Navigation Key to reach the desired sub-menu option.
- Press the Select / OK Key to select the desired sub-menu option.

To exit menu,

- Press Cancel Key.
  or
  Go ON-Hook.

Call Waiting Indication

During an on-going call, if there is another incoming call, an indication will be provided to you for the waiting call.

The call waiting indication depends on the **Call Waiting Tone (for SPARSH VP248/VP310/VP510)** option you select in General Parameters under SIP Extensions. See “Configuring Matrix SPARSH VP510” for instructions.

Connecting SPARSH VP510

For detailed instructions to connect SPARSH VP510, see “Connecting SPARSH VP510 as Extended SIP Extension” in SIP Extensions.

For detailed instructions on how to configure SPARSH VP510, see “Configuring Matrix SPARSH VP510”.

Operating SPARSH VP510

Please refer the *EON510_SPARSH VP510 User Guide* for instructions on operating the features of ANANT UCS.

Matrix SPARSH VP210

SPARSH VP210, the Entry Level IP Phone sets the benchmark for quality performance with elegant design and crystal-clear voice. SPARSH VP210 features a 3.1” Graphical LCD Display, SIP Line Keys, High Quality speaker-phone and high definition audio quality.

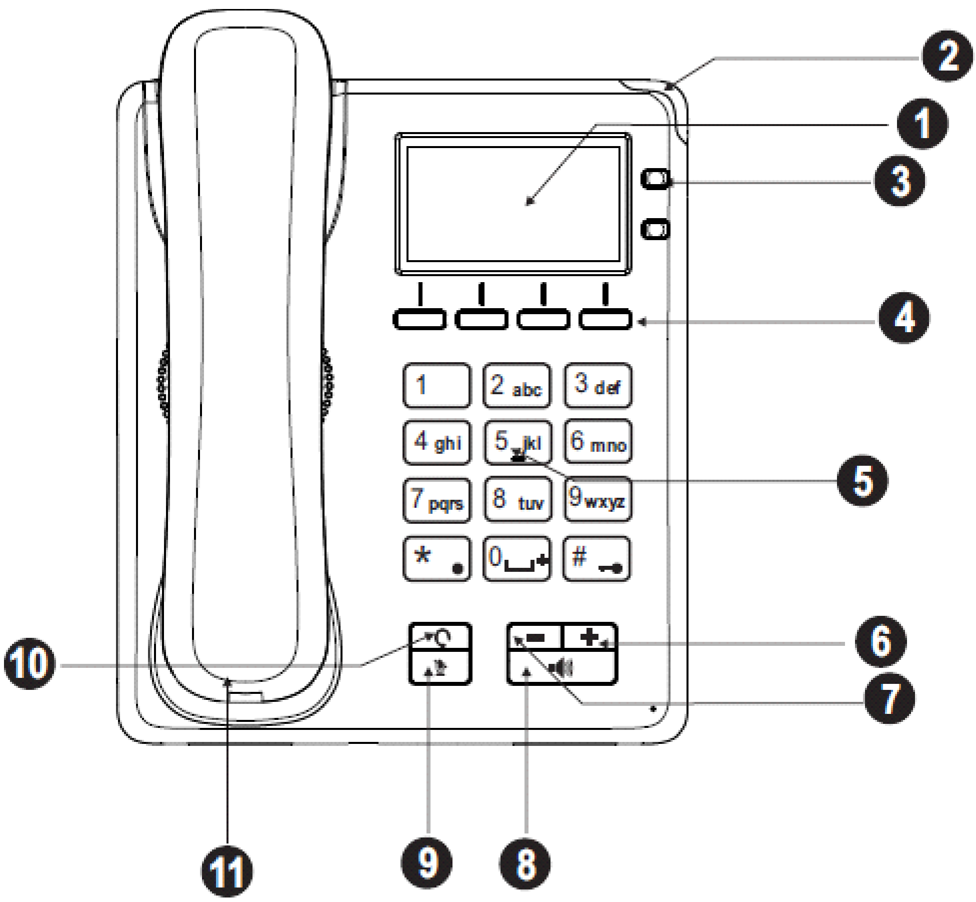
The State-of-the-art Deskphone is best suited for usage in lobbies, cafeterias, conference centers wherein the basic level endpoint security is sufficient. It can also be used by Administrative Staff, Hospitality guest rooms, knowledge workers etc.These phones offer flexibility to streamline communication and attain higher return over investment.

Key Features

- 128 x 64 Graphical LCD
- LED for Call and Message Wait Indication
- HD Voice, HD Handset, HD Speaker
- 4 Context Sensitive Keys
- 3 feature keys: Headset, Mute, Hands-free speaker phone

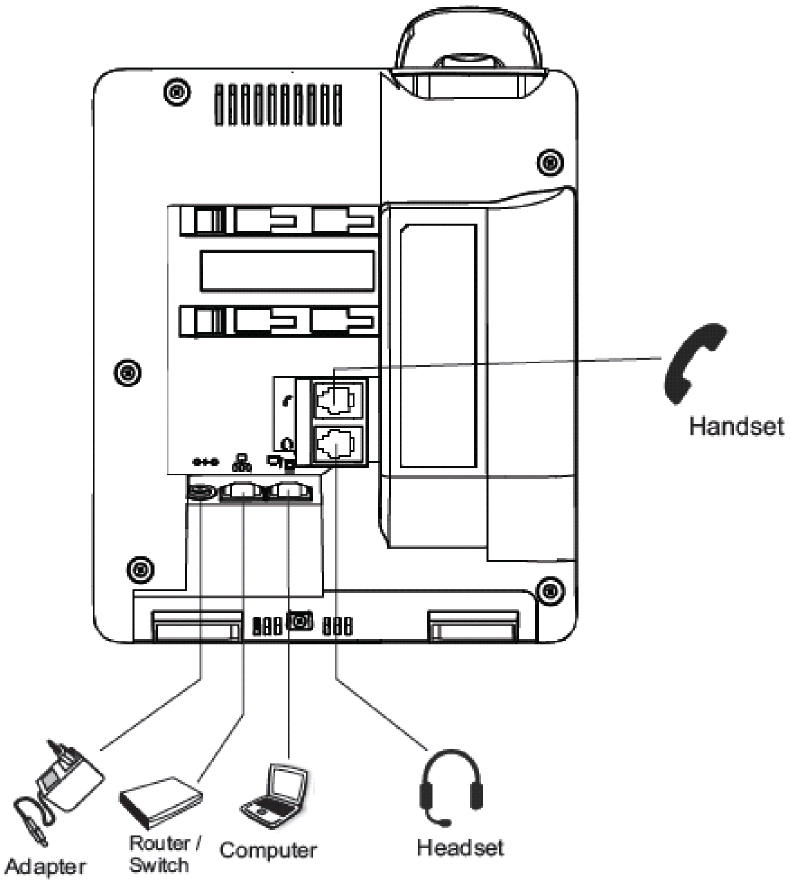
- Fixed Function Keys (Without LED) — Hold, Conference, Redial, Transfer
- Tight integration with Server over SIP protocol /Proprietary Protocol
- HTTP Auto Provisioning
- Dual Color illuminated LED for line status
- One Touch Transfer
- Call logs
- Ringtone selection
- Wideband Codec : G722
- Narrowband Codec: G.711(A/μ), G.729, G.726, G.723
- VAD, CNG, AEC, AJB, AGC
- Full Duplex speaker phone with AEC
- IP Assignment : Static / DHCP
- TCP/ DNS-SRV
- AEC encryption for config file
- IEEE802.1x
- RJ9 headset port
- Dual port 10/100 Mbps Ethernet
- Stand with 2 adjustable angles
- PoE (IEEE 802.af) class2

Front View



1	LCD Screen
2	Ringer LED
3	Navigation/Notification Keys
4	Context Sensitive Key
5	Dial Pad
6	Volume Increase Key
7	Volume Decrease Key
8	Speaker Key
9	Mute
10	Headset Key
11	Handset

Bottom View



It is a PoE enabled phone and can be powered over Ethernet by connecting it to a PoE enabled LAN Switch (IEEE 802.3af Compliant). If you do not want to use PoE, plug in the connector of the Adapter into the power jack (DC Jack) at the bottom of the phone. The IP phone should be used with Matrix original power adapter (5V/0.6A) only.

LCD Display

The LCD display of the phone is Dot Matrix Graphic LCD. The LCD backlight can be turned on and off as well as adjusted for contrast and brightness from the Phone Menu.

Ringer LED

The Ringer LED will glow in Blue (1 sec ON – 500 msec OFF) to indicate incoming internal and external calls.

Feature Keys

here are 3 Feature Keys. Each Feature Key is accompanied by a feature icon that describes its function. Default features assigned to these keys are as follows.

Feature icon	Assigned Feature	LED	Programmable
	Headset	No	No
	Mute	No	No
	Speaker	No	No

Navigation/Notification Keys

There are 2 Navigation Keys, Up/Down Keys.

When the phone is in idle state these keys are used for accessing the Notifications - Call Back, Auto Redial, Trunk Reservation, Contact Sync.

You can navigate sideways using the context keys, that is, **Left Navigation <** Key or **Right Navigation >** Key.

Dial Pad/Key Pad

The dial pad consists of 12 fixed keys for the digits 0, 1-9, and the characters Star (*), Hash (#), Lock ( ), Plus (+) and Dot. The dial pad is used for dialing numbers of stations or external parties.

Speaker Key

The speaker key sets the phone in 'Speaker mode' for hands-free operation.

Volume Keys

- **"+" (plus)**: This is the increase key, to raise the volume of speech while talking and to increase the Ringer volume, when the phone is ringing.

- **"-" (minus)**: This is the decrease key, to lower the volume of speech while talking and to decrease the Ringer volume when the phone is ringing.

Headset Connectivity

The phone provides an RJ9 connector at the bottom of the phone body to connect a headset.

To use the Headset, a Headset Key is assigned on the phone. Make sure you have enabled the **Use Headset** option.

Phone Menu

You can access the following PBX and phone features from the Menu of the phone:

Menu option	Description
Call Logs	To view call history of internal and external Missed, Answered and Dialed calls.
Contacts	To add, edit, delete names and numbers of contacts in the Global Directory Part 1.
Call Forward	To set and cancel Call Forward - Unconditional, Call Forward - Busy, Call Forward - No Reply, Call Forward - Busy/No Reply, Call Forward - Not Registered.
Dynamic Lock	To change the Toll Control level of the phone.
User Status	To set User Present or Absent.
Keypad Lock	To lock the keypad of the phone.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming internal and external calls.
Hotline	To set/cancel Hotline and Delayed Hotline.
Alarm & Reminder	To set/cancel Personalized and Automated Alarms/Reminders.
One Touch Transfer	To set/clear the fixed destination number for One Touch Transfer.
Pickup	To configure as well as access Group/Selective Call Pick-up
Voicemail	To access your Mailbox.
Dial-In Conference	To schedule as well as establish the Conference.
Call Retrieve	To retrieve a call parked in the Personal or General Orbit.
CLIR	To set/cancel CLIR.
Call Supervision	To configure as well as access Call Supervision.
Message Wait	To set/cancel Message Wait.
Paging	To configure the Page Zone and make the announcement.
Meet Me Paging	To access Meet be Paging.
Room Monitor	To configure and access Room Monitor.
Intercom	To configure and access Intercom.
Follow Me	To set Follow Me.
Walk-in	To set/cancel Walk-in.
PIN Dialing	To make calls using PIN.
Department Group Call Forward	To set/cancel Department Group Call Forward.

Menu option	Description
Open Cosec Door	To open the Cosec Door Lock.
Settings	To change the following settings: • User Password: To change User Password. • Phone Settings: To customize settings of the phone. • Network Settings: To change Network Settings. • PCAP: To Start/Stop PCAP
Phone Info	Displays the phone information.

Navigating the Phone Menu

To navigate the menu,

- Press the **Menu** Key when the phone is idle.
- Scroll by pressing the **Up/Down Navigation** Key to reach the desired Menu option.
- Press the **Select** Key to select the desired Menu option.
- Scroll by pressing the **Up/Down Navigation** Key to reach the desired sub-menu option.
- Press the **Select** Key to select the desired sub-menu option.

To exit menu,

- Press **Back** Key.
  or
  Go ON-Hook.

  To scroll Up or Down you need to use the **Up/Down Navigation** Keys. To scroll sideways, you need to use the **Left Navigation <** Context Key or **Right Navigation >** Key.

Context Specific Keys (CSK)

SPARSH VP210 has the provision to program the four Context Keys. These keys enable you to access the most frequently used functions/features at the press of a single button.

You can configure these Keys from ANANT UCS Jeeves only.

The screens — Idle Screen, Ringing Screen, Busy Screen, Call Screen, Conversation Recording Screen, all have different set of features that can be accessed. SPRASH VP210, enables you to customize these by allowing you to set the priorities of the features in each type of screen as per your preference. You can assign the features to the Context Keys depending on the state of the call.

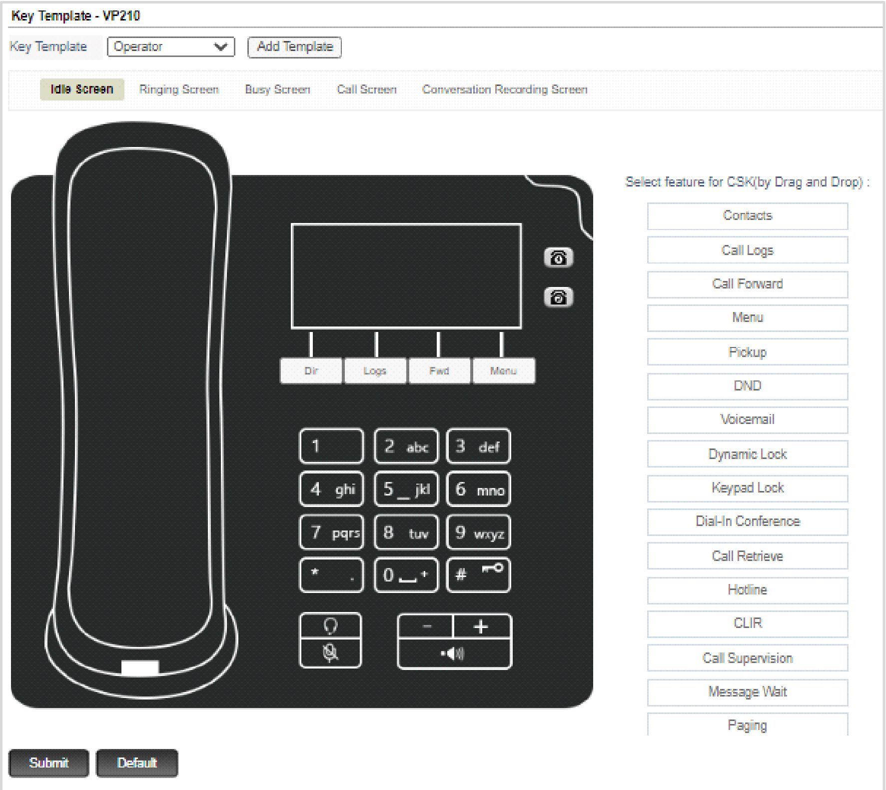
- In the Idle Screen you can assign the desired feature/function to the Context Keys as well as set their priorities as per your requirement.
- In the other Screens you can only set the priorities of the features.

Refer to “Key Maps”, “Customizing Extended IP Phone Templates” as well as “DSS Key Settings” in “Configuring Matrix SPARSH VP210”.

Key Maps

As SPARSH VP210 may be the extension of the Operators and Executives in an enterprise to meet the varied requirements of each user group, these key maps can be customized to match the exact requirement of individual users. For instructions on customizing the Key Maps, see “Customizing Extended IP Phone Templates”. You can also personalize the key maps for each location, for instructions, see “DSS Key Settings” in “Configuring Matrix SPARSH VP210”.

Matrix Extended IP Phone, SPARSH VP210 Key Template (default)



The key maps of the Operator and Executive 1, 2, 3, Hotel Attendant as well as Guest are the same.

By using Key Templates you can prepare and assign common key maps to all or as many Extended IP Phones as you want, at one go.

ANANT UCS also offers the flexibility to personalize the Key Maps of each Extended IP Phone, instead of using the Key Templates. For example, if you have assigned a common Executive Key Template to 12 Extended IP Phones, but you want to reassign some of the keys on two of these Extended IP Phones, ANANT UCS allows you to selectively personalize the key maps of these two Extended IP Phones.

Connecting SPARSH VP210

For detailed instructions to connect SPARSH VP210, see *“Connecting SPARSH VP210 as Extended SIP Extension”*.

Configuring SPARSH VP210

For detailed instructions on how to configure SPARSH VP210, see *“Configuring Matrix SPARSH VP210”*.

Operating SPARSH VP210

Refer the *SPARSH VP210 (Extended) User Guide* for instructions on operating the features of ANANT UCS.

Matrix Extended SPARSH VP710

Extended SPARSH VP710, the Smart Video IP Deskphone is engineered to deliver a seamless communication solution to the user with experience of an android touch screen. Extended SPARSH VP710 is an integration of SPARSH VP710, an android based deskphone with VARTA ADR100 application. This tight integration of the UC Client, VARTA ADR100 with SPARSH VP710 offers advance calling capabilities.

The IP Phone provides an easy way of managing the modern communication needs for meeting the day to day business requirements. With sophisticated looks and innovative features, it is the next step towards collaboration for offering flexibility and convenience in day to day communication. Deskphone is the perfect client to extract the capability of server and with a smarter deskphone you can optimize the return over investment by utilizing the UC features of the IP Phone. Once it is registered with the ANANT UCS, you can start operating the IP Phone.

Key Features

- **Enhanced Call Management:** Dedicated one touch feature keys and intuitive user interface provides quick access to full range of PBX call management features including Call Hold, Call Park, Call Transfer, Conference and Voicemail.
  The IP Phone also provides an easy way for businesses to integrate their enterprises' voice solutions within the Android OS family.

- **System Extension:** The IP Phone becomes an extension of the System. It provides users with a quicker and more user-friendly access to phone facilities, helping businesses optimize their employees' productivity.

- **Dial by Extension:** Flexibility to communicate with colleagues by dialing their respective extension numbers.

- **Smart Directory Access:** Provides you with the easy and quick way to access the extensions and other contacts through Smart Directory.

- **Presence:** You can set your presence status and view other extension users' presence statuses.

- **Voicemail Access:** Access to the corporate Voicemail System from any location ensures no opportunity is lost.

- **Multiple Call Support:** Easy handling of multiple incoming calls by keeping the ongoing call on hold and attending the higher priority call first. It also supports merging of calls to initiate a conference or splitting the conference to attend the calls separately.

- **Video Calling:** Video calling provides you the facility to make video calls to anyone, anywhere in the world. This makes it easier to conduct business meetings, discussions, demonstrations and presentations between people working at different locations.

- **Handover and vice-versa:** Using handover you can automatically move an active call from the IP Phone to your cellular number on the cellular network and vice-versa, without disconnecting the call and/or having to redial.

- **Busy Lamp Field (BLF)**: Using BLF you can monitor the status of another extension or trunk and confirm whether it is available, busy, ringing or on hold.

- **IM and SMS:** Using this feature, you can send/receive IMs and SMSs to/from remote users.

- **One Touch Transfer:** You can transfer the ongoing call to a fixed extension without entering the number of that extension and without putting the call on hold. Similarly, you can also transfer a call from the fixed extension to your IP Phone.

- **Better Voice Quality:** Using customized codec settings, enhanced voice output is available. If you are aware of the bandwidth and the network criteria of your location, you can select the appropriate codec to get high quality voice output.

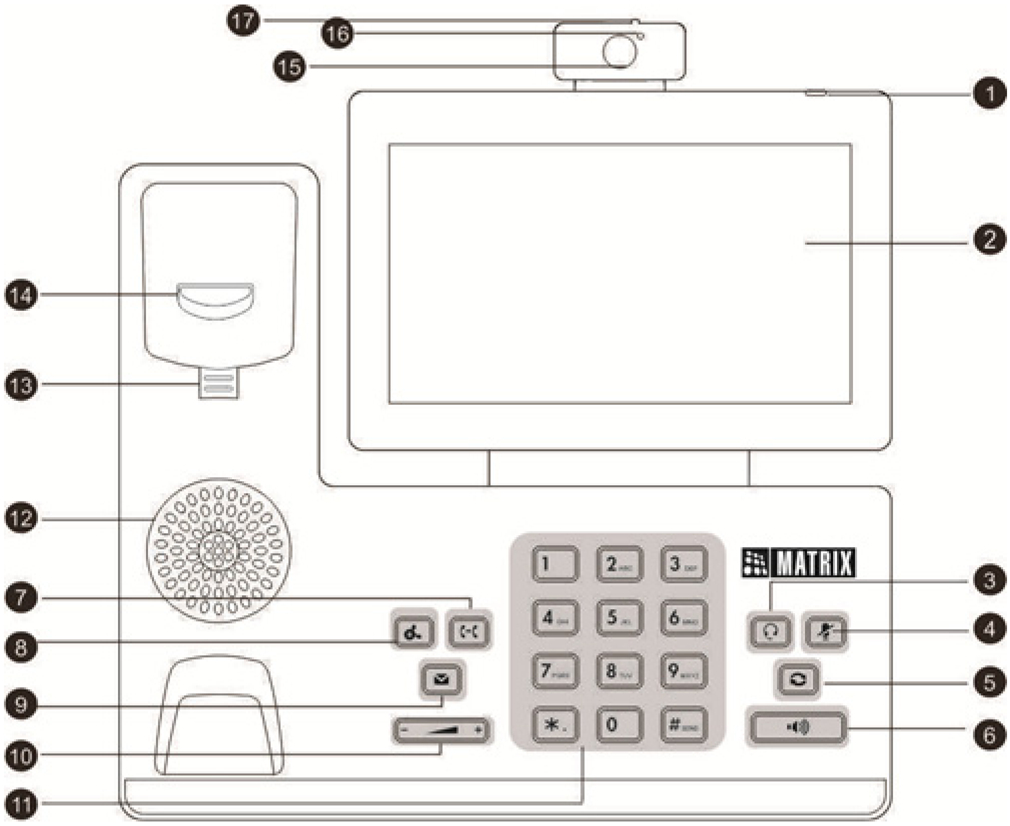
- **Standard Phone Features:** Provides intuitive access to Keypad, Contacts, Call Logs and more based on the Native Android design. One-touch access to call feature options during VoIP (Voice over IP) calls including Adding a New Call, Mute, Hold, Transfer and Speaker-phone.

- **Cost Effective Calling:** If you are using the enterprise Wi-Fi or Ethernet network to register the IP Phone with the System, calls made will be almost free.

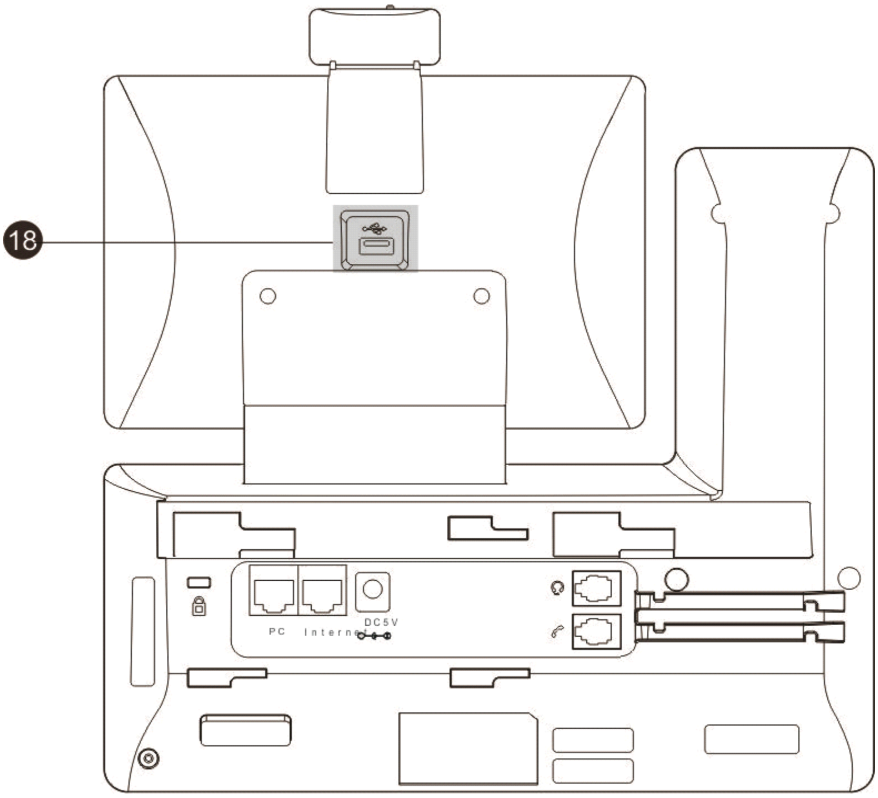
- **Wi-Fi Support:** Supports Wi-Fi (WLAN) connectivity using which the IP Phone provides seamless connectivity to the corporate Wi-Fi network and offers flexibility to work from anywhere in the office. If your installation setup does not meet the requirements of suitable wired Ethernet connectivity due to any reason, then you can register the IP Phone through the Wi-Fi Network.

- **Advanced Call Capabilities:** Provides access to the features such as Callback, Dial-in Conference, Conversation Recording and many more.

Extended SPARSH VP710, Front View



Extended SPARSH VP710, Back View



Key Label	Item	Description
1	Power Indicator LED	Indicates the status of calls, messages and voicemails. Also displays the registration status of the IP Phone.
2	Touch Screen	7 inch (1024 x 600) capacitive (5 point) touch screen. Tap to select and highlight screen items.
3	Headset Key	Toggles and indicates the headset mode. The key LED illuminates solid green when you activate the headset mode.
4	Mute Key	Toggles and indicates mute feature. The key LED illuminates solid red when you mute a call.
5	Call Log Key	Displays all the missed, received and dialed calls.
6	Speaker Key	Toggles and indicates the speaker mode. The key LED illuminates solid green when you activate the speaker mode.
7	Transfer Key	Transfers a call to another party.
8	Hold Key	Places a call on hold or resumes a held call.
9	Voicemail Key	Accesses voice mails.
10	Volume Key	Adjusts the volume of the handset, headset, speaker, ringer or media.
11	Keypad	Provides the digits and special characters.
12	Speaker	Provides speaker audio output.

Key Label	Item	Description
13	Hookswitch Tab	Secures the handset in the handset cradle when the IP Phone is mounted vertically.
14	Hookswitch	Picking up the handset from the handset cradle, the hookswitch bounces and the phone connects to the line. Laying the handset down on the handset cradle, the phone disconnects from the line.
15	Camera Lens	2 Mega-pixel camera. Provides near-site video. The better distance between camera and images you want to capture should be in the range of 0.35 meters (1 foot) to 2 meters (6 feet).
16	Camera Indicator LED	Indicates the status of camera and video calls.
17	Shutter Switch	Covers and uncovers the camera. When the camera is switched off, the video image is black.
18	USB2.0 Port	Allows you to connect the USB flash drive/USB headset to the phone.

LED Indications

Power Indicator LED

LED Status	Description
Solid Red	When the Phone is not registered.
Fast Flashing Red	When the Phone is in ringing state.
Slow Flashing Red	When the Phone receives a missed call, message or voice mail.
Off	When the Phone is powered off. When the Phone is in busy state. When the Phone is idle. When the call is placed on hold. When the call is muted.

Camera Indicator LED

LED Status	Description
Solid Green	When the Phone is powered on and the camera is connected properly. When the camera is idle. When the phone receives an audio call.
Solid Red	When the Phone receives a video call. When there is an active video call. When the video call is muted. When the video call is placed on hold.
Off	When the Phone is powered off. When the camera is not connected properly. When the camera shutter switch is closed.

Connecting Extended SPARSH VP710

For detailed instructions to connect Extended SPARSH VP710, see "Connecting Extended SPARSH VP710 as Extended SIP Extension" in SIP Extensions.

Configuring Extended SPARSH VP710

For detailed instructions on how to configure Extended SPARSH VP710, see "Configuring Matrix Extended SPARSH VP710".

Operating Extended SPARSH VP710

Please refer the *EXTENDED SPARSH VP710 User Guide* for instructions on operating the features of ANANT UCS.

Matrix VARTA ADR100 UC Client

Matrix VARTA ADR100 is a proprietary SIP (Session Initiation Protocol) based UC Client running on Android Phones/Tablets, delivering full-array of Matrix ANANT UCS features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise mobility features of the ANANT UCS, Matrix VARTA ADR100 provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using UC features like Presence, IM, IM to SMS, Corporate VMS access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or traveling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhance speed of communication and collaboration with office users and customers.

Make sure the phone/tablet in which you install Matrix VARTA ADR100, runs on Android V4.0.3 or later.

To use MATRIX VARTA ADR100 make sure you have:

- Purchased and activated the VARTA Essential or VARTA Professional or VARTA Collaboration license. For more details, see *“Licenses Supported in ANANT UCS”* and *“License Management”*.
- Assigned the desired license to the SIP Extension. For more details, see *“VARTA License Management”*.

Key Features

- **System Extension:** Matrix VARTA ADR100 becomes a mobile extension of the ANANT UCS. As an increased number of business professionals are using the collaborative tools found in smartphone devices to help them in their work activities, this application offers businesses an easy way to integrate their enterprises’ voice solutions within the Android OS family.

- **Advanced Call Capabilities:** Access to features such as Callback, Dial-in Conference, Conversation Recording and many more.

- **Mobility:** Matrix VARTA ADR100 provides you the mobility that you need in today's highly competitive business environment; with the ability to access ANANT UCS features easily once you are connected to either Wi-Fi or 3G network. Considering the case of roaming users, one can register Matrix VARTA ADR100 with the ANANT UCS using the enterprise Wi-Fi network when working within the office (that is within the organization's dedicated Wi-Fi coverage area). While working out of the office (where Wi-Fi network may not be available), one can register the application using the Mobile Data (3G) network.

- **Single Number Reach:** Retains the identity of the corporate phone system while working away from the office; so enhances business collaboration and lowers communication delays.

- **Dial by Extension:** Flexibility to reach to office users with direct extension number dialing.

- **Corporate Directory Access:** Enhance business collaboration with one-touch access to the Corporate Directory contacts using the ANANT UCS Global Directory.

- **Video Support:** The application offers the added advantage of Video Calling.

- **Presence:** Supports changing your Presence status as well as you can view the Presence status of other extension users.

- **Busy Lamp Field (BLF):** Using BLF you can monitor the status of another extension or trunk and confirm whether it is available or busy or ringing or on hold.

- **IM and SMS:** The application allows you to send IM and SMS to remote users.

- **One Touch Transfer:** You can transfer an ongoing call to a fixed extension without entering the number of that extension and without putting the call on hold. Similarly, you can also transfer a call from the fixed extension to your application.

- **Voicemail Access:** Access to the corporate Voicemail System from any location ensures no opportunity is lost.

- **Multiple Call Support:** With multiple call support, you can easily handle multiple incoming calls, merge and split calls apart, and place users on hold with a simple tap. With this Android Application, it’s like taking your deskphone on the road.

- **Wi-Fi to Cellular Handover and vise versa:** The application can automatically move an active call from the application to your cellular number on the cellular network and vise versa, without disconnecting the call and having to redial.

- **Better Voice Quality:** Using customized codec settings, enhanced voice output is available. If you are aware of the bandwidth and the network criteria of your location, you can select the proper codec from the application to get high quality voice output.

- **Standard Telephone Features:** Provides intuitive access to Keypad, Contacts, Call Logs and more, based on the Native Android design. One-touch access to call feature options during VoIP (Voice over IP) calls including Adding a New Call, Mute, Hold, Transfer and Speakerphone. Also provides DTMF support to enter numbers using an Auto Attendant.

- **Cost Effective Calling:** If you are using the enterprise Wi-Fi network to register Matrix VARTA ADR100 with the ANANT UCS; calls made from the application will be almost free. Even if you are using the application via 3G network during roaming, external calls can be made using the ANANT UCS trunks and thus reducing mobile calling and roaming charges.

- **Multiple Language Support:** The application can be viewed in six different languages including English, French, German, Spanish, Portuguese and Italian.

- **Application diagnostics:** Supports logging and sending of log files to concerned recipients by e-mail. These logs are used by the system engineer and/ or Matrix support engineers for troubleshooting.

Installing VARTA ADR100

For detailed instruction to install VARTA ADR100, refer to the *Matrix VARTA ADR100 User Guide*.

Configuring VARTA ADR100

For detailed instructions on how to configure VARTA ADR100, see “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

Operating VARTA ADR100

Refer to *Matrix VARTA ADR100 User Guide* for instructions on operating the features of ANANT UCS.

Matrix VARTA AMP100 UC Client

Matrix VARTA AMP100 is a proprietary SIP (Session Initiation Protocol) based UC Client running on iPhones, delivering full-array of Matrix ANANT UCS features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise mobility features of the ANANT UCS, Matrix VARTA AMP100 provides advanced call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these, you can take the advantage of using UC features like Presence, IM, IM to SMS, Corporate VMS access to enhance your overall mobile experience.

Mobile workers can use any Wi-Fi or cellular data networks to stay connected with business communications while working from office, home or travelling to any location. An innovative and easy to understand user interface delivers all productivity features at fingertips that enhances speed of communication and collaboration with office users and customers.

Make sure the phone in which you install Matrix VARTA AMP100, runs on **iOS9** and above.

To use MATRIX VARTA AMP100 make sure you have:

- Purchased and activated the VARTA Essential or VARTA Professional or VARTA Collaboration license. For more details, see “Licenses Supported in ANANT UCS” and “License Management”.
- Assigned the desired license to the SIP Extension. For more details, see “VARTA License Management”.

Key Features

- **System Extension:** Matrix VARTA AMP100 becomes a mobile extension of the ANANT UCS. As an increased number of business professionals are using the collaborative tools found in smartphone devices to help them in their work activities, this application offers businesses an easy way to integrate their enterprises’ voice solutions within the iOS family.

- **Advanced Call Capabilities:** Access to features such as Callback, Dial-in Conference, Conversation Recording and many more.

- **Mobility:** Matrix VARTA AMP100 provides you the mobility that you need in today's highly competitive business environment; with the ability to access ANANT UCS features easily once you are connected to either Wi-Fi or 3G network. Considering the case of roaming users, one can register Matrix VARTA

AMP100 with the ANANT UCS using the enterprise Wi-Fi network when working within the office (that is within the organization's dedicated Wi-Fi coverage area). While working out of the office (where Wi-Fi network may not be available), one can register the application using the Mobile Data (3G) network.

- **Single Number Reach:** Retains the identity of the corporate phone system while working away from the office; so enhances business collaboration and lowers communication delays.
- **Dial by Extension:** Flexibility to reach to office users with direct extension number dialing.
- **Corporate Directory Access:** Enhances business collaboration with one-touch access to the Corporate Directory contacts using the ANANT UCS's Global Directory.
- **Video Support:** The application offers the added advantage of Video Calling.
- **Presence:** Supports changing your Presence status as well as viewing Presence status of other extension users.
- **Busy Lamp Field (BLF):** Using BLF you can monitor the status of another extension or trunk and confirm whether it is available or busy or ringing or on hold.
- **IM and SMS:** The application allows you to send IM and SMS to remote users. It also supports the Emoji keyboard to add Emoticons (Smileys) in your messages.
- **One Touch Transfer:** You can transfer an ongoing call to a fixed extension without entering the number of that extension and without putting the call on hold. Similarly, you can also transfer a call from the fixed extension to your application.
- **Voicemail Access:** Access to the corporate Voicemail from any location ensures no opportunity is lost.
- **Multiple Call Support:** With multiple call support, you can easily handle multiple incoming calls, merge and split calls apart, and place users on hold with a simple tap. With this iPhone application, it’s like taking your deskphone on the road.
- **Cellular to Wi-Fi Handover and vice-versa:** You can move an active call from the Cellular number (on the Cellular network) to your application (registered using Wi-Fi network) without disconnecting or redialing the number. Similarly Wi-Fi to Cellular Handover is also possible where you can manually handover your call from the Wi-Fi to the Cellular network.
- **Better Voice and Video Quality:** Using customized codec settings and video quality preferences, enhanced voice output and video rendering are available. If you are aware of the bandwidth and the network criteria of your location, you can select the proper codec and video quality option from the application to get optimum audio and/or video output.
- **Standard Telephone Features:** Provides intuitive access to Keypad, Contacts, Call Logs and more. One touch access to call feature options during VoIP (Voice over IP) calls including Adding a New Call, Mute, Hold, Transfer and Speakerphone. Also provides DTMF support to enter numbers using an Auto Attendant.
- **Cost Effective Calling:** If you are using the enterprise Wi-Fi network to register Matrix VARTA AMP100 with the ANANT UCS; calls made from the application will be almost free. Even if you are using the application via 3G network during roaming, external calls can be made using the ANANT UCS trunks which reduces calling and roaming charges to a significant amount.

- **Multiple Language Support:** The application can be viewed in six different languages including English, French, German, Spanish, Portuguese and Italian.

- **Application Diagnostics:** Supports logging and sending of log files to concerned recipients by e-mail. These logs are used by the system engineer and/or the Matrix support engineers for troubleshooting.

Installing VARTA AMP100

For detailed instruction to install VARTA AMP100, refer to the *Matrix VARTA AMP100 User Guide*.

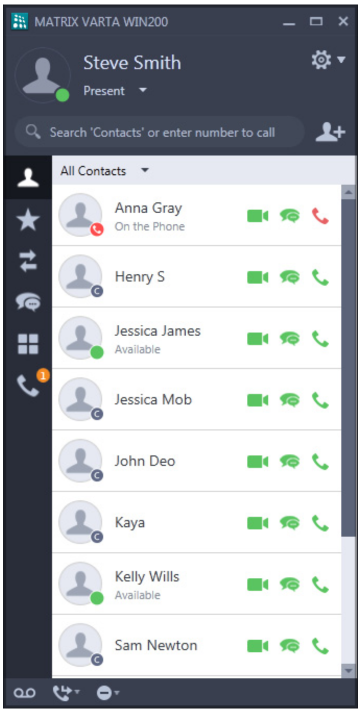
Configuring VARTA AMP100

For detailed instructions on how to configure VARTA AMP100, see “Configuring Matrix VARTA ADR100/AMP100 UC Clients”.

Operating VARTA AMP100

Refer to *Matrix VARTA AMP100 User Guide* for instructions on operating the features of ANANT UCS.

MATRIX VARTA WIN200



MATRIX VARTA WIN200, is a SIP (Session Initiation Protocol) based Unified Communication Client running on Windows OS, delivering full-array of the System features to the user on-the-go along with an added advantage of video calling. Through tight integration with the enterprise features of the System, UC Client provides advance call capabilities including Conferencing, Corporate Directory Access (Global Directory), Call Logs and Conversation Recording with one-touch access. Other than these you can take the advantage of using UC features like Presence, IM, IM to SMS, Corporate VMS access to enhance your overall mobile experience.

To use MATRIX VARTA WIN200 make sure you have:
- Purchased and activated the VARTA Essential or VARTA Professional or VARTA Collaboration license. For more details, see *“Licenses Supported in ANANT UCS”* and *“License Management”*.
- Assigned the desired license to the SIP Extension. For more details, see *“VARTA License Management”*.

Installing MATRIX VARTA WIN200

For detailed instruction to install MATRIX VARTA WIN200, refer to the *MATRIX VARTA WIN200 User Guide*.

Configuring MATRIX VARTA WIN200

For detailed instructions on how to configure MATRIX VARTA WIN200, see “Configuring Matrix VARTA WIN200 UC Client”.

Operating MATRIX VARTA WIN200

Refer to *MATRIX VARTA WIN200 User Guide* for instructions on operating the features of ANANT UCS.

Firebase Cloud Messaging (FCM) Support

Firebase Cloud Messaging (commonly referred to as Android Push Notification or FCM) is a platform notification service created by Google LLC that enables third party application developers to send notification data to their applications installed on Android devices.

Previously, VoIP applications needed to maintain a persistent connection in order to receive calls. Keeping a connection open in the background, drains the battery as well as causes all kinds of problems when the application crashes or is terminated by users.

In Android 4.1 and above, Google has introduced FCM as part of their effort to improve battery life, performance and stability for VoIP applications such as Skype, WhatsApp, etc. FCM offers high-priority push notification with a large payload. The VoIP application receives the notification in the background, sets up the connection and displays a local notification to the user.

ANANT UCS supports FCM for VARTA ADR100 Application only. Push Notifications will be sent for calls, new messages as well as for voicemail. Push Notifications will be sent to the MATRIX VARTA ADR100 Application only if it is in the background and when there is persistent internet connection. You will receive the Push Notifications even after you exit the application provided the *Calls and Messages after exit* check box is enabled in the VARTA ADR100 Application. For details refer to the VARTA ADR100 User Guide.

How it works

Pre-requisites for Push Notifications:

- Make sure that the server has a persistent internet connection and there is connectivity with the FCM Server. To check the connectivity, refer "FCM Connectivity".
- Make sure the Date and Time of the server is synchronized with the NTP Server.
- To receive IM and IM notifications make sure the application is registered at Location 1. For more details, refer "Configuring Matrix VARTA ADR100/AMP100 UC Clients".

Let us see how the notifications will be sent by the server when MATRIX VARTA ADR100 application is registered with the server as a SIP Extension and it is in the background. There is an incoming call or message:

- You can check the status of the SIP Extension user. It will display Registered (as the device is in the background) and under the respective Contact 1, 2, 3, it will display the time remaining for the expiry of the VARTA Client Inactivity Timer. The default value of the VARTA Client Inactivity Timer is 10 days. To configure this timer, refer to "System Timers and Counts".

- The server will send a Push Notification to the MATRIX VARTA ADR100 application (client).

- The server will wait for 15 seconds after sending the Push Notification:

    - if the client registers with the server within this time, the call will be connected or the message will be delivered. The status of the SIP Extension will display Registered and under the respective Contact 1, 2, 3 it will display the SIP ID, IP Address and the Registration Expiry Timer.

    - if the client does not register with the server within this time, the call will be disconnected or the message will be rejected.The status of the SIP Extension will display Registered and under the respective Contact 1, 2, 3 it will display the time remaining for the expiry of the VARTA Client Inactivity Timer.

- The server maintains a configurable timer, VARTA Client Inactivity Timer which is set as 10 days. Till the expiry of the timer the server will send Push Notifications to the application.
- If for this duration, the server does not receive any registration request from the application and the timer expires, the server will consider the application as unregistered and will stop all Push Notifications to the application.The status of the SIP Extension will display Not Registered and under the respective Contact 1, 2, 3 the details will be cleared. Calls and messages will be rejected.

  The server will start sending notifications to the application in the background after the application is bought in the foreground once and a registration request is received by the server.

Feature Interactions when the Application is in the Background

Call Forward when not Registered:

If the application user is a member of any Routing Group or Department Group, the Call Forward functionality will not be applicable.

To know more, see “Call Forward-When Not Registered”.

Handover and Handoff:

VARTA ADR100 users will be able to use Handover but Handoff will not be possible. To know more, see “Handover and Handoff”.

System Restart

After System Restart the VARTA Client Inactivity Timer will be reset.

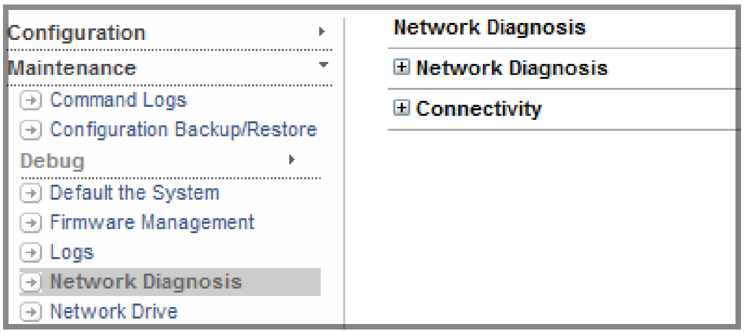
FCM Connectivity

A connectivity between the system and the FCM Server is required so that the Push notifications can be sent to the clients.

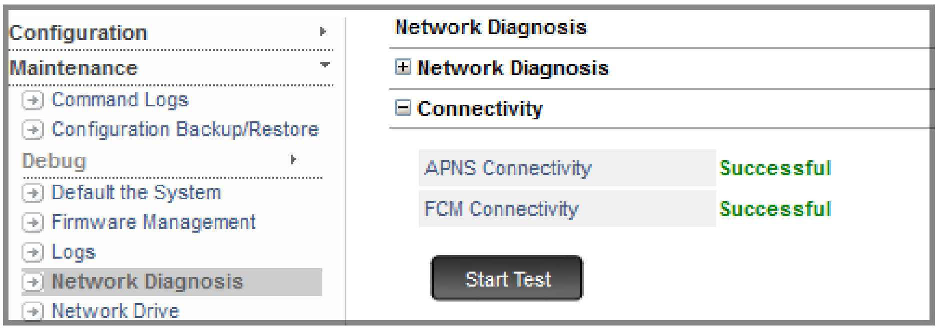
To check the FCM connectivity status,

- Login as System Engineer.
- Click **Maintenance.**

- Under Maintenance, click **Network Diagnosis.**



- Click **Connectivity** to expand.



- Click **Start Test**.

- In **Connectivity**, the status is displayed as:
  - Successful - if the connectivity between the system and the APNS/FCM Server is established
  - Timeout - if there is no connectivity.

*If the **Connectivity** Test of either of the servers (APNS or FCM) with ANANT UCS is not successful, then Push Notifications will not be sent to the Mobile Clients — MATRIX VARTA ADR100 / AMP100 application.*

Flashing on Trunks (Continued Dialing)

ITSP's support features like call waiting, call forward, etc. To be able to use these features, users need to dial certain codes during speech.

When a system is connected between the user and the ITSP's, the codes for dialing the features of the ITSP's may clash with the codes for accessing the features of the system, making it difficult for users to access the features of the ITSP's while in speech.

To overcome this, ANANT UCS supports Flashing on Trunks (Continued Dialing), which informs the system about the codes dialed for the features of the ITSP's on trunks by extension users.

How to configure

To be able to use this feature, *Continued Dialing* must be allowed to the extension in its "Class of Service (CoS)".

How to use

For Extended IP Phone Users

While in speech on trunk,

- Press 'Transfer' Key, dial * and the Desired Service Provider Code.
  Or
- Press DSS Key assigned to Flashing on Trunks (if programmed).
- Dial the Desired Service Provider Code.

Flexible Numbers

ANANT UCS offers Flexibility to assign a code of your choice to access an extension. This code is called Flexible number. For example, to access first SIP Extension one can dial 2001. It is possible to change this code to any other number of your choice.

ANANT UCS offers the following types of extensions, SIP Extensions, Virtual Extensions and Department Groups. The system loads default access codes to all extensions on first power ON. Later on the extensions can be assigned default Flexible numbers using a command.

The Default Access Codes for the Extensions are given below:

Software Port	Default Access Codes (Flexible Numbers)
0001 to 5000	Blank for SIP Extensions (This is the SIP ID)
01 to 64	Blank for Virtual Extensions
01 to 16 17 to 32	3901 to 3916 for Department Groups Blank

ANANT UCS also allows you to assign a Flexible Number to each extension individually or to a range of extensions simultaneously.

Configuring Flexible Numbers (Access Codes)

Assigning Access Codes to Extensions

You can configure the Flexible Numbers using Jeeves. For instructions, see “Configuring SIP Extension Settings as per the Extended Phone Type”, “Virtual Extension” and “Department Call”.

- *It is possible to have maximum of 6 digit flexible numbers.*
- *It is possible to clear the flexible number of a extension, range of extension and all extensions.*
- *Flexible numbers are the codes dialed from dial phase to call another extension. These flexible numbers should be unique and should not match with any of the features available from the dial phase.*
- *Flexible number having common digits can be assigned to another extension. Please refer “Conflict Dialing” for more details.*
- *Same flexible number cannot be assigned to two different extensions.*
- *Flexible numbers for all the features can be used from User mode and SA mode.*

Assigning Access Codes to a Range of Extensions

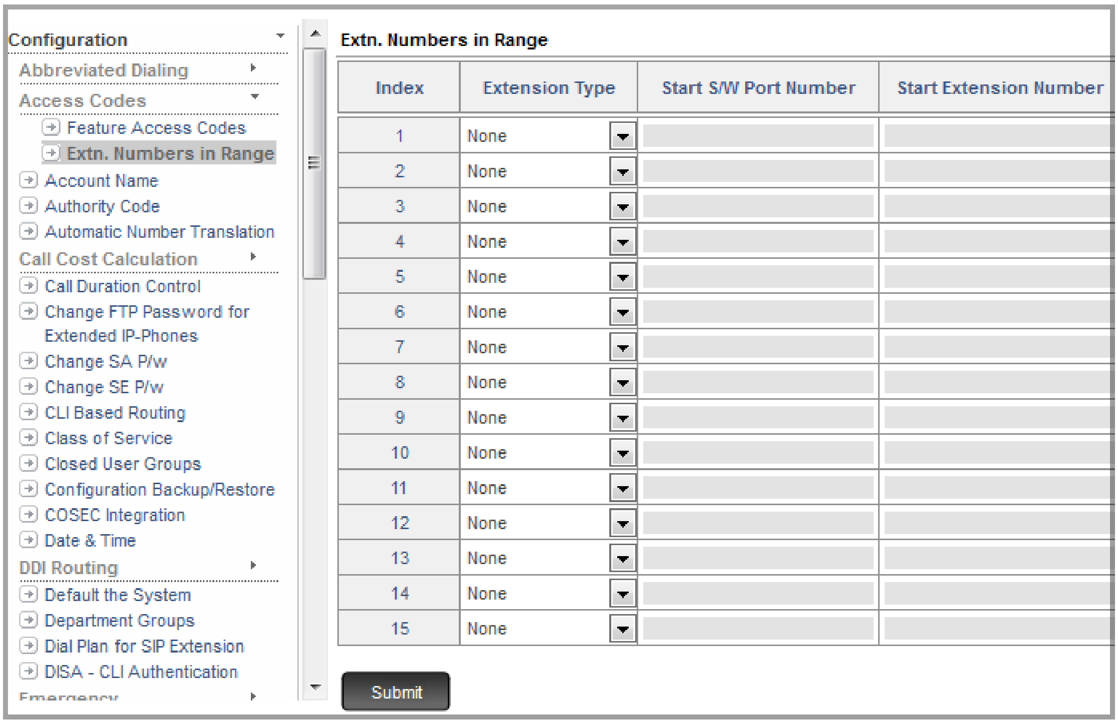
You can assign Access Codes to a range of Extensions.

If you are assigning a range of Extension Numbers (Access Codes) to the desired ports, and a match is found for the same extension numbers, the system will clear these extension numbers from the existing database. The new extension numbers will be assigned according to the given range.

Assigning Extension Numbers through **Extn. Numbers Range** has a priority over Extension Numbers assigned on individual ports.

To assign a range of Extension Numbers,

- Login as System Engineer.
- Under **Configuration**, click **Access Codes.**
- Click **Extn. Numbers in Range**.



- Against each Index configure the following:
  - **Extension Type:** Select the Extension Type. You can select SIP Extension, Department Group or Virtual Extension.
  - **Start S/W Port Number:** Enter the Software Port Number from which you want the system to start assigning the desired extension numbers.
  - Define the range of Station Access Codes (Extension Number/Flexible Number/Access Code) that you wish to assign to the Extension Type you selected in **Start Extension Number** and **End Extension Number**.
    For the given range of extension numbers, the system will assign extension numbers from the software port number specified in Start S/W Port Number for this particular entry. The range of extension numbers will be assigned in ascending order of the Software Port Number.

    For example:
    Extension Type you selected is SIP Extension
    Start Software Port Number is 1
    Start Extension Number is 2001
    End Extension Number is 2100

The system will assign Extension Number 2001 to SIP Extension 1, 2002 to SIP Extension 2 and so on. The system will assign the last Extension Number 2100 to SIP Extension 100.

If you want to assign access codes starting with ***#*** *or* ***** *to a range of extensions, make sure both* ***Start*** *and* ***End extension numbers*** *begin with* ***#*** *or* *****.

- Click **Submit**.

To check the Extension Numbers (Access Codes) assigned by the system,

- Click the Extension Type, in this case, under **Configuration** click **SIP Extension Settings**.

- Similarly you can assign the Access Codes (Flexible Numbers) to Department Group or Virtual Extension.

  The other parameters need to be configured manually on the respective pages of the SIP Extension/ Department Group/ Virtual Extensions.

Floor Service

The Floor Service feature allows you to provide a common access code to extension users which they can dial to call floor service.

Essentially a hospitality feature, Floor Service is also useful in offices. Floor service can be any administration or service department in the building, such as a stationery room, back office, backroom, photocopy/ mail room, secretarial assistance, concierge/janitor, Storeroom.

Just as all extension users can reach the Operator by dialing the common access code '9', they can reach the floor service by dialing a common access code, '38'. This is the default Floor Service access code, for all geographical regions where ANANT UCS is installed.

This feature can be used in:

- Multi-storied buildings, which have floor service (pantry, mail sorting, house keeping, janitor, coffee room, refreshment area) for each floor. The ANANT UCS can be configured to land calls made by extension users dialing the common access code '38' on the floor service extensions of their respective floors.
- Offices that have a centralized floor service, instead of one on each floor. ANANT UCS can be configured to land calls made from all extension phones by dialing '38' on the common floor service extensions.

This feature requires a license. To use this feature you must purchase the license for Hospitality. Refer the topic "Licenses Supported in ANANT UCS" to know more.

How it works

For example, Midas Towers houses different departments on each floor. Each floor has Floor Service.

Extensions 2001 to 2010 are on the first floor, 2011 to 2020 on the second floor, and 3001 to 3010 on the third floor. The floor service extensions are numbered as 2012 on the first floor, 2022 on the second floor and 3012 on the third floor.

With the Floor Service configured for each floor, when the extension user 2001 dials '38', the call will land on the service extension 2012, assigned to room service on the first floor. Similarly, when the extension user 3008 on the third floor dials '38', the call will land on the service extension 3012 on the third floor.

If Midas Towers had a single floor service extension 2012 for all floors, with Floor service configured, calls made from all extensions by dialing '38' would land on extension 2012 only.

How to configure

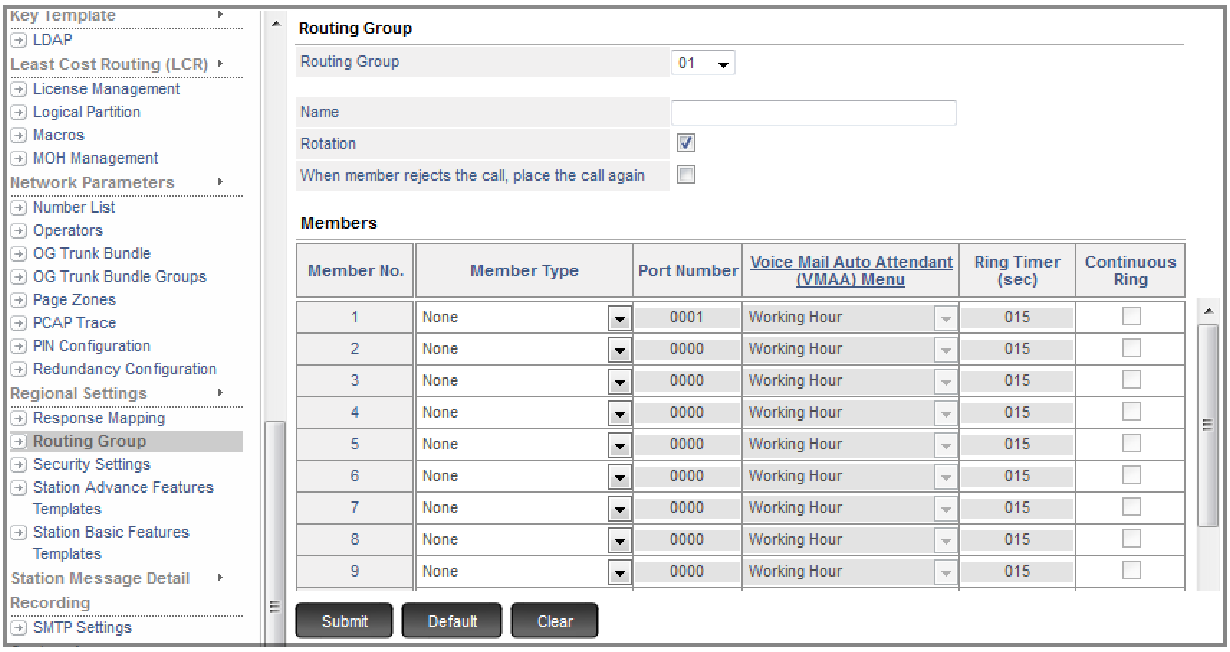
Configuring Floor Service feature involves the following steps:

1. Creating a routing group for each floor. Include Floor service extensions of a floor in a routing group prepared for that floor.
2. Assigning a routing group (number) in the Floor Service feature in the Station Advanced Feature Template. Prepare a different Station Advance Feature Template for each floor.
3. Applying the Station Advanced Feature Template (with the Floor service group programmed) to the extension. This will assign the extensions to the routing group programmed in the Template.

If the Enterprise/Building has centralized floor service, you only need to create a single Routing Group with service extensions, as required. This routing group number can be configured on a common Station Advanced Feature Template which will be applied to all extensions.

Configuring Floor Service

- Login as System Engineer.
- Under **Configuration**, click **Routing Group**.



- You can create 96 Routing Groups with 32 members in each group.
- Select a **Routing Group** Number from 1 to 96.
- For each Group you must configure the following:
    - Configure the **Name** you wish to assign to the Routing Group.
    - Select the **Rotation** check box to enable rotation of calls in the routing group which has multiple 'member' extensions. When enabled, each fresh call will land on the extension which is next to the one that received the last call. This ensures equal distribution of incoming calls to all the destinations within the routing group. The option is not relevant if the routing group has only one member extension.
    - If any member extension rejects an incoming call and the system again checks the routing group sequence, you can allow or restrict placing the same call on this extension. Select the **When member rejects the call, place the call again** check box, if you want the system to place the call again on the extension.
    - To configure Members in the Group,
        - Select the **Member Type**. You can select SIP Extension, Virtual Extensions, OTBG or the Voice Mail Auto Attendant.

Configure only as many extensions as you want in the routing group and set the remaining Member Types to 'None'.

For example: if you want to program only one extension in the routing group, set the Member Type in the remaining columns (Member 02-Member 32) to 'None.'You can program different routing groups for different floors. In each routing group you can program maximum 32 service extensions as 'members'.

If you have selected OTBG then enter the OTBG number here.

- If you have selected the *Voice Mail Auto Attendant* as the Port Type, select the **Voice Mail Auto Attendant (VMAA) Menu** to assign to the respective routing group.

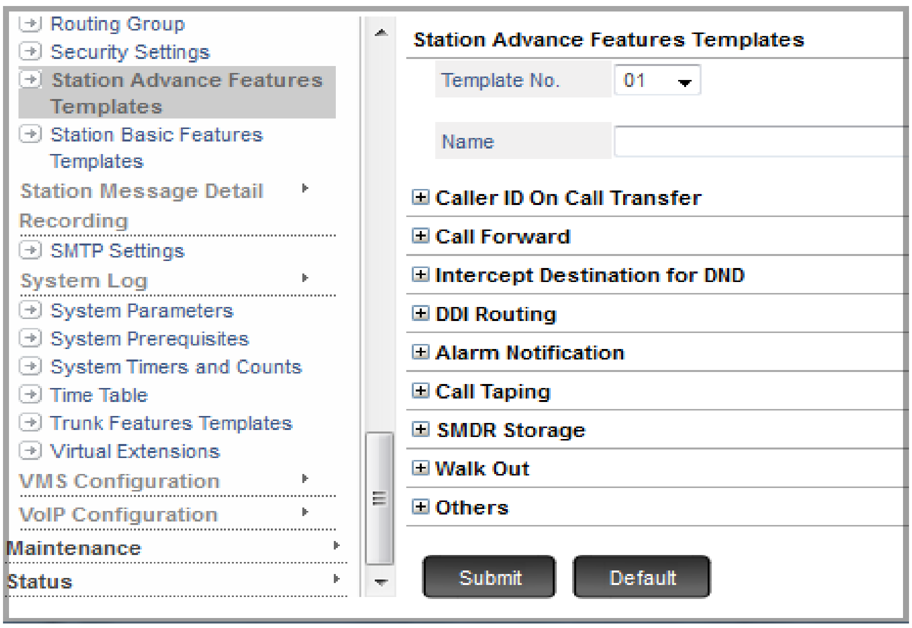
  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see “Voice Mail Auto-Attendant Menu”.

- In **Ring Timer (sec)**, configure the time for which the extension, on which the call lands, should ring. By default, the ring timer is set to 015 seconds and can be changed.

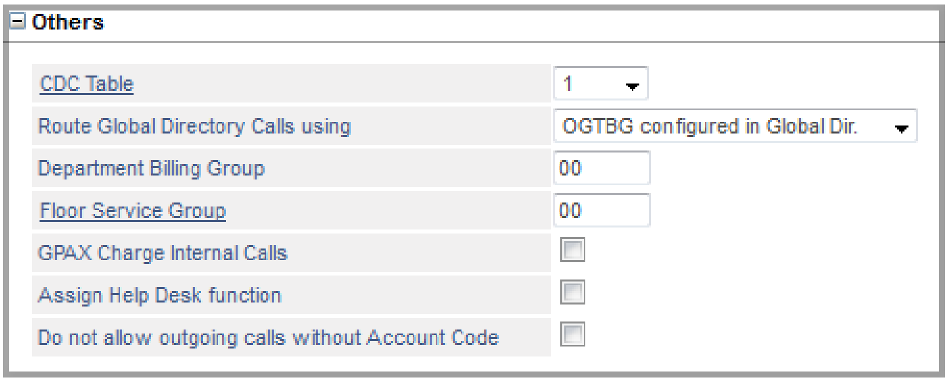
- Select the **Continuous Ring** check box, if you want an extension to ring continuously until the call is answered. The first extension will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This parameter is not relevant, if there is only one member extension in a routing group.

- Repeat the above steps to include other floor service extensions in the routing group
- Click **Submit**.
- Click **Station Advanced Feature Template** to open the page.

All extensions are assigned the Advanced Feature Template 01, by default. If the enterprise requires separate floor-service group for each floor, program a separate Station Advanced Feature Template for the extensions of each floor.



- **Template Number:** Select the **Template Number**, from 1 to 50 that you wish to configure.
- **Name:** Configure the **Name** you wish to assign to the Template you selected.
- Click **Others** to expand.



- **Floor Service Group:** This parameter is related to the Floor Service feature. Floor Service can be floor-wise or centralized. Floor Service requires you to configure the Routing Groups as landing destinations for extension calls.

  Configure the Floor Service (Routing) Group first and enter this Floor Service (Routing) Group number here. There are 96 different Routing Groups that can be configured as Floor Service Groups. By default, no Routing Group is assigned to Floor Service in the Template ('00').

  Calls from the extension will land on the Floor Service (Routing) Group you have assigned here.
- Click **Submit.**
- Now, apply the Station Advanced Feature Templates (with floor service routing groups programmed) to the extensions of the respective floors. For example: Template number 02 on extensions 2001 to 2010 are on the first floor, Template number 03 on extensions, 2011 to 2020 on the second floor, and Template number 04 on extensions 3001 to 3010 on the third floor.

  If extensions are SIP Extensions, assign the Template on the **SIP Extensions Settings** page.

  If extensions are Virtual Extensions, assign the Template on the **Virtual Extensions** page.

  Refer the topic “Station Advanced Feature Template” for instructions.

How to use

To be able to use floor service, extension users may dial the default Access Code defined for Floor Service: 38.

Check with your System Engineer if this access code has been changed and dial the new access code obtained from the System Engineer.

For Extended IP Phone Users

- Go OFF-Hook.
- Press DSS Key assigned to Floor Service (if programmed)
  OR
- Dial Access Code '38'
- Talk
- Go ON-Hook

Follow Me

Using this feature, extension users can make your calls follow you wherever you go. Extension users can receive their calls on another extension, whenever they want.

How it works

- A's extension number is 2001.
- B's extension number is 2003.
- A is currently at B's extension.
- A wants to receive calls from extension 2001 on extension 2003.
- A sets Call Follow Me on extension 2003.
- All calls landing on A's extension 2001 will be forwarded to extension 2003.
- When A returns to extension 2001, A cancels Call Follow Me.

- *The extensions dial tone changes to feature tone if its calls are forwarded.*
- *Multiple users can use 'Follow Me' from the same extension.*
- *Follow Me can be overwritten. Extension A sets Follow-Me on extension B. After a period of time; goes to extension C. A can receive calls on extension C by setting Follow Me on extension C. Follow Me set by A on extension B will be cancelled.*
- *Follow Me cannot be chained. If extension A sets Follow Me to extension B. And extension B sets Follow Me on extension C, Calls for A will land on B only and calls for B will land on C only.*
- *DND is given priority over Call Follow Me feature.*

Also see "Call Forward", "Class of Service (CoS)" and "Do Not Disturb (DND)".

How to configure

To be able to use Follow Me, extension users must have Call Forward feature enabled in their Class of Service for the time zone. For instructions, see "Class of Service (CoS)" and "Station Basic Feature Template".

How to use

For Extended IP Phone Users

To set Follow Me from another extension,

- Press 'Forward' Key of the other extension.
  OR
  Dial **135**.
- Enter your extension number.
- Enter your user password.

To cancel Follow me, from your extension,

- Press ‘Forward’ Key of your extension phone.
- Select ‘Cancel’
  OR
  Dial **130**.

Forced Answer

Extension users can force other extension users to answer their calls when there is no response from the called extensions.

How it works

Forced Answer can be requested by the calling extension. The calling extension may be an Extended IP Phone. However, the called extension (being forced to answer) must be an Extended IP Phone.

- Extension user A calls extension user B.
- B's phone is ringing, but B does not answer.
- A dials Forced Answer feature code.
- The speaker of B's phone's is turned on (goes OFF-Hook).
- A may now talk to B.

Forced Answer can be used when the called extension is idle or ringing.

How to configure

To be able to use Forced Answer, extension users must have this feature enabled in their Class of Service for the time zone. For instructions, see "Class of Service (CoS)" and "Station Basic Feature Template".

How to use

For Extended IP Phone Users

To use forced answer on an extension:

- Dial the desired extension number.
- Press the DSS Key assigned to Forced Answer on Ring Back tone.
  OR
  Dial **5** on Ring Back Tone.
- The Ring Back Tone stops.
- The called extension's speaker is turned on.
- You are in speech with the called extension.
- You may talk.

.

You can also dial '5', the feature code for Forced Answer, immediately after dialing the desired extension number, instead of dialing it during the Ring Back Tone. This way, you can talk to the desired extension user without waiting for the called extension user to answer your call.

Forced Call Disconnection

Forced Call Disconnection enables extension users to disconnect a busy extension at will, and free the system resources (access to extension) for themselves.

How it works

Forced Call Disconnection of an Extension:

- A, B and C are extensions.
- A and B are in speech.
- C calls B and finds it busy.
- C uses Forced Call Disconnection by dialing the feature command.
- C gets confirmation tone, while A and B get error tone.

Forced Call Disconnection of a Trunk:

- A and B are extensions. C is the external party.
- A is in speech with C on Trunk 1.
- B grabs Trunk 1 using *Selective Port Access*, but gets busy tone.
- B uses Forced Call Disconnection by dialing the feature command.
- B gets confirmation tone. A gets disconnected and gets error tone.
- B must grab Trunk 1 again to get the dial tone of the network.

- *To be able to use Forced Call Disconnection, the extension user must have a higher "Priority" than the extension user whom he/she tries to forcibly disconnect.*

- *To be able to use Forced Call Disconnection on a busy trunk, the extension user must have grabbed that trunk using "Selective Port Access". If the extension user has grabbed the trunk using a Trunk Access Code, the feature code to dial Forced Call Disconnection will not work.*

How to configure

To be able to use Forced Call Disconnection, the extension must have:

- **Forced Release** feature enabled in the Class of Service. For instructions see "Class of Service (CoS)" and "Station Basic Feature Template".

- As Forced Call Disconnection on a busy trunk is possible only if the extension user has grabbed that trunk using **Selective Port Access**, this feature must be enabled in the Class of Service of the extension. For instructions see "Class of Service (CoS)" and "Station Basic Feature Template".

- Higher **Priority** assigned than other extensions. See "Priority" for instructions.

How to use

For Extended IP Phone Users

To forcibly disconnect a busy extension/trunk[104]:

- Press the DSS Key assigned to Forced Call Disconnection on Busy tone.
  OR
  Dial **#*** on Busy tone.
- You get confirmation tone.
- You may now dial the extension number/grab the trunk.

You are advised to restrict access to this feature only to important extension users. Extension Users who are allowed this feature are advised to use it judiciously.

104. *Only if you have grabbed this trunk using Selective Port Access.*

Gain Settings

To avoid noise or echo during speech, you must set the speech volume levels on the SIP Trunks/Extensions. ANANT UCS allows you to set the speech volume levels for the SIP Trunks/Extensions.

The speech volume levels can be adjusted by increasing or decreasing the Gain Settings provided on the Trunk ports.

How it works

A call received on the SIP Trunk 1 can be placed on SIP Trunk 2. The speech volume levels may differ accordingly. Hence, on the SIP Trunk 1 you must set the speech volume levels for different SIP Trunks.

In this case, let us assume that the call on the SIP Trunk 1 is to be placed on the SIP Trunk 2.

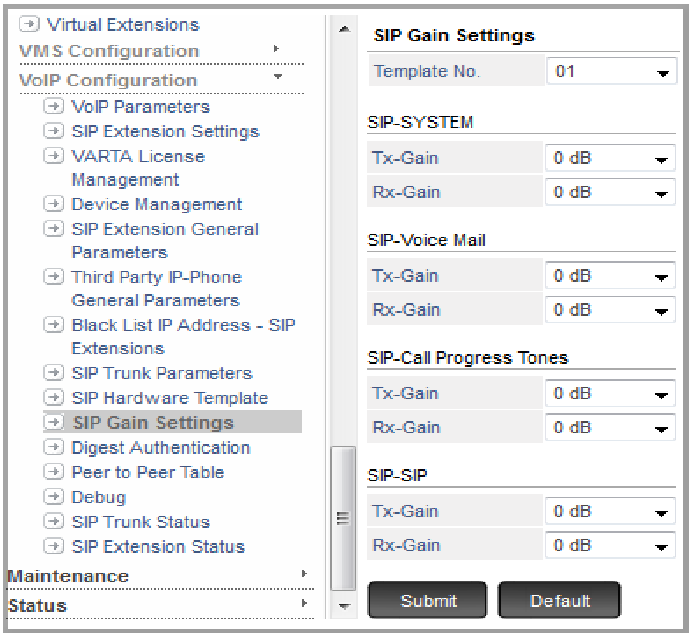
Before placing the call on the SIP Trunk, the system applies the SIP to SIP Gain Setting (Receive and Transmit Gain settings) configured on the SIP Trunk to adjust the speech volume level.

When the call is placed on the SIP Extension, the SIP to SIP Gain Settings (Receive and Transmit Gain settings) on the SIP Trunk are applied to adjust the speech volume level.

Hence, you can set different speech volume levels for each SIP Trunk and the system automatically detects and applies these gain settings for each SIP Trunk.

How to configure

- Login as System Engineer.
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Gain Settings**.



- **Template Number:** Select the desired **Template Number**, from 1 to 4 that you wish to assign in the SIP Hardware Template.
- **SIP - System (Tx-Gain and Rx-Gain)**: Configure the Gain Settings that you want the system to apply on the SIP Port with respect to the system (for example Call Conference). Valid Range for Tx Gain and Rx Gain: -24dB to +24dB.
- **SIP - Voice Mail (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP port when the incoming calls on the SIP port are being answered by the VMS. Valid Range for Tx Gain and Rx Gain: -24dB to +24dB
- **SIP - Call Progress Tones (Tx-Gain and Rx-Gain)**: Configure the Gain Setting that you want the system to apply on the SIP port while playing Call Progress Tones. Valid Range for Tx Gain and Rx Gain: -24dB to +24dB
- **SIP - SIP (Tx-Gain and Rx-Gain)**: Configure the Gain setting that you want the system to apply on the SIP port when the SIP port is connected to another SIP Port of the system during an incoming or outgoing call. Valid Range for Tx Gain and Rx Gain: -24dB to +24dB
- Click **Submit.**

Assigning the Gain Settings Templates

- Assign the SIP Gain Settings Template you configured in the SIP Hardware Template, see “SIP Hardware Template”.

Handover and Handoff

Handover allows you to move an active VARTA Mobile UC Client call from the Wi-Fi network to the cellular number (in the cellular network). This is useful when you have an ongoing call and you leave the Wi-Fi network, or if there are voice quality issues over the Wi-Fi network.

Handoff is when you are back into the Wi-Fi network, you can move the call from the cellular number to the VARTA Mobile UC Client. The call is moved without being disconnected and redialing the number.

- *ANANT UCS will serve the Handover request only if:*
  - *the Cellular Number is configured in the Application Settings in the Mobile Client. Refer to the respective Mobile Client User Guide for details.*
    OR
  - *the Mobile Number is configured in the system. See* *"Configuring Matrix VARTA ADR100/AMP100 UC Clients".*

- *When the VARTA ADR100/ VARTA AMP100 application is in the background, handoff will not be possible. Refer VARTA ADR100 User Guide and VARTA AMP100 User Guide to know more.*

How to configure

To use this feature, make sure the *Basic Features* are enabled in the Class of Service assigned to you. For instructions, see "Class of Service (CoS)" and "Station Basic Feature Template".

Help Desk

An organization may have a Centralized Information Office which provides information related to different departments such as HR, IT, or General information. For each department in the organization, an extension number can be defined as a Help Desk.

How it works

- Extension 2002 is defined as Help Desk for HR policies and general rules.
- Extension 2016 calls the Help Desk extension 2002.
- If the Help Desk extension is busy/not responding, an Auto Callback request is set automatically on the Help Desk extension.
- The system will serve the auto callback request as soon as the Help Desk extension is available.
- The Help Desk extension calls back extension 2016.

Also see “Auto Call Back (ACB)”.

How to configure

You can define an extension as ‘Help Desk’ by selecting the ‘Assign Help Desk function’ check box in its “Station Advanced Feature Template”.

Holiday Table

The Holiday Table feature of ANANT UCS enables you to configure incoming call management for holidays.

Using the Holiday Table feature of ANANT UCS you can,

- define the landing destination for incoming calls on trunks on holidays.
- greet callers with customized holiday messages.
- determine the way extensions must work on the holidays.

You can configure a list of holidays in a single table.

How it works

In the Holiday Table, you need define the following:

- The Start and the End Dates and the Time to be considered as Holiday.

- The Time Zone to be considered for operating the time-zone based trunk and extension features[105] during the Date and Time configured as Holiday. The Time Zone for Holiday can be defined as *Non-Working Hours*, *Working Hours*, or *As per Time Table*.

If you are using the Voice Mail Auto Attendant as the landing destination for calls, on holidays, you can play customized greeting messages to callers. For each holiday, you can play a different message.

For example, A company, ABC Ltd., has the following requirements:

- December 23 to December 31, all the employees will be on holiday. The callers must be greeted with a holiday message.
- January 1 to January 4, few employees will be attending the office.

In this case you must define the following in the Holiday Table:

At Index 1,

- In **Start**, enter the starting date and time of the holiday in DD-MMM-HH-MM format, that is 23 DEC, 00:00 and in **End** enter the last date and time of the holiday, that is 31 DEC, 23:59.

- In **Time Zone for Holiday**, select Non-working hours.

- In **Holiday Message** select the customized holiday message number. The system will greet the callers with this message.

At Index 2,

- In **Start**, enter the starting date and time of the holiday in DD-MMM-HH-MM format, that is 31 DEC, 00:00 and in **End** enter the last date and time of the holiday, that is 4 JAN 23:59.

---

105. *Trunk Landing Group, Auto-Attendant, DISA are time-zone based features of Trunks configured in the Trunk Feature Template assigned to trunks. Class of Service, Toll Control, and OG Trunk Bundle Group, are time-zone based features of extensions that are configured in the Station Basic Feature Template assigned to extensions.*

- In **Time Zone for Holiday**, select As per Time Table.

- In **Holiday Message** select **None**.

Index	Holiday								Name	Time Zone for Holiday	Holiday Message
	Start (DD-MMM-HH-MM)				End (DD-MMM-HH-MM)						
1	23	DEC	00	00	31	DEC	23	59	Christmas Holidays	Non-Working Hours	01
2	31	DEC	00	00	04	JAN	23	59	Christmas Holidays	As per Time Table	

After you have defined the above parameters, this is how the feature Holiday Table works,

- On the set date and time, when ANANT UCS detects a day as a holiday, it checks the configured Time Zone for Holiday and whether Holiday Message is configured.

- When the Time Zone for Holiday is defined as Non-working hours,
  - the system checks for and routes the incoming call on the trunk as per the Trunk Landing Group and the Auto Attendant defined for the trunk for Non-Working hours in the Trunk Feature Template.
  - customized Holiday Message, if configured, is played to the caller. The Holiday message is played in place of the Welcome message for Non-Working hours.
  - Extensions work according to the Class of Service, Toll Control and Outgoing Trunk Bundle Group assigned to them for Non-working hours in the Station Basic Feature Template.

Similarly, if the Time Zone for Holiday is defined as Working Hours, the system will operate the time-zone based features of the trunks and extensions according to Working Hours.

- When Time Zone for Holiday is As per Timetable,
  - the system routes the call as per the Trunk Landing Group and Auto Attendant defined for the current time zone in the Trunk Feature Template.
  - customized Holiday message, if configured, is played to the caller in place of the Welcome message for the current time zone.
  - extensions work according to the Class of Service, Toll Control and Outgoing Trunk Bundle Group assigned to them for the current time zone in the Station Basic Feature Template.

Feature Interaction:

Day/Night Mode: You can set the system in Day/Night Mode, even when you have configured the Holiday Table. In that case, the mode you select, Day (Working Hours) or Night (Non-working Hours) will override the Time Zone you have selected for Holiday.

In order for the Time Zone for Holiday to come into effect, you must set the Day/Night Mode in System Parameters to ***Operate System as per Timetable assignment***.

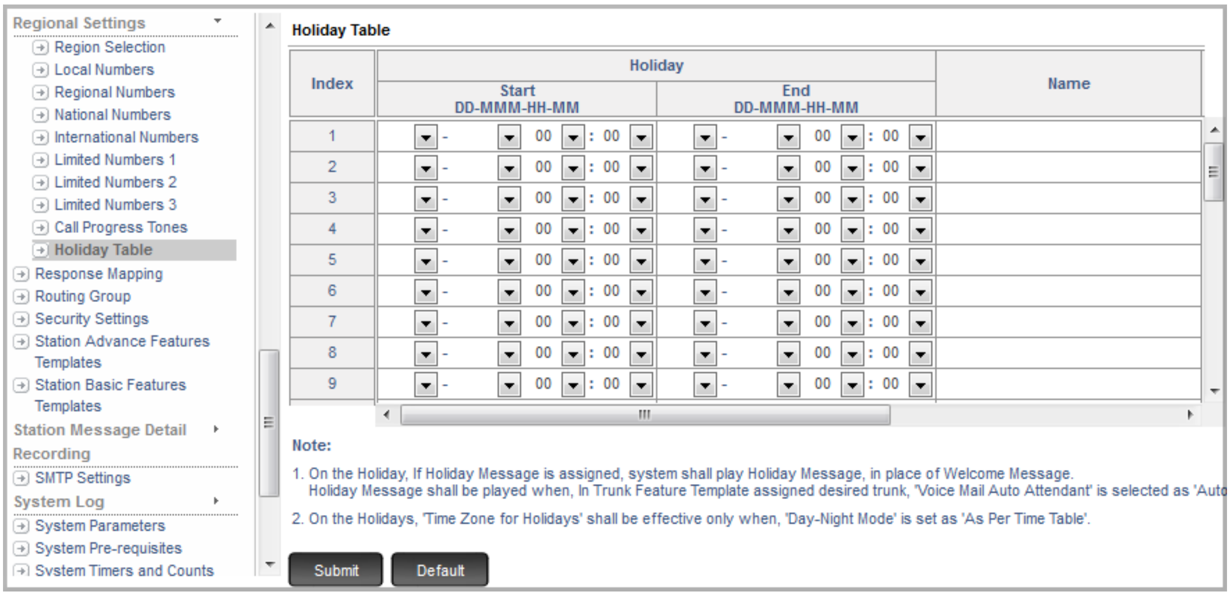
See “Day Night Mode” and “System Parameters” to know more about this feature.

How to configure

You can configure the Holiday Table from the SE as well as the SA mode.

Configuring Holiday Table

- Login as System Engineer.
- Under **Configuration**, click **Regional Settings**.
- Click **Holiday Table**.



Against each Index configure the following parameters:

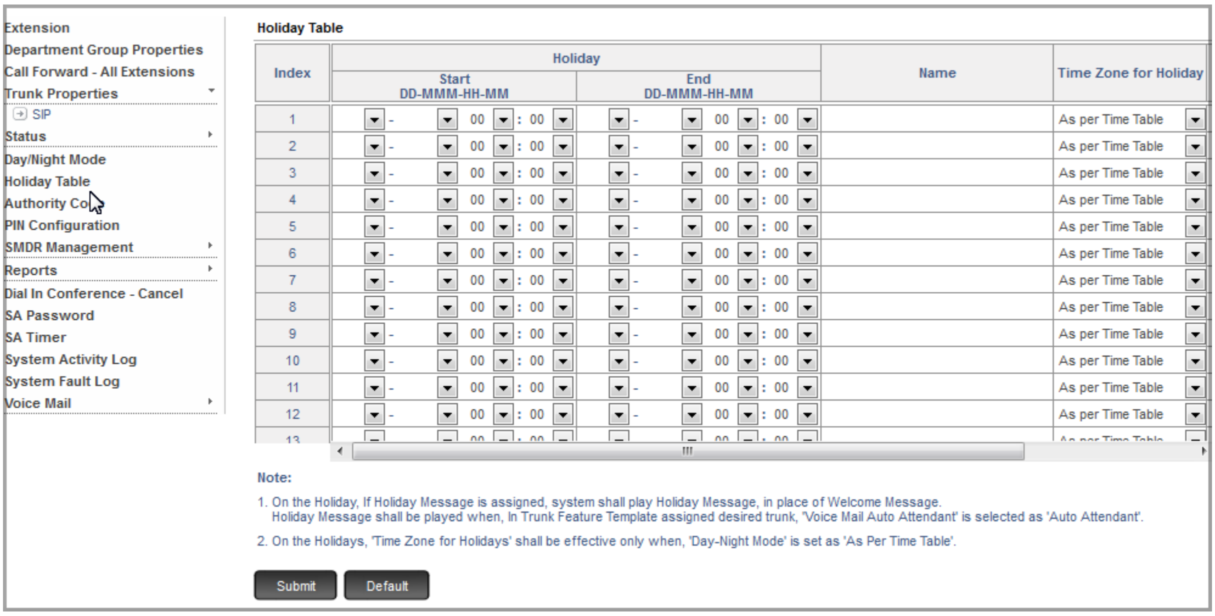
- In **Holiday** configure the **Start** Date, Month and Time of the holiday and **End** Date, Month and Time of the holiday.

  Default, Start and End Date are Blank. Valid Range is from 01 to 31.
  Start and End Month are Blank. Valid Range is from January to December.
  Start and End Time is 00:00 (Hours:Minutes). Valid Range is from 00:00 to 23:59

- You can assign a **Name** to each time period you have defined as Holiday. For example, Christmas, Independence Day, Thanksgiving. Default: Blank
- As the **Time Zone for Holiday** select the time zone for the period you have defined as Holiday: Working Hours, Non-working Hours or As Per Time Table. Default: As Per Time Table.
- If the incoming calls are routed to the VMS Auto Attendant, select the **Holiday Message** number that you want the system to play to the callers. Default: Auto_Attendant_01.
- Click **Submit**.

You can also configure the Holiday Table from the SA mode also. To do this,

- Login as System Administrator.
- Click **Holiday Table**.



- Follow the same steps as given above to configure the Holiday Table.
- Click **Submit.**

Configuring Holiday Messages

To greet callers with customized holiday messages, you need to do the following configuration:

- Ensure that Voice Mail Auto Attendant is selected in the Trunk Feature Template. For detailed information, see “Trunk Feature Template”.
- You can either use the default Holiday messages or customize your messages by recording messages of your preference. To know more about the default Holiday Messages and how to record customized Holiday Messages, see “Recording Voice Messages”.

Hotline

The Hotline feature connects the extension user immediately to a particular number or trunk, whenever the extension user goes OFF-Hook.

You can set Hotline to connect immediately to another extension, to a Department Group, to an external number or to an outgoing trunk.

Hotline set for external numbers and outgoing trunks is referred to as *Hot Outward Dialing*.

ANANT UCS offers two types of Hotline/Hot Outward Dialing:

- **Immediate:** As soon as the extension user goes Off-Hook, the user gets connected to the desired hotline extension number, department group, external number, or outgoing trunk. For this the *Hotline Timer* must be set to ‘00’ seconds (default: 3 seconds).
- **Delayed:** When the extension user goes OFF-Hook, the system plays Dial Tone to the extension user and waits for the *Hotline Timer* (default: 3 seconds). On the expiry of this timer, it connects the extension user to the desired hotline extension number, department group, external number or outgoing trunk.

How it works

- Hotline/Hot Outward Dialing can be set from an Extended IP Phone extension, if the extension has Hotline in its Class of Service.
- To be able to use Hotline/Hot Outward Dialing, extension users must do the following:
  - Select the type of Hotline they want to set on their extension; whether to an internal Extension Number, a Department Group, or an External Number or Outgoing Trunk.
  - Configure the *Hotline Timer*. For *Immediate Hotline*, extension users must set the Hotline Timer to ‘00’ seconds. For *Delayed Hotline*, extension users can set the Timer as per their requirement.
- Here is an example of how Hotline works:

  A frequently dials the number of B. So, A sets Hotline for B’s number and also sets the Hotline Timer to 5 seconds (Delayed Hotline).

  - A goes Off-Hook
  - ANANT UCS plays dial tone and waits for 5 seconds
  - If A dials a number within the Hotline Timer, ANANT UCS outdials the number dialed by A.
  - If A does not dial any digit within this time, ANANT UCS dials B’s number.
  - A gets connected to B.

If ‘Dial Tone’ timer of the system is less than Hotline Timer, the Hotline Timer will override the ‘Dial Tone’ timer. To know more about these timers, see *“System Timers and Counts”.*

  - If A had set the *Hotline Timer* to ‘00’ seconds (Immediate Hotline), A would be connected to B as soon as A goes Off-Hook.

- If A sets delayed Hot Outward Dialing for a Trunk or an External Number, the system will play dial tone to A and wait for the duration of the Hotline Timer for A to dial digits. If A does not dial any digits within this timer, the system connects A to the Trunk/External Number.

- If A sets immediate Hot Outward Dialing (Hotline Timer set to ‘00’ seconds), A will be connected to the Trunk/External number as soon as A goes Off-Hook.

- Delayed Hotline/Hot Outward Dialing allows extension users to dial out other numbers or grab another trunk, without having to cancel the Hotline/Hot Outward Dialing they have set for a particular number or trunk.

How to configure

To be able to use Hotline, extension users must have this feature enabled in their “Class of Service (CoS)” for the time zone, as required.

How to use

Hotline can be set/canceled by users for their own extension, or for any other extension from the SA mode.

Hotline when set/canceled from the SA mode, will not depend on the assigned CoS.

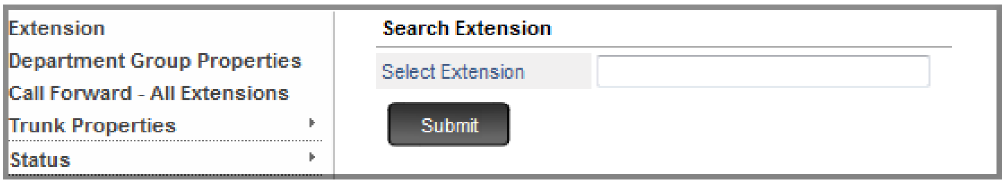
Set/Cancel Hotline for Extension Users

The Operator or any extension user having access to System Administrator mode can set or cancel Hotline for other extension users.

Setting Hotline

- Login as System Administrator.

- Click **Extension**.

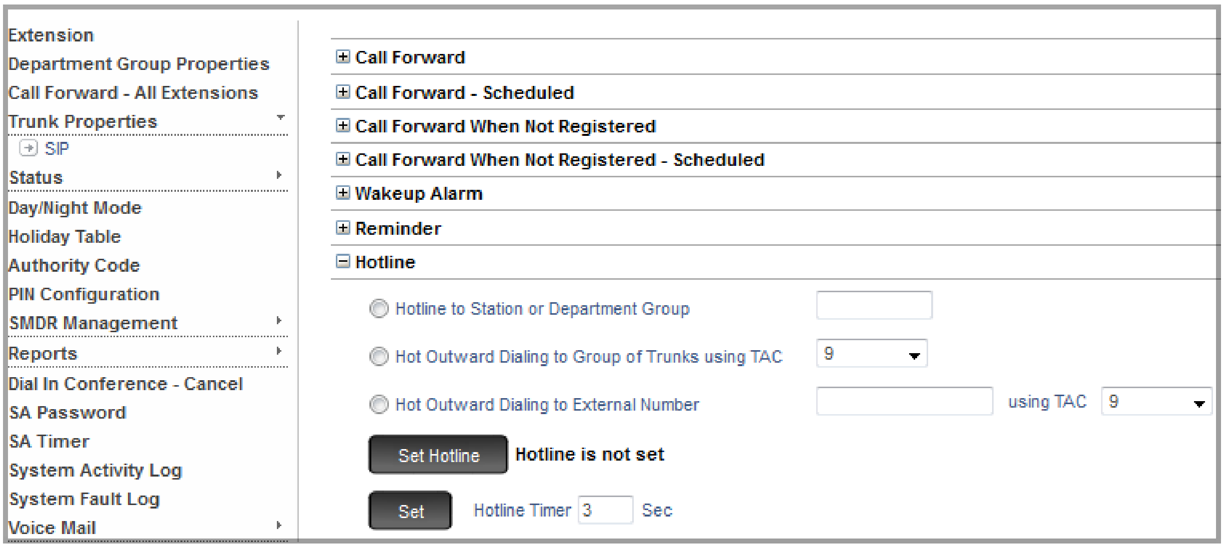


- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.

- Click **Submit.**

- The searched extension users details appear on your screen.

- Click **Hotline** to expand.



- Select the type of Hotline you want to set for the extension user from the following:

  - To set Hotline for an Extension or Department Group, select **Hotline to Station or Department Group**. Enter the Extension number or the Department Group Number in the corresponding box. Default: Blank.

  - To set Hotline for a group of trunks, select **Hot Outward Dialing to Group of Trunks using TAC** and select the Trunk Access Code from the corresponding drop down list.

    Each Trunk Access Code has a group of trunks for which Hotline will be set.

  - To set Hotline for an external number, select **Hot Outward Dialing to External Number.** Enter the external number in the corresponding box and in **Using TAC** select a TAC from the drop down list. Using a free trunk from this TAC the external number will be dialed out by the system.

- Click the **Set Hotline** button to set Hotline.

  The message “Hotline is set” appears.

- To set Delayed Hotline, in **Hotline Timer** enter the desired time in seconds.

  On the expiry of this timer, it connects the extension user to the desired hotline extension number, department group, external number or outgoing trunk.

- Click the **Set** button to set Delayed Hotline.

- To cancel Hotline, click the **Cancel Hotline** button.

When you set the Hotline Timer to ‘00’ seconds (for immediate Hotline), you will not be able to dial any digits, not even the feature code to Cancel Hotline.

Set/Cancel Hotline by Extension Users

For Extended IP Phone Users

To set Hotline on an Extension / Department Group,

- Press DSS key assigned to Hotline.
- Scroll to select ‘Set Hotline: Stn/ Dept’, press enter key
- Enter Extension / Department Group Number
  OR
- Dial **151** and enter Extension Number / Department Group Number

To set Hot Outward Dialing for Trunk,

- Press DSS key assigned to Hotline.
- Scroll to select ‘Set Hotline: OG Trunk’, press enter key
- Enter TAC
  OR
- Dial **152-TAC**

To set Hot Outward Dialing to External Number,

- Press DSS key assigned to Hotline.
- Scroll to select ‘Set Hotline: Ext Num’, press enter key
- Enter TAC
- Enter External Number #*
  OR
- Dial **153-TAC-External Number#***

You cannot set Hotline and Hot Outward Dialing on the same extension at the same time.

To set Hotline Timer,

- Press DSS key assigned to Hotline
- Scroll to select ‘Set Hotline Timer’, press enter key
- Enter Hotline Timer:000-255 seconds
  OR
- Dial **154-seconds (000-255)**

  The default value of Hot Line Timer is 3 seconds.

To Cancel Hotline / Hot Outward Dialing,

- Press DSS key assigned to Hotline
- Scroll to select ‘Cancel Hotline’, press enter key.
  OR
- Dial **150**

The cancellation code must be dialed from the dial tone. You have to be very quick in dialing the cancellation code, if the delay in the Hotline Timer is set to 1 or 2 seconds.

Incoming CLI Modification[106]

For System users in countries, where the Calling Line Identification (CLI) received must be suitably modified before it can be used to dial out the number, ANANT UCS offers the feature 'Incoming CLI Modification'.

The Incoming CLI is received with the Country or Area Code, or both. However, the dialing pattern of the public network may require the received CLI to be prefixed with additional digits, to dial out the same number. Or the dialing pattern of the public network may require the CLI to be stripped off the prefixed digits to dial out the same number.

With the feature 'Incoming CLI Modification' configured, ANANT UCS detects whether the incoming CLI is a local number, a national, or an international number. It modifies the incoming CLI accordingly, by adding or stripping off the prefixed digits so that the number can be dialed out as per the dialing pattern supported by the public network.

The modified CLI is presented to the extension phones and is stored in the "Call Logs", and SMDR (see "Station Message Detail Recording (SMDR)"). Extension users can call a number in the Call Logs without having to modify the CLI manually.

How it works

- Incoming CLI Modification parameters must be configured in the system considering the dialing pattern supported by the local public network.
- Accordingly, ANANT UCS matches the CLI received with the configured parameters.
- It detects whether it is an international, national or local number.
- It modifies the CLI according as per the Modification parameters configured.
- It presents the modified CLI to the extension; stores the modified CLI in the SMDR and in the Call Logs of the extension.
- When the received CLI is dialed out by the extension user from Call Log, ANANT UCS dials out the same number.

How to configure

For this feature to work, **Enable Incoming CLI Modification** check box in "System Parameters" and **Allow Incoming CLI Modification** check box in SIP trunk must be enabled. Refer "Configuring SIP Trunks" to know more.

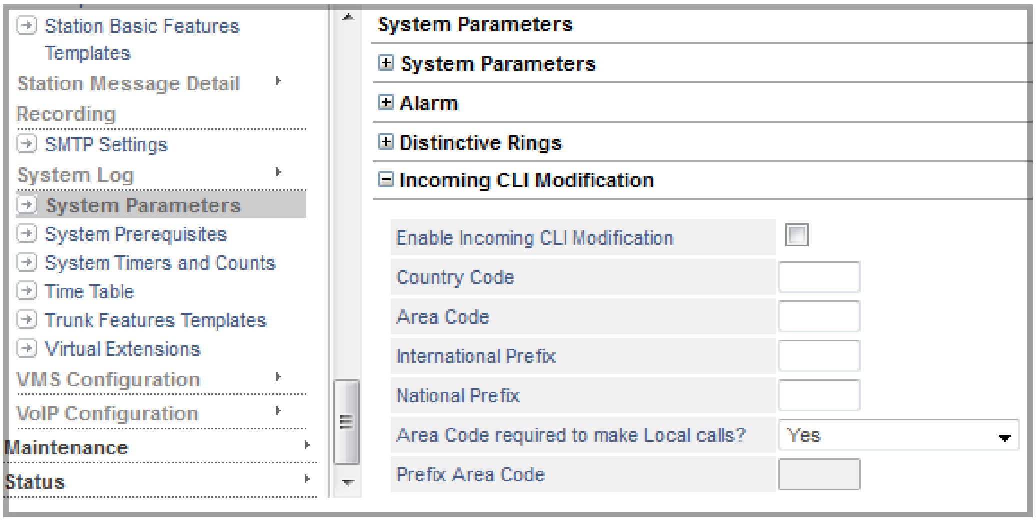
Configuring Incoming CLI Modification

- Login as System Engineer.

- Under **Configuration**, click **System Parameters.**

- Click **Incoming CLI Modification** to expand.

106. To comply with the Indian Government Laws and Regulation, this feature is not provided for India Region.



Now, configure these parameters:

- **Enable Incoming CLI Modification:** Select this check box if you want to use the Incoming CLI Modification feature. By default, this check box is disabled.

If you receive CLI in dial-able format, there is no need to use this feature. In such case, keep the check box *disabled. You do not need to configure any of the CLI modification parameters.*

- **Country Code:** Enter the Country Code of the country where ANANT UCS is installed. The Country Code helps ANANT UCS detect whether the Incoming CLI received is a national or an international number. Do not enter any prefix for the Country Code. By default it is Blank.

- **Area Code:** Enter the Area Code of the place where the ANANT UCS is installed. The Area Code helps ANANT UCS detect whether the Incoming CLI received is a local number. Do not enter any prefix for the Area Code. By default it is Blank.

- **International Prefix:** Enter the digits that are required as Prefix for dialing International Numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. By default it is Blank.

- **National Prefix**: Enter the digits that are required as Prefix for dialing long distance, National (within the country) numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. By default it is Blank.

- **Area Code required to make local calls?:** Depending on the dialing pattern of your local public telephone network, you may choose from the following options:

  - **No (Area Code not required)**: select this option if your public telephone network does not require the dialing of Area Code for local numbers.

  - **Yes (Area Code is required)**: select this option if your public telephone network requires you to dial the Area Code for local numbers.

  - **Yes (Area Code with Prefix required):** select this option if you public telephone network requires you to dial Area Code with a particular Prefix for local numbers. If you select this option, you must also program the Prefix digits for the Area Code.

  By default, the option, ‘No (Area Code not required) is selected.

- **Prefix Area Code:** If you have enabled the ‘Area Code required to make local calls’ check box in the previous parameter, enter the prefix digits for the area code for local calls in this field.

- Click **Submit.**

Intercom

The Intercom feature of ANANT UCS enables extension users to connect quickly with any desired extension, without waiting for the called extension to answer.

- *ANANT UCS will serve an Intercom call made by an extension only if:*
  - *the called extension is in idle state.*
  - *the called extension is able to identify the incoming call as an intercom call (applicable in the case of Standard SIP Phones).*
  - *the calling extension has Intercom in its Class of Service.*
  - *the Priority of the calling extension is higher than that of the called extension.*
- *ANANT UCS supports Intercom using Call-INFO / Alert-INFO Message. For a list of IP phones on which this feature has been tested, see* *"ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.*

How it works

- A's extension number is 3001 with Priority Level '7' and 'Intercom' feature enabled in the Class of Service.
- B's extension number is 3003 with Priority Level '5'.
- A wants to quickly connect to B. A dials the Intercom feature code ***5** followed by B's number, 3003.
- ANANT UCS places the Intercom call on B.
- B's extension is idle at the time of the call, and the speaker of B's phone goes OFF-Hook, creating a speech path between A and B.
- A can now talk with B.

Feature Interactions

- **Do Not Disturb (DND):** If the called extension has set DND, ANANT UCS will not place the intercom call on the called extension.
- **Privacy from DND Override:** If DND as well as Privacy from DND Override is enabled in the Class of Service of the called extension, ANANT UCS will reject the Intercom call.
- **Call Forward-Unconditional:** If the called extension has set Call Forward-Unconditional, ANANT UCS will forward the intercom call to the forwarded destination number. The call placed on the forwarded destination will not be an Intercom call.
- **Call Forward-No-Reply:** If the called extension has set Call Forward-No-Reply, ANANT UCS will not forward the intercom call to the forwarded destination number on the expiry of the No-Reply Timer.
- **Call Forward-Busy:** If the called extension has set Call Forward-Busy, ANANT UCS will place the call on the forwarded destination number. However, this call (placed on the forwarded destination) will not be an Intercom call.

- **User Absent/Present:** ANANT UCS will place the Intercom call on the called extension only if the status of the called extension is ‘Present’.

- **Auto Call Back:** When the Intercom call is generated and the called extension is busy, the calling extension can set Auto Call Back on the called extension. When the called extension is free, ANANT UCS will serve the Auto Call Back request set by the calling extension. The ACB call placed on the called extension will be a normal call.

- **DISA:** An Intercom call can also be generated from DISA mode.

- **Priority**: The calling extension must have a higher Priority level than the called extension.

When the Intercom call is generated on SIP Extension having multiple call appearance and already a call is present on the SIP Extension then the ANANT UCS will place the Intercom call as normal call on the SIP Extension.

How to configure

To provide this feature to extension users, you must enable this feature in their Class of Service. For instructions, see “Class of Service (CoS)” and “Station Basic Feature Template”.

How to use

For Extended IP Phone Users

To use intercom to call an extension:

- Press the DSS Key assigned to Intercom.
  OR
  Dial *5
- Dial the desired extension number.
- The called extension’s speaker goes OFF-Hook.
- You are in speech with the called extension.
- You may talk.

Internal Call Restriction

Using this feature of ANANT UCS, the operator will be able to allow/restrict internal calls. The operator has flexibility to allow calls during day time and restrict calls during night time.

The operator can allow/restrict internal calls by assigning 'Guest Group' to the desired extensions.

How it works

Assigning 'Guest Group' to the extensions

In an enterprise, there is a special room assigned to visitors. In general, most of the visitors need to communicate only with the receptionist or the concerned person. S/he is not required to dial the numbers of the other employees in the organization. For example, a visitor in the office needs to communicate with the receptionist and not with the employees in the marketing department.

Sometimes, it may also happen that a group of employees may occupy different floors in an organization. In such cases members of the group need to communicate only among themselves and not with members of other department. For Example, employees in Marketing department need to communicate with the marketing employees and not with the employees in the RnD department.

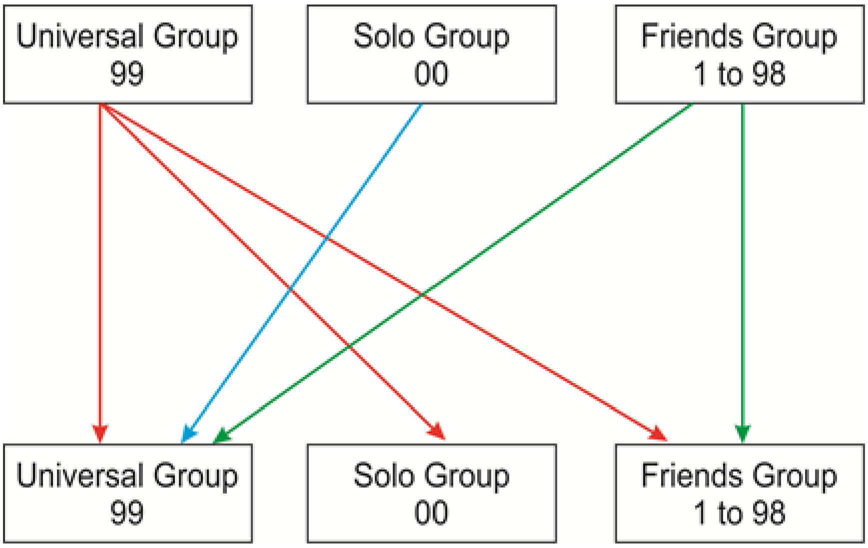
Essentially we need following logic:

- Certain extensions should be able to dial any other department or any visitors (for example Receptionist).
- Certain extensions should be able to dial only service departments (for example, visitors calling the receptionist).
- A group of extension users need to dial amongst each other and the service departments (for example: marketing employees communicating among themselves and the receptionist).

Above requirements are implemented as follows:

- Universal Group (For example: Receptionist and other services staff).This is classified in group 99.
- Solo Group (for example, visitors). This is classified in group 00.
- Friend's Group (for example: Group of employees in marketing department). Such a group of extension users can be assigned to any group number from 01 to 98.

Following diagram depicts this logic:

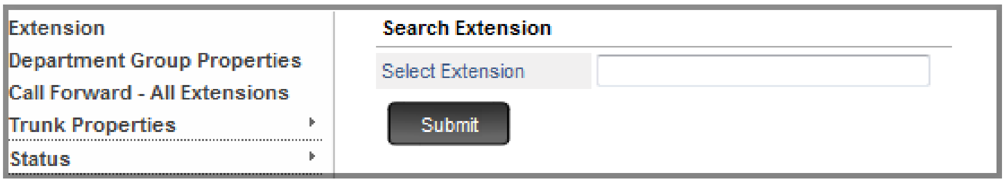


How to configure

To be able to use this feature, you must assign the guest group to the desired extensions for the SA Mode.

Assigning Guest Group

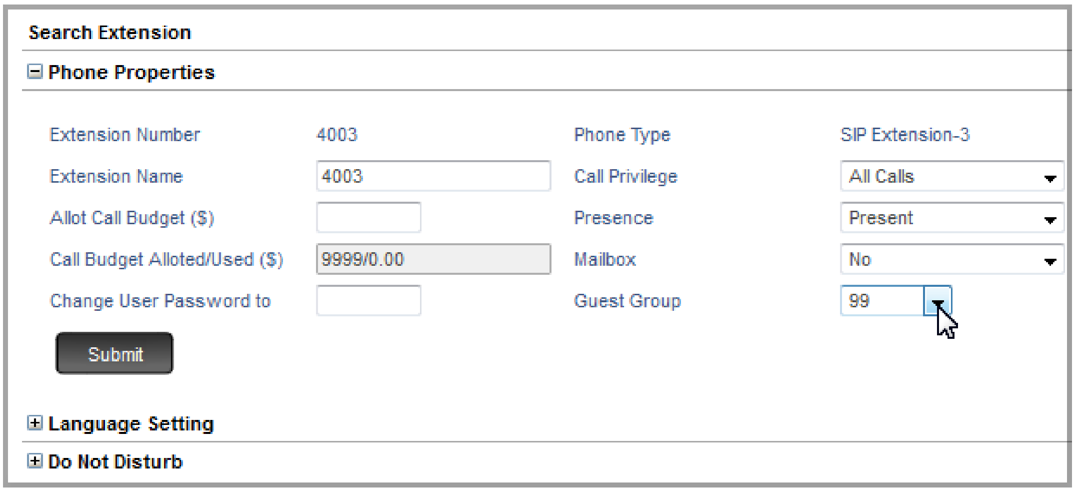
- Login as System Administrator.
- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit**.

  The searched extension users details appear on your screen.

- Click **Phone Properties** to expand



- Assign the desired group number to the extension from the **Guest Group.**

- Click **Submit.**

Interrupt Request (IR)

Interrupt Request allows you to break into an on-going conversation after intimating the extension user about the interruption.

In case of an important or urgent trunk call the operator can put the call on hold, interrupt the busy extension user to inform about the urgent call and then transfer the urgent call.

How it works

- A, B and C are extension users.
- A and B are talking to each other.
- C calls A.
- C gets busy tone.
- C dials Interrupt Request feature code.
- C gets Ring Back tone (RBT) and A gets beeps indicating a new call.
- To answer C's call, A must dial **Flash/Transfer** before the expiry of the Interrupt Request Timer. A will be in speech with C. B will be put on hold and will get music on hold.
- If A does not dial Flash/Transfer before expiry of the Interrupt Request Timer, C's call will be disconnected.

Feature Interactions

- **Call States:**
    - Interrupt Request works only if the dialed extension is busy. The dialed extension may be busy with another extension or trunk (external number).
    - Interrupt Request works only if the user about to be interrupted in is in a two-way normal speech with another user or external party. However, it will not work if the conversation is being recorded.
    - It will not work if the busy signal is due to the user being Off-hook, or in the middle of dialing, or accessing a feature of the ANANT UCS.
- **"Call Toggle":** Once A and C comes in speech with each other, A can toggle between B and C using Call Toggle feature.
- **Privacy against Interrupt Request:** If the feature 'Privacy against Interrupt Request is enabled for an extension, it cannot be interrupted. See "Privacy".
- **"Priority":** No Interaction with Interrupt Request. If 'A' has lower priority than 'B' but has Interrupt Request enabled; A can interrupt B.
- **"Call Taping"**: Interrupt Request will not work when the two-way conversation between the users is being taped.

How to configure

To be able to use Interrupt Request, extension users must have this feature enabled in their “Class of Service (CoS)” in their “Station Basic Feature Template”.

For instructions on configuring the SIP extension, refer “Configuring SIP Extensions”.

If required you may also change the default value of the *Interrupt Request Timer.* For instructions see “System Timers and Counts”.

How to use

For Extended IP Phone Users

When dialed extension is busy,

- Press DSS Key assigned to Interrupt Request
  OR
- Dial **3** on Busy Tone

Last Caller Recall

ANANT UCS offers a facility—Last Caller Recall— to trace the extension that last made the call to your extension.

How it works

- A's extension number is 2001.
- A wants to know who made the last call to extension 2001.
- A dials Last Caller Recall feature access code (Default:1092).
- The extension that last called 2001 rings.
- When the called extension answers, speech is established between A and the called extension user.

On SIP extensions, ANANT UCS supports Last Caller Recall using text (lcr) as access code.For a list of IP Phones on which this feature has been tested, see *"ANANT UCS Features tested on IP Phones of different Brands" in the Appendix.*

How to use

For Extended IP Phone Users

- Press DSS Key assigned to Last Caller Recall.
  OR
- Go OFF-Hook
- Dial **1092**

  The system dials out the extension number that last called your extension.

Last Number Redial

The system redials the last number string (external/internal) dialed from an extension. By default the system stores only the external numbers in the Last Number Redial List. If you want the system to store internal calls in this list, make sure you enable the **Store Internal Calls in Redial Call Log** check box in the “System Parameters”.

How it works

- Extension A dials the feature access code for ‘Redial’.
- The last 16 external numbers dialed by Extension A are displayed on the phone’s LCD.
- Extension A may select the number to be dialed out. The system will dial out this number using the same trunk access code used for dialing this number.

If Extension A has ‘Dynamic Lock’ set and uses Redial feature, the system will check for Toll Control as per the Lock Level set for Extension A before dialing out the number.

How to configure

No particular configuration is required for this feature to work. Redial is included in the Basic Features allowed to all extensions by default in their “Class of Service (CoS)”. So, all extensions can use the Redial feature.

How to use

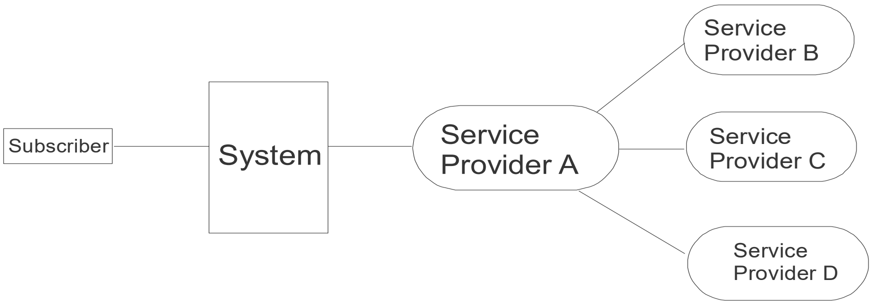
For Extended IP Phone Users

- Press Redial Key.
  Or
- Dial **7.**
  A List of last 16 external numbers dialed will appear on your phone’s display.
- Scroll to select the desired number.
- Press Enter key.
- The system dials out the external number.

Least Cost Routing-Carrier Pre-Selection

This type of Least Cost Routing is used in countries where the same service provider offers local call and long distance calling services. These service providers allow subscribers to select the service provider or Carrier for long distance calling.

For example,



The subscriber of Service Provider A must grab trunk lines of Service Provider A to call other subscribers in the local area.

However, when the subscriber of Service Provider A wants to make a long distance call, the subscriber must dial a prefix to select the a carrier (trunk) of the desired long distance, Service Provider B, C and D. Thus, the subscriber accesses a secondary service provider by dialing a short code or prefix for long distance calling.

This feature works on the basis of "Automatic Number Translation". Using Automatic Number Translation, ANANT UCS adds the code of the appropriate secondary Service Provider to the number string dialed by the extension user to route the call to the desired secondary Service Provider.

How to configure

To use this feature, you must do the following:

- Select an Automatic Number Translation Table Number from 1 to 8.
- Configure the Automatic Number Translation Table. In the Automatic Number Translation Table, in the Dialed Number String column, enter the long distance numbers that extension users will dial. In the Add Prefix column, enter the digits which are to be added as prefix to the Dialed Number string by the system before dialing it out and in the Strip Digits column enter the number of digit(s) to be stripped off by the system from the Dialed Number string before dialing it out. For example, the code '961' for Service Provider B must be prefixed to the number '2630555' dialed by extension users, you must enter '2630555' in the Dialed Number String column, '961' in the Add Prefix column and '0' in the Strip Digits column.
- Enable "Automatic Number Translation" feature on the SIP trunk of your Primary Service Provider in the "Outgoing Trunk Bundle" and apply the ANT Table number you configured to this OG Trunk Bundle.
- Create "OG Trunk Bundle Group" that includes the OG Trunk Bundle you created with Automatic Number Translation as member.
- Assign the OG Trunk Bundle Group (containing the OG Trunk Bundle with ANT) to the extensions for the Time Zones, that is, Working Hours, Non-Working Hours, Break Hours, in the "Station Basic Feature Template" of the extensions.

License Management

This section includes step-by-step instructions for activating different licenses supported in ANANT UCS.

To know about the different types of licenses in ANANT, refer to *"Licenses Supported in ANANT UCS"*.

If you have purchased a License Key with support of Hospitality features supported, then make sure that you have upgraded the system with Firmware Version V2.1 or later.

Instructions for Matrix Channel Partners

Your license voucher may be a paper or a PDF (protected) file. You may activate your License Online. For this, keep the following items ready:

- The License Voucher containing the 16-digit PIN.
- A valid, unique User ID and Password from the Matrix License Support Centre.
- Access to Internet.
- Current License Key of the system.

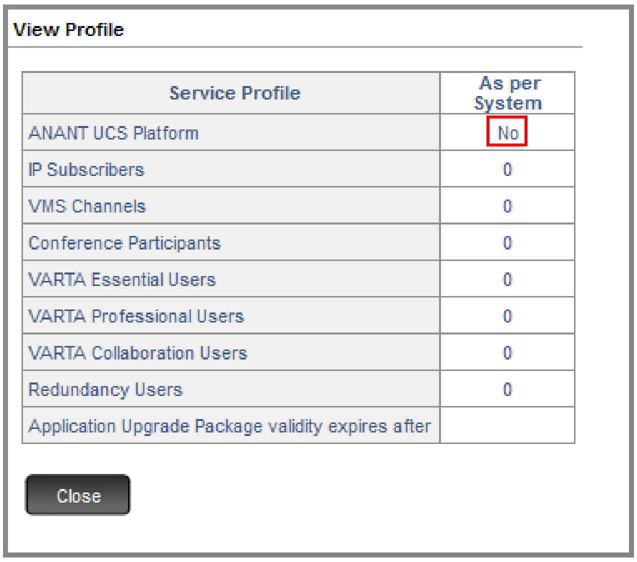
To activate the License Key *online*,

- Login as System Engineer.
- Under **Configuration**, click **License Management**.

  The License Management page opens.

- Note down or copy the Current License Key on this page.

- If you wish to view the current Service Profile, click **View Profile.**



View Profile

Service Profile	As per System
ANANT UCS Platform	No
IP Subscribers	0
VMS Channels	0
Conference Participants	0
VARTA Essential Users	0
VARTA Professional Users	0
VARTA Collaboration Users	0
Redundancy Users	0
Application Upgrade Package validity expires after	

Close

A new window opens which displays the features and functionalities that are currently available to you.

- Click **Close.**

- Keep your Current License Key and the License Voucher (paper or PDF) ready.

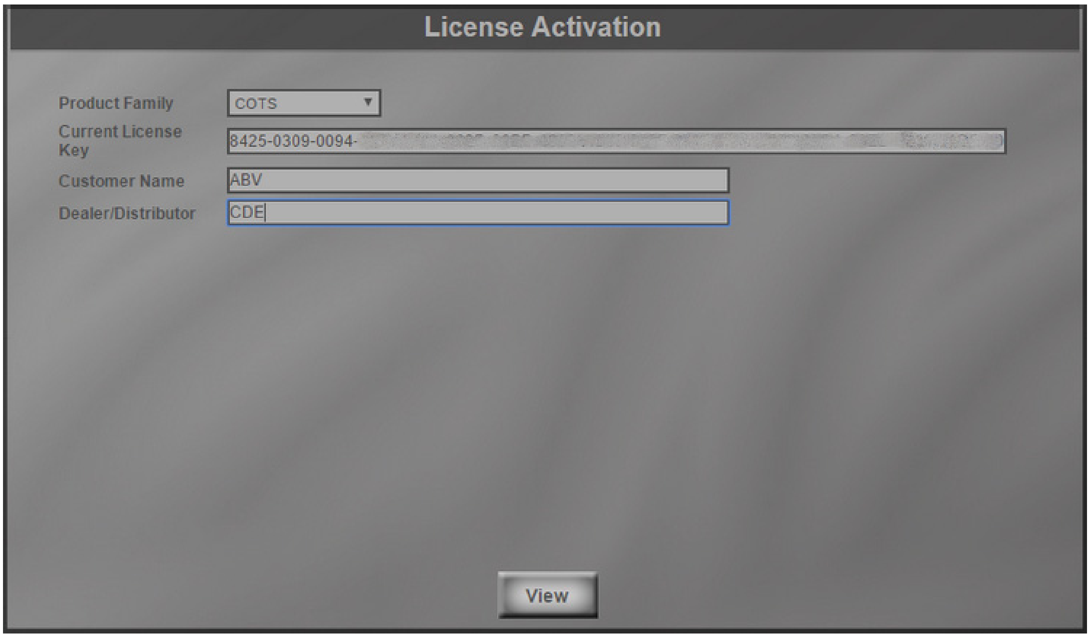
- Open a new window on your browser.

  Enter **http://www.matrixcomsec.com/support/license-activation/** in the address bar.

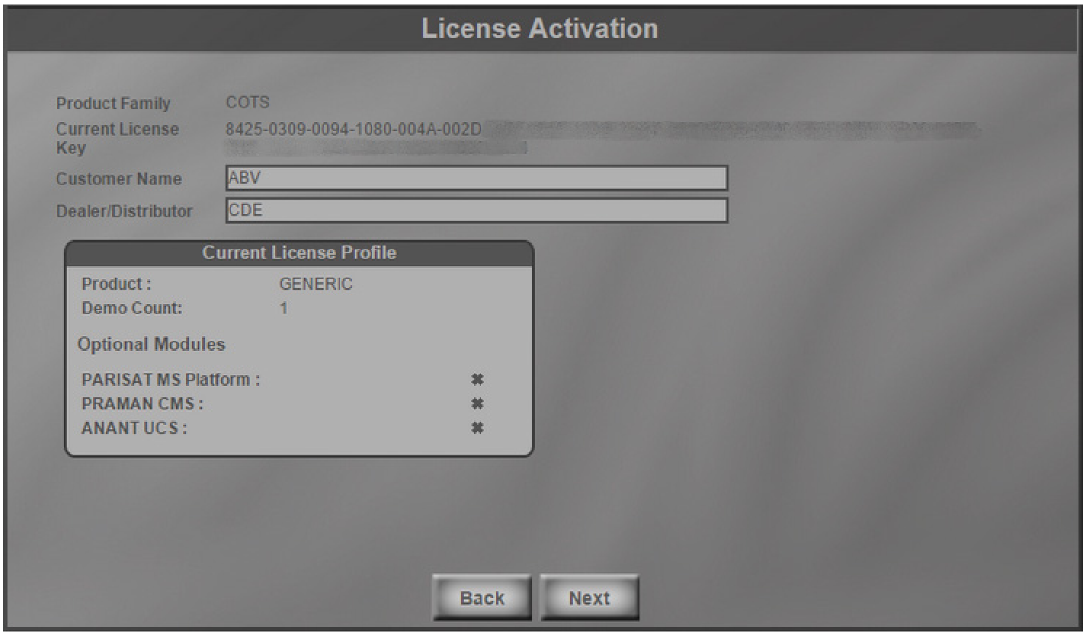
  The Login to Access page will open.

- Enter your **User Name** and **Password** provided by Matrix and click the **Login** button.

On successful login, the **License Activation** page will open.

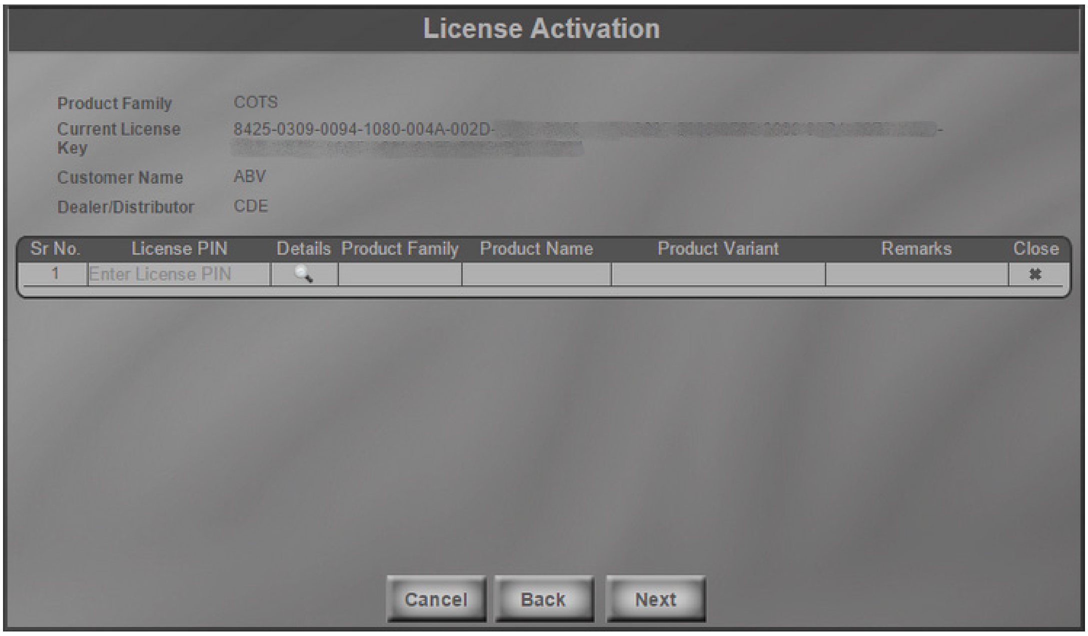


- As **Product Family**, select **COTS**.
- In the **Current License Key** field, type the Current License Key you noted or paste the key you copied from the **License Management** page of Jeeves.
- Click **View**.

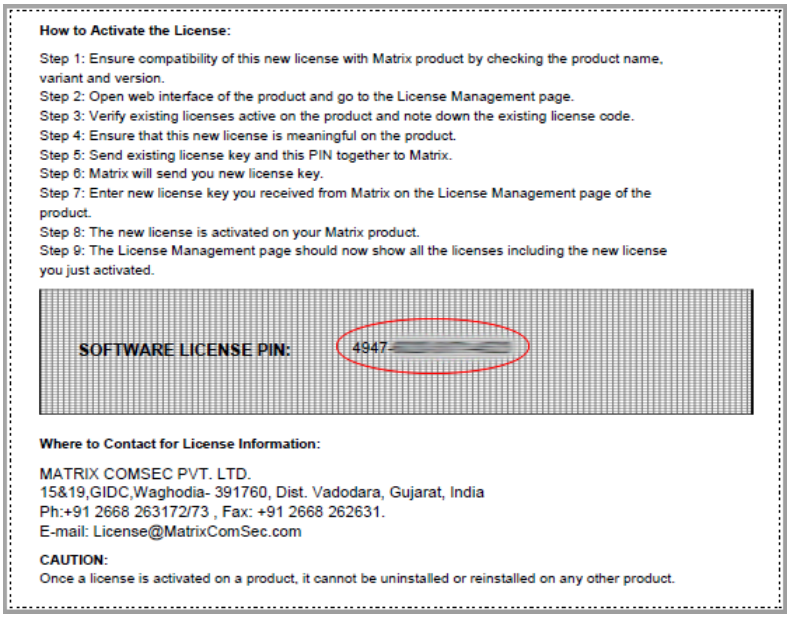


- The page displays the **Current License Profile**. Click **Next** to continue.

The **License Activation** page opens.



In the **License PIN** field on this page, enter the 16-digit License PIN from the Voucher.



How to Activate the License:

Step 1: Ensure compatibility of this new license with Matrix product by checking the product name, variant and version.
Step 2: Open web interface of the product and go to the License Management page.
Step 3: Verify existing licenses active on the product and note down the existing license code.
Step 4: Ensure that this new license is meaningful on the product.
Step 5: Send existing license key and this PIN together to Matrix.
Step 6: Matrix will send you new license key.
Step 7: Enter new license key you received from Matrix on the License Management page of the product.
Step 8: The new license is activated on your Matrix product.
Step 9: The License Management page should now show all the licenses including the new license you just activated.

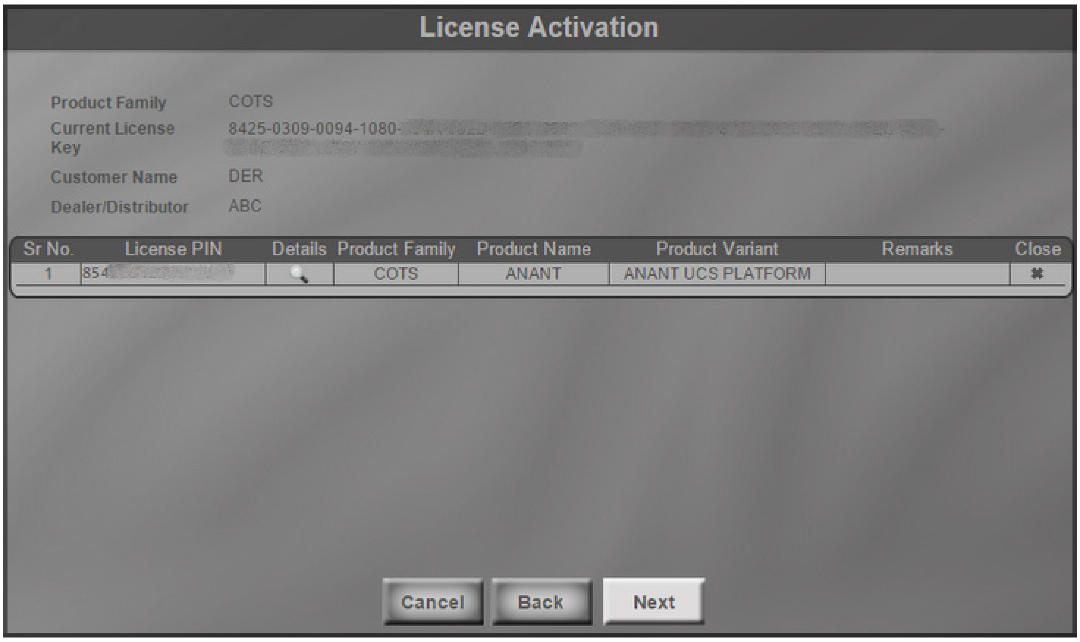
SOFTWARE LICENSE PIN: 4947-

Where to Contact for License Information:

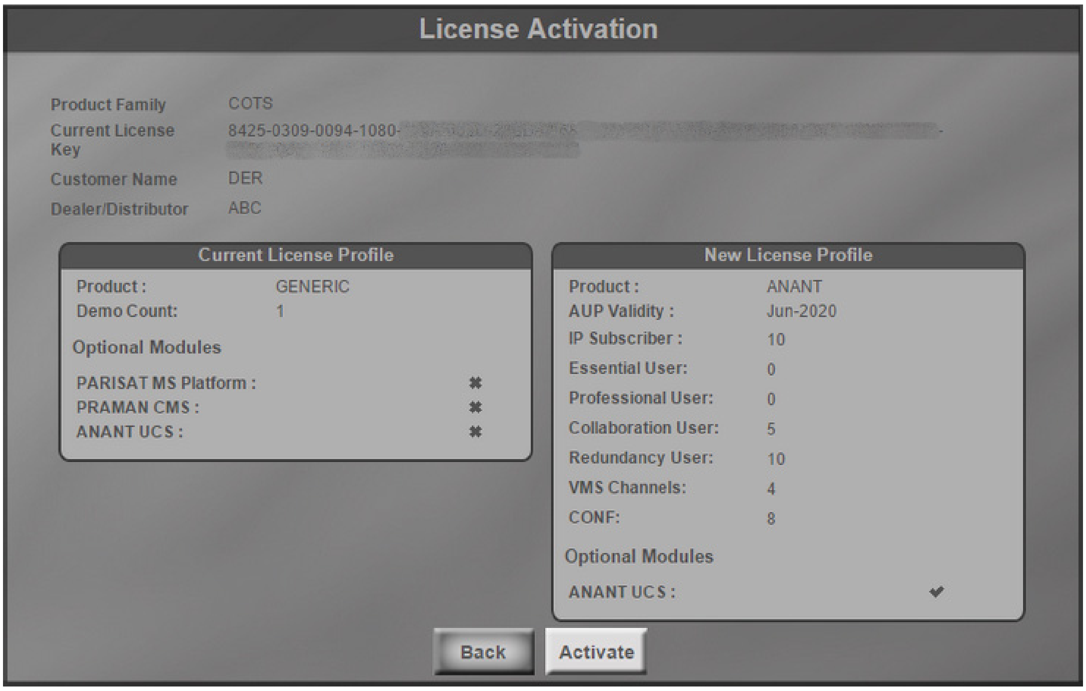
MATRIX COMSEC PVT. LTD.
15&19,GIDC,Waghodia- 391760, Dist. Vadodara, Gujarat, India
Ph:+91 2668 263172/73 , Fax: +91 2668 262631.
E-mail: License@MatrixComSec.com

CAUTION:
Once a license is activated on a product, it cannot be uninstalled or reinstalled on any other product.

- Click **Details.** The details appear in the fields **Product Family, Product Name**, **Product Variant**.



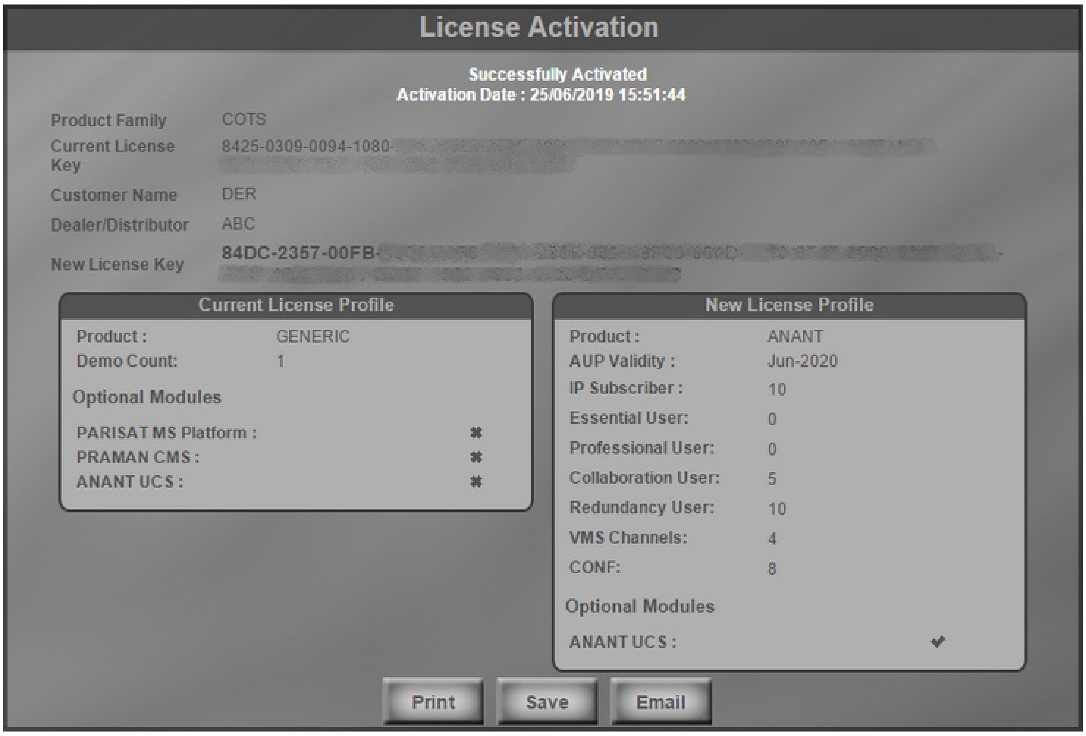
- Click the **Next** button. Your **Current License Profile** and your **New License Profile** will appear on this page.



- Click the **Activate** button and wait for a few seconds, as the activation is initiated.

  On successful activation, the confirmation message will appear on your screen along with the activation date and time.

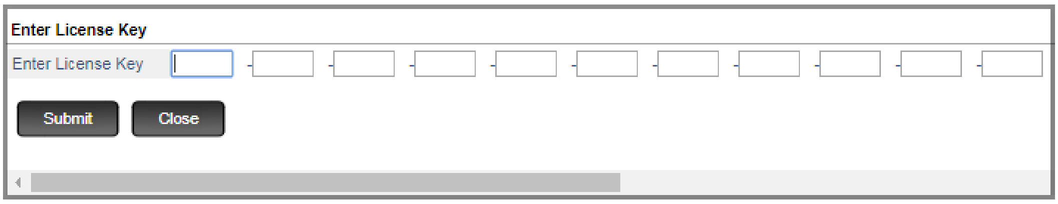
A confirmation mail will also be sent to your e-mail ID (registered with Matrix).



You may **Save**, **Print**, or **Email** this information for your records, by clicking the relevant button.

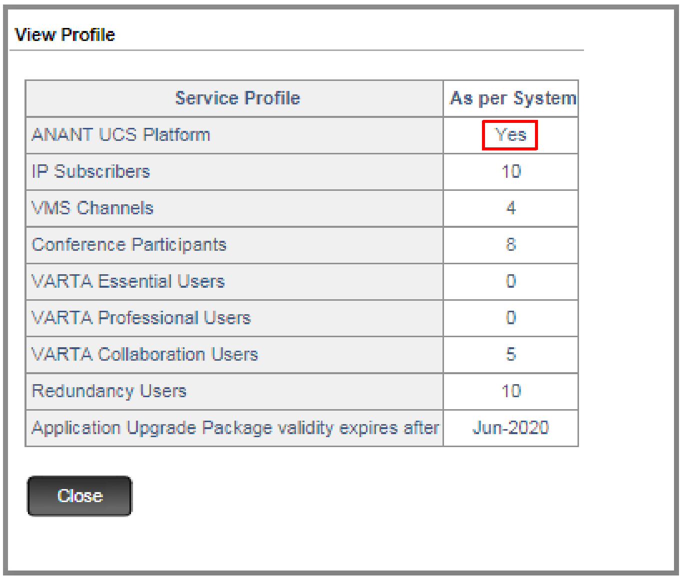
- Note down or copy the New License Key generated on this page.
- Go back to the Jeeves window (or login as System Engineer again, if your session has ended).
- Under **Configuration**, click **License Management**.

- Click **Enter License Key**. A new window opens.



- In **Enter License Key**, paste or enter the New License Key generated.
- Click **Submit.**

- On successful activation, a confirmation message will appear on your screen. Click **OK**.
- On the License Management page, click **View Profile.**



View Profile

Service Profile	As per System
ANANT UCS Platform	Yes
IP Subscribers	10
VMS Channels	4
Conference Participants	8
VARTA Essential Users	0
VARTA Professional Users	0
VARTA Collaboration Users	5
Redundancy Users	10
Application Upgrade Package validity expires after	Jun-2020

Close

A new window opens which displays the updated service profile.

- Click **Close.**

If you are unable to use Online Activation of the License Key or have no internet access, contact the Matrix License Support Centre for assistance in generating the new License key.

Instructions for Customers

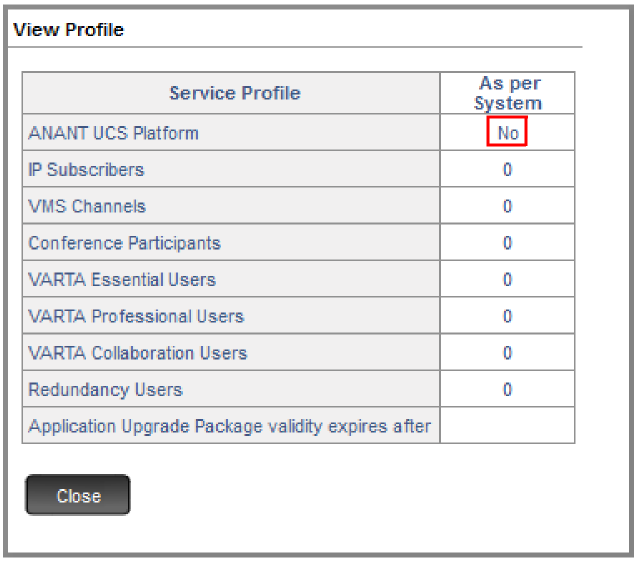
To activate your License, you would need the License Voucher containing the 16-digit License PIN. Contact your Dealer/Distributor in this regard. Your License Voucher may be a paper or a protected PDF file.

- Open Jeeves.
- Login as System Engineer.
- Under **Configuration**, click **License Management**.

The License Management page opens.

- Note down the current **License Key** on this page.

- If you wish to view the current Service Profile, click **View Profile.**

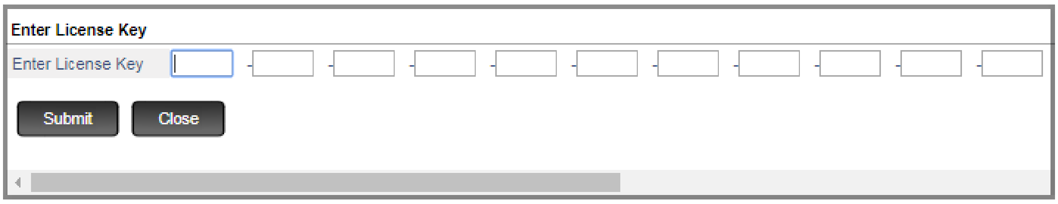


Service Profile	As per System
ANANT UCS Platform	No
IP Subscribers	0
VMS Channels	0
Conference Participants	0
VARTA Essential Users	0
VARTA Professional Users	0
VARTA Collaboration Users	0
Redundancy Users	0
Application Upgrade Package validity expires after	

A new window opens which displays the features and functionalities that are currently available to you.

- Click **Close.**
- Send your Current License Key and the License PIN (on the Voucher) to the Matrix License Support Centre.
- You will receive a new License Key.
- Go back to the Jeeves (or login as System Engineer, again, if your session has ended).
- Login as System Engineer.
- Under **Configuration**, click **License Management**.

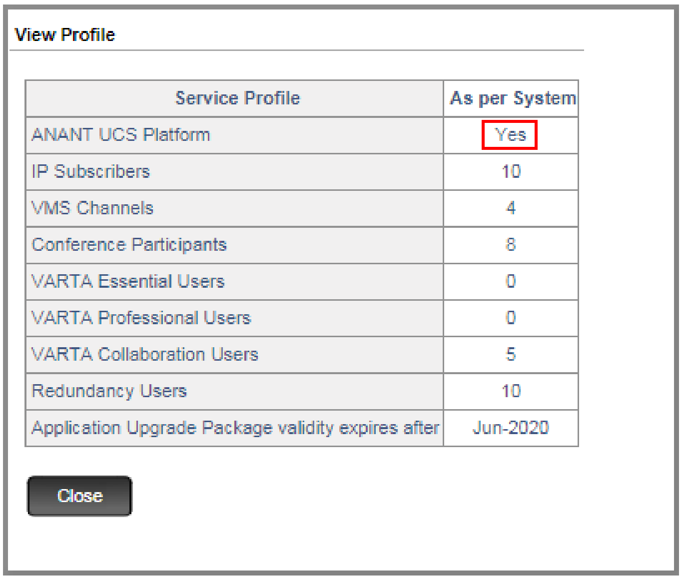
- Click **Enter License Key**. A new window opens.



- In **Enter License Key**, enter the New License Key you obtained from Matrix.
- Click **Submit**

On successful activation, a confirmation message will appear on your screen. Click **OK**.

- On the **License Management** page, click **View Profile.**



View Profile

Service Profile	As per System
ANANT UCS Platform	Yes
IP Subscribers	10
VMS Channels	4
Conference Participants	8
VARTA Essential Users	0
VARTA Professional Users	0
VARTA Collaboration Users	5
Redundancy Users	10
Application Upgrade Package validity expires after	Jun-2020

Close

A new window opens which displays the updated service profile.

- Click **Close.**

Similarly, you can activate all the other licenses available for ANANT UCS.

The current License Key and Service Profile will remain unchanged when the system is set to default or the firmware is upgraded.

Lightweight Directory Access Protocol (LDAP)

LDAP (Lightweight Directory Access Protocol) is an application protocol used by the system for accessing and maintaining information services for the distributed directory over an IP network.

The advantage for LDAP is that you can access the central LDAP directory of the organization using your system. Therefore, you do not have to maintain the local directory. The LDAP serveLIGr differentiates the data in its entries based on specific attributes. ANANT UCS functions as the LDAP client and once it synchronizes with the LDAP server, it saves the fetched LDAP entries to the Global directory. You can search and dial out the desired contact from the Global directory.

This feature is beneficial for organizations where multiple servers are deployed at various locations and all these servers need to maintain a centralized directory to facilitate ease of inter-department communication.

How it works

Pre-requisites

To be able to use this feature,

- Make sure that the third-party LDAP server supports LDAP Protocol Version 3.

  The following LDAP servers have been tested successfully:

  - Microsoft Active Directory
  - Open LDAP Directory Server

- The required details of the contacts are stored in the LDAP server directory.

- Make sure the LDAP parameters have been properly configured. For details, refer to “Configuring LDAP”.

- Manually synchronize your system with the LDAP server.

Once you have fulfilled the above pre-requisites, the sequence of the events that occur are as follows,

- When you manually synchronize, the system sends request to the LDAP server.

- The LDAP server asks for Authentication ID and Password depending upon the authentication settings in the server. Consult your SE for more details.

- After successful authentication, the LDAP server sends response to the system with the required contacts and their details as per the search criteria configured.

- The fetched results are stored in the Global Directory. The total number of search results stored depends upon the number of Global Directories synchronized with the LDAP server.

- Now, any extension user can easily dial out the desired contact from the Global Directory. To know more about making an outgoing call from the Global Directory, refer to “Global Abbreviated Dialing”.

- *Once the required contacts are synchronized with the LDAP server and loaded in the Global directory, these cannot be edited or deleted unless the LDAP option is disabled.*
- *Extension users can edit a number before making an outgoing call, but the changes will not be applied to the details stored in the LDAP directory and in the Global Directory which is synchronized with LDAP.*

LDAP Attributes

LDAP database consists of various details — first name, last name, common name, multiple phone numbers etc. To distinguish it, each type of detail is assigned a specific attribute in LDAP database. Contact your LDAP server Administrator to know the detailed list of attributes maintained.

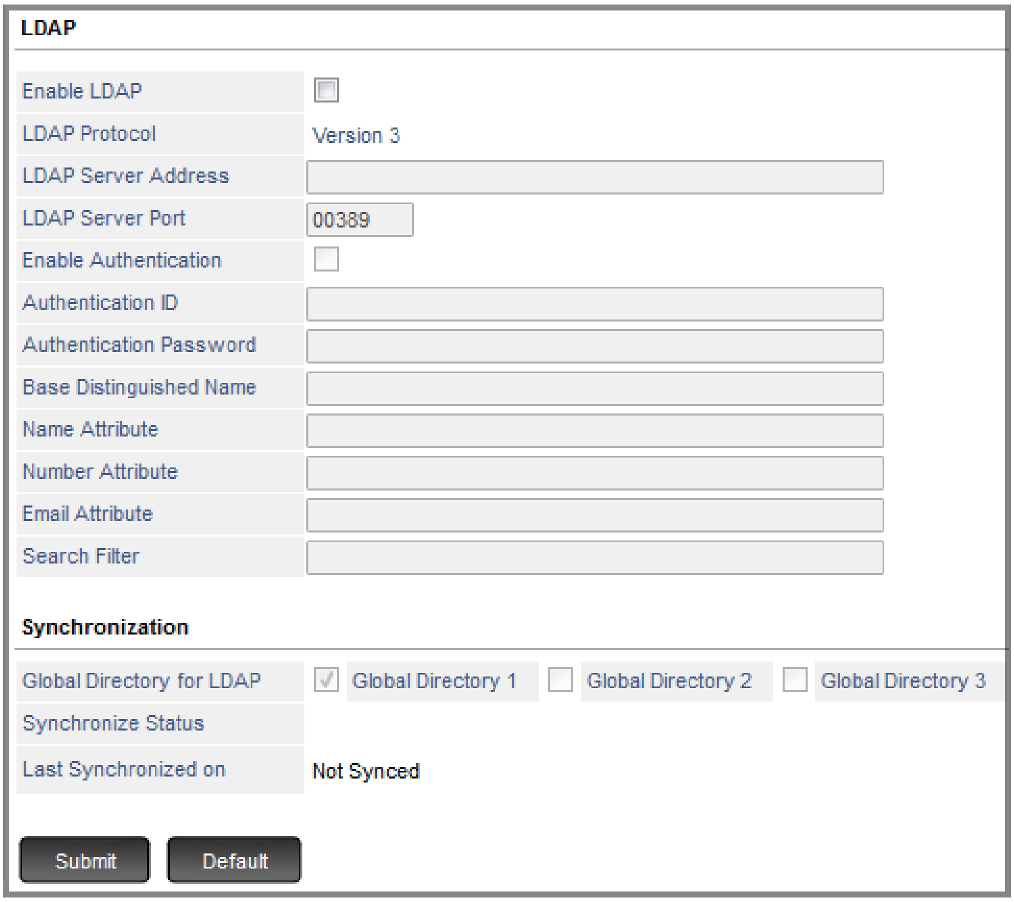
A few examples are listed below.

Abbreviation	Name	Description
gn	givenName	First name
cn	commonName	Full name
sn	surname	Last name or family name
dn	distinguishedName	Unique identifier for each entry
dc	dc	Domain component
-	company	Company or organization name
-	telephoneNumber	Office phone number.
mobile	mobilephoneNumber	Mobile or cellular phone number
mail	emailid	Email address

How to configure

Configuring LDAP

- Login as System Engineer.
- Under **Configuration**, click **LDAP**.



LDAP

Make sure you Contact your LDAP Server Directory Administrator for the LDAP Server configuration details.

- **Enable LDAP:** Enable this check box if you wish to synchronize contacts from the LDAP server. When this check box is enabled, ANANT UCS behaves as the LDAP client. By default, it is disabled.
- **LDAP Protocol:** Displays the LDAP Protocol Version supported. By default, it is Version 3.
- **LDAP Server Address:** Configure the IP address or Domain name of the third-party LDAP server here.
- **LDAP Server Port:** Configure listening port of LDAP Server here. By default, the value is 00389.
- **Enable Authentication:** This parameter enables you to set permission for validating the credentials (Authentication ID and password) of the system by the LDAP server. By default, it is disabled.

- **Authentication ID:** Enter the ID for authentication. The Authentication ID entered here should match with the Authentication ID configured in the LDAP server. You cannot keep this field blank. The ID may be a string of maximum 40 characters. Default: Blank.

- **Authentication Password:** Enter the Password for authentication. The Authentication Password entered here should match with the Authentication Password configured in the LDAP server. You cannot keep this field blank. The Password may be a string of maximum 24 characters. Default: Blank.

*Authentication ID and Password can only be configured when **Enable Authentication** check box is enabled. Contact your SE for more details.*

- **Base Distinguished Name:** Enter the specific location in the LDAP Server from where the search should begin. You can also specify the subtree as the base entry. You cannot keep this field blank. Default: Blank.

- **Name Attribute:** You can define the Name attribute to be fetched from the LDAP server directory here.You cannot keep this field blank. Default: Blank.

- **Number Attribute:** You can define the Number attribute to be fetched from the LDAP server directory here. You can define only one attribute and hence the system will fetch only one number per contact from the search results. You cannot keep this field blank. Default: Blank.

- **Email Attribute:** You can define the Email Attribute to be fetched from the LDAP server directory here. This needs to be configured only if you need the Email-id of the required contacts.

- **Search Filter:** You can use the Search filter when specific entries from a particular Distinguished Name are required. Enter the details after contacting you LDAP Server Administrator.

Synchronization

The Synchronization with the LDAP server will function only if the parameters mentioned under LDAP have been configured correctly.

You can now proceed further with configuring the Synchronization parameters:

- **Global Directory for LDAP:** Select the Global directory in which the search results obtained from the LDAP server should be stored. You can select either one or combination of — Global Directory 1, Global Directory 2, Global Directory 3. By default, Global Directory 1 is selected.

- **Synchronize Status:** Displays the synchronization status (In Progress, Fail or Pass) of the system with LDAP server.

- **Last Synchronized on:** Displays the date & time when the system last synchronized with LDAP server.

- Click on **Sync Now** button to manually synchronize the system with the LDAP Server.

- **Test:** You can test the parameters configured in the system for LDAP synchronization. To do so,
  - click the **Test** button. A new window for Test is displayed.
  - Select the desired attribute and enter the Name, Number or Email-id.
  - Click the **Test** button.
  - If the connection is successfully established then search results fetched from the LDAP server will be displayed.

- ***Test** button will be displayed only when the mandatory fields in LDAP have been configured.*
- *When configuration backup is restored then the synchronized LDAP contacts in the Global directory will not be restored. Hence, you must manually synchronize the Global directory contacts with LDAP server again (if LDAP contacts are required).*

- Click **Submit.**

How to use

To make an outgoing call to an LDAP contact from the synchronized Global Directory, refer to “Global Abbreviated Dialing”.

Live Call Supervision

Using Live Call Supervision, any extension can know the last external number dialed by another extension, even when that extension is in speech with an external party.

This feature is useful for supervisors who want to know where their subordinates are calling.

How it works

- A is the supervisor of B.
- A wants to know where B is calling, A can use Live Call Supervision.
- When B dials an external number it is stored in the system's memory.
- When A requests Live Call Supervision for B's extension, the system retrieves the last external number dialed by B and presents it on the display of A's phone.
- If B has not dialed any external number, A will get error tone with the message 'No Calls to Supervise' displayed on the LCD.
- If the B has dialed internal as well as external numbers, the last external number dialed by B will be displayed on LCD of A's phone.

Live Call Supervision can be used also when the extension being supervised is in speech with an external party.

How to configure

To be able to use Live Call Supervision, extension users must have this feature enabled in their "Class of Service (CoS)" in their "Station Basic Feature Template" for the required time zones.

How to use

For Extended IP Phone Users

- Press DSS key assigned to Live Call Supervision.
  OR
- Dial **1098.**
- Enter the Extension number to be supervised.

Logical Partition

Logical Partitioning is used to restrict the flow of call traffic between Private Networks as well as between VoIP networks.

This feature may be used when you want to allow calling between certain networks / IP Addresses only.

Logical Partition is applied on the SIP Trunks as well as SIP Extensions.

How it works

Let us understand this with the help of the following example:

- You want to allow calling from all the IP addresses within networks
- You want to allow calling from a single IP address only.

To allow incoming and outgoing calls from all the IP addresses in the network,

- you need to disable the VoIP to VoIP calling.
- configure the particular IP address, 192.168.5.0 as the IP address.
- configure the Subnet as 255.255.255.0

To allow incoming and outgoing calls from a particular IP addresses in the network,

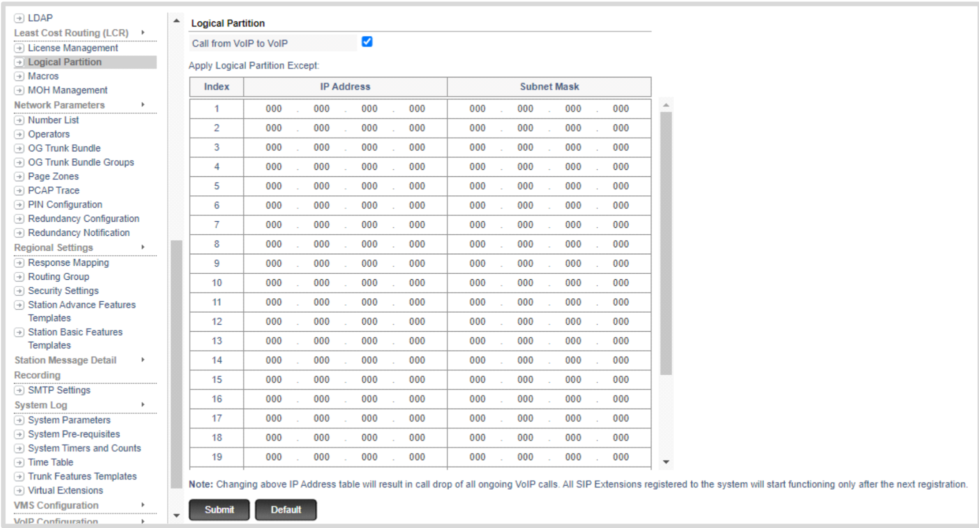
- you need to disable the VoIP to VoIP calling.
- configure the particular IP address, 192.168.5.0 as the IP address.
- configure the Subnet as 255.255.255.255

In this way the calling can be restricted as per your requirement.

Matrix recommends you to enable the logical partition in order to avoid any toll bypass wherever necessary. Matrix will not be responsible in case of any discrepancy related to this.

Configuring Logical Partition

- Login as System Engineer.
- Under **Configuration**, click **Logical Partitioning**.



- **Call from VoIP to VoIP:** By default, this check box is enabled. That is, there is no restriction for outgoing and incoming calls.

  Disable this check box if you wish to restrict calling. Make sure the desired IP addresses are configured in the table.

  If the check box is disabled and no IP address is configured in the table, then no calls will be made or received.

- **IP Address:** Referring the example above, if you want to allow calls between two networks, then configure the particular IP address, 192.168.5.0 here and in the **Subnet Mask** configure 255.255.255.0

  If you want to allow calls from a particular IP only, configure the particular IP address, 192.168.5.0 as the **IP Address** and configure the **Subnet Mask** as 255.255.255.255.

- Click **Submit.**

Macros

Extension users often have to dial access codes for specific functions like dialing a feature code, making an internal call, making an external call, etc.

ANANT UCS supports Macros, using which, you can abbreviate long number strings for regularly used functions in to macros and assign them to a DSS key on a Extended IP Phone extension.

How it works

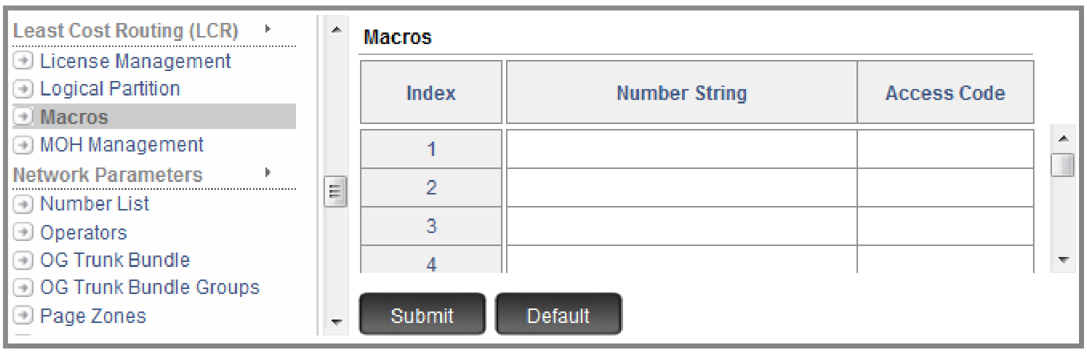
- Extension 2001, frequently sets *Call Forward-All Calls* to an external number 26550333.
- To do this, each time, Extension 2001 must dial ***131-Trunk Access Code-26550333-#****.
- Instead of having to dial this lengthy number string, a Macro can be created for *Call Forward-All Calls* to External number.
- If Extension 2001 is an Extended SIP Phone, the Macro can be assigned to a DSS key on the phone.
- Instead of dialing this number string, the user of Extension 2001 can simply press the DSS key on which this Macro is assigned.
- Thus when the DSS key on which a Macro is assigned is pressed, the corresponding access code is executed.
- The system sets call forward to the external number automatically.

How to configure

You can create as many as 50 Macros.

Creating Macros

- Login as System Engineer.
- Under **Configuration**, click **Macros**.



- Each macro is stored against an index number. By default the Macros Number String and Access Code are blank.

- In the **Number String**, enter the strings the system should consider as command when the DSS Key on the Extended IP Phone is pressed.

- Click **Submit**.

When ANANT UCS is operated in the Hotel Mode (see Customer Profile under "System Parameters"), Number Strings and Access Codes can be assigned for features such as Front Desk, Room Service, Voice Guided Alarm, Reservation Desk, Voice Mail System, Retrieve Message. To know more, see the topic Customer Profile in the ANANT UCS Hospitality System Manual.

Now, assign this Macro to a DSS Key on the Extended IP Phone.

- When you assign a Macro to a DSS Key on the Extended IP Phone,

- You must select **Macro** as **Function Type**

- As **Offset**, you must select the Macro Index number (1-50) at which the number string is stored. In this example, it would be Macro Index 1.

For instructions on assigning a macro to a DSS key of the Extended IP Phone, see "DSS Key Settings" in "Configuring Matrix SPARSH VP330", "DSS Key Settings" in "Configuring Matrix SPARSH VP248", "DSS Key Settings" in "Configuring Matrix SPARSH VP310" and "DSS Key Settings" in "Configuring Matrix SPARSH VP510" for instructions.

Meet Me Paging

While a Paging[107] announcement is being made, any extension user of ANANT UCS can get connected to the Paging extension, by dialing the Meet Me Paging feature code and the number of the Paging extension.

This feature is useful to Operators. Using this feature they can locate extension users who are away from their desks and get connected to them at their current location.

How it works

- A calls B's extension, but B is away from the desk.
- A uses Paging and makes an announcement asking B to call A's extension.
- To get connected to A's extension, B may use Meet Me Paging from any extension, while the announcement is being made.
- B dials Meet Me Paging code and A's number during the announcement.
- B gets connected to A.

- *Paging is an announcement made to a group of extensions within a Page zone. Extension users, (including those who are outside the Page zone) who want to use Meet Me Paging to answer the Paging call, will need to know the extension number they must call. Therefore, extension users who are paging are advised to announce their extension number.*

- *Meet Me Paging can be used only if the Paging call is active. Therefore, extension users who are paging must keep their call active, if they want their call to be answered using Meet Me Paging.*

How to configure

No configuration is required for Meet Me Paging. However, extension users who are using Paging to make their announcement, must have the feature "Paging" allowed in their Class of Service.

How to use

You can use Meet Me Paging to answer a Paging call from any extension, if you know the number of the Paging extension, and if the call is still active.

For Extended IP Phone Users

To answer a Paging call from any extension other than the Paged extensions,

- While the announcement is being made,
- Press the DSS key assigned to Meet Me Paging.
  OR
- Dial **1093.**
- Dial the number of the paging extension.
- You get connected to the Paging extension user.

107. This feature is not applicable for UC Clients — Matrix VARTA ADR100, VARTA AMP100 and VARTA WIN200.

To answer a Paging call from the same extension that is paged,

- While the announcement is being made,
- Go ON-Hook and then go OFF-Hook.
- Press the DSS key assigned to Meet Me Paging.
  OR
- Dial **1093.**
- Dial the number of the paging extension.
- You get connected to the Paging extension user.

Message Wait

The Message Wait feature of ANANT UCS enables extension users/Operator to set Message Wait on other extensions to deliver important messages.

If the extension user has a mailbox assigned, the Message Wait feature indicates to the extension user, the arrival of new messages in the user's mailbox.

Thus, Message Wait can be set by extension users as well as by the Voice Mail System.

You can set multiple Message Wait, but on different extension users. However, only one Message Wait can be set on one extension. A Maximum of 4 Message Wait can be set on an extension.

How it works

Message Wait set by Extensions/Operator

The Operator/any extension user can set Message Wait on another extension.

- The Operator calls Extension A.
- Extension A is not at the desk to attend the call.
- The Operator has an important message to communicate. So, the Operator sets Message Wait on Extension A, using the Message Wait key (if configured) or by dialing the feature access code.
- Extension B tries to reach Extension A, and sets Message Wait on Extension A, using the Message Wait key (if configured) or by dialing the feature access code.
- If Extension A is an Extended IP Phone and has DSS key assigned for Retrieve New Message, the LED of this key will glow to indicate new message wait.
- Now, Extension A can dial the feature access code to retrieve Message Wait, or press the Retrieve Message Wait Key, if assigned.
- The system will call the extension that first set Message Wait on Extension A. In this case, the Operator. If the Operator is busy, the busy message will be displayed on your LCD. The system will try to call the extensions that set Message Wait until the call is answered.
- If the Operator is not answering the call, disconnect and press the Message Wait key again, the system will place the call on Extension B. The system will try to call the extensions that set Message Wait until the call is answered.
- The extension that set Message Wait on A gets the CLI of A as Message Wait. A can now deliver the message.
- The LED of the Retrieve Message Wait key, if assigned, on Extension A will be turned off after all message wait set by other extensions on Extension A have been served.

Message Wait set by the Voice Mail System

The Voice Mail System (VMS) sets Message Wait on the extension, whenever a new message arrives in its personal Mailbox.

- Extension A is assigned a Personal Mailbox.
- There is a new message in A's Mailbox. The VMS indicates this to Extension A as per the option selected in *Message Wait Indication* set for Extension A.
- If Extension A is an Extended IP Phone, the Voice Mail key on the phone will also glow to indicate the arrival of a new message.
- If the Retrieve Message Wait key is assigned on the Extended IP Phone of Extension A, the LED of this key will also glow simultaneously to indicate arrival of the new voice mail.
- To listen to the new message, Extension A can
    - press the Voice Mail key
      or
    - the Retrieve New Message key (if assigned)
      or
    - dial the feature access code for Retrieve New Message.

  The VMS answers the call. After Extension A has listened to the new messages, the LED of the Voice Mail key is turned off.

  The LED of the Retrieve Message Wait key, if assigned, will also be turned off.

Message Wait Indications

The system gives Message Wait indication to extensions according to the option selected in the *Message Wait Indication* for that extension. The type of Message Wait Indication offered by ANANT UCS are:

- **Stuttered Dial Tone:** This Tone will be played as the Message Wait Notification to the extension user, when s/he goes OFF-Hook.
- **Silence:** The extension user will not hear any Message Wait Notification, when s/he goes OFF-Hook.

How to configure

To provide this feature to extensions, you must do the following configuration on the extensions:

- Enable the Message Wait feature in the "Class of Service (CoS)" of the "Station Basic Feature Template" of the extensions. This allows the extensions to set and cancel Message Wait on other extensions. Only those extensions that have this feature in their CoS can set or cancel Message Wait on other extensions. By default, this feature is enabled in the CoS of all extension types for all the time zones.
- Select the desired **Message Wait Indication** in the "Extension Voice Mail Settings" of SIP extensions and Department Groups.

- You may also assign DSS Keys to ‘Message Wait’ and ‘Retrieve Message Wait’ on an Extended IP Phone extensions.

  For instructions on assigning these features to DSS keys of the Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310”, “DSS Key Settings” in “Configuring Matrix SPARSH VP210”and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”.

How to use

For Extended IP Phones

To set Message Wait:

- Press DSS Key assigned to 'Message Wait'.
  OR
  Dial **1076**.
- Dial the extension number on which you want to set Message Wait.
- Select the option ‘Set Message Wait’
- Press Enter Key.

To cancel Message Wait:

- Press DSS Key assigned to 'Message Wait'.
  OR
- Dial **1076**.
- Dial the extension number on which you want to cancel Message Wait.
- Select the option ‘Cancel Message Wait’
- Press Enter Key.

To retrieve Message Wait:

- Press DSS Key assigned to 'Retrieve Message Wait' or the Voice mail Key when the LED glows.
  OR
- Dial **1077.**

Mobility Extension

ANANT UCS offers mobility to its extension users whose nature of work keeps them away from their desks frequently and for longer durations.

Using mobility extensions, the extension users of ANANT UCS can make and receive their calls from their current (remote) location, placing calls through the system, and can access the system just as any other normal extension of ANANT UCS.

ANANT UCS offers mobility using:

- VARTA UC Clients — VARTA ADR100 or VARTA AMP100. For details refer to the respective User Guide.
- DISA — For details, refer to the explanation given below.

How it works

ANANT UCS supports two types of users:

- **Extension Users:** They are extension users of ANANT UCS to whom a dedicated physical station is assigned on their desk.

- **Virtual Users:** They are extension users of ANANT UCS who share a physical station, or may not have any physical station allotted to them.

  The facility of Mobility Extension is provided to both virtual and normal extension users using the features "Direct Inward System Access (DISA)" and "Call Forward".

How to configure

To provide Mobility Extension to users, follow the steps described below.

- List out the Extension Users and Virtual Users and configure them first.

- Make sure that extensions which are to be provided Mobility Extensions have the features Call Forward and DISA enabled in their "Class of Service (CoS)"[108]. Class of Service is to be configured in the "Station Basic Feature Template" assigned to the Mobility Extensions.

- Make a list of External numbers to which the Mobility Station users will forward their calls. Configure these numbers in the 'Allowed List' of Local, Regional, National and International Numbers, as appropriate.

- The Toll Control assigned to the extension will be applied when a call is forwarded to an external number. Make sure that the extensions which are to be provided Mobility Extension have the required "Toll Control" level for Call Forward to the External Numbers (the numbers you have configured in the Allowed List). Toll Control level is to be configured in the "Station Basic Feature Template" assigned to the Mobility Extensions.

108. *It is possible to map the virtual user's flexible number (extension number) to a physical station. With this mapping, whenever a virtual user's number is dialed, the call is placed to the assigned physical extension. The physical extension can be assigned by defining it as the 'Landing Destination for Virtual Users' in the Station Advanced Feature Template assigned to the virtual user.*

- Configure the parameter ‘Allow External Call Forward for' in the “Station Advanced Feature Template” assigned to the Mobility Extensions. This parameter defines the types of call for which the External Call Forwarding is to be applied. Select any one of the options, Internal Calls Only, Trunk Calls Only, Internal + Trunk Calls as required.

- Configure the parameter 'DISA' in the “Trunk Feature Template” of the trunk lines which Mobility Extensions users are to be provided access to. Make sure you select the option 'CLI Auth.- Multiple Calls' in the 'DISA' parameter of the 'Trunk Feature Template'.

- Make a list of numbers which the Mobility Extension users will use to access the ANANT UCS from DISA mode. Configure this list of numbers in the "DISA - CLI Authentication Table".

  Configure this list of numbers in the 'Calling Party’s Number' field of the Authentication Table. Configure the 'Port Type' and 'Port Number' of the Extension assigned to Mobility Extension Users in the 'Auto Login As' for the respective 'Calling Party’s Number' field. Refer the topic “Direct Inward System Access (DISA)” to know more.

How to use

The Mobility Extension Users of ANANT UCS can use the features of ANANT UCS from a remote location as described below.

Receiving calls

To receive calls, the Mobility Extension User must set Call Forward on his extension with an external number (mobile number, landline number, etc.) as the destination number.

To make calls ring on the extension and the external number simultaneously, the Mobility Extension User must activate the Call Forward-Dual Ring feature on his extension.

The Mobility Extension User can also choose where he wants to receive the calls during a particular time of the day. For example, he can receive calls during a particular time of the day, that is, Time Zone on his external number and have his calls received by his Voice mail or the Operator or any other number during another Time Zone. To do this, he must set **“Call Forward-Scheduled"** on his extension. Dual Ring can also be set for Call Forward-Scheduled.

Making calls

The Mobility Extension User should make a call on the DISA enabled trunk of ANANT UCS from the external number and the system will provide the dial tone to the user after authenticating the external number with the help of the DISA-CLI Authentication table.

On getting the dial tone, the Mobility Extension User can make internal as well as external calls as per the “Toll Control” and “Class of Service (CoS)” assigned to his Extension.

The Mobility Extension User can also dial codes of the Personal directory and Global directory numbers to use the feature Abbreviated Dialing.

Accessing Features

The Mobility Extension User can access the system features by dialing specific codes after making calls on the DISA enabled trunk of ANANT UCS, or after answering the calls received on his external number.

These codes are listed below.

Activity	Code to be dialed
On-Hook	***#0***
Off-Hook	***#1***
Flash	***#2***
Pause	***#3***
Terminate the call	***#9***

Described below are instructions for Mobility Extension users on using different call management features.

Call Hold

To put a call on hold,

- When in speech with Party A, Mobility Extension User dials ***#2.***
- The call with Party A is put on Hold.
- Press Flash (***#2***) to retrieve the held call and talk to Party A again.

Call Transfer

To conduct a screened Call Transfer,

- When in speech with Party A, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User gets Feature Tone.
- Dial Trunk Access Code (TAC) followed by the number of Party B. [To make external call]
  OR
- Dial Internal station number [to make call on a station]
- When in speech, dial ***#0*** to go on hook.
- Party A and Party B will get connected.

To perform Call Transfer-While Ringing,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User will get Feature Tone.
- Dial TAC - number of Party B. [To make external call]
  OR
- Dial Internal station number [to make call on station]
- Dial ***#0*** on receiving Ring Back Tone (RBT).

To perform Call Transfer-On Busy Station,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User will get Feature Tone.
- Dial number of Station user.
- Dial ***#0*** on receiving Busy Tone.

To perform a Call Transfer-Trunk-to-Trunk,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The Call of Party A is put on Hold.
- Mobility Extension User will get Feature Tone.
- Dial Trunk Access Code to grab the Trunk
- After receiving the Dial Tone of the trunk, dial number of Party B.
- After Party B answers the call, dial ***#0.***
- Party A will get connected with Party B.

Making a Second Call

To make a second call by putting the current call on hold,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The call of Party A is put on hold.
- Mobility Extension will get Feature Tone.
- To make an external call: dial TAC - number of Party B.
  OR
- To make an internal call: dial extension number.

Call Toggle (Call Splitting)

To toggle between two calls,

- Mobility Extension User is in speech with Party A.
- During speech, Mobility Extension User dials ***#2.***
- The call of Party A is put on hold.
- Mobility Extension will get Feature Tone.
- Dial number of Party B.
- Make speech with Party B.
- Dial ***#2-1*** to put Party B in hold and speech with Party A.
- Again dial ***#2-1*** to put Party A in hold and speech with Party B.

Call Pick Up

To pick up the call of same Call Pick Up-Group,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After receiving dial tone, dial ***4.***
- The call ringing on the station of the same Call Pick up-Group will get connected with Mobility Extension User.

To pick up the selective call,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After receiving dial tone, dial ***12-Station Number.***
- The call ringing on the dialed station will get connected with Mobility Extension User.

3-Party Conference

To conduct a 3-Party Conference,

- Make the first call.
- Dial ***#2.***
- Make the second call.
- Dial ***#2-*3*** to establish conference.

Multiparty Conference

To create a Multiparty Conference,

- Mobility Extension User has generated three-party conference with B and C.
- Dial ***#2*** from Mobility Extension user.
- Dial number of Party D.
- After speech, dial ***#2-*3.***
- Mobility Extension User, B, C and D are in Multi-party Conference.
- Dial number of Party E.
- Dial ***#2*** from Mobility Extension user.
- After speech, dial ***#2-*3.***
- Mobility Extension User, B, C, D and E are in Multi-party Conference.
- Dial ***#2*** from Mobility Extension user.
- Dial number of Party F.
- After speech, dial ***#2-*3.***
- Mobility Extension User, B, C, D, E and F are in Multi-party Conference.
- All Parties will get connected with each other.

To terminate the Multiparty Conference,

- When in the middle of a Multiparty Conference,
- Dial ***#2***.
- Dial ***190*** after getting feature tone.

To temporarily leave the Multiparty Conference,

- When in the middle of a Multiparty Conference,
- Dial ***#2***.
- Dial ***191*** after getting feature tone.

To rejoin the Multiparty Conference,

- Dial **#1** to go Off Hook.
- Dial ***191***.

To permanently leave the Multiparty Conference,

- While in multi-party conference, dial **#0** to go Off Hook.

When Dialed Extension is busy

To make a call on an internal extension which is busy,

- During busy tone
- Dial ***3*** for interrupt request.
- Dial ***4*** to Barge-In
- Dial ***5*** to Raid.

Call Forward-Unconditional

To set Call Forward-Unconditionally,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***131-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-Unconditionally on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***131-TAC-external number.***
- The system will give confirmation tone.

Call Forward-Busy

To set Call Forward-Busy,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***132-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-Busy on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***132-TAC-external number.***
- The system will give confirmation tone.

Call Forward-No Reply

To set Call Forward-No Reply,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***133-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-No Reply on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After getting Dial Tone, dial ***133-TAC-external number.***
- The system will give confirmation tone.

Call Forward-Busy/No Reply

To set Call Forward-Busy/No Reply,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***134-Station/Department Group/VMS.***
- The system will give confirmation tone.

To set Call Forward-Busy/No Reply on external number,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***134-TAC-external number.***
- The system will give confirmation tone.

To cancel Call Forward,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***130.***
- The system will give confirmation tone.

Call Forward-Dual Ring

To set Call Forward-Dual Ring,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***136-1.***
- The system will give confirmation tone.

To cancel Call Forward-Dual Ring,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***136-1.***
- The system will give confirmation tone.

Call Forward-Scheduled

To set/cancel Call Forward-Scheduled,

- Make a call on DISA enabled Trunk from Mobility Extension User.
- After the Dial Tone, dial ***1175-Time Zone-Call Forward Type-Number.***
- The system will give confirmation tone.

- *Mobility Extension users can have Call Forward and Call Forward-Scheduled set on their extension by the Operator or by another extension user.*
- *Using "Call Forward-Remote" and by setting "Call Forward-Scheduled" from the SA mode, the extension user/Operator can set Call Forward and Call Forward-Scheduled for any Mobility extension user.*

Multi-Stage Dialing

The Multi-Stage Dialing feature of ANANT UCS is typically used in applications like Calling Card, where extension users are required to dial digits in stages when making a call using the calling card.

The Multi-Stage feature enables extension users to directly dial the number they want to call, and the system dials out the number at different stages of the call by suitably modifying the number.

How it works

A typical example of Multi-Stage Dialing is the use of prepaid Calling Cards. Here, the person using a calling card must dial a fixed number string before dialing the actual number. When using a calling card,

- Users must first dial the number of the Calling Card server, for example: 1602233 (7 digits).
- After the call is answered by the Calling Card server, users must dial the PIN provided by the calling card service provider, for example 1132121234.
- After dialing the PIN number, users can dial the number they want to call, for example 0014125126508.

Thus, when using a Calling Card, users must dial a very lengthy number string, each time they need to make a call using the Calling Card.

The use of Multi-Stage Dialing saves the time and effort of dialing out lengthy digits in stages.

The Multi-Stage Dialing makes use of the “Automatic Number Translation” table. This table must be configured on the trunk from which extension users will make calls using Calling Cards.

To take the above example further,

- Say, extension users are allowed to make international calls using Calling Card from the SIP Trunk1. “Automatic Number Translation” table must be configured on SIP Trunk1 in the OG Trunk Bundle.
- The Automatic Number Translation table consists of Dialed Number Strings, Strip Digit and Add Prefix Number strings.
- In Dialed Number, you must configure ‘00’, the prefix for international numbers.
- In Add Prefix you must configure the Calling Card server number and the PIN Number.
- Keep Strip Digits as 00.
- As the system must wait for the Calling Card server to answer before dialing the PIN, you must configure Wait for Answer between the Calling Card server number and the PIN number.

  You must insert a delay by configuring the Pause Timer after the PIN number and the destination number.

- The Automatic Number Translation table would look like this:

Index	Dialed Number String	Strip Digits	Add Prefix
1	00	00	1602233**W**1132121234**P**
:			
32			

- When the Automatic Number Translation table is configured, the Extension user can simply dial the Trunk Access code (0/5/61/62/63/64) and the destination number (0014125126508).

- The system matches the dialed number with the Dialed Number String of the ANT table, the number matches with the entry '00' stored in the table.

- The system dials the Add Prefix Number string 1602233 (number of the calling card server). It waits for the calling card server to answer the call.

- When the call is matured, i.e. the calling card server has answered the call, the system dials the PIN number 1132121234 and waits for the Pause Timer before dialing the destination.

Thus, the extension user directly dials the desired destination number and the system substitutes this number by adding the Calling Card server number and PIN number and dials these numbers in two stages.

How to configure

Configuring Multi-Stage Dialing

To be able to use Multi-Stage Dialing, you must configure the following:

- Automatic Number Translation table on the trunk you want to use this feature. For instructions on configuring ANT on SIP Trunk, see "Automatic Number Translation" and to assign the ANT Table number see "Outgoing Trunk Bundle".

- Configure the **DTMF Out Dial** on the SIP Trunk. Set the **DTMF ON Time** and the **DTMF Inter Digit Pause** and **Pause Timer** to the required values. For instructions on configuring DTMF Out Dial on SIP Trunk, see, "Configuring SIP Trunks".

- If required you may configure the **Call Proceeding Tone for Multi-stage Dialing** as **Network tone**, **Pseudo Tone**, or **Silent**. For instructions, see "System Parameters".

Music on Hold (MoH)

The music played to extension users and external callers who are put on hold is called Music on Hold (MoH).

How it works

ANANT UCS plays the default MoH to the hold party. No special configuration is required for using the default MoH. It is automatically played to the hold party.

You can also upload the customized MoH. The customized MoH may be a piece of music or voice message (recorded from the external source). The message may contain any promotional information about your company or services provided by your organization, etc.

MOH will not be played, if the RTP Mode is set as RTP Relay or Direct RTP and none of the selected MoH Vocoder Preference is negotiated. For details, refer to “Configuring VoIP Parameters”.

How to configure

You can record a piece of music or a message of maximum 10 minutes using an external source. For details on how to record a voice message, refer “Recording Voice Messages”.

You can upload this recorded message from **MOH Management**. The MOH file must be a .wav file.

Refer “Uploading/ Downloading MoH” for instructions on uploading the MOH file.

- *The default MOH file will be over-written with the new uploaded MOH file.*
- *If the option **Routing Group** is selected as the **Alarm Notification Type** for an extension, when the extension goes Off-hook to answer the alarm call, and the extensions in the Routing Group for Alarm Notification are busy, Music-on-Hold will also be played to the extension answering the alarm call.*
- *If your SIP Extension supports multiple call appearance and you want the system to play MOH to internal callers, when your extension is busy, enable the **Play MOH to Queued Internal Calls on SIP Extension** check box in “System Parameters”. As soon as your extension is free, Ring Back Tone will be played to the caller.*

MoH Management

ANANT supports the playing of a custom MoH instead of a default MoH.

No special configuration is required for using the default MoH. However, if you want to use the customized MoH, you must first:

- record the message
- upload the new recorded message file

- *Default MOH file cannot be downloaded or deleted.*
- *The default MOH file will be over-written with the new uploaded MOH file.*

Recording Voice Messages

When you record MoH of your choice, consider these important points:

- The MoH file is in WAV format, so the customized MoH must also be in the same format and can be of maximum 10 minutes.

- The MoH file has a unique name MoH.wav, make sure the audio file of the custom MoH you have recorded must have the same unique file name as the existing default audio file. The filename can be a maximum of 64 characters. The allowed characters for the filename are A to Z, a to z, 0 to 9, - and _.

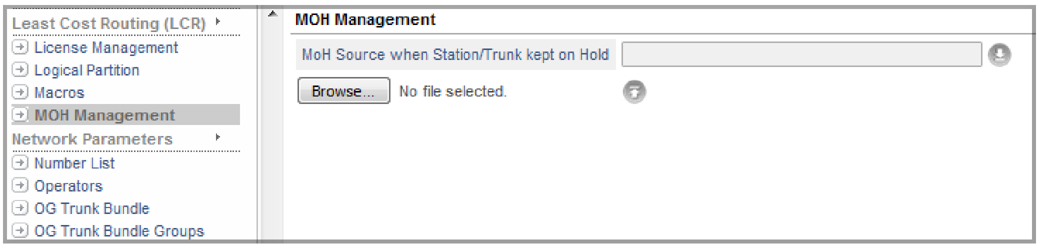
- You can record custom MoH from any external source and upload the audio file using Jeeves.

- When you record the MoH from any external source and upload it, make sure that the audio file is recorded in.*wav* file format, with the attributes listed below:
    - Audio Format: u-law
    - Channel: 1 (mono)
    - Sampling Frequency: 8 KHz
    - Audio Sample Size: 8 bit

Uploading/ Downloading MoH

- Login as System Engineer.

- Under **Configuration**, click **MOH Management.**



To upload the MoH file:

- Click **Browse** button to select the desired MoH file from the local disk on the Computer.

- The system displays the name of the MoH File. Click to upload the file.
- The system prompts you with the message: "*It will overwrite existing MOH file after successfull upload*"

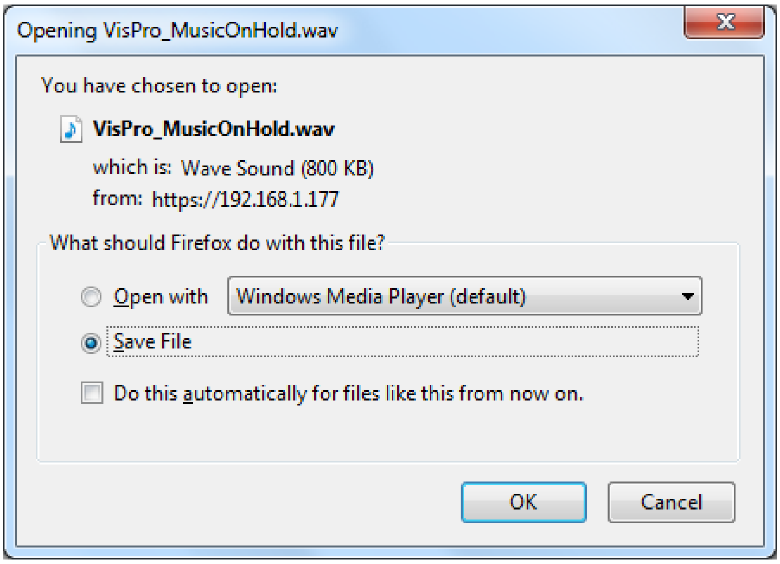
  Click **Ok** to upload or **Cancel** to discard uploading of the MoH file.
- The uploaded MoH file will be displayed in **MoH Source when Station/Trunk kept on Hold.**

*While the system is uploading the MoH file, you can click the **Abort** button to stop the process, if required.*

To download the MoH file:

- Click , to download the MoH file.

The **Opening Filename.wav file** window will open.



- You can either open the .wav file or save the file to a location.

- *The above window display depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
  OR
  If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.
- *If you are using Mozilla Firefox (version 3.5.1 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files.***

Mute

This feature helps the extension user to disconnect the speech transmission path in the middle of a conversation. The extension user can still listen to the opposite party because the receiving path remains connected. Mute is useful when you want to consult someone in the middle of a conversation, but do not want the opposite party to listen to your discussion. You can Mute a call before making a call or during speech.

How it works

- A is in speech with B.
- A wants to consult to C in the room, but does not want B to hear their conversation.
- A presses the Mute Key.
- The transmit speech path from A to B is disconnected. The receive path remains connected.
- So, A will be able to hear B, but B will not be able to hear the conversation between A and C.
- When A has finished consulting C, to resume speech with B, A presses the Mute key again.
- The transmit speech path from A to B is restored. A and B are in speech again.

How to use

For Extended IP Phone Users

To mute a call before making the call:

- Press the ‘Mute’ Key
  The LED of the key glows.
- Dial a number on confirmation tone.
  OR
- Dial **1052**
- Dial desired number.

To mute a call during speech:

- Press the ‘Mute’ Key to silence outgoing speech.
  OR
- Press Transfer Key.
- Dial **1052**

To resume outgoing speech:

- Press the ‘Mute’ Key.
  The LED of the key is turned off.
  OR
- Press Transfer Key.
- Dial **1052**

Number Lists

A 'Number List' is a group of number strings. ANANT UCS uses Number Lists to support different features as Call Duration Control, Call Taping, Call Back on Trunk Ports, Station Message Detail Recording (SMDR).

ANANT UCS supports 16 Number Lists. Each Number List can contain upto 999 number strings. Each number string consist of a maximum of 16 characters.

The number strings are stored against Location Index numbers in the Number List. The Location Index numbers start from 001 to 999.

The default values of the Number Lists in the system are shown below:

Default Number Lists

Location Index	List Number															
	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16
001		00	0	00	*	*	*	*	*	*	*	*	*	*		
002		0	*	*	#	#	#	#	#	#	#	#	#	#		
003		1	#	#												
004		2														
005		3														
006		4														
007		5														
008		6														
009		7														
010		8														
011		9														
012		*														
013		#														
014		F														
015		+														
016																
:	:	:	:	:	:	:	:	:	:	:	:	:	:	:	:	:
999																

Use of Number Lists in various Features of ANANT UCS

Call Back on Trunk Ports

When Call Back on is set on any of the SIP Trunks, you must program two lists:

- Call Back Incoming Number List: this list defines the numbers which are eligible for a call back.
- Call Back Outgoing Number List: this list define the number to which the call back is to be made.

By default, Number List 15 is assigned to Call Back Incoming Number List, and Number List 16 is assigned to Call Back Outgoing Number List.

Call Duration Control

To set Call Duration Control (CDC) feature on an extension, you must program the Call Duration Control Table, in which you must program the numbers on which CDC must be applied as well as the numbers on which CDC must not be applied.

This requires you to program and assign two Number Lists: the Apply CDC to Number List and the Do Not Apply CDC to Number List.

By default, Number List 02 (For CDC Table 1, for others CDC Tables it is 7) is assigned to Apply CDC Number List, and Number List 08 is assigned to Do Not Apply CDC Number List.

Refer "Call Duration Control (CDC)" to know more.

Call Taping

Call Taping allows you to record conversations of incoming and outgoing internal and external calls. When you set Call Taping on an extension for external calls, the ANANT UCS uses two Lists: Number List - Incoming Calls: this list has the list of numbers of external callers whose conversation is to be recorded.

Number List - Outgoing Calls: this list has phone numbers of external called parties whose conversation is to be recorded. The system matches the incoming and outgoing numbers with the respective lists to apply Call Taping.

By default, Number List 09 is assigned to Incoming Calls Number List, and Number List 10 is assigned to Outgoing Calls Number List.

Refer the feature description "Call Taping" to know more.

Station Message Detail Recording (SMDR)

ANANT UCS supports 'Destination wise' storage of outgoing calls as an SMDR Filter, using which it is possible to store calls made from and received on selected destination numbers. To set Destination wise storage of calls in the SMDR Buffer, you must program and assign a Number List containing the phone numbers whose call details need to be tracked and stored. SMDR of an outgoing call will be tracked and stored only if it matches with an entry in the Number List assigned.

Similarly, SMDR Reports of Incoming and Outgoing Calls can be generated and printed for selected numbers by programming and assigning a Number List.

By default, Number List 02 is assigned to Destination Wise storage of SMDR and SMDR Report Printing.

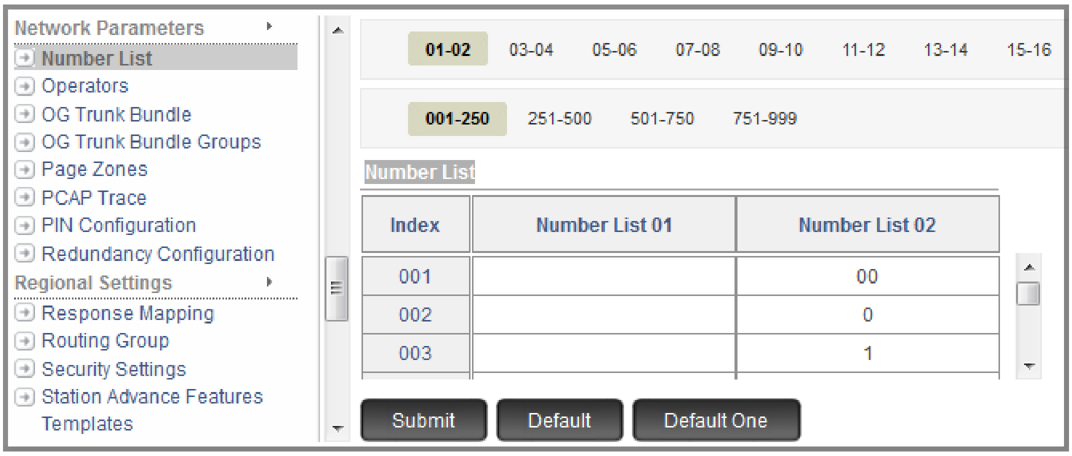
Refer the topics "Station Message Detail Recording (SMDR)", "Station Message Detail Recording-Storage", "Station Message Detail Recording-Report".

How to configure

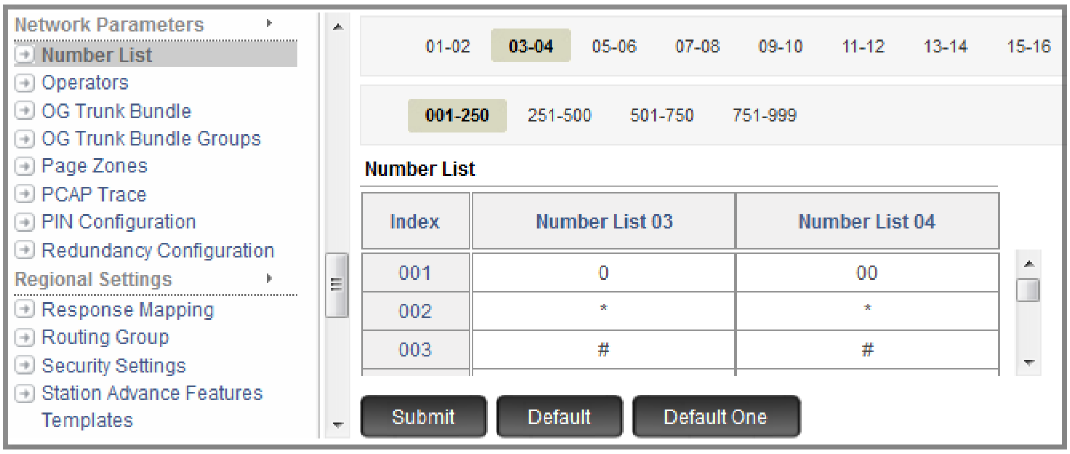
Take a pen and a paper. Decide which of the above-mentioned seven features are to be used. Number List according to the feature for which it is to be used.

Configuring Number List

- Login as System Engineer.
- Under **Configuration**, click **Number List**.



- Select the desired List Number. For example, Number List 03-04. Now click '001-250' of Number list 03-04. The location Index 001 to 250 will appear for both lists on the page.



- Enter each number string against a Location Index (refer to the list you prepared).
- Click **Submit.**
- Assign the Number List you prepared to the relevant feature.

One Touch Transfer

ANANT supports the UC feature, One Touch Transfer. Using this you can transfer an ongoing call from one extension to another mobile/fixed extension without putting the call on hold or dialing the destination extension number.

ANANT UCS will serve the One Touch Transfer request made by a SIP Extension (Matrix Extended IP Phones or Mobile Clients) only.

How it works

Consider the scenario, wherein you have the following registered as your extensions:

1. VARTA Mobile UC Client is registered as SIP Extension 1 at location 1.
2. Extended IP Phone is registered as SIP Extension 1 at location 2.

- You are in speech with A using your Extended IP Phone and you want to move away from your desk.
- In this case, you can transfer the ongoing call on your VARTA Mobile UC Client using One Touch Transfer.
- Press the DSS key assigned to One Touch Transfer (make sure the extension number of SIP Extension 1 is pre-configured as destination number for One Touch Transfer).
- You will receive the call on your VARTA Mobile UC Client.
- Answer the call from the VARTA UC Client. Speech is established and the call on your Extended IP Phone disconnects.
- You can now move away from your desk but your call can continue.

Similarly, you can transfer a call to another fixed extensions also.

- *You can access One Touch Transfer only during an ongoing 2-way speech. A held or a waiting call cannot be transferred using One Touch Transfer.*
- *For using One Touch Transfer between Extended IP Phone/VARTA UC Clients registered on different locations of the same SIP Extension, make sure the Call Appearance of the SIP Extension is more than 1.*

How to configure

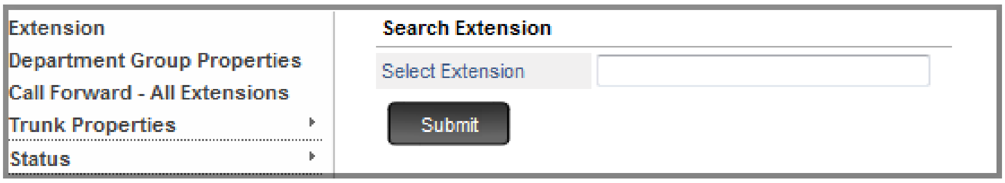
To access this feature,

- Make sure the *Basic Features* are enabled in the Class of Service assigned to you. For instructions, see “Class of Service (CoS)” and “Station Basic Feature Template”.
- Make sure you have configured a DSS Key for One Touch Transfer. For instructions on configuring DSS Key, see the topic “DSS Keys Programming”.
- Configure the destination extension number for One Touch Transfer. This number can be configured by extension users themselves from the Phone Menu or for any other extension from the SA mode.

Configuring One Touch Transfer

- Login as System Administrator.

- Click **Extension**.

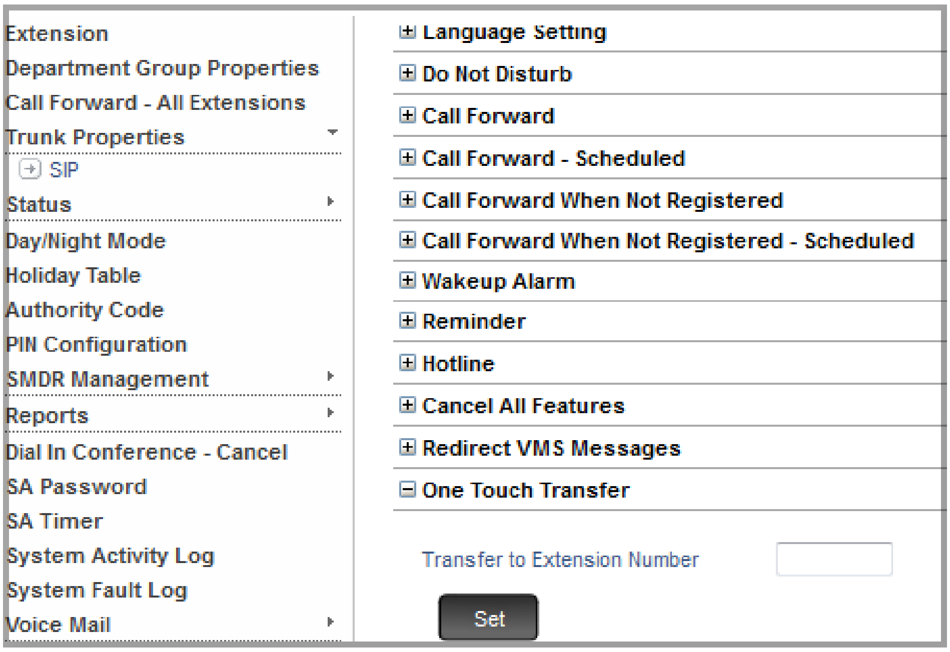


- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.

- Click **Submit**.

  The searched extension user details appear on your screen.

- Click **One Touch Transfer** to expand.



- In **Transfer to Extension Number**, enter the destination extension number on which you want the call to be transferred, when the feature One Touch Transfer is accessed.

Configuring One Touch Transfer by Extension Users using Phone Menu

To configure the Destination Extension Number,

- Scroll down the Phone Menu.
- Select **One Touch Transfer** by pressing the Enter Key.
- Select **Set Transfer Number** by pressing the Enter Key.
- Enter the destination extension number on which you want the call to be transferred.
- You hear Confirmation tone.

How to use

For Extended IP Phone Users

To use One Touch Transfer:

- During an ongoing call, press the DSS Key assigned to One Touch Transfer.
- The destination extension number configured for One Touch Transfer starts ringing.
- Answer the ringing call.
- You will be in speech with the remote party.

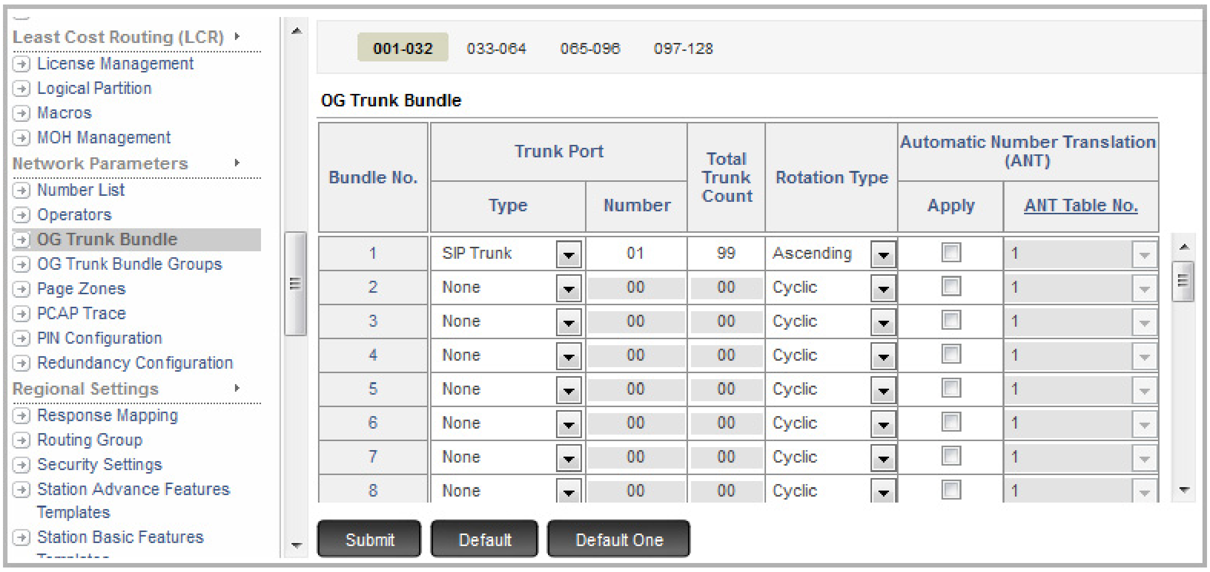
Outgoing Trunk Bundle

The Outgoing (OG) Trunk Bundle is set of parameters that completely define the grouping of the SIP trunks. These Bundles are further used to form an OGTBG. ANANT UCS supports 128 OG Trunk Bundles.

With the default OGTB assigned in the OGTBG outgoing calls will not be possible.You must change the configurations as per your requirement.

Configuring OGTB

- Login as System Engineer.
- Under **Configuration**, click **OG Trunk Bundle**.



Configure the following parameters for each Bundle Number:

- **Trunk Type:** You can select SIP Trunk or None
- **Trunk Number:** If you have selected SIP Trunk as the Trunk Type, enter the SIP Trunk number from which you want to begin the grouping here.
- **Total Trunk Count:** Multiple trunks can be there in a single group. Enter the maximum number of trunks you want in a group here. For example if you have entered the Trunk number as 2 and the total as 10, then trunks 2 to 11 will form one bundle.
- **Rotation Type:** This parameter shows which Trunk should be selected when the next call lands on that port.
  - Select **Ascending** if you want the system to check trunks from 01-99 to find a free trunk.
  - Select **Descending** if you want the system to check trunks from 99-01 to find a free trunk.
  - Select **Cyclic**, if you want the system to always select the next trunk for a new OG call.

- **ANT Apply:** Select the check box if you wish to use ANT feature. By default it is disabled.
- **ANT Table No.:** Enter the number of the ANT table you configured for this trunk, in which you have configured the Dialed Numbers and their corresponding Strip Digits and/or Add Prefixes. This table number is assigned to the specific trunk from which the number is to be dialed. Refer chapter “Automatic Number Translation” for more details.
- Click **Submit**.

OG Trunk Bundle Group

OG Trunk Bundle Group provides efficient allocation of trunks to different extensions.

All the trunks connected to the system can be bunched in different groups called OG Trunk Bundle Group. Maximum 8 Trunk Bundles can be put in one OG Trunk Bundle Group and 32 such OG Trunk Bundle Groups can be formed. These OG Trunk Bundle Groups can be allotted to each individual extension. An Extension can be allotted different OG Trunk Bundle Group during different timings of the day.

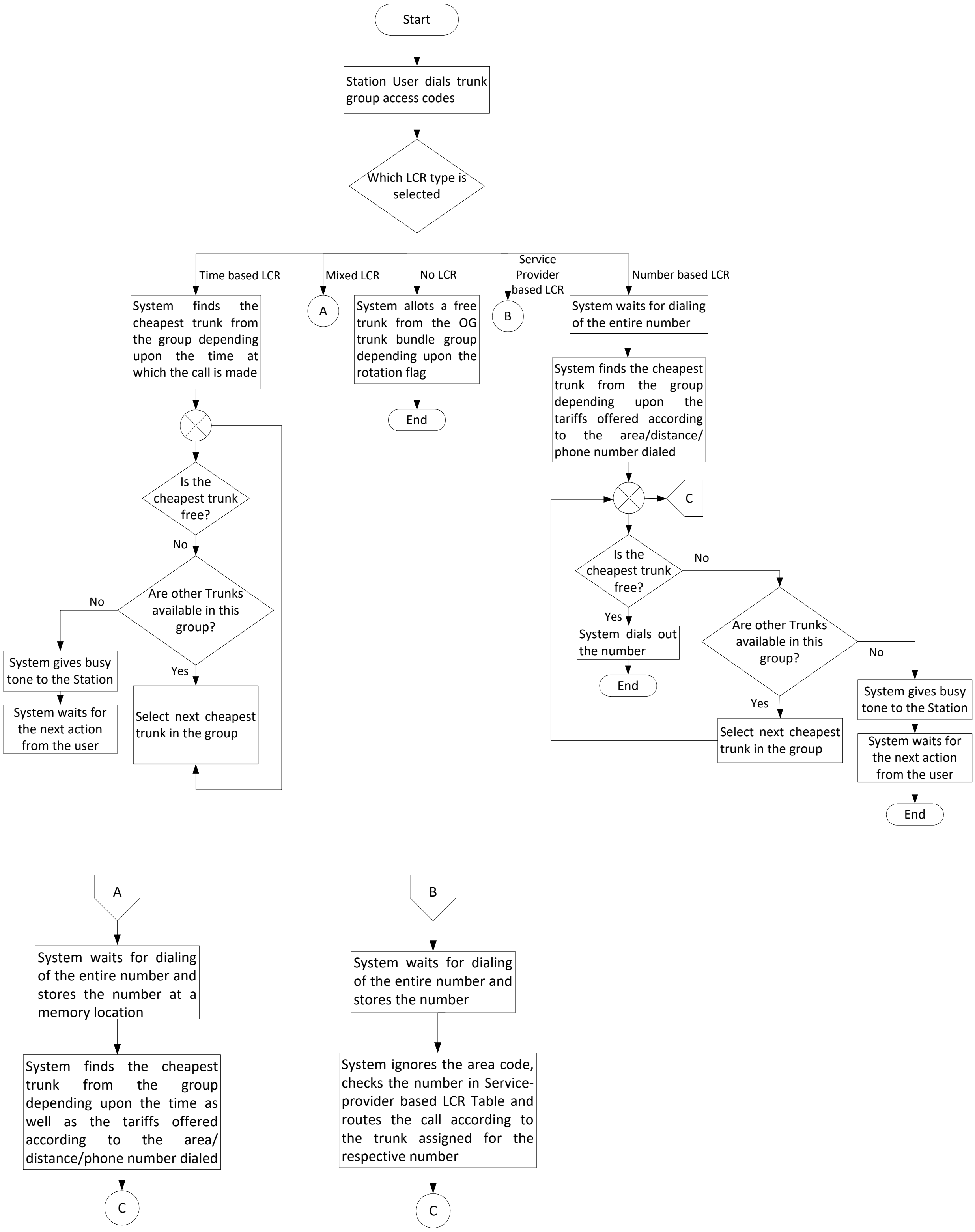
With the default OGTB assigned in the OGTBG outgoing calls will not be possible.You must change the configurations as per your requirement.

How it works

System uses two methods while selecting a trunk from the OG Trunk Bundle Group: ‘Remember last trunk’ and ‘Don’t Remember last trunk’.

In ‘Remember last trunk’ method, the system remembers the last trunk used and allots next trunk in the group to the extension. In ‘Don’t remember last trunk’ method, the system searches for a first free trunk from the group.

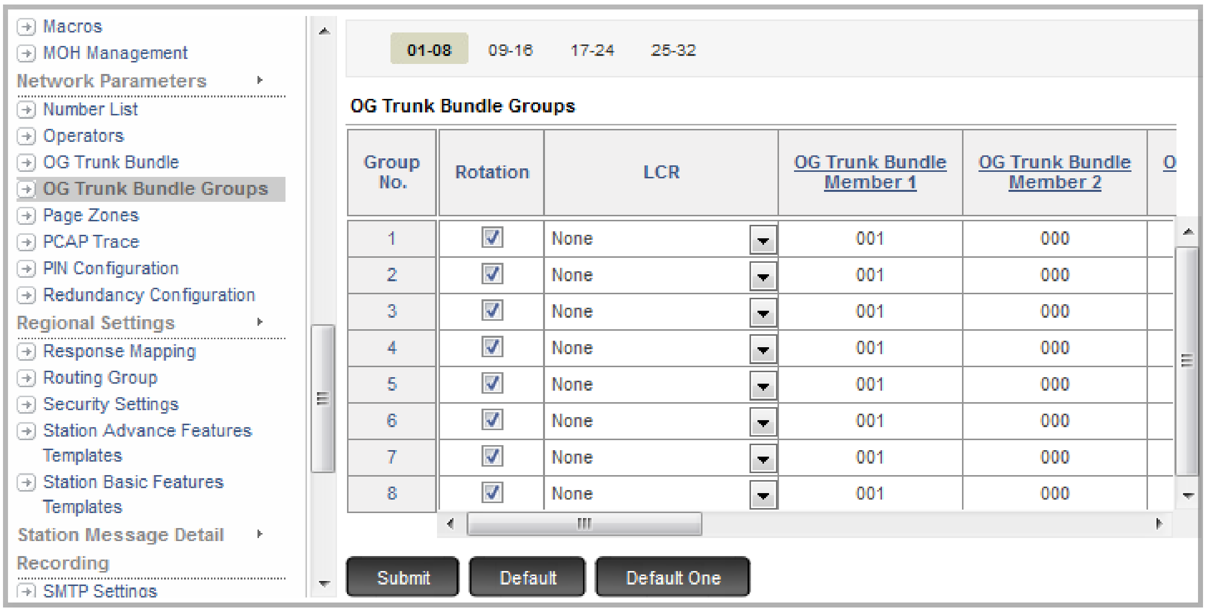
Following flow chart depicts the chronology of events when an extension grabs a trunk.



Configuring OGTB Groups

- Login as System Engineer.

- Under **Configuration**, click **OG Trunk Bundle Groups**.



Choose the OG Trunk Bundle Group number (01-08) you want to use. In each group you can configure maximum 8 'members'.

- Now configure the following parameters for the selected OG Trunk Bundle Group:

  - **Rotation:** Select the Rotation check box to enable rotation for routing outgoing calls in the group which has multiple 'member' trunks. When enabled, each outgoing call will be routed through the next member to the one that routed the previous call. This ensures equal distribution in routing outgoing calls. The check box has no relevance if the group has only one member. For detailed instructions, see “Outgoing Trunk Bundle”.

  - **Least Cost Routing:** If required select the desired method — Time-based LCR, Number based LCR, Time and Number based LCR or Service Provider-based LCR — as per your requirement. For details, see “Configuring LCR”.

  - **OG Trunk Bundle Member1 to 8:** Select the desired OG Trunk Bundle numbers you have already created. If you have not created the OG Trunk Bundles, for instructions see “Outgoing Trunk Bundle”.

    Configure only as many OG Trunk Bundles as members in the group as you want and set the remaining Members as '000'.

  - Click **Submit**.

- Repeat the same steps to create another OG Trunk Bundle Group.

How to use

1	Lift the handset.	Dial tone
2	Dial ***0/5/61 to 64.***	Dial tone of the trunk.
3	Dial desired external number.	

- To grab OG Trunk Bundle Group 3, the user should dial ‘61’.
- To grab OG Trunk Bundle Group 1, the user should dial ‘0’.

How the Rotation check box work?

- Eight OG Trunk Bundles can be configured as members of each OGTBG.
- OGTBG has “Rotation” check box, and each OG trunk bundle has Rotation type “Cyclic/Descending/ Ascending”.
- These two check boxes don’t have any relation with each other, and so, will work in isolation.

If Rotation is ON in the OGTBG:

- The first call will get routed using the “OG Trunk bundle Member1”. The trunk port from the OG Trunk Bundle member 1 will get selected using the rotation type selected for the OG trunk bundle programmed as member 1.
- When there is a second call, it will get routed using the “OG trunk bundle Member 2”, and the trunk port from the OG trunk bundle member 2 will get selected according to the rotation type programmed in the OG Trunk Bundle programmed as member 2.
- Accordingly, the member of the OGTBG will be accessible to the station user accessing the OGTBG in sequence.

If the Rotation is OFF in the OGTBG:

- The calls will always get routed from the “OG trunk Bundle member 1” if any trunk in it is free. If all the trunks of the “OG trunk bundle member 1” are busy, then the call will get routed using the “OG trunk bundle member 2”.
- Now the trunk to route the call will get selected as per the rotation type configured for the OG trunk bundle used as member 2. When the trunk ports of OG trunk bundle configured in member 1 and member 2 all are busy, the OG trunk bundle member 3 will be used to route the call.
- Thus the Rotation check box of OGTBG will be used to select the OG trunk bundle member1 to member 8 as per call basis while the rotation type check box associated with the OG trunk bundle will decide the rotation mechanism to select the trunk port from the particular OG trunk bundle.

Default OG Trunk Bundle Group Table (Table-1):

Group No.	Rotation	LCR	OGTB Member 1	OGTB Member 2	OGTB Member 3	OGTB Member 4	OGTB Member 5	OGTB Member 6	OGTB Member 7	OGTB Member 8
01	√	None	00	00	00	00	00	00	00	00
02	√	None	00	00	00	00	00	00	00	00
03	√	None	00	00	00	00	00	00	00	00
04	√	None	00	00	00	00	00	00	00	00
05	√	None	00	00	00	00	00	00	00	00

::	√	None	00	00	00	00	00	00	00	00
21	√	None	00	00	00	00	00	00	00	00
22	√	None	00	00	00	00	00	00	00	00
23	√	None	00	00	00	00	00	00	00	00
24	√	None	00	00	00	00	00	00	00	00
25	√	None	00	00	00	00	00	00	00	00
26	√	None	00	00	00	00	00	00	00	00
27	√	None	00	00	00	00	00	00	00	00
28	√	None	00	00	00	00	00	00	00	00
29	√	None	00	00	00	00	00	00	00	00
30	√	None	00	00	00	00	00	00	00	00
31	√	None	00	00	00	00	00	00	00	00
32	√	None	00	00	00	00	00	00	00	00

Paging

Paging allows you to make announcements to groups of extension users and to make public announcements over a public address system. You can deliver a message to a mass of people at once by just lifting the handset of your phone and dialing a code.

This feature is useful when you want to call several people at once; for example, to inform them about a meeting you have scheduled. If the persons you want to call have Matrix Extended IP Phones or Standard SIP Phones as their extensions, you can use paging instead of calling them up one by one.

ANANT UCS supports paging where announcements are made on SIP extensions.

The extensions which are to be paged must be included in 'Page Zones'.

- *You can start paging from any SIP Extension and the paged extensions must also be an Extended IP Phones or Standard SIP Phones. The Standard SIP Phones on which you are paging must support Call-Info or Alert-Info header for Paging.*
- *When the Paging call is generated in SIP Extension having multiple call appearance and already a call is present on the SIP Extension then the* ANANT *UCS will place the Paging call as normal call on the SIP Extension (as headers required in INVITE for paging call and intercom call are same).*
- *Paging is a one-way communication. As the mic of the paged extensions is muted during Paging, the users of the paged extensions cannot speak to the paging extension.*

How it works

The Pre-requisites

- Page Zones must be created. Each Page Zone accommodates up to 32 SIP extensions. You can create 12 different Page Zones of 32 SIP extensions.
- Paging must be enabled in the Class of Service allowed to the SIP extension from which this feature is to be used.

The Process

- A user of a SIP Extension having Paging in its Class of Service, dials the Access code for Paging and the Number of the Page Zone to which the user wants to make the announcement.
- The system activates the speakers of the SIP Extensions programmed in the Page Zone number. The system activates the speakers only of those SIP extensions in the Page Zone that are free.
- The calling SIP Extension user makes the announcement.
- All SIP Extensions in the Page Zone can hear the announcement. But as the mic of their phones is muted, their speech will not be heard by the calling SIP extension user.
- To answer the Paging call the desired extension user must use Meet Me Paging while the Paging call is active. For details refer, “Meet Me Paging”.

- If no reply is received via Meet Me Paging, the calling SIP extension goes ON-Hook after the announcement.
- The system deactivates the speakers of the SIP Extensions.

How to configure

For this feature to work, you must create Page Zones and enable this feature in the Class of Service of the extensions which are to be allowed this feature.

Allowing Paging in Class of Service

In the default factory settings, Station Basic Feature Template Number 01 is assigned to all the stations of ANANT UCS. Station Basic Feature Template 01 is assigned CoS group 01 which has Paging enabled. So, all stations of ANANT UCS can page.

If you want to allow Paging to all stations, retain CoS group 01 in Station Basic Feature Template 01.

However, if Paging is to be disallowed to some stations then follow these steps:

a. Define a CoS group with Paging disallowed.

b. Prepare a Station Basic Feature Template with this CoS group applicable in all the "Time Zones".

c. Assign this new Template to the stations to which Paging is to be disallowed.

Refer the topics "Class of Service (CoS)" and "Station Basic Feature Template" for detailed instructions and configuration.

Creating Page Zones

Decide how many Page Zones you want to create. You can create as many as 12 different Page Zones. You can include up to 32 SIP extensions.

For each Page Zone number, decide and assign the SIP extensions.

On a sheet of paper create a three-column table for each page zone, as shown below. Enter the Type of extension-SIP, and the Software Port number of the extensions you want to include in the page zone.

Page Zone 1

Member Number	Type of Extension	SIP Extension Port Number
1	SIP	002
2	SIP	003
3	SIP	008
4	SIP	009
:		
:		
32		

Page Zone 2

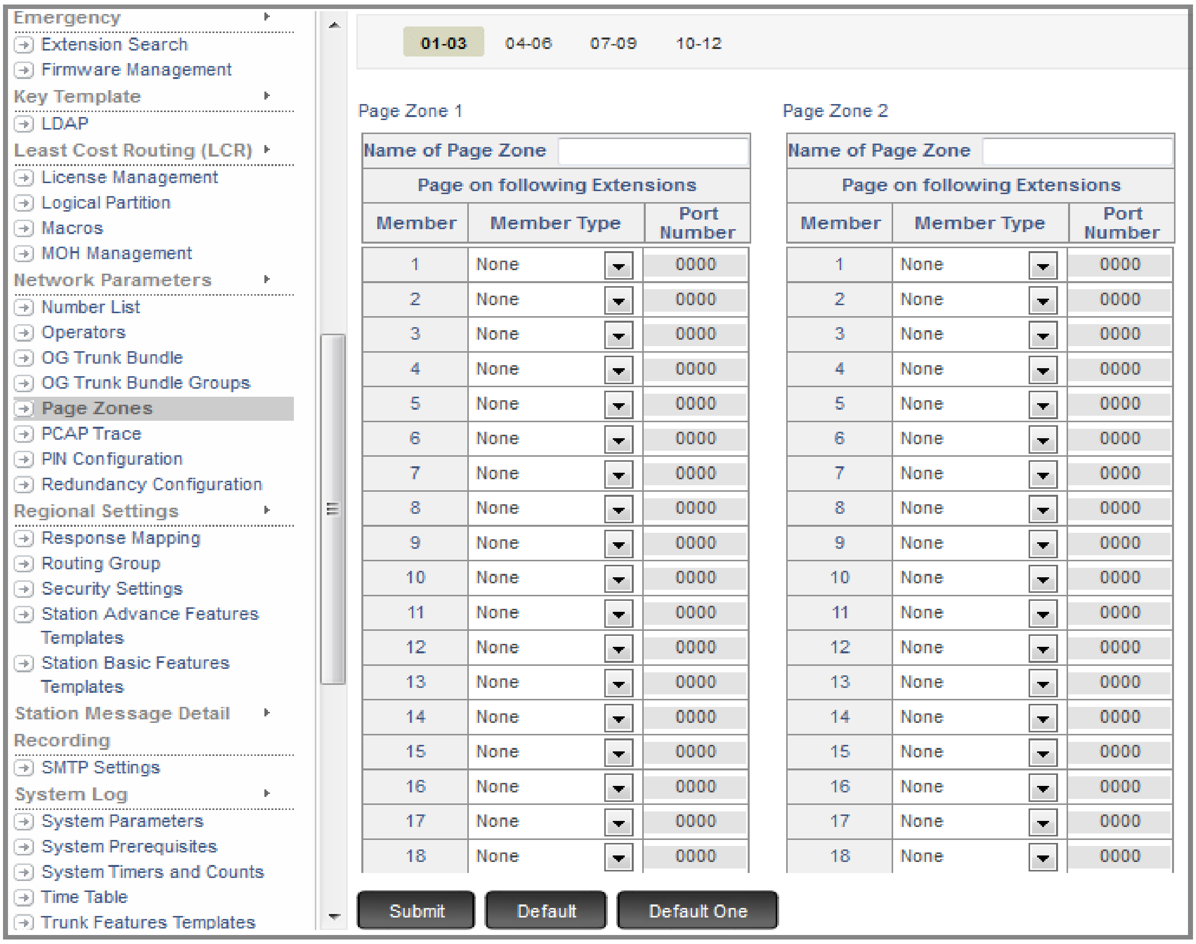
Member Number	Type of Extension	SIP Extension Port Number
1	SIP	007
2	SIP	010
3	SIP	012
4	SIP	013
:		
:		
32		

Now, configure Page Zones using Jeeves.

Configuring Page Zones

- Login as System Engineer.
- Under **Configuration**, click **Page Zones**.
- The first three page zones appear on this page.
- Refer to the Page Zones you created on paper and program the following parameters in each Page Zone:
  - **Name of Page Zone:** Enter the name you want to assign to the Page Zone.
  - **Member Type:** Select the extension type you want to include in the Page Zone: SIP extension.

- **Port Number:** Enter the number of the Software Port of the extension you selected in the Port Type.



- Click **Submit**.

- To go to other Page Zones, you may click the hyperlinked page zone numbers on the top of the Jeeves screen.

How to use

For Extended IP Phone Users

- Press DSS Key assigned to Paging.
  OR
- Dial **1074**
- Enter Page Zone Number on the prompt on your phone's display.
- You get the prompt on your phone's display: Start Paging <Page Zone Number>.
- Make your announcement.
- Go ON-Hook at the end of your announcement.

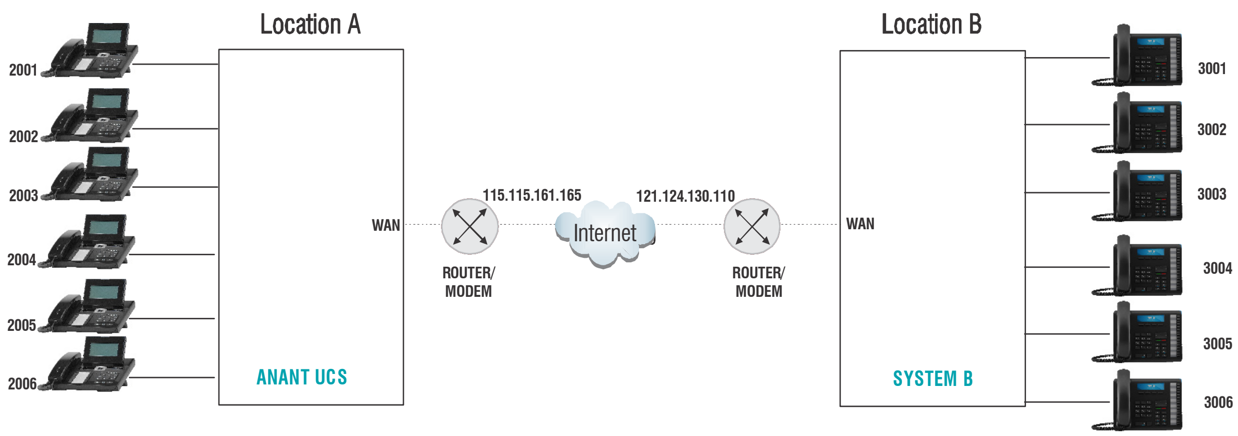
Peer-to-Peer Calling

Making an IP call without the intervention of a proxy server is called Peer-to-Peer Calling. As Peer-to-Peer calling does not require a proxy server, voice communication using this application can be done virtually free of cost. The major cost savings offered by this application makes it a very attractive mode of inter-branch or intra-office voice communication.

For dialing a number use '' in-place of '.' This feature is not supported in Standard SIP Phones.*

How it works

Let us understand how to use Peer-to-Peer Calling with the following illustration:



- Two offices are directly connected over the IP network.
- ANANT UCS is installed at Location A.
- The System at Location B may also be ANANT UCS.
- Peer-to-Peer calls can be made between the two locations with suitable configuration of ANANT UCS and the System.
- **At Location A**, you need to do the following configuration in the ANANT UCS:
  - Select a SIP Trunk to be used for this application and enable it. For example, SIP1.
  - Set the **SIP Trunk Mode** of this trunk to **Peer-to-Peer**.
  - Keep the **SIP ID** field of the SIP trunk blank.
  - Set the **Treat Incoming call as** option on the SIP trunk to **Station**.

  For detailed instruction, see "Configuring SIP Trunks".

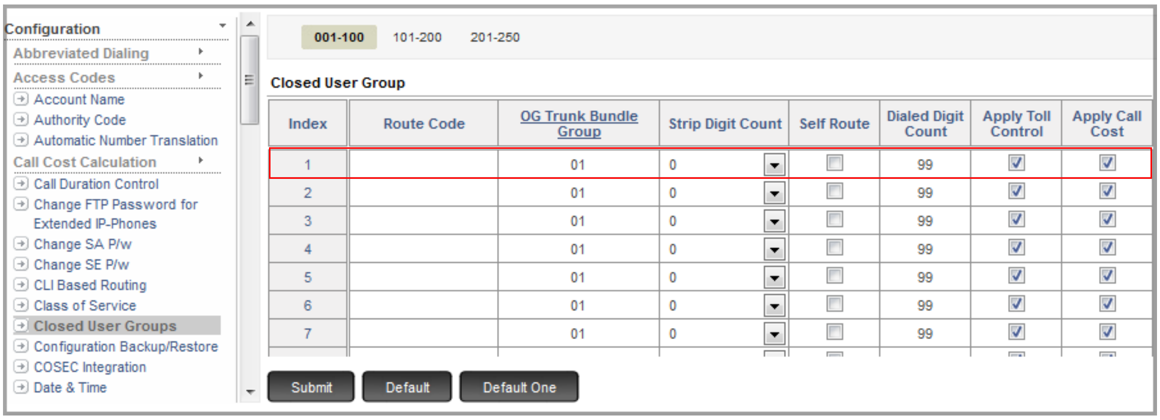
In the Router, you must configure the same SIP and RTP Ports as configured in the ANANT UCS. In other words, you must configure Port Forwarding for SIP and RTP on the Router.

- You must also configure the **Trusted IP Address/es** table for this SIP Trunk to receive incoming calls. If you do not configure this table, all incoming calls on this SIP Trunk will be rejected. For instructions, see “Configuring SIP Trunks”.

- Set the **Send CLI Option** on the SIP trunk as **Calling Party Wise**. See “Configuring SIP Trunks” for instructions.

- You may also configure the **Closed User Group (CUG) Table** to avoid dialing the Trunk Access Code for the outgoing calls made from this SIP Trunk, i.e. SIP 1.

  At Location A, you may configure the CUG table as follows:



  - In **Route Code**, enter the extension numbers of the System at Location B. Instead of entire number strings, you can configure a single digit, the starting digit of the extension numbers as Route Code. In this case, you may configure Route Code as ‘3’, as all extensions at Location B start with ‘3’.

  - Keep the **Strip Digit Count** as ‘**0**’.

  - Keep the **Self Route** check box disabled.

  - In **Dialed Digit Count**, enter the digit length of the extension numbers at Location B. In this case, ‘**4**’.

  - In **OG Trunk Bundle Group**, select 01. Configure SIP Trunk1 as the only member in this group. The calls will be routed through this SIP Trunk only.

  - When Self Route check box is disabled, system will check **Apply Toll Control** parameter. By default, it is enabled. The system will apply toll control to all the outgoing calls.

    Disable this check box, if you do not want to apply toll control to the CUG numbers dialed by you.

  - By default, **Apply Call Cost** check box is enabled. Select this check box for Peer to Peer calls if you require the call cost calculation.

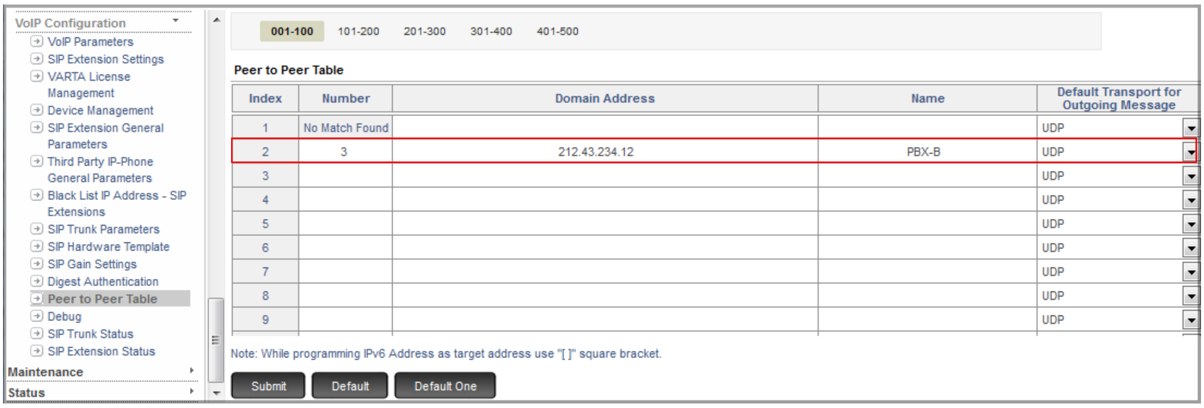
  For detailed instructions, see “Closed User Group (CUG)”.

If the System’s at both the offices have the same extension numbers, the users need to dial the Exchange ID of the System along with the extension numbers. For detailed instructions, see “Closed User Group (CUG)”

- Now, configure the **Peer-to-Peer Table**.

  The Peer-to-Peer table stores up to 500 entries. Each entry consists of the parameters Number, Domain Address, Name and Default Transport for Outgoing Message.

  At location A, you would have to configure the Peer-to-Peer table as follows:



  - In **Number**, enter the digit you configured as **Route Code** in the CUG Table for calling the extension numbers of the System at Location B. In this case ‘3’.

  - In **Domain Address**, enter the **IP Address** of the WAN Port of the Router at Location B.

  - In **Name**, enter ‘Location B’ for identification.

  - Keep **Default Transport for Outgoing Message** as ‘UDP’.

- At Location B, you may do a suitable configuration of the System.

- When an extension user 2001 of ANANT UCS at Location A dials 3001, an extension number of System B, it checks the CUG table to match the dialed digits with the Route Code and the Dialed Digit Count. As a match is found, it selects the SIP trunk defined for routing the Route Code, i.e. SIP 1.

- As SIP 1 is set to Peer-to-Peer mode, the system checks the Peer-to-Peer Table configured. It finds a match for the digit ‘3’ and will route the call to the IP Address configured for this number. In this case, to the IP Address of the Router at Location B (121.124.130.110).

- Further, the Router will forward the call to System B. With suitable configuration done in System B, the call will be routed to the desired destination i.e. extension 3001.

Configuring Peer-to-Peer table

Peer to Peer Table is used, for deciding the destination IP Address for routing calls using non-proxy SIP trunks (Registrar server address shall be programmed as blank for these trunks). The Peer-to-Peer table stores upto 500 entries. Both IPv4 and IPv6 addresses are supported. This table is common for all the SIP trunks.

For instructions on configuring the SIP Trunk for the Peer-to-Peer application—SIP Trunk Mode, SIP ID, Treat incoming calls as Station, Trusted IP Address/es table—see “Configuring SIP Trunks”.

For instructions on configuring the CUG Table, see “Closed User Group (CUG)”.

To configure the Peer-to-Peer table using Jeeves,

- Login as System Engineer.

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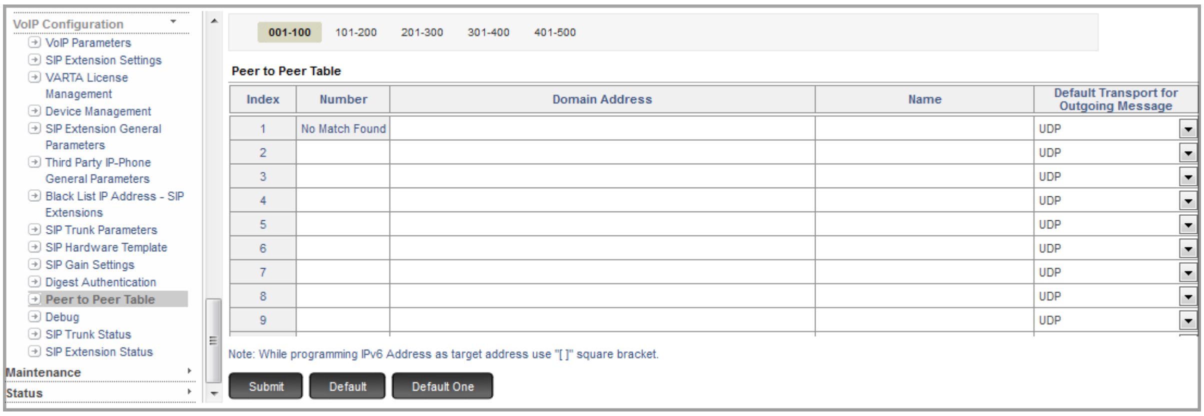
- Under **VoIP Configuration**, click **Peer-to-Peer Table**. The Peer-to-Peer table opens.



The first entry is reserved for No Match Found.

- In **Number**, enter the peer-to-peer number string—prefix or entire number—that will be dialed. The number string must not exceed 8 digits. Default: Blank.

- In **Domain Address,** enter the domain name or IP Address to where the call is to be placed. Both IPv4 and IPv6 addresses are supported. The Domain Address may consist of up to 48 characters (maximum). Default: Blank.

  The Destination Address can also be in the form of Address: Port number.

- In **Name**, enter a name to identify the number string you configured. It may be the name of your contact or any name you wish to assign to the number string. The name may consist of 24 characters (maximum). Default: Blank.
  The name you configure here will not be used in SIP signaling.

- In the **Default for Outgoing Message** field, select the option for transporting outgoing SIP messages. You can select UDP, TCP or TLS.

- Click **Submit.**

PIN Dialing

PIN is a unique four digit code with an associated Class of Service and Toll Control, which can be assigned to the extension user.

PIN Dialing allows an extension user to make outgoing calls from any extension according to the toll control assigned to his/her PIN. PIN Dialing must be enabled in the Class of Service of the extension from where the outgoing call using PIN is to be made. ANANT UCS supports maximum 500 PINs for each type of toll control.

Calls made using PIN can be logged in the ANANT UCS and you can print the report online or later as and when required.

How it works

Let us understand how this feature works with the help of an example.

User A is denied dialing of International Numbers from his/her extension but has PIN Dialing enabled in his/her CoS. A is assigned a PIN 1234 with toll control level as international call. Now, User A can make international calls in two ways:

1. User A must dial the feature access code to access PIN Dialing i.e. *2 (configurable, see “Access Codes”) followed by his/her PIN. As soon as ANANT UCS detects a valid feature access code and PIN, it gives trunk dial tone to A. Now user A can dial the desired international number and talk.

2. User A can directly dial an international number from any extension that is assigned a “Station Basic Feature Template” with PIN Dialing enabled in the Class of Service and LCR enabled in the OGTB group.

   ANANT UCS will prompt A to dial his/her PIN. As soon as s/he dials a valid PIN, ANANT UCS will dial out the number and connect A to the called party.

   This method of calling is useful while making calls through Standard SIP Phones, as it prevents the PIN from being stored in the call logs of the phone.

- *You cannot Redial or set Auto Redial for the numbers dialed out using PIN.*
- *While dialing the Access Code and PIN from the desired extension, the **Extension - Inter Digit Wait Timer (sec)** will be applicable. See “System Timers and Counts” for more details.*

How to configure

To be able to use this feature,

- Make sure, PIN Dialing is enabled in the “Class of Service (CoS)” of the Extensions from where you want to make outgoing calls using PIN.

- Enable the parameter *Store Outgoing Calls* in the “Station Basic Feature Template” assigned to the extensions from where you want to make outgoing calls using PIN.

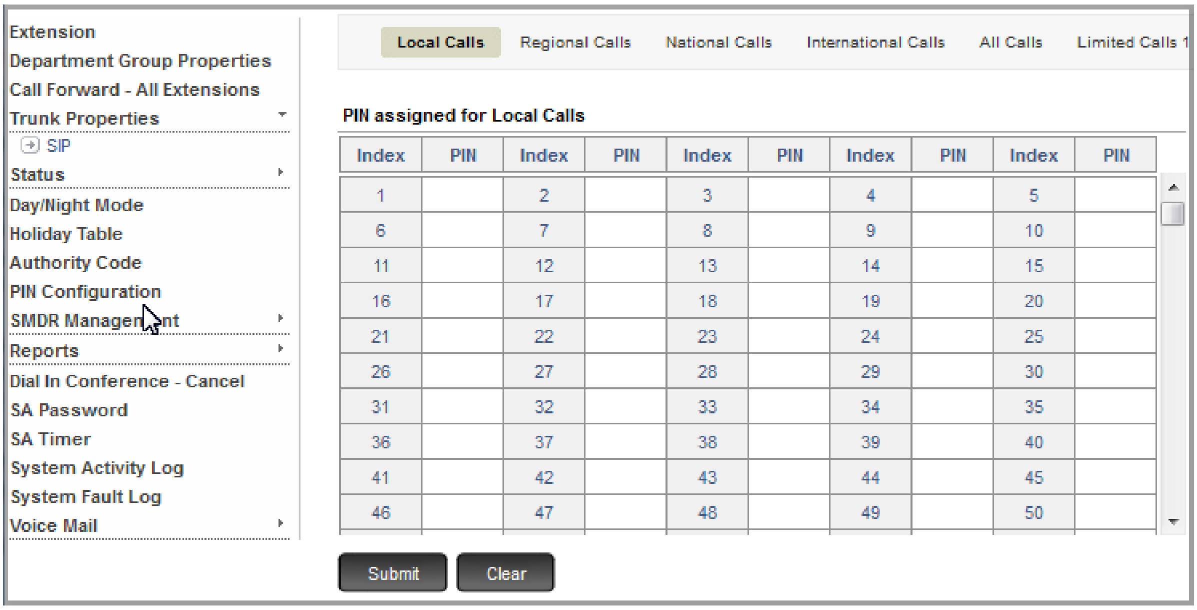
- Assign LCR enabled OGTB Group in the “Station Basic Feature Template” of the Extensions from where you want to make outgoing calls using PIN.

- Configure the PIN table for each type of Toll Control level from the SA Mode. See “Configuring PIN Table”.

- Select the OGTB Group and type of LCR for making outgoing calls using PINs, for each Toll Control Level from the SE Mode. See “Configuring OGTB Group and LCR”.

Configuring PIN Table

- Make a list of users to whom you want to assign PINs.
- Login as **System Administrator**.
- Click **PIN Configuration**.



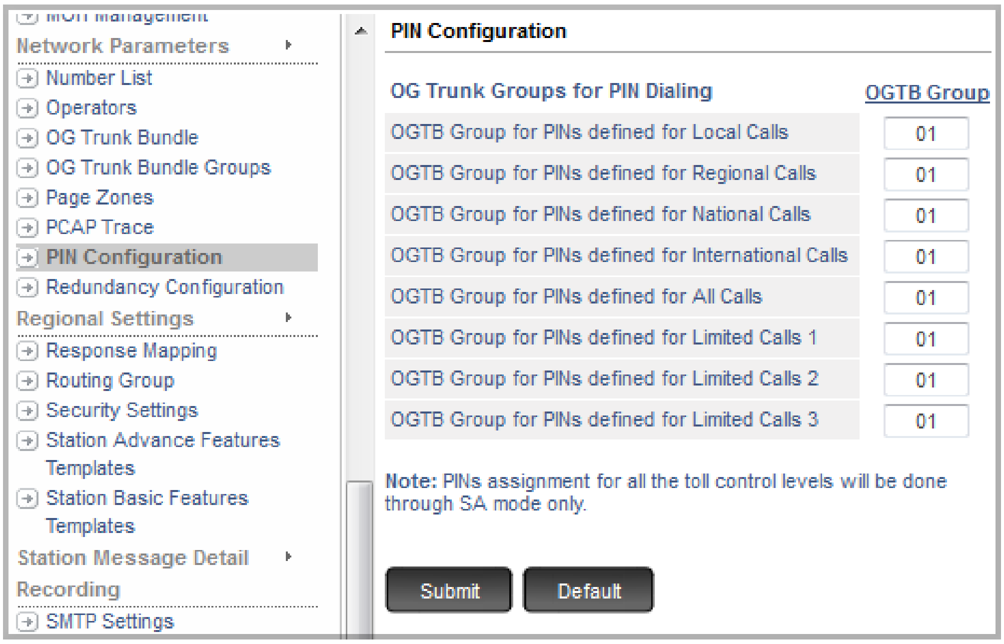
- Select the desired Toll Control level. Each Toll Control level includes the Allowed and Denied numbers as configured by SE. See “Toll Control” for more details.
- Configure the PINs at a desired index’s. The PIN must be of 4 digits. Valid digits are 0 to 9.

- *Each PIN must be unique. You cannot assign same PIN to two different index within the same Toll Control level.*
- *If you re-assign the same PIN to a different Toll Control Level, it will not be valid for the previously assigned Toll Control Level.*
- *To avoid unauthorized access, we recommend you to change the PIN regularly. Make sure it is strong and is kept confidential.*

Configuring OGTB Group and LCR

- Login as **System Engineer**.

- Under Configuration, click **PIN Configuration**.



- In **OG Trunk Groups for PIN Dialing**, enter the desired **OGTB Group** number for each Toll Control level.

- To select the desired trunks in the OGTB Group and the type of LCR, see “OG Trunk Bundle Group” for instructions.

- To configure the LCR Table, see “Configuring LCR” for instructions.

How to use

For Extended IP Phone Users

To use PIN Dialing from any extension:

- Go OFF-Hook.
- Press DSS Key assigned to ‘PIN Dialing’.
  OR
- Dial ***2**
- Dial the PIN.
- You will hear the trunk dial tone.
- Dial the desired number.
  OR
- Go OFF-Hook.
- Dial TAC.
- Dial the desired number.
- You get Feature tone.
- Dial the PIN.
- Speech with the Called Party.

Printing Reports of Outgoing Calls made using PIN

You can print call details of users who made outgoing calls using the PIN. For this, you will need to:

- Enable **Store Outgoing Calls** in the Station Basic Feature Template of the extension from where you want to make outgoing calls using PIN.
- Enable **Print Calls made using PIN** and set the PIN range in **Calls made using PIN** under Calls filter in Outgoing Call Report.
- Configure the **Destination IP Address and Port** for SMDR-Outgoing Call Report.

Refer “Station Message Detail Recording-Report”, for detailed instructions on printing reports using filters.

You may also print online report of calls made using PIN, when ANANT UCS is interfaced with a third party Call Accounting Software (CAS). For this, you must set the parameter PIN in SMDR - Posting. See “Station Message Detail Recording-Posting” for more details.

Presence

Presence is an important UC feature as it helps you to know the availability status of other users. Depending on the status of users you can decide whether to initiate a conversation or find an alternative way to contact the desired user.

For example, an extension user may want to leave his desk for an indefinite period, but does not want to use Call Forward or set Do Not Disturb. He wants to indicate to callers about his absence. Similarly, extension users who are present at their desk may want to hide their presence from other users; or they may want to show their current activity to the other extension users like they are Busy, or are away from their desks, or on the phone with someone on another call, etc.

With the Presence feature of ANANT UCS, extension users, including the Operator, can 'publish' their presence to callers from other extensions. By doing so, they can indicate to the other extensions about their availability.

In the same way, the Presence feature allows extension users to view the 'Presence' status (availability) of the extensions that they want to call, before making the call or when their call is not answered.

How it works

Publishing Presence

Any SIP Extension User can 'publish' their presence by setting any of the messages listed in the following on their phone, by dialing the access code for this feature.

SIP Extension users who want to publish their presence have two options:

- *Using the PUBLISH feature supported by the SIP Client.*
- *Using the feature access code for Publish Presence supported by ANANT UCS.*

The first option requires the parameter 'PUBLISH' to be enabled in the SIP Extension Settings. Refer *“Configuring SIP Extensions”**. By default, this parameter is disabled.*

Publishing Presence Messages

1. **Absent:** When an extension user sets 'Absent' as the message, all incoming internal as well as external calls will be blocked from landing on his/her extension.

   When any other extension user calls this extension, the text message 'User Absent' will appear on the caller's phone display.

   If the extension phone that has set 'Absent', the letter 'A' appears on the phone's display to indicate absence.

   The letter 'A' disappears when the extension user sets a presence message other than 'Absent'.

- *External callers who call the extension, on which 'Absent' is set, will get an error tone only.*
- *Outgoing calls can be made from the extension which has set 'Absent'. Only incoming calls are restricted.*
- *If more than one extension is configured as "Operator" (routing group), incoming calls will be blocked only on the Operator extension which has set User Absent.*

2. **Present:** When an extension user sets 'Present', all incoming calls will be received as normal on this extension.

   If previously set as 'Absent', when an extension user sets 'Present' the letter 'A' will disappear from the phone's display.

   When any other extension user calls this extension, the name of the extension user will be displayed on the caller's phone display, when the called extension is ringing.

3. **Auto Detect:** When an extension user sets 'Auto Detect', the system will detect the state of the phone; depending on the call state, it will publish the presence message to the other extensions. Three types Publish Presence messages are possible, with Auto Detect:

   a. ***Idle:*** When the system detects the extension phone to be ON-Hook, it indicates the status of the phone to other extensions 'idle'.

   b. ***On the Phone:*** When the system detects the phone to be OFF-Hook, or in speech with another party or if it detects an incoming call placed on the phone, it will indicate to the other extensions that this extension user is 'On the Phone' with another party.

   c. ***DND Text message:*** When the system detects that the extension phone has Do Not Disturb (DND) set on it with a DND Text message, it will display to the calling extension, the DND message set by the called party (this may be the default DND message or the DND Text message set by the called extension).

3. **Away:** When an extension user sets 'Away', the system will display this message to the other extensions.

4. **On the Phone:** When an extension user sets 'On the Phone', the system will display this message to the other extensions.

5. **Do Not Disturb:** The extension user can set this message to be published to other extensions, if s/he wants to work uninterrupted.

   Unlike the DND Feature, the extension user who has set this message will continue to receive calls both internal as well as external calls, as the system considers this extension as 'present'.

6. **I am Mobile:** The extension user can set this message to be displayed to other extensions, when s/he is not at the desk.

7. **In Meeting:** The extension user can set this message to be displayed to the callers, if s/he is busy in a discussion or meeting.

8. **Out for Meal:** The extension user can set this message to be displayed to other extensions when going on a lunch break.

9. **Out of Office:** The extension user can set this message to be displayed to the callers when s/he leaves the office temporarily.

- *When an extension user sets any Publish Presence message other than 'Absent', the system will consider the user as 'Present'. All incoming and outgoing calls will be allowed on this extension.*
- *It is possible to customize another message in place of Publish Presence messages listed from 6 to 9: I am Mobile, In a Meeting, Out for Meal, Out of Office.*
- *Publish Presence messages can be set or changed for any extension from the System Administrator (SA) mode.*

Viewing Presence

- Extension users can know the status of another extension user before calling or when the extension user does not answer the call.
- Generally, when SIP extension users call another extension, the name of the called extension is displayed on the calling SIP extension. Now, if the check box *'Display Presence status during call on Extended IP Phone'* is enabled in the System Parameters, when SIP extension users call another extension, the calling SIP extensions will be displayed the 'presence' status message published by the called extension[109].
- SIP extension users can use the Presence feature of ANANT UCS to view the presence status of other extensions. For this, they must dial the feature access code and the number of the desired extension.
- SIP extension users who want to view the status of other extensions using the feature supported by their SIP Client, must have 'Presence Subscription' enabled in their SIP Extension Settings. Refer "Configuring SIP Extensions".

How to configure

This feature involves configuring the following parameters:

- **'Display Presence status during call on Extended IP Phone' check box:** SIP extension users will be able to view the presence status for the called extension only if this check box is enabled in the System Parameters.
- **PUBLISH:** SIP extension users who want to publish their presence using the feature supported by their SIP client will be able to publish their presence status only if this feature is enabled in their SIP Extension Settings. This parameter is not necessary, if they want to publish presence using the feature of ANANT UCS.
- **Presence Subscription:** SIP extension users who want to view the presence of other extensions using the feature supported by their SIP client must have this feature enabled in their SIP Extension Settings. This parameter is not necessary, if they want to view presence using the feature of ANANT UCS.
- **Publish Messages:** It is possible to customize the Publish Messages listed above from 6 to 9 viz.: 'I am Mobile', 'In Meeting', 'Out for Meal', 'Out of Office'.

The above parameters, with the exception of 'Publish Messages', can be configured using Jeeves.

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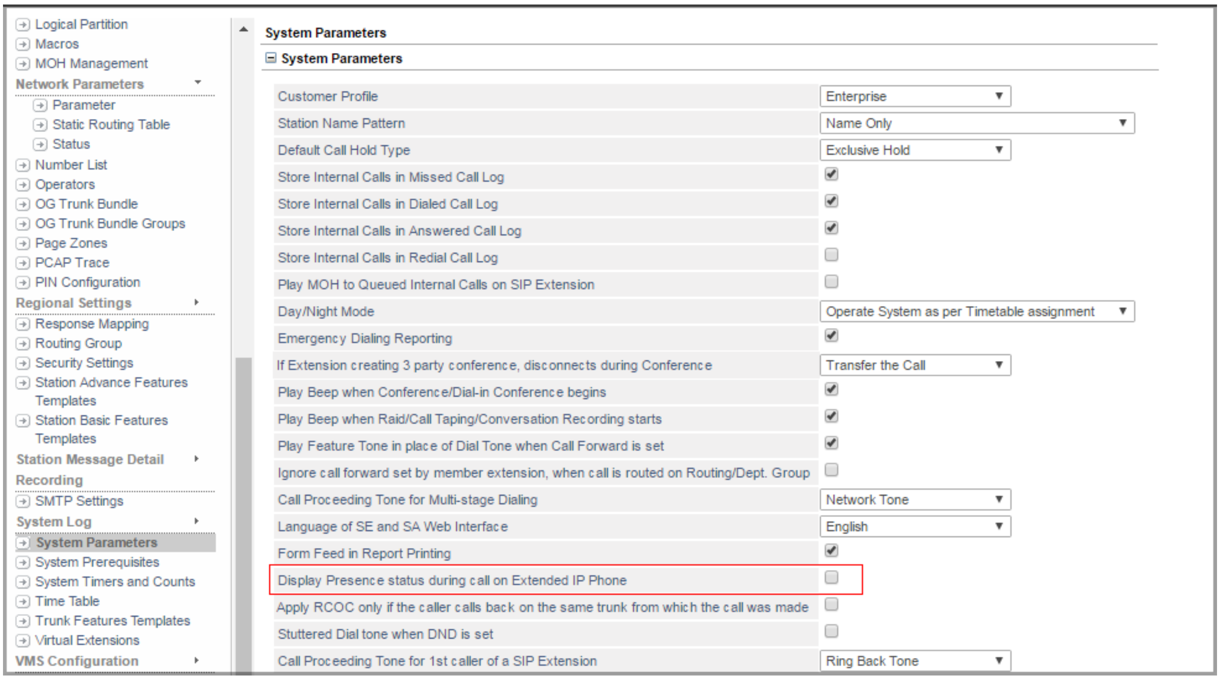
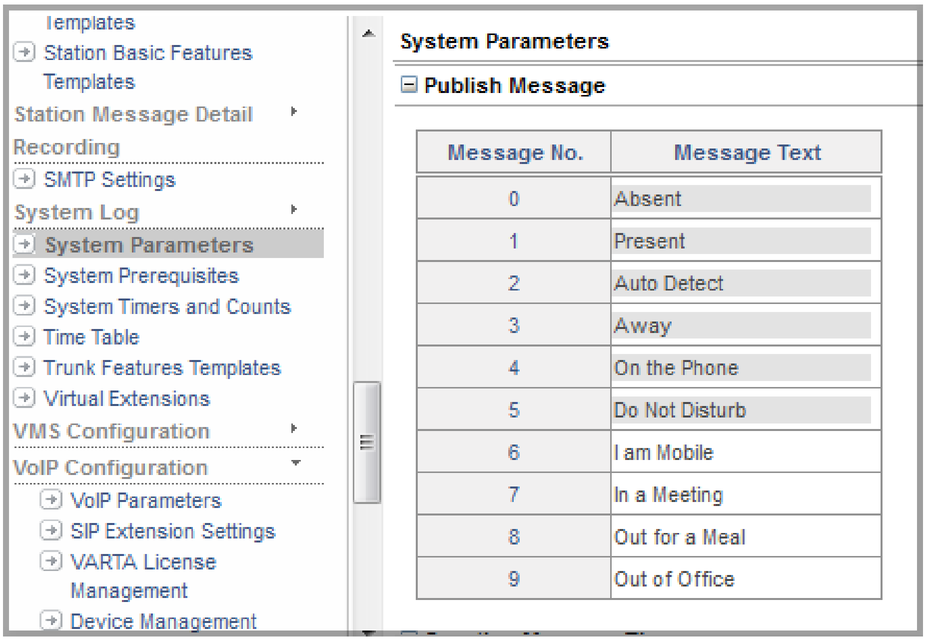
109. *SIP users can also dial a feature access code and the number of the extension to see the status of that extension on their extension. But this would not be required, if the 'Display Presence Status during call on extended IP Phone' check box is enabled in the System Parameters.*

Enabling Presence

- Login as System Engineer.
- Under **Configuration**, click **System Parameters**.
- Enable the check box **Display Presence status during call on Extended IP Phone**.



- Click **Submit.**
- Click **Publish Message** to expand.



- You can change message number 6 to 9 as desired. The string may consist of a maximum of 16 characters. All ASCII characters except <, >, :, ", /, \, |, ?, * are allowed.

- Click **Submit.**
- Under **Configuration**, click **VoIP Configuration**.
- Click **SIP Extension Settings.**
- Now, go to the desired SIP Extension number for which you want to enable the features **PUBLISH** and **Presence Subscription**. By default both features are disabled. Click the respective check boxes to enable the features.
- Click **Submit.**

How to use

This feature requires you to dial your User Password. The default User Password 1111 is not accepted. Please change the User Password first.

Publish Presence can be set for an extension also from the System Administrator mode.

For Extended IP Phone Users

Publishing Presence by Extension User

- Press DSS Key assigned to PUBLISH presence.
  OR
- Dial **104**
- Enter User Password on the prompt.
- Scroll to the desired Publish message from the menu:
  - Absent
  - Present
  - Auto Detect
  - Away
  - On the Phone
  - Do Not Disturb
  - I am Mobile
  - In Meeting
  - Out for Meal
  - Out of Office
- Press Enter key to select message.
- You get the confirmatory tone.

Publishing Presence from SA Mode

- Press DSS Key assigned to PUBLISH presence.
- Enter Destination Number, that is, the number of the extension Publish Presence is to be set.
- Scroll to the desired Publish message from the menu:
  - Absent
  - Present
  - Auto Detect
  - Away
  - On the Phone
  - Do Not Disturb

- I am Mobile
    - In Meeting
    - Out for Meal
    - Out of Office
- Press Enter key to select message.
- You get the confirmatory tone.

To view Presence Status

- Press DSS Key assigned to Display Presence Status.
  OR
- Dial **1097*.***
- Enter Extension number
- The status of the extension number you dialed will be displayed on your phone's LCD.
- Go ON-Hook.

Preset Call Forward

ANANT UCS supports the Preset Call Forward. This feature is useful when Call Forward is not set by users, as their calls will automatically be forwarded to the selected destination. This feature is independent of the Class of Service assigned to the extension users.

Preset Call Forward options can be configured for each time zone by the SE only. The calls will be forwarded to the selected destination— Voicemail, Extension or Department Group as per the Preset Call Forward type selected.

If users set Call Forward from their extensions, it will have a priority over Preset Call Forward. When the users cancel Call Forward from their extensions, the Preset Call Forward option will be applicable automatically.

The Preset Call Forward feature of ANANT UCS offers the following forwarding options:

- **When Busy** - Calls are forwarded to the destination phone number only when the called party's phone is busy.
- **When No Reply** - Calls are forwarded to the destination phone number only when the called party does not answer the phone. The default time is 30 seconds for all extensions and can be changed by customizing the Call Forward No-Reply Timer.
- **When Busy or No Reply** - Calls are forwarded to the destination phone number when the called party's phone is either busy or does not reply.

How it works

A has set Preset Call Forward When No Reply to the Voicemail.

- The system waits for the Call Forward No-Reply Timer to expire and forwards all incoming calls to A’s Voicemail.

A has set Preset Call Forward When Busy to B’s extension.

- The system forwards the call for A to B on detecting Busy signal from A.

B has set Preset Call Forward-No Reply on A and A belongs to a Department Group.

- The Preset Call Forward request will be served and the call will land on A.

If the Ignore call forward set by member extension, when call is routed on Routing/Dept. Group option is enabled in System Parameters, then Preset Call Forward request will not be served. See “System Parameters” for more information.

A has set Preset Call Forward to Department Group.

- The system forwards the call for A to the Department Group. The free member in the group answers the call.

A has set Preset Call Forward When Busy or No Reply to the Voicemail.

- Whenever there is a call for A, if the system does not detect a busy signal from A, it waits for the Call Forward No-Reply timer to expire.
- The system forwards the call to the Voice Mail System.

If A wants to change the Call Forward destination temporarily, then A must set Call Forward from his/her extension. For detailed instructions, see "Call Forward". In this case, the Preset option will not be applicable. But as soon as A cancels Call Forward from his/her extension, the Preset option will be applicable.

- *Preset Call Forward cannot be canceled by the users.*
- *The system supports only single-point Preset Call Forward, which means, if the destination extension has also forwarded its calls, the call will not follow the forwarding path. For example: Calls for extension A are forwarded to extension B. Preset Call Forward is also set on extension B with C as the destination number. In this case, Calls for A will land on B and calls for B will land on C. Calls for A will not land on C.*
- *Only one Preset Call Forward Type can be set for each Time Zone. Every new Preset Call Forward Type set overrides the previous one.*

How to configure

The functioning of this feature is controlled by the following parameters: Preset Call Forward configured in the 'Station Advanced Feature Template' assigned to the extension user and 'Call Forward No-Reply Timer'.

When Preset Call Forward No-Reply is set, if required the Call Forward No-Reply Timer needs to be configured.

Configuring Preset Call Forward in the Station Advanced Feature Template

The default Station Advanced Feature Template 01 is assigned to all extension users of ANANT UCS. In this template Preset Call Forward is disabled.

Decide which extensions are to be allowed 'Preset Call Forward'. If you want to allow Preset Call Forward to all extensions, retain the default Station Advanced Feature Template 01 and configure the Preset parameters. However, if you want to allow Preset Call Forward only to selected extensions, select another Station Advanced Feature Template and configure the Preset parameters in that template.

Now, to assign the template to selected extensions, follow these steps:

1. Prepare a Station Advanced Feature Template with Preset Call Forward configured for each time zone.
2. Assign this newly prepared template to the desired extensions.

Refer the topics "Station Advanced Feature Template" for detailed configuring instructions on how to customize a Station Advanced Feature Template, configure the Preset Call Forward parameters and how to apply this template on extensions.

Call Forward No Reply Timer

When using Call Forward-No Reply, each extension can set a different Time after which the incoming call on the extension should get forwarded when there is no reply from the extension. For this, the Call Forward No-Reply Timer must be configured. By default, this timer is set to 30 seconds.

The Call Forward No-Reply Timer is to be configured in the “Station Advanced Feature Template” applied on the extensions which are allowed Call Forward in their CoS.

If you want to set this timer to the same duration for all extensions, simply set the Call Forward No-Reply Timer in the default Station Advanced Feature Template 01 which is assigned to all extensions.

If you want to set different Timer duration for different extensions, then prepare separate Station Advanced Feature Templates with the desired Timer durations and assign different Templates (with different Timer durations) to the extensions as desired.

Refer the topic “Station Advanced Feature Template” for instructions on customizing the template and applying the template to extensions.

Priority

Priority is the precedence given to certain trunks and extensions over others in being answered by the destination extension.

When Priority is assigned to trunks and if there are incoming calls on multiple trunks at the same time, the call on the trunk with highest priority will be answered by the landing destination extension/Operator first.

When Priority is assigned to Extensions, calls from extensions with highest priority will have precedence in landing on the destination extension.

You can set priority levels from 1 to 9 as given in the table below.

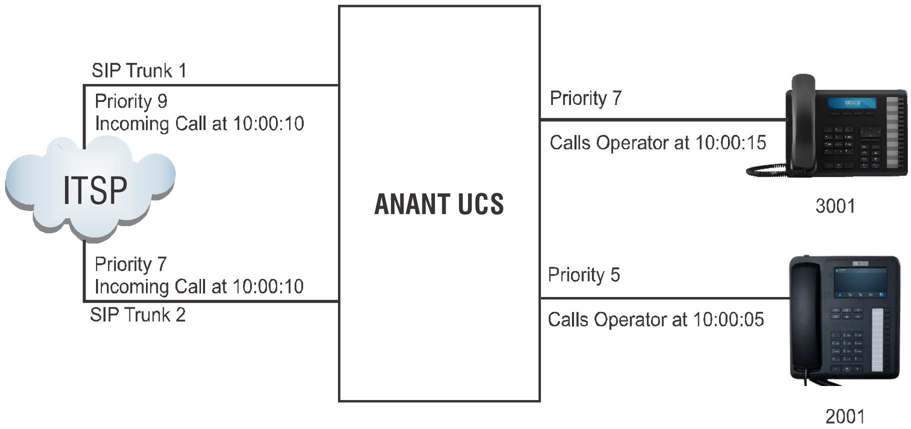
Priority Level	Meaning
1	None
2	Lowest
3	Lower
4	Low
5	Normal
6	Medium
7	High
8	Higher
9	Highest

Highest Priority can be assigned to Extensions of important or higher ranking persons in an organization; for example, calls from senior managers or top executives in an organization can be allowed to be answered first by the destination extension.

Highest Priority can be assigned to particular Trunks, such as special or private trunk lines, trunk lines dedicated as help lines or emergency trunks, or trunks designated as hotlines, so that when there are incoming calls on different trunks at the same time, the call on these trunks gets answered first by the destination extension.

Priority can be assigned to both trunks as well as extensions.

To understand how this feature works, consider this illustration:



Here,

- There are two incoming calls, one on the SIP Trunk 1 and other on the SIP Trunk 2 at the same time, but the SIP Trunks have different priorities set.
- The operator also has incoming calls from extension users which have different priorities set.
- Now, on the Operator extension, which is the landing destination, the incoming calls from the trunks and the extensions will land in the following chronological order:

Caller	Time of the Call	Priority
SIP Extension 2001	10:00:05	5
SIP Trunk 1	10:00:10	9
SIP Trunk 2	10:00:10	7
SIP Extension 3001	10:00:15	7

- These incoming calls, however, will appear on the Display of Operator's phone (Extended IP Phone) in the order of priority:
  - SIP Trunk 1
  - SIP Trunk 2
  - SIP Extension 3001
  - SIP Extension 2001
- Now, when the Operator goes Off-hook (pressing speaker key or picking up the handset), the call on SIP Trunk 1 will be answered first, as SIP Trunk 1 has the highest priority.
- The Operator goes On-hook and then Off-hook, the call on SIP Trunk 2 will be answered. Though SIP Trunk 2 and SIP Extension 3001 have the same priority, '7', SIP Trunk 2 will be answered first, following the chronological order.
- When the Operator goes On-hook after answering the call on SIP Trunk 2, the call from SIP Extension 3001 will be placed on the Operator phone with a *Priority Ring* (configurable; default: Triple Ring).
- Similarly, when the Operator goes On-hook after answering the call from SIP Extension 3001, the call from SIP Extension 2001 will be placed on the Operator phone with a *Priority Ring* (configurable; default: Triple Ring).

- *Priority Ring will be played only if the Extended Phone is SPARSH VP248.*
- *Priority is relevant only when there is more than one call on the destination.*
- *When there are multiple incoming calls, the call with the highest priority will be displayed first on the Operator's screen*

How to configure

To assign Priority to Trunks, you must set the priority in their "Trunk Feature Template". See "Configuring Trunks".

To assign Priority to Extensions, see instructions for configuring the respective Extension port type:

- "Configuring SIP Extensions"
- "Virtual Extension"

If required, you may change the Ring Pattern of the *Priority Ring*. See "Distinctive Rings" for instructions.

Privacy

Extensions of ANANT UCS can be protected from the intrusions by other extensions or from trunk calls by activating Privacy.

How it works

Intrusions can occur on an extension when another extension invokes the following features:

- DND Override
- "Interrupt Request (IR)"
- "Barge-In"
- "Raid"

To prevent such intrusions, ANANT UCS enables you to set the following types of Privacy:

- **Privacy from Interrupt Request, Barge-In, DND Override:** This type of Privacy protects an extension from intrusions by other extensions using Interrupt Request, Barge-In or DND Override.

  For example: Extension A has Privacy from Interrupt Request, Barge-In and DND Override.

  Extension A and B are in speech, Extension C attempts to intrude the conversation Interrupt Request or Barge-In. Extension C's call will be blocked and C will get error tone.

  Now, Extension A has set DND and Extension B attempts to override it using DND Override. Since A has Privacy from DND Override, B's call will be blocked and B will get error tone.

- **Privacy from Raid:** This type of Privacy protects an extension from intrusions by other extensions using Raid.

  For example: This type of Privacy is set on Extension A. Extension A and B are in speech, Extension C uses Raid to intrude the conversation. Extension C's call will be blocked and C will get error tone.

How to configure

By default, Privacy from Raid is enabled in the Class of Service for SIP Extensions. So, none of the extensions can raid the other. You may disable this feature in the Class of Service of extensions, which you want to protect from Raid.

By default, Privacy from Interrupt Request, Barge-In and DND Override are disabled in the Class of Service of all Extensions. You may enable this feature on extensions which you want to protect from intrusions using any of these features.

For instructions, see "Class of Service (CoS)", "Station Basic Feature Template" and "Configuring SIP Extensions".

Quick Dial

Quick Dial provides Extended IP Phone users the facility of 'One-touch' dialing of numbers stored in their Personal Directory and the Global Directory.

How it works

Quick Dial is based on "Abbreviated Dialing".

To be able to Quick Dial a number,

- The number must exist in the Personal or Global Directory assigned to the extension.
- Personal and Global Directory dialing must be allowed in the Class of Service of the extension.
- On the Extended IP Phone, DSS keys must be configured with the Short Codes or Abbreviated Numbers that are to be dialed out. These short codes are derived from the Index numbers of the Personal Directory and the Memory Location Index of the Global Directory.
- You can Quick Dial a number simply by pressing the DSS key.
- The system locates the number to be dialed out in the Personal/Global Directory on the basis of the Index Number/Memory Location Index configured on the DSS Key.

How to configure

See "Abbreviated Dialing" for instructions on configuring and assigning the Personal and Global Directories.

To assign the Short Codes or Abbreviated Numbers to be used for Quick Dial on DSS keys, for each Extended IP Phone extension,

- List down the numbers from the Personal Directory and Global Directory to be used for Quick Dial.
- If the number is from the Personal Directory assigned to the extension, note the Index number at which it is stored in the Personal Directory: 0001 to 0025.
- If the number is from the Global Directory assigned to the extension, note the Memory Location Index at which it is stored in the Global Directory: 0100 to 2999.
- Now, configure the Quick Dial numbers on the DSS keys of the Extended IP Phone.

  For instructions on configuring DSS Keys on Matrix Extended IP Phone, see "Configuring SIP Extension Settings as per the Extended Phone Type" under "Configuring SIP Extensions".
- On the desired key, select **Quick Dial** as **Function Type**.
- As **Offset**, select the Index Number against which the number is stored in the Personal/Global Directory.

How to use

For Extended IP Phone Users

- Press the DSS Key assigned to the Quick Dial numbers.
- The number will be dialed out.
- Talk when the called party answers.

Raid

Raid allows you to interrupt a telephone conversation between two extension users, turning the conversation into a three-way call.

You can use Raid to land in a conversation between two extension users, or between an extension user and an external caller, with a warning beep to the extension user. The extension user will hear a beep when you raid and you will enter in to three-way speech with both parties.

You may also Raid a conversation without any warning by disabling the beep.

How it works

- A, B and C are extension users.
- A and B are in speech.
- C calls A.
- C gets busy tone.
- C dials the feature access code for Raid.
- Beep is played. Three-way speech is established between A, B and C.
- If any of these three parties disconnects, two-way speech is established between the remaining parties.

Feature Interactions

- Raid works only if the dialed extension is busy in two-way speech. The two-way speech may be with another extension or with an external number on a trunk. However, it will not work if the conversation is being recorded.
- You cannot Raid on Trunks, that is, if two external numbers are in two-way speech. In this case, C cannot raid the conversation.
- Raid will not work if **Privacy against Raid** is enabled in the Class of Service of the extension being raided. In this case, if Extension A has Privacy against Raid in its Class of Service, C will not be able to Raid the conversation between A and B. To know more about this feature, see "Privacy".
- The extension using Raid must have higher Priority assigned to it than the extension being raided. In this case, C must have higher Priority than A to be able to invoke Raid.
- Raid will not work when the two-way conversation between the users is being taped.

Raid is a sensitive feature. You are advised to restrict access to this feature to select extension users.

How to configure

To be able to use Raid, extension users must have this feature enabled in the "Class of Service (CoS)" assigned to them for the time zones in their "Station Basic Feature Template".

By default, beep is played as a warning to the extension being raided. If required, you may disable the beep played during Raid, by clearing the **Play Beep when Raid/Call Taping/Conversation Recording starts** check box in the **System Parameters**. For instructions, see "System Parameters".

How to use

For Extended IP Phone Users

When dialed extension is busy,

- Press DSS Key assigned to Raid.
  OR
- Dial **5** on Busy Tone.

RCOC (Return Call to Original Caller)

Generally, extensions users of the System are given a trunk access to make outgoing calls from their phones. It is also common for a group of extensions to share the same trunks to make outgoing calls.

When an extension user of the System makes an outgoing call and the called party does not answer the call or is busy on another line, it is possible for the called party to return the call (made by the extension user) on the basis of the CLI number received.

However, when the called party returns the call, this incoming call is most likely to land on the Operator extension, as incoming calls are usually routed to the Operator.

Now, the Operator has no way of knowing which extension made the call so as to transfer the call to that extension.

Instead, the Operator must either ask the called party whom they wish to speak to and transfer the call or put the called party on hold and find out the extension that made the call. This is an unwieldy process for all concerned - the Operator, the called party and the extension user who originally made the call.

This can be overcome if the System is able to route the returned call to the original caller's extension.

ANANT UCS makes this possible with the Return Call to Original Caller feature.

How it works

The Prerequisites

- RCOC is enabled on the desired Trunk/s
- RCOC is enabled in the Class of Service group assigned to the extension.

RCOC Table

RCOC Table is maintained internally by ANANT UCS and it is non-configurable.

- ANANT UCS can keep a record of 512 entries in the RCOC Table.
- Each entry is kept for the duration of the RCOC Record Delete Timer (configurable; default: 999 minutes). Whenever a record is stored in the RCOC database, the Record Delete Timer for that entry is activated. On the expiry of the Timer, the entry is deleted by the system.
- If a same external number is dialed using 3 different SIP Trunks with RCOC enabled on all the Trunks and Extensions, then if the called party calls back, the call will land on the original callers using FIFO logic.
- Each record is deleted from the database either after the call is returned or on expiry of the Record Delete Timer.
- The RCOC database remains unaffected during power outages.

The Process

- When an extension having RCOC feature in its Class of Service makes an outgoing call, the system checks if RCOC is enabled on the trunk through which the outgoing call is routed.

- If RCOC is enabled on the trunk, the system stores the record of the outgoing call in the RCOC Table.

- The system sets RCOC for the outgoing call in the following conditions,

  - The Destination Port is a SIP Trunk, RCOC is set when:
    - called party is busy.
    - called party does not answer the call.
    - caller (extension that made the call) goes ON-Hook before the called party answers the call.

  - When ANANT UCS acts as Gateway (RCOC),
    - RCOC will be set, if it is enabled on Destination Port.
    - RCOC shall be set only if the Calling Party's Number is available. If calling party number is missing, then RCOC shall not be set.

*While returning call to the original caller, if you want ANANT UCS to match the Trunk Port type and Trunk Port number of the incoming call with the Trunk Parameters of the entry stored in the RCOC table, then you must enable **Apply RCOC only if the caller calls back on the same trunk from which the call was made** option in the System Parameters.*

- Whenever there is an incoming call on any trunk, the system checks the ***Apply RCOC only if the caller calls back on the same trunk from which the call was made*** *check box in the System Parameters.*

  - If enabled, the system matches the Trunk Port number and the Trunk Port type of the incoming call with the entry stored in the RCOC Table.

  - If disabled, the system matches the CLI of the incoming call with the entry stored in the RCOC Table.

- If a matching record entry is found, the system routes the call to the original caller and clears the record entry from the RCOC Table.

- The return call rings on the original caller's extension for the period of the Ring Back Tone Timer (configurable; default 45 seconds). If the original caller does not answer the call within this Timer, the call is routed to the Trunk Landing Group configured for that trunk.

  The Ring Back Tone Timer is common to all internal calls; calls made from one extension will ring on the destination extension till the end of this timer. Change in the Ring Back Tone Timer for RCOC returned calls on original caller's extension will also be applied on Ring Back Tone Timer for all internal calls. So, change this Timer taking this into consideration.

- If no match is found in the RCOC Table or the extension or the original caller is busy, the call will be routed according to the incoming call logic configured (as configured in the assigned Trunk Feature Template) in the system.

- *As RCOC is a "Class of Service (CoS)" dependent feature, extensions that are not allowed this feature in their COS cannot have their calls returned; even if this feature is enabled on the Trunk they used to make the call.*
- *In case of Call Transfer, RCOC will be set for the extension on which the call is transfered.*

Feature Interaction: RCOC and DISA CLI Authentication

- When DISA CLI Authentication (Multiple Calls or One Call) is enabled on a trunk, whenever there is an incoming call on the trunk, the system will first check the DISA CLI Authentication Table.

- If a matching entry is found in the DISA CLI Authentication table, the system will give dial tone to the caller.

- The caller can now invoke RCOC feature by dialing ** (pressing Star key twice).
  OR
- The caller can make calls to a station or an external number or use a feature as required.

- If the caller invokes RCOC feature by dialing ** (pressing Star key twice), the system will check the RCOC Table.

- If a matching record entry is found, the system routes the call to the original caller and clears the record entry from the RCOC Table.

How to configure

For this feature to work, it must be enabled on the Trunk and in the Class of Service of the extensions. See "RCOC on Trunk",and "RCOC in Class of Service" for instructions.

If desired, the related Timers, that is, the RCOC Record Delete Timer and the Ring Back Tone Timer may also be changed. See "System Timers and Counts" for instructions.

If you want ANANT UCS to match the Trunk Port type and Trunk Port number of the incoming call with the Trunk Parameters of the entry stored in the RCOC table, then you must enable **Apply RCOC only if the caller calls back on the same trunk from which the call was made** check box in the System Parameters. See "System Parameters" for instructions.

RCOC on Trunk

Enabling RCOC on Trunk

- Login as System Engineer.

- Under **VoIP Configuration**, click **SIP Trunk Parameters**.

- Click **Advance** at the bottom of the page.

- Click **Outgoing Call** to expand.

  Select the 'Return Call to Original Caller (RCOC)' check box to enable this feature on the desired trunk port.

- Click **Submit**.

RCOC in Class of Service

The feature 'RCOC' must be enabled in the "Class of Service (CoS)" group assigned to the extensions for returned calls to land on them.

In the default Station Basic Feature Template 01 assigned to all extensions of ANANT UCS, the default Class of Service group 01 has the feature "RCOC" enabled. So, all extensions of ANANT UCS are by default allowed this feature.

There is no need to configure this feature if all extensions are to be allowed this feature.

However, if you want to deny this CoS feature to certain extensions and allow this feature to all other extensions, follow these steps:

1. Define a CoS group with RCOC disabled.
2. Prepare a Station Basic Feature Template with this CoS group applicable in all the time zones.
3. Assign this newly prepared Template to the extensions on which 'RCOC' is to be disabled.

Refer the topic “Class of Service (CoS)” and “Station Basic Feature Template” for instructions.

Real Time Clock (RTC)

Various features and facilities supported by ANANT UCS, such as Alarms, Station Message Detail Records, System Activity Log, Time Zones, Daylight Savings, certain Voice Mail features need the correct time and date for their proper functioning.

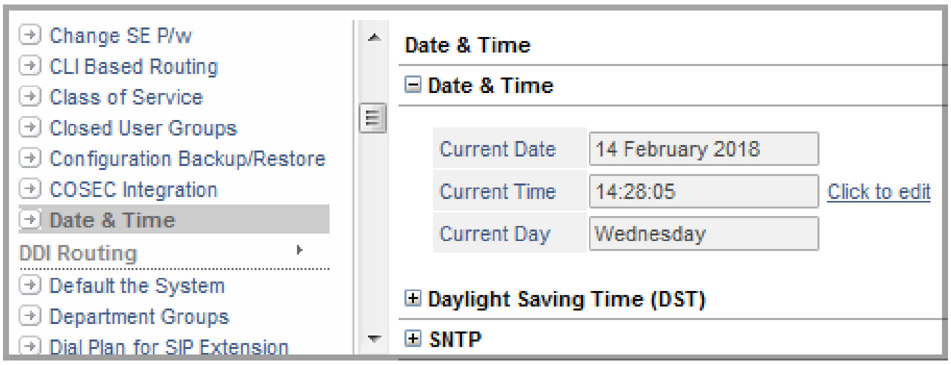
ANANT UCS uses the Real Time Clock (RTC) of the Bare Machine (on which it is installed) to maintains date and time. When you select Region, the RTC is automatically set to the current date and time of the country/region where ANANT UCS is installed.

Since the RTC circuit may drift over a period, it is recommended that you check and reset RTC values at least once every month to correct this drift. The RTC takes care of the leap years.

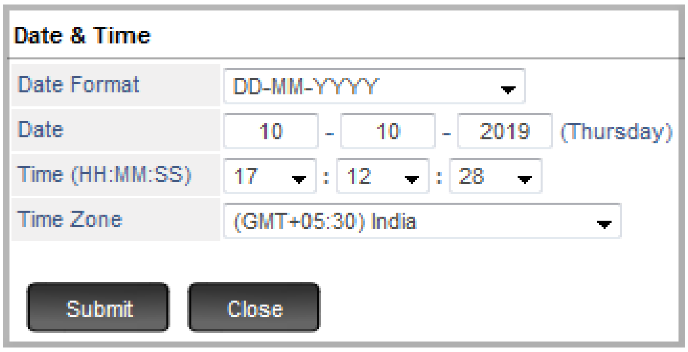
How to Configure

Configuring RTC

- Login as System Engineer.
- Click **Configuration**, click **Date and Time**.
- Click **Date and Time** to expand.



- Click on **Click to edit** to set the date and time as per your requirement.



- Select the desired **Date Format**:
- Enter the current **Date**

- The current **Day** will appear as per the date you set.
- Set the current **Time** in HH:MM:SS (Hours-Minutes-Seconds) format.
- Select **Time Zone** of the country/region where ANANT UCS is installed.
- Click **Submit.**

Time Zones

Time Zone	Index
(GMT-12:00) US Minor Outlying Islands (Baker Island, Howland Island)	135
(GMT-11:00) American Samoa	126
(GMT-10:00) United States (Hawaii)	116
(GMT-09:30) French Polynesia	130
(GMT-09:00) United States (Juneau)	115
(GMT-08:00) United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	114
(GMT-08:00) Mexico (Tijuana)	072
(GMT-08:00) Canada (Vancouver)	031
(GMT-07:00) United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	113
(GMT-07:00) Mexico (Chihuahua)	071
(GMT-07:00) Canada (Calgary)	030
(GMT-06:00) United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	112
(GMT-06:00) Mexico (Mexico City)	070
(GMT-06:00) Costa Rica	035
(GMT-06:00) Canada (Winnipeg)	029
(GMT-05:00) United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	111
(GMT-05:00) Peru	085
(GMT-05:00) Cuba	037
(GMT-05:00) Colombia	034
(GMT-05:00) Canada (Montreal, Ottawa, Toronto)	028
(GMT-05:00) Brazil (Acre)	022
(GMT-05:00) Bahamas	009
(GMT-04:30) Venezuela	118
(GMT-04:00) Paraguay	084
(GMT-04:00) Guyana	047

Time Zone	Index
(GMT-04:00) Chile	032
(GMT-04:00) Canada (Halifax)	027
(GMT-04:00) Brazil (Manaus)	021
(GMT-04:00) Bolivia	015
(GMT-04:00) Antigua & Barbuda	003
(GMT-03:30) Canada (St. John's)	026
(GMT-03:00) Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	020
(GMT-03:00) Argentina	004
(GMT-02:00) Brazil (Fernando De Noronha)	019
(GMT-01:00) Cape Verde (Cabo Verde)	129
(GMT) Ireland	054
(GMT) Portugal	088
(GMT) United Kingdom	110
(GMT+01:00) Algeria	002
(GMT+01:00) Austria	008
(GMT+01:00) Belgium	013
(GMT+01:00) Bosnia & Herzegovina	016
(GMT+01:00) Cameroon	025
(GMT+01:00) Cote d'Ivoire	125
(GMT+01:00) Croatia	036
(GMT+01:00) Czech Republic	039
(GMT+01:00) Denmark	040
(GMT+01:00) France	044
(GMT+01:00) Germany	045
(GMT+01:00) Italy	056
(GMT+01:00) Namibia	076
(GMT+01:00) Netherlands	078
(GMT+01:00) Nigeria	080
(GMT+01:00) Norway	081
(GMT+01:00) Poland	087
(GMT+01:00) Slovakia	095
(GMT+01:00) Spain	097
(GMT+01:00) Sweden	100
(GMT+01:00) Switzerland	101

Time Zone	Index
(GMT+02:00) Belarus	012
(GMT+02:00) Botswana	017
(GMT+02:00) Bulgaria	023
(GMT+02:00) Cyprus	038
(GMT+02:00) Egypt	041
(GMT+02:00) Finland	043
(GMT+02:00) Greece	046
(GMT+02:00) Hungary	049
(GMT+02:00) Israel	055
(GMT+02:00) Jordan	058
(GMT+02:00) Lebanon	065
(GMT+02:00) Libya	066
(GMT+02:00) Mozambique	074
(GMT+02:00) Romania	090
(GMT+02:00) South Africa	096
(GMT+02:00) Syria	102
(GMT+02:00) Turkey	106
(GMT+02:00) Ukraine	108
(GMT+02:00) Yugoslavia	121
(GMT+02:00) Zambia	122
(GMT+02:00) Zimbabwe	123
(GMT+03:00) Yemen	120
(GMT+03:00) Bahrain	010
(GMT+03:00) Iraq	053
(GMT+03:00) Kenya	060
(GMT+03:00) Kuwait	063
(GMT+03:00) Qatar	089
(GMT+03:00) Russia (Moscow, St. Petersburg)	091
(GMT+03:00) Saudi Arabia	124
(GMT+03:00) Sudan	099
(GMT+03:00) Uganda	107
(GMT+03:30) Iran	052
(GMT+04:00) Mauritius	069
(GMT+04:00) Oman	082

Time Zone	Index
(GMT+04:00) United Arab Emirates	109
(GMT+04:30) Afghanistan	001
(GMT+05:00) Maldives	068
(GMT+05:00) Pakistan	083
(GMT+05:00) Tajikistan	104
(GMT+05:00) Uzbekistan	117
(GMT+05:30) Sri Lanka	098
(GMT+05:30) India	050
(GMT+05:45) Nepal	077
(GMT+06:00) Bangladesh	011
(GMT+06:00) Bhutan	014
(GMT+06:00) Kazakhstan	059
(GMT+06:00) Kyrgyzstan	064
(GMT+06:00) Russia (Novosibirsk)	092
(GMT+06:30) Myanmar	075
(GMT+07:00) Cambodia	024
(GMT+07:00) Indonesia	051
(GMT+07:00) Thailand	105
(GMT+07:00) Vietnam	119
(GMT+08:00) Australia (Perth)	005
(GMT+08:00) Brunei	018
(GMT+08:00) China	033
(GMT+08:00) Hong kong	048
(GMT+08:00) Malaysia	067
(GMT+08:00) Mongolia	073
(GMT+08:00) Philippines	086
(GMT+08:00) Singapore	094
(GMT+08:00) Taiwan	103
(GMT+08:45) Australia (Eucla)	127
(GMT+09:00) Japan	057
(GMT+09:00) Korea-North	061
(GMT+09:00) Korea-South	062
(GMT+09:30) Australia (Adelaide)	006
(GMT+10:00) Australia (Brisbane, Canberra, Melbourne, Sydney)	007

Time Zone	Index
(GMT+10:00) Russia (Vladivostok)	093
(GMT+10:30) Australia (Lord Howe Island)	128
(GMT+11:00) Solomon Island)	134
(GMT+12:00) Fiji	042
(GMT+12:00) New Zealand (Auckland, Wellington)	079
(GMT+12:45) New Zealand (Chatham Islands)	132
(GMT+13:00) Samoa	133
(GMT+14:00) Kiribati	131

Redundancy

Redundancy is a process in which upon failure of one System, the other System takes control of the active operation. Redundancy can be used only if you have two systems — Primary System and Backup System connected in the same subnet and you have the *"Redundancy Users License"* activated in one of the systems. The system that takes charge of the operation is known as the Active Server and the other is the Standby Server.

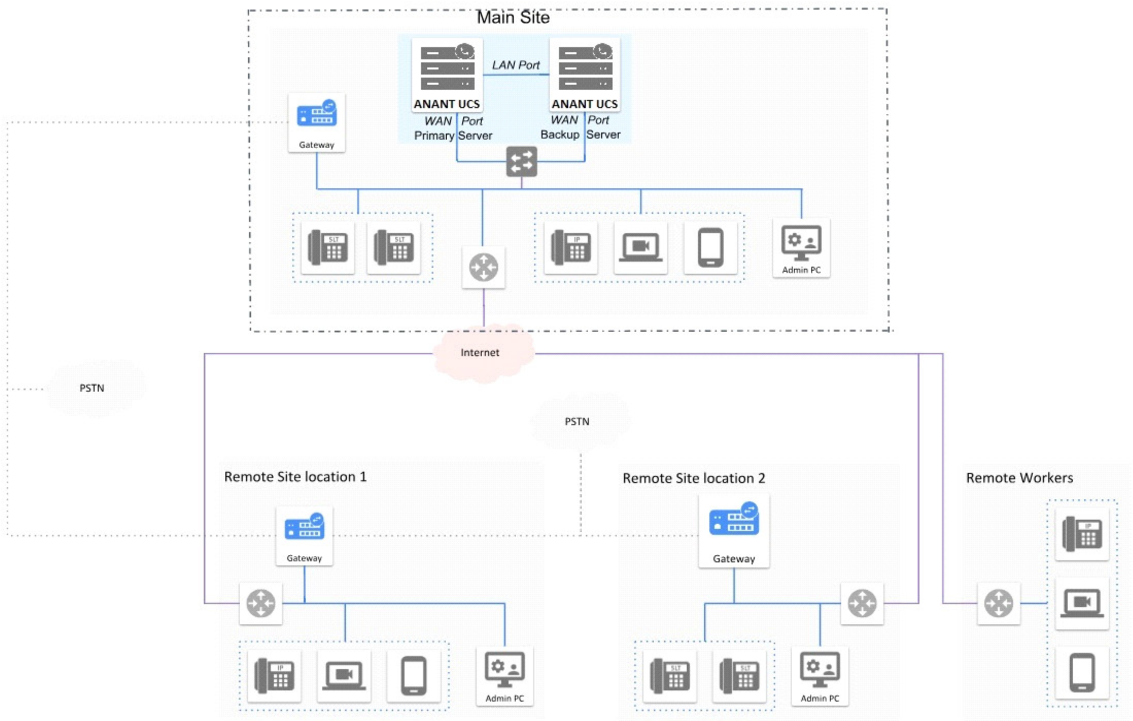
Hence, redundancy feature ensures system's availability even while experiencing technical difficulties or natural calamities.This feature is extremely useful in organizations where persistent network communication link needs to exist. For example — banks, hospitals, defense services etc. Any sudden discontinuation in communication in such organizations may cause unfavorable results.

Redundancy prevents such situations by transferring all the configuration and call information from the Active System to the Standby System. Thus, it reduces the downtime thereby providing high availability and uninterrupted communication.

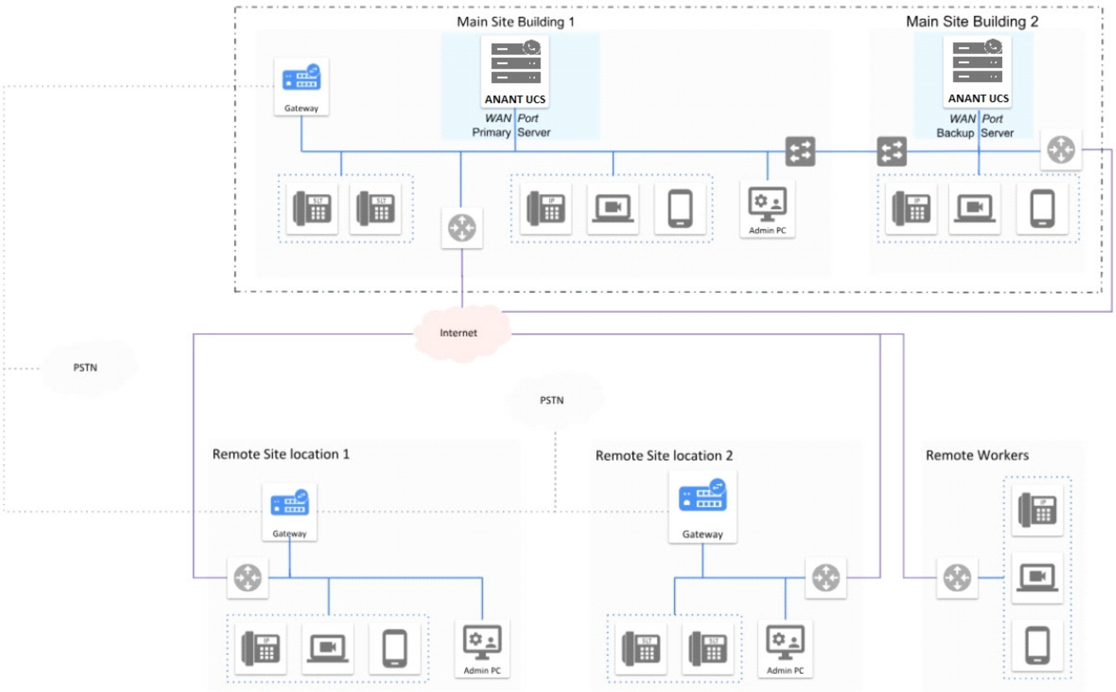
How it works

ANANT UCS supports **Redundancy in the same subnet** in two ways, over LAN port or over WAN port. However, it is highly recommended to use the LAN for better bandwidth utilization and avoiding network congestion.

The following example illustrates how Redundancy is applied over LAN port of networks.



The following example illustrates how Redundancy is applied over WAN port of networks.



Let us understand Redundancy over the LAN Port:

- Here, one server is the Primary Server (with Redundancy and other licenses activated), while the other server is the Backup Server.
- The Primary Server manages the communication, whereas the Backup Server is synchronized and ready to transit to active state. The Backup Server continuously monitors the health of the Primary Server through the LAN port. In this case, the Primary Server is Active whereas the Backup Server is Standby. All the configuration and call information is transfered from the Active to Standby Server through this port.
- A Virtual IP is essentially a 'floating' IP address that is not linked physically with any one server. It keeps on switching between the two servers based on their availability, that is, it is the IP Address of the Active Server. When the Primary Server fails, the Backup Server takes control of the functionality through a Virtual IP. Thus, mature calls stay in progress even when the server in active state gets switched to standby and vice-versa.
- The client devices — IP Phones, VARTA UC Clients — communicate with the Primary server through the WAN port. Make sure the Virtual IP Address is configured in all the clients.

Redundancy Users License

Redundancy Users License is required to support redundancy between the two server — Primary Server and Backup Server. Make sure Redundancy Users license is activated in the Primary Server in which you have the other licenses activated as well. To activate the Redundancy Users License, refer to *"Licenses Supported in ANANT UCS"*.

Redundancy Users License activated in the Primary Server is shared between the Primary and Backup Server. Thus, allowing you to use all the licensed features and functionalities that were supported by the Primary Server during Redundancy. Hence, you need not purchase a separate Redundancy license for the Backup Server.

To view or manage the Redundancy License,

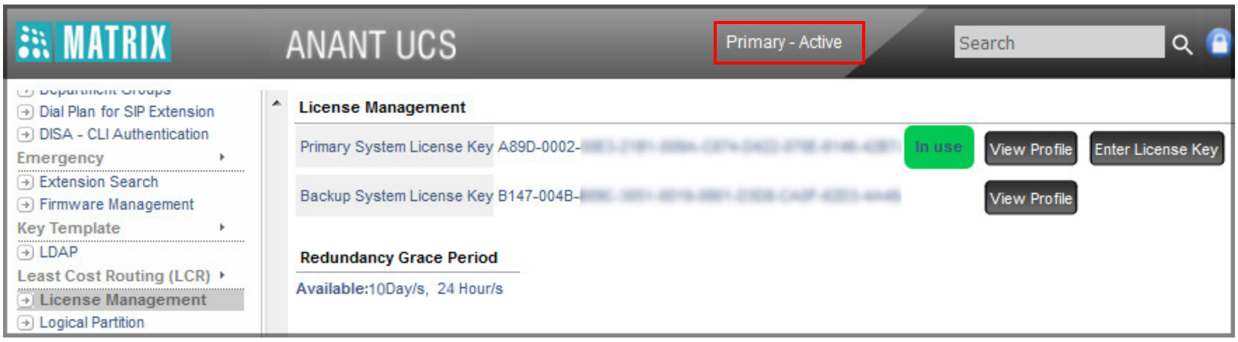
- Initially, configure the redundancy parameters in Primary Server and Backup Server. To know more, refer to "Configuring Redundancy Parameters".

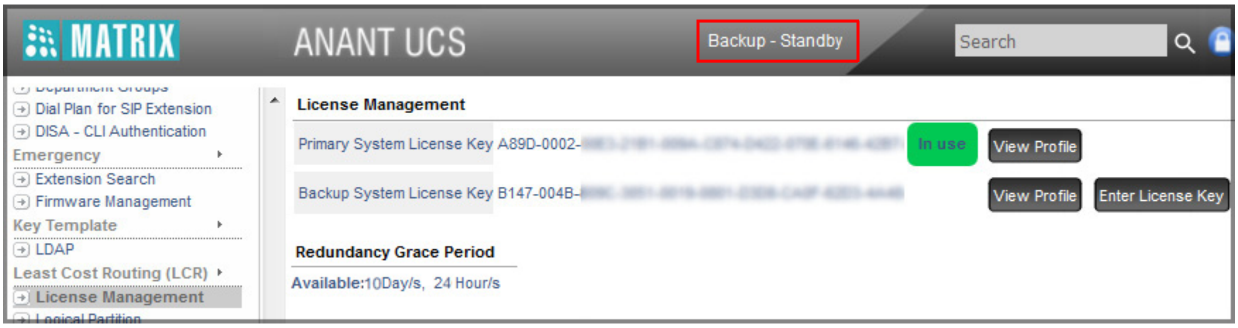
- Login as System Engineer.

- Under **Configuration**, click **License Management**.

  The License Management page opens.

  Here, the Primary System is in the Active state and the Backup System is in the Standby state.





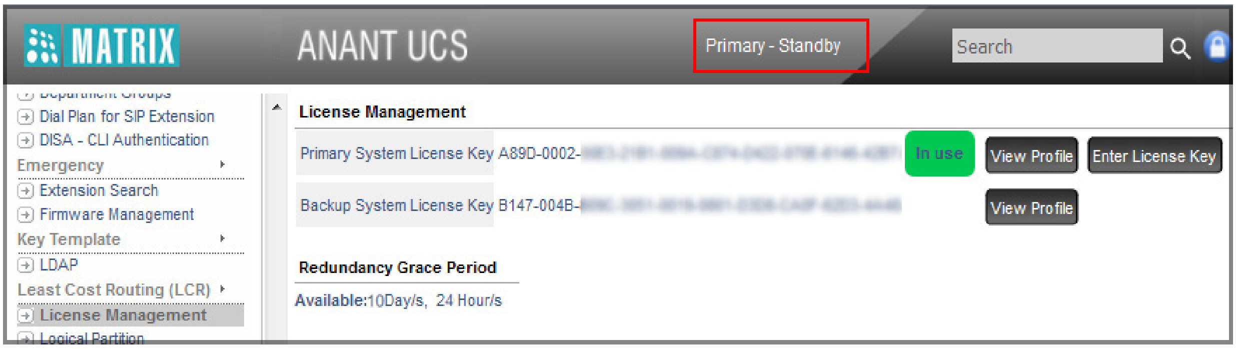
To view the list of licenses activated in Primary System, click the **View Profile** button of Primary System. Similarly, you can also view the default licenses present in the Backup System. To do so, click the **View Profile** button of Backup System.

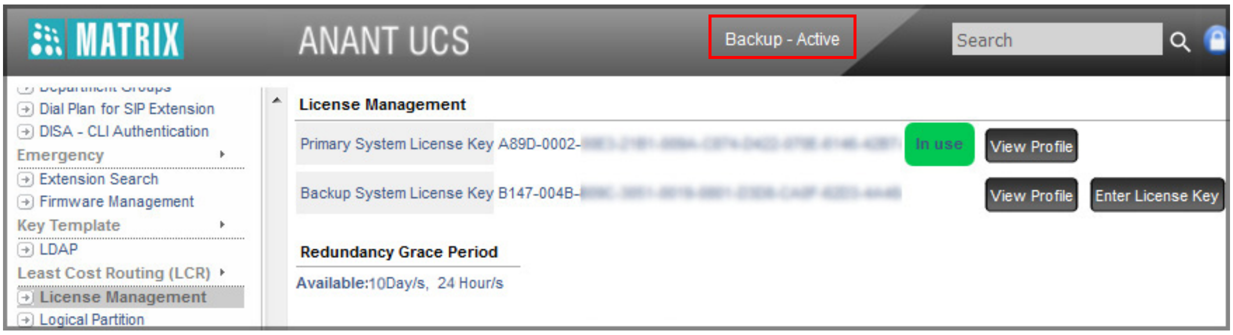
The 'In Use' tag indicates the license key of the System which is currently being used. This implies the system will use the licenses as activated in this license key.

You can also compare the licenses activated in both the systems, to do so, click the **View Profile** button of any desired system and click the **Compare All Licenses** check box. Compare All Licenses check box will function only when the License Dongle is connected in both the servers.

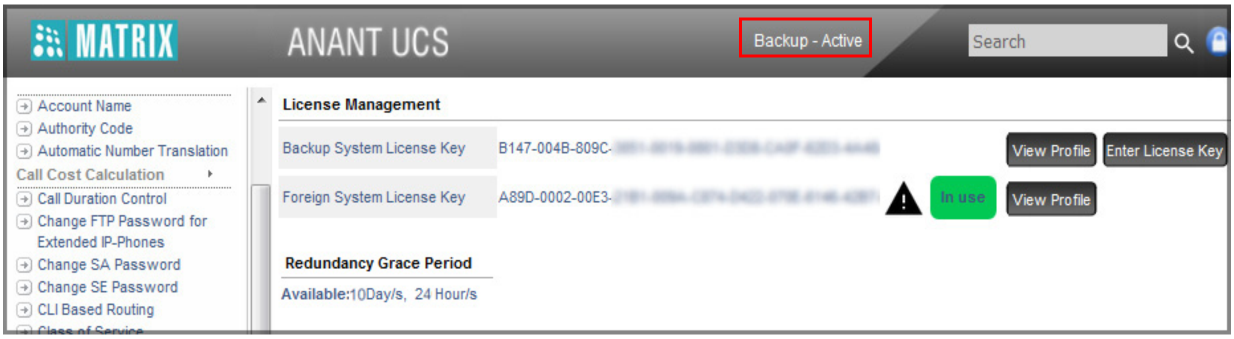
- The Redundancy User License is activated in the Primary System only and the Backup System does not have any license activated.

Case 1: Primary Server with Redundancy License fails
Backup Server is in Active state.





Case 2: Primary Server with Redundancy License Fails and this system is not available (cannot be detected or is sent for repair)
Backup Server is in Active state.



In this case, the Backup System becomes active and the **Redundancy Grace Period** starts. During this process, the Primary System's License Key becomes the **Foreign System License Key.**

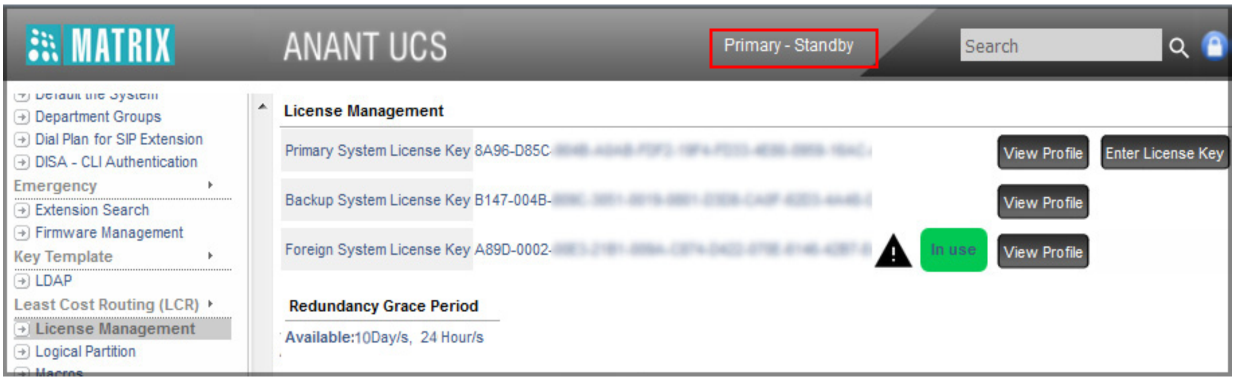
The **Redundancy Grace Period** allows Backup System to use all the licensed features of Primary System free of cost for limited duration of 10 days.

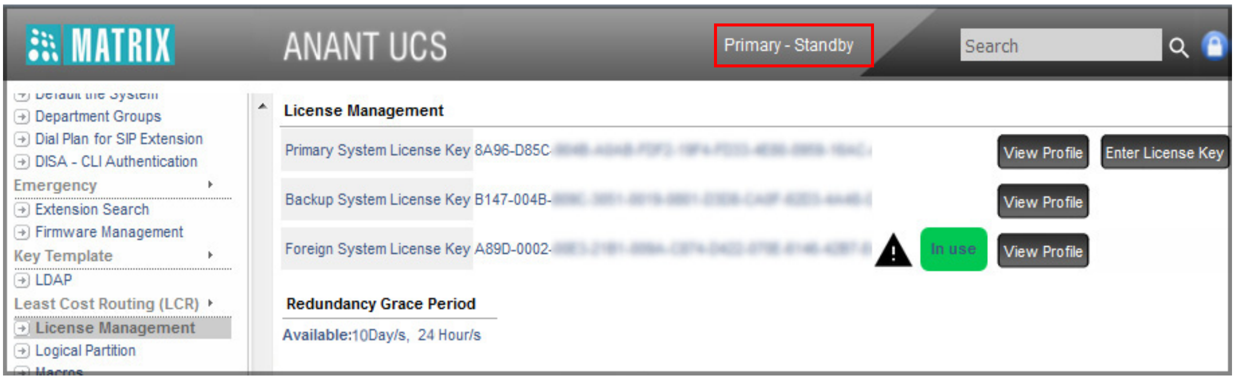
To support Redundancy during the Redundancy Grace Period,

- Install another System (New Primary System) as the Primary System and configure its parameters. For detailed information, refer to "Configuring Redundancy Parameters".
- To use Redundancy make sure you connect the LAN/WAN port of the systems to communicate with each other in the network as per your installation scenario.

- After you connect New Primary System in the network, click **License Management.**

  The License Management opens.





- In this case, New Primary System is in standby state whereas Backup System is in active state.

  Both the systems use the **Foreign System License Key** (This is the Primary System's License Key).

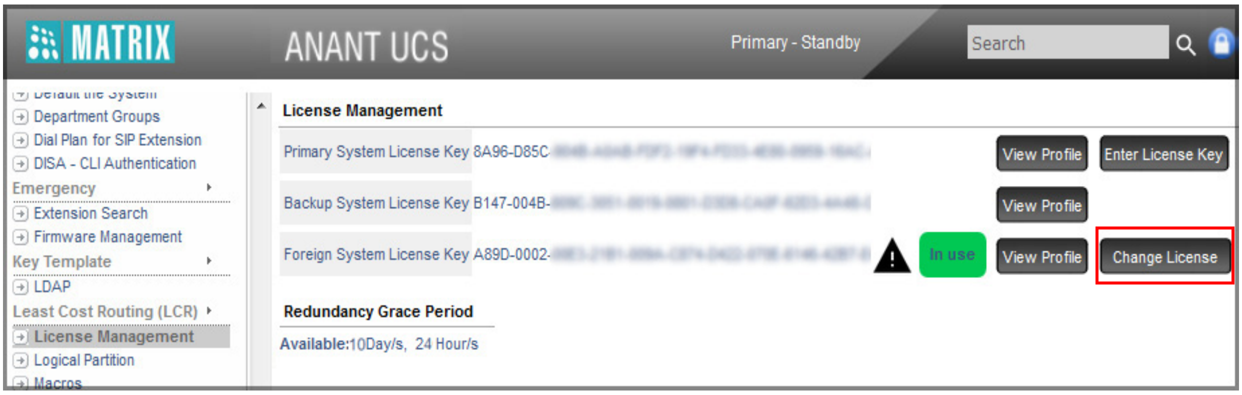
- *If you want to use Redundancy license feature before the expiry of the Redundancy Grace Period, you must purchase the Redundancy license key and activate it in either — New Primary or Backup System.*
- *If the Primary System is repaired and sent back, then for Firmware Version later than V2.2 the Grace period will be incremented till it reaches 90 days for both the systems.*

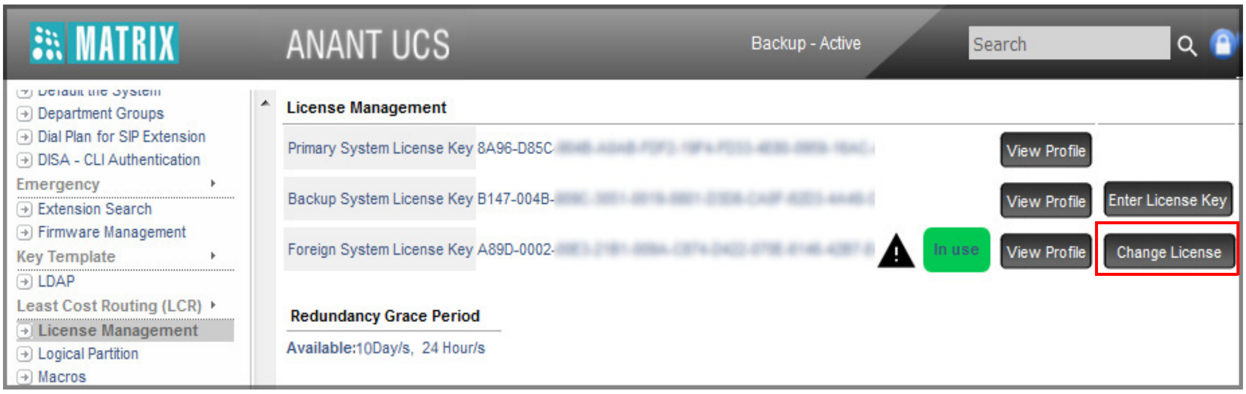
Assume that you have activated Redundancy License in the New Primary System,

Now, there are two Redundancy Licenses available — Redundancy License of the new Primary System and Foreign System License Key. You can opt to change the license you wish to use.

To change the Redundancy License,

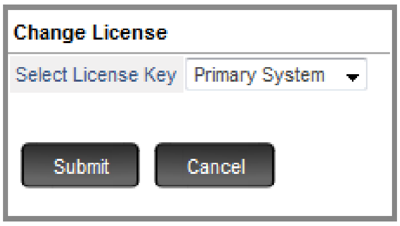
Click **License Management.** The **License Management** page opens.





It is recommended to change the license before the expiry of Redundancy Grace Period to prevent Redundancy from getting disabled.

- Click **Change License.** A new window opens.



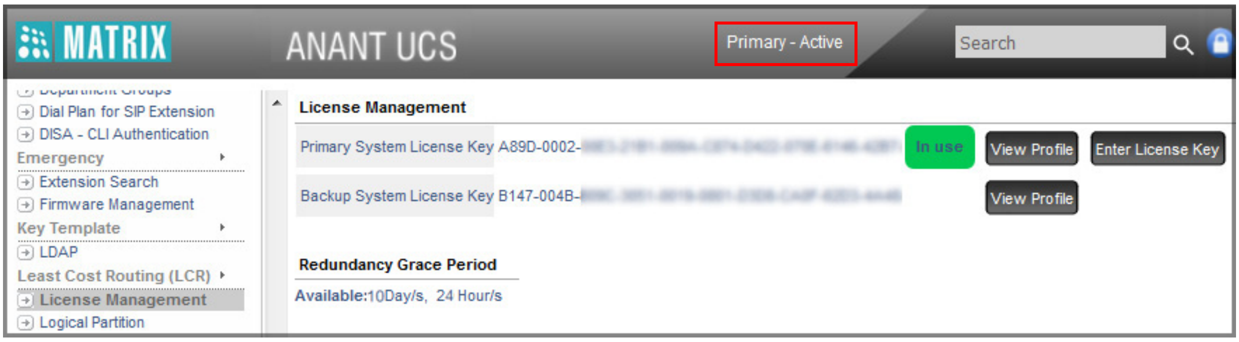
- Select the desired option displayed in **Select License Key.**

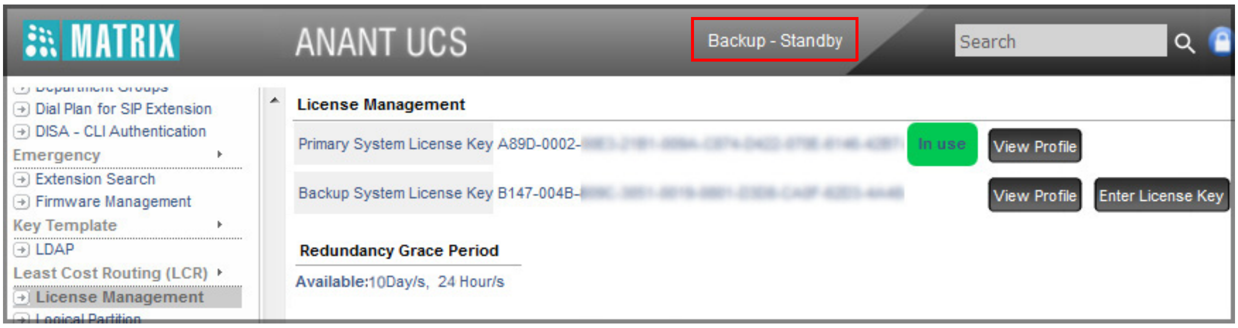
  In this case, the Foreign System License Key is being used. Hence, it displays the other available option - **Primary System**.

- Click **Submit.**

Once you have changed the license,

The **License Management** page is updated.





After the redundancy process is completed, ANANT UCS intimates you about the same via a redundancy notification call. To know how to receive the notification call, refer to “Redundancy Notification Call”.

Redundancy Notification Call

When the active server fails, the standby server takes over the control and becomes active. This process of restoring the functionality of the server by transferring the active operation to the standby server is known as redundancy. To know more, see “Redundancy”.

As redundancy is an important event, the intimation of the same can be provided to the concerned entities in an organization via an auto-generated notification call, known as Redundancy Notification Call.

This redundancy notification call is generated by the server and sent to the desired landing destination/s, after the redundancy process is completed.

You can either select a dedicated SIP extension or a routing group as the landing destination for the redundancy notification call.

You can also define different landing destinations for each time zone, that is Working Hours (WH), Break Hours (BH) and Non-working Hours (NH) as per your requirement.

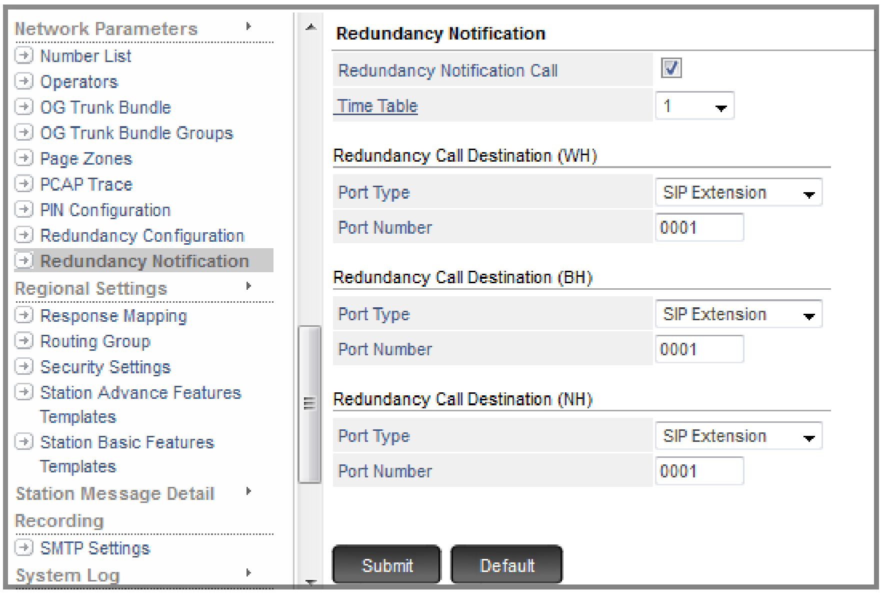
How it works

- After the redundancy process is completed, the extension configured to receive the redundancy notification call rings[110], displaying the message “Redundancy Notification Call” on the LCD screen.
- When the extension user answers the redundancy notification call, s/he is played a piece of music or a voice message as uploaded in MoH management. To know more, refer to “MoH Management”.
- The extension user can now acknowledge the redundancy notification call. The acknowledged redundancy notification call is logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)” with the details of the extension that acknowledged the call.
- If the extension configured to receive the redundancy notification call is busy, then the system will place the notification call only after the extension becomes idle.
- If the extension user does not answer or reject or answer but does not acknowledge the redundancy notification call till the expiry of the Ring Back Tone Timer, then the system makes one more attempt after an interval of 10 minutes and place the notification call on the landing destination again. Now, if the redundancy notification call is not acknowledged, then the notification call is logged in the “System Activity Log” as well as “Simple Network Management Protocol (SNMP)”.

110. *The extension rings for the duration of the Ring Back Tone Timer (configurable, default:45 minutes). See “System Timers and Counts”.*

How to configure

- Login as System Engineer.
- Under **Configuration**, click **Redundancy Notification**.



- **Redundancy Notification Call:** Enable this check box if you want the system to place the Redundancy Notification Call on extension/s after the redundancy process is completed. By default, this check box is enabled.
- **Time Table:** A Time Table is a schedule for the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours for a week

  You can define and select the Time Table for the Redundancy Notification Call as per you requirement. There are 8 different Time Table templates to select from. By default, the Time Table 1 is selected.

  In Time Table 1, six days of the week - Monday to Saturday - have working hours from 9:00-18:00, break hours from 13:00-14:00 hours and non-working hours from 18:00 to 09:00. Sunday is a holiday, with all three Time Zones set to 00:00 hours.

  You may also customize the default Time Table 1 OR customize and assign a different Time Table as per your requirement. Refer to “Time Tables” for more details.

  You can select different landing destinations for each time zone, that is Working Hours (WH), Break Hours (BH) and Non-working Hours (NH) as per your requirement.

  For example, redundancy notification call can be routed to the Operator’s extension during working hours and the security personnel extension’s during the non-working hours.
- **Port Type:** Select the landing destination on which you want to place the Redundancy Notification Call. It may be a dedicated SIP extension or a Routing Group.

  If you select routing group as the landing destination for the redundancy notification call, then you must configure a routing group. To know how to create a routing group, refer to “Routing Group”.

By default, SIP Extension is selected as the landing destination for all the three time zones.

- *This page displays only those Extension Port Type that are pre defined in the System Pre-requisites page. For instance, if the value selected for **SIP Extensions** under **Number of Ports Used** is zero in the **System Pre-requisites** page, then SIP Extension will not be displayed as one of the Port Type. To know more, refer to “Configuring System Pre-requisites”.*

- *The Redundancy Notification Call will be displayed in the call logs, if you have configured the UC Clients (VARTA ADR100/ VARTA AMP100/ VARTA WIN200) or SPARSH VP330 or SPARSH VP210 as the landing destination. However, the notification call will not be displayed in the call logs, if you have configured, SPARSH VP310/ SPARSH VP510 as the landing destination.*

- *Redundancy Notification Call is not supported on SPARSH VP248.*

- *Redundancy notification call will not be applicable, if you select Voice Mail Auto Attendant or OTBG as the members in the Routing Group.*

- **Port Number:** Enter the port number (software port or routing group number) on which the landing destination is configured.

  If you have selected *SIP Extension* as the Port Type, enter the software port number of the SIP Extension.

  If you have selected *Routing Group* as the Port Type, enter the routing group number (01 to 96) in this field.

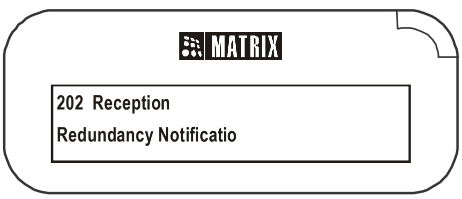
  Similarly, you can configure the landing destination for the Break Hours and the Non-Working hours as per your requirement.

- Click **Submit**.

- *The Redundancy Notification call is generated 5 minutes after the redundancy process is completed.*
- *If you have set DND or Call Forward on a extension that is configured to receive the redundancy notification call, then in such case, the system will override these features and will place the redundancy notification call on the extension.*

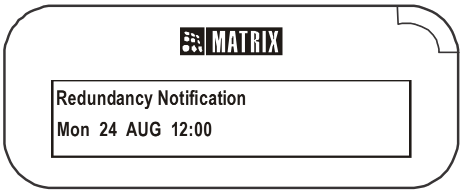
Acknowledging Redundancy Notification Call from SPARSH VP310 Extended IP Phone

- You will receive a Redundancy Notification Call, after the redundancy process is completed.

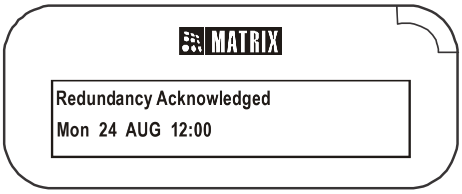


- Lift the Handset/ press the Speaker Key/ press the Headset Key/ CA Key to answer the Redundancy Notification Call.

- You will hear a piece of music or a voice message.



- Press the digits (**0-9, * or #**) to acknowledge the Redundancy Notification Call.



- The Redundancy Notification Call is acknowledged.

Refer the respective User guide of the Extended IP Phones and UC Clients, to know how to acknowledge the Redundancy Notification Call.

Even though ANANT UCS supports Redundancy Notification Call, it is advisable to enable the Email Notification for the system related activities and faults, so that, in case you miss the notification call, you still are notified about the redundancy process via an Email. To know how to enable the Email Notifications, refer to "System Log Notification".

Reminder

Reminders are a variation of the “Alarms” feature, requiring the Date and Time to be set for each Reminder call.

Reminder calls are useful for extension users who wish to be reminded of important tasks or appointments.

For Reminder calls, date and time are set in the following format:

Date is set, according to Date Format you selected in the “Real Time Clock (RTC)” parameters, as:

- Day-Month-Year (DD:MM:YYYY)
  Or
  Month-Date-Year (MM:DD:YYYY).
- Reminders can be set and canceled by:
  - the Operator from Jeeves.
  - extension users from their phones.
- Multiple Reminders can be set for an extension by the Operator and/or by the extension user.
- The mechanism for serving Reminders calls can be configured as 'Personalized' or 'Automated'.
- Reminders can be voice-guided.
- ANANT UCS can register as many as 999 Reminders set by the Operator and extension users.

How it works

Personalized Reminder

When the Reminder call serving mechanism is configured as 'Personalized',

- The Operator Phone rings first[111], displaying the number of the extension to which the reminder call is to be served.
- When the Operator answers this call, a call is placed on the extension on which the reminder call is set.
- The extension phone rings for the duration of the Alarm Ring Timer.
- When the extension user answers the call, the Operator greets the extension user with the reminder message.
- If the extension user does not answer the call till the *Alarm Ring Timer* has elapsed, the Operator phone will display a text message notifying 'No Reply' from the extension. The Reminder is now considered as served.

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111. The Operator phone rings for the duration of the Alarm Ring Timer. If the Operator does not answer the call, the ANANT UCS will make two more Alarm Attempts at an Alarm Attempt Interval of 5 minutes to call the Operator.

- If the extension is busy, the Operator phone will display a text message notifying that the extension number is 'Busy'.

- The Operator can now choose to

  - inform the extension user about the Reminder in person or send someone to do it.
    OR
  - try the busy extension again.
    OR
  - set “Auto Call Back (ACB)”.

Personal Reminders will work even if the extension user has set DND or Call Forward.

Automated Reminder

When the Alarm serving mechanism is configured as 'Automated',

- The extension phone rings at the set time till the end of the Alarm Ring Timer. If the extension phone is the Matrix Extended IP Phone, Reminder message will appear on its display.

- When the extension user answers the call, the user may be played music-on-hold, or be connected to the Voice Mail, or routed to the Operator, depending upon the Alarm Notification Type you have configured for the extension.

- If the extension user does not answer the reminder call, the ANANT UCS makes two more attempts (in all, 3 attempts) at an interval of 5 minutes between each attempt, to call the extension.

- If all Reminder call attempts go unanswered, the ANANT UCS places the call on the Operator Phone. The Operator Phone rings till the end of the Alarm Ring Timer. The Operator Phone displays the number of the extension with the message 'No Reply'. The Reminder call is now considered as served.

- If the extension phone is busy, the ANANT UCS will continue to make the Reminder call Attempts at the Alarm Interval configured. When all Alarm Attempts go unanswered, ANANT UCS will place a call on the Operator phone. The Operator Phone will display the number of the extension phone with the message 'Busy'.

Snooze

The Snooze function can be added to Automated Reminders to ensure that the extension user answers the call. Snooze is a system-wide feature; when set, this function will be added to all Automated Reminder calls.

When Snooze is activated,

- The extension phone rings for the Number of Alarm Attempt configured, at the set Alarm Attempt Interval.

- The extension stops ringing when the user answers the call and dials **0** to acknowledge the Reminder call. This reminder call Acknowledgment Code **0** is non-configurable.

- *Reminder can be set for Operator phones also.*
- *Reminder settings will be retained in the system during power down and system upgrades.*
- *When multiple reminder requests have been set by an extension user, the extension user cannot selectively cancel a particular reminder request. Only the Operator can selectively cancel Reminders set for an extension user from the System Administrator pages of Jeeves.*
- *It is not possible to modify—change the date and time—of a reminder request. So, you may cancel the Reminder request and set a new one*.
- *Consider you have set a reminder with snooze enabled and Number of Alarm Attempts set as three (configurable). If this reminder call is not acknowledged by the extension user at the first attempt and due to some reason, the system restarts, then the pending two attempts will not be served. However, this reminder will be displayed under the pending reminder list.*

Reminder Status Report

The Operator can know the details of Reminders that have not been served on the *Reminder* page of Jeeves, from the System Administrator (SA) mode or by pressing the DSS key assigned to Wakeup Call Log. To view the log using DSS key, see “Alarm Status Report”.

The status of Reminders set by Operator as well as extension users appears on this page, with details of time (hours and minutes), and serving mechanism (personalized, automated).

The Operator can view the Reminder report whenever required and can also print this report.

How to configure

The configuration of Reminders is the same as *Alarms*.

To configure Reminders feature, do the following:

- Select the **Alarm Notification Type** for the extensions.
- Configure, as required, the Alarm Call related parameters: **Alarm Ring Timer**, **Number of Attempts**, **Alarm Attempt Interval**, **Configurable Alarm Type** and **Configurable Alarm Category**, and **Snooze**.

  For instructions, see the topic “How to configure” under “Alarms”.

How to use

Reminders can be set by the extension users by themselves. The extension users can also ask the Operator to set the Reminder for them.

ANANT UCS offers voice-guided Reminders using VMS to extension users and the Operator.

Voice-guided reminders lead users through a menu, helping them set the alarm in a step-by-step manner.

Voice-Guided Reminders set/canceled by Operator

The Operator can set voice-guided Reminders for extension users

For Extended IP Phones

- Press DSS Key assigned to 'Remote Voice-Guided Reminder' function
- Follow Voice Mail System Prompts to set/cancel Reminder.

Voice-Guided Reminder set/canceled by Extension Users

For Extended IP Phone Users

- Press the key assigned to Voice-guided 'Reminder' function.
  OR
- Dial **164**
- Follow Voice Mail System prompts.

Non-Voice Guided Reminders set/canceled by Operator

For Extended IP Phones

To set Reminder for the extension user,

- Press the DSS Key assigned the 'Remote Reminder' function.
- Enter the Extension Number.
- Enter Date and Time in the format:
  DD: MM: YYYY: HH: MM
  OR
  MM: DD: YYYY: HH: MM (users in USA)
- Select 'Personalized' or 'Automated'.
- Press 'Enter' key to set Reminder.
- You get a confirmation tone and message.

To cancel Reminder set for the extension user,

- Press key assigned for ‘Remote Reminder’ function.
- Enter Extension Number.
- Select 'Cancel All'.
- Press 'Enter' Key.

- *To cancel reminder calls selectively, go to 'Reminder Status' page from the System Administrator of Jeeves.*
- *If the 'Configurable Alarm Category' check box is enabled, only then the system will give the option of selecting 'Automated' or 'Personalized' as the serving mechanism. By default the serving mechanism is Automated.*

Non-Voice Guided Reminders set/cancel by Extension Users

For Extended IP Phone Users

To set Reminder:

- Press the 'Reminder' key.
  OR
- Dial **162**
- Enter Date and Time in the format
  DD:MM:YYYY:HH:MM
  OR
  MM:DD:YYYY:HH:MM (users in USA)
- Press 'Enter' key.
- You get a confirmatory text message and confirmation tone.
- Go Idle.

To cancel Reminder:

- Press 'Reminder' Key again.
  OR
- Dial **162**
- Select 'Cancel All'.
- Press Enter Key.

Viewing and Printing Reminder Status

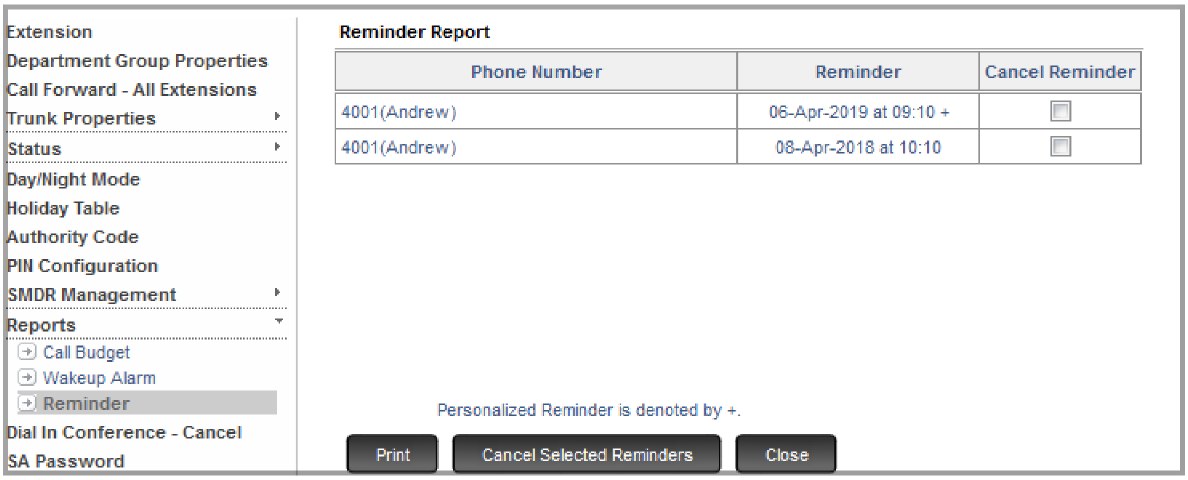
The Operator can view the status of Reminders that are yet to be served from the System Administrator pages of Jeeves.

To view Reminder Status from the System Administrator pages of Jeeves,

- Login as System Administrator.

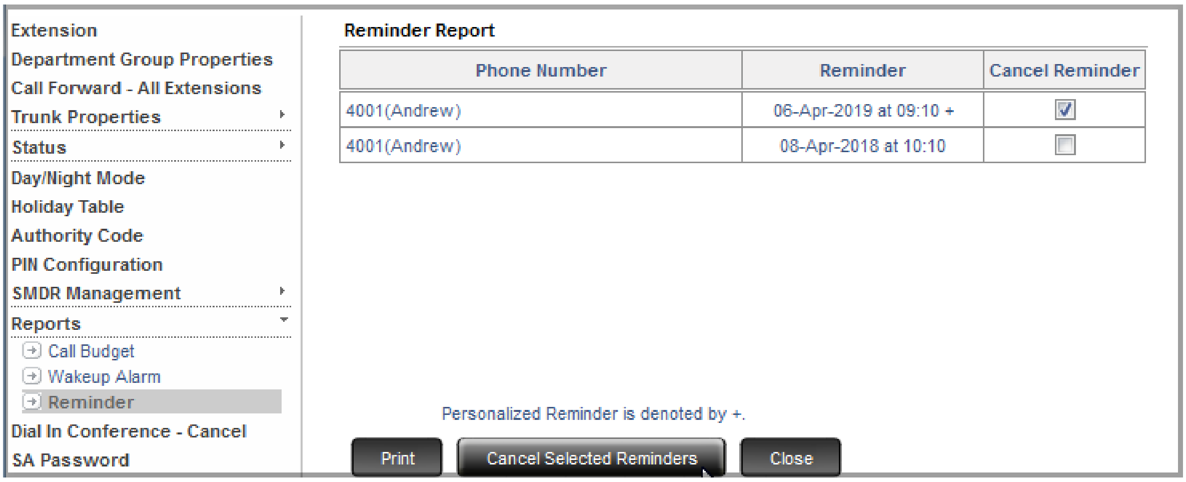
- Click the **Reports**.

- Under Reports, click the **Reminder**.



The unserved Reminders appear on the page.

- To cancel any of the unserved Reminders,
  - Select the **Cancel Reminder** check box of the extension number for which you want to cancel the reminder.



  - Click the **Cancel Selected Reminders** button at the bottom of the page.
- To print this page, click the **Print** button.
- Click **Close** to exit the page.

Room Monitor

This feature enables the Extended IP Phone Extension users to listen to the conversations taking place in another location where an Extended IP Phone is present.

Room Monitor can be used to monitor activities on the Shop Floors / Manufacturing areas from another location.

- *Use this feature in accordance with the local privacy laws.*
- *Matrix Comsec is not responsible for any mis-/abuse of this feature by users.*

How it works

- A is a supervisor in a Manufacturing unit.
- A's room is on the second floor. The manufacturing area is on the ground floor.
- To keep track of the activities in the plant on the ground floor, there must be an Extended IP Phone at the place where the activities are to be monitored, and A's extension must have higher "Priority" than the extension at the monitored location.
- If there is an Extended IP Phone at the desired location, A can activate Room Monitor.

  A can activate Room Monitor only if the Extended IP Phone at the desired location is idle.
- When A activates Room Monitor, the microphone of the Extended IP Phone on the ground floor goes Off-hook. A can now hear all the sounds taking place on the ground floor, without anyone present there coming to know that they are being monitored.
- To end Room Monitor, A must disconnect.
- Room monitoring will be terminated on the Extended IP Phone on the ground floor, if someone lifts the handset of this phone or if there is a call on this phone from another extension.

You can activate Room Monitor from any extension port type, but the extension being monitored must be an Extended IP Phone.

How to configure

To be able to use Room Monitor, extension users must have this feature enabled in the "Class of Service (CoS)" in the "Station Basic Feature Template" assigned to their extensions.

How to use

For Extended IP Phone Users

To enable Room Monitor on an extension,

- Press the DSS Key assigned to Room Monitor.
- Dial Extension Number to be monitored.

OR

- Dial **1073.**
- Enter Extension Number that you want to monitor.

Routing Group

Routing Group is a group of extensions used for landing incoming calls as a Trunk Landing Group, as Alarm Notification Routing Group, as Floor Service Group and as Department Group

How it works

ANANT UCS supports the formation of 96 Routing Groups. In each group you can have upto 32 members.

The *member* of a Routing Group can be SIP Extensions, Virtual Extensions, Voice Mail Auto Attendant and Outgoing Trunk Bundle Group.

These groups can be used:

- as Trunk Landing Groups to route incoming calls.
- as Alarm Notification Routing Groups to server Alarm Notifications.
- as Floor Service Groups to provide Floor Service.
- as Department Groups to route incoming calls to a particular department.

This is how a Routing Group works,

- There is an incoming call on SIP Trunk1.
- Routing Group 1 is assigned as the Trunk Landing Group for SIP Trunk1. The Routing Group has extensions 2001, 2002 and 2003 as landing destinations.
- By default incoming calls will be placed on the members in rotation, that is first call on 2001, second call on 2002 and so on.

  If you want incoming calls to be placed on extension 2001 always, you must disable Rotation.
- By default, an incoming call will be placed on 2001. 2001 rings for the duration of the Ring Timer, if the call is unanswered the system re-directs the call to 2002 and so on, till the call is answered.

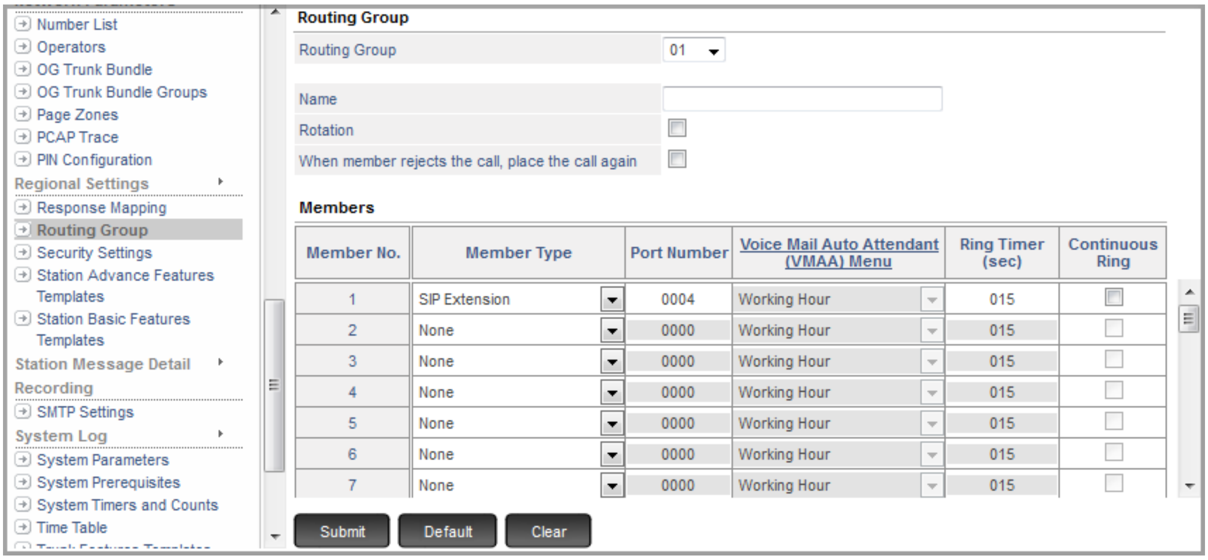
  If you want all the extensions to ring continuously till the call is answered by any member, you must enable *Continuous* Ring. 2001 will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to 2001 once again.
- In this way the system places the incoming calls on the member extensions in a Routing Group till the call is answered.

If you have selected Voice Mail Auto Attendant or OTBG as members in a Routing Group, the parameters Ring Timer and Continuous Ring are not applicable.

How to configure

Configuring Routing Groups

- Login as System Engineer.
- Under **Configuration**, click **Routing Group**.



- You can create 96 Routing Groups with 32 members in each group.
- Select a Routing Group Number from **1 to 96.**
- For each Group you must configure the following:
  - Configure the **Name** you wish to assign to the Routing Group.
  - Select the **Rotation** check box to enable rotation of calls in the routing group which has multiple 'member' extensions. When enabled, each fresh call will land on the extension which is next to the one that received the last call. This ensures equal distribution of incoming calls to all the destinations within the routing group. The option is not relevant if the routing group has only one member extension.
  - If any member extension rejects an incoming call and the system again checks the routing group sequence, you can allow or restrict placing the same call on this extension. Select the **When member rejects the call, place the call** again check box**,** if you want the system to place the call again on the extension.
  - To configure **Members** in the Group,
    - Select the **Member Type**. You can select SIP Extension, Virtual Extensions, OTBG or the Voice Mail Auto Attendant.

      Configure only as many extensions as you want in the routing group and set the remaining Member Types to 'None'.

      For example: if you want to program only one extension in the routing group, set the Member Type in the remaining columns (Member 02-Member 32) to 'None'.

- In **Port Number**, enter the software port number on which the SIP or Virtual Extension is connected.

  If you have selected OTBG then enter the OTBG number here.

- If you have selected the *Voice Mail Auto Attendant* as the Port Type, select the **Voice Mail Auto Attendant (VMAA) Menu** to assign to the respective routing group.

  You may click the *Voice Mail Auto Attendant (VMAA) Menu* link to edit the parameters of desired VMAA Menu. For details, see "Voice Mail Auto-Attendant Menu".

- In **Ring Timer (sec)**, configure the time for which the extension, on which the call lands, should ring. By default, the ring timer is set to 015 seconds and can be changed.

- Select the **Continuous Ring** check box, if you want an extension to ring continuously until the call is answered. The first extension will continue to ring even as the system hunts for other extensions in the routing group to land the call. If the call still remains unanswered, the system will return the call to the first extension once again. This parameter is not relevant, if there is only one member extension in a routing group.

- Click **Submit**.
- Repeat the same steps when you select other Routing Group Number.
- To route incoming calls on a trunk, you must assign a Routing Group in the Trunk Landing Group in a Trunk Feature Template assigned to the trunk.
- To assign a Routing Group as a Trunk Landing Group, under "Configuring Trunks", see "Trunk Feature Template".
- To assign a Routing Group as a Department Group, see "Department Call".
- To assign a Routing Group as a Alarm Notification Group, under "Alarms", see "How to configure".
- To assign a Routing Group as a Floor Service Group, see "Floor Service".

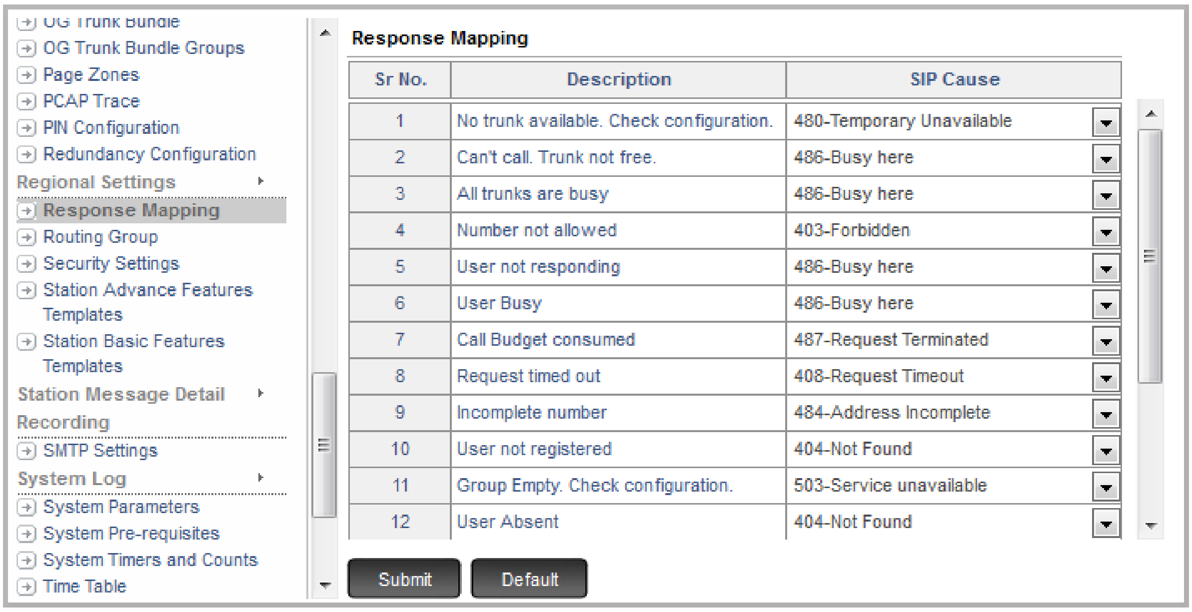
Response Mapping

The mapping of system disconnect/release events to SIP causes/errors in the SIP networks are vital aspects for VoIP functionality.

ANANT UCS supports configurable event cause mapping. It allows you to select the disconnects/release cause, that the system can send as a SIP response (cause/error) to the remote SIP Peer.

Configuring Response Mapping

- Login as System Engineer.
- Under **Configuration**, click **Response Mapping**.



- The **Description** column contains the list of the System events. For each event you can select the SIP cause/error that you want the system to send to the SIP network. By default, the most appropriate SIP cause/error is mapped against the event.
- Click **Submit**.

Security Settings

The feature Security Settings enables you to restrict unauthorized access to ANANT UCS using Remote login, Web, Third Party Auto Configuration as well as SIP Extension registration.

ANANT UCS also supports TLS (Transport Layer Security) protocol. Based on the TLS version configured in the server, TLS negotiation takes place. This enables the SIP Extensions and Web server to connect securely with the system over TLS protocol.

For Remote Login, Web, Third Party Auto Configuration

Remote Login

For the Remote Login, the Jeeves provides you the facility to generate the keys. Once these keys are generated, you need to contact the Matrix Technical Support Team for Remote Login.

Web and Third Party Auto Configuration

When any user attempts to access Web Server, Third Party Auto Configuration using false credentials for 5 times consecutively within 10 minutes, ANANT UCS blocks such IP Address for 10 minutes.

To allow access to Web Server and Third Party Auto Configuration, to specific trusted IP Address/es, you must configure them in the *Trusted IP Address/es* table. For instructions, see “How to configure” below.

For SIP Extensions

When any user attempts to register as a SIP Extension using false credentials— Authentication ID or Authentication Password and the authentication attempt fails for 10 times consecutively, ANANT UCS blacklists the IP Address and port used for registration. See “Black List IP Address - SIP Extensions” for more details.

You are recommended to configure the trusted IP Addresses in the *Trusted IP Address/es* table to avoid blacklisting. For instructions, see “How to configure” below.

However, if any IP Address is already blacklisted, it will be stored in the **Black List IP Address - SIP Extensions** table. To allow access to such blacklisted IP Address, you must remove it from the **Black List IP Address - SIP Extensions** table manually.

How it works

For this feature to work,

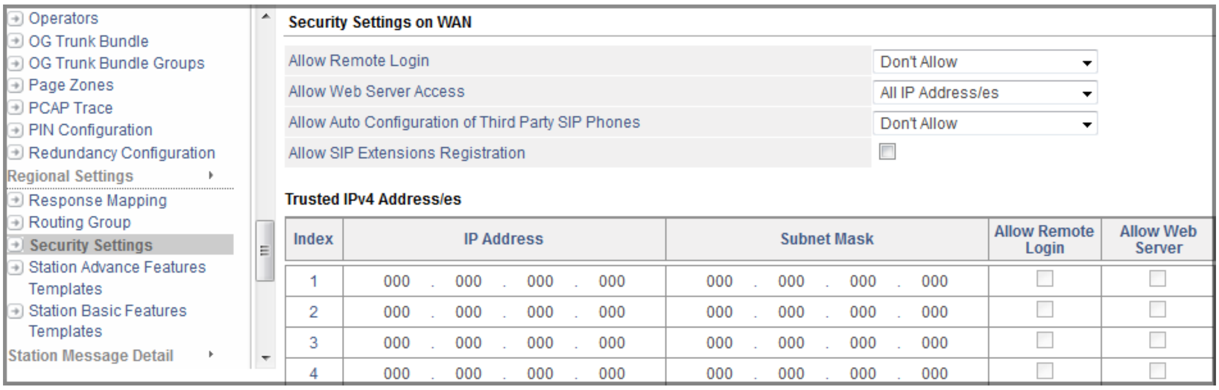
- the *Trusted IP Address/es* table must be configured. You can configure a maximum of 25 addresses in this table.
- determine the facilities you want to allow to each IP Address — Remote Login, Web, Third Party Auto Configuration, SIP Extension registration.
- with this table configured,
    - access to Remote Login, Web Server, Third Party Auto Configuration will be allowed only to the configured Trusted IP Address/es.

- Trusted IP Addresses configured for the registration of the SIP Extension, will not be blacklisted.

The successful attempt to access ANANT UCS using Web will be logged in the System Activity Log.

How to configure

- Login as System Engineer.
- Under **Configuration**, click **Security Settings**.



Trusted IPv4 Address/es

- In **Allow Remote Login**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

  If you want to allow access to generate the password for the Remote Login from all IP Addresses, select **All IP Address/es** option and click **Submit**.

  If you want to allow access to generate the password for the Remote Login from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.
  - enable the **Allow Remote Login** check box in the **Trusted IPv4 Address/es** table.
  - click **Submit**.

  When you select **All IP Address/es** or **Only Trusted IP Address/es** option, you will be provided an option to generate key. To do so,

  Click **Generate Key** link. The Key Generation window opens.

  In **Key Count**, select the number of keys you wish to generate. You can generate maximum 10 keys.

  Click **Generate**.

  The number of keys you selected are generated and saved in a file. Using these keys you can remotely login into the system.

  For further assistance for Remote Login, along with this file contact the Matrix Technical Support Team.

- In **Allow Web Server Access**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, All IP Address/es is selected.

*The option to select **Don't Allow** in the parameter **Allow Web Server Access** will be displayed only when both the LAN and the WAN Ports are configured in the system.*

  If you do not want to allow access to the Web Server from any IP Addresses, select **Don't Allow** option.

  If you want to allow access to the Web Server from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.
  - enable the **Allow Web Server** check box in the **Trusted IPv4 Address/es** table.

- In **Allow Auto Configuration of Third Party SIP Phones,** you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

  If you want to allow Auto Configuration of Third Party SIP Phones from all IP Addresses, select **All IP Address/es** option.

  If you want to allow Auto Configuration of Third Party SIP Phones from specific IP Addresses only:
  - select **Only Trusted IP Address/es** option.
  - configure the IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.
  - enable the **Allow Auto Configuration of Third Party SIP Phones** check box in the **Trusted IPv4 Address/es** table.

- By default, the **Allow SIP Extensions Registration** check box is disabled. If you want to allow SIP extensions registration from the WAN Port, select the check box.

  - If you allow SIP Extension Registration, you can **Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts**.

    By default, **All IP Address/es** option is selected. If you do not want to include the Trusted IP Addresses, select **Except Trusted IP Address/es** option.

    If you select **Except Trusted IP Address/es,**

    - configure the Trusted IP Address/es and their respective Subnet Mask in the **Trusted IPv4 Address/es** table.

    - enable the **Black List SIP Extension IP Address:Port except** check box for those IP Addresses which you do not want to blacklist.

- Click **Submit.**

Trusted IPv6 Address/es

- In **Allow Remote Login**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es option. By default, **Don't Allow** is selected.

  If you want to allow access to generate the password for the Remote Login from all IP Addresses, select **All IP Address/es** option and click **Submit**.

  If you want to allow access to generate the password for the Remote Login from specific IP Addresses only:

- select **Only Trusted IP Address/es** option.
- configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).
- enable the **Allow Remote Login** check box in the **Trusted IPv6 Address/es** table.
- click **Submit**.

When you select **All IP Address/es** or **Only Trusted IP Address/es** option, you will be provided an option to generate key. To do so,

Click **Generate Key** link. The Key Generation window opens.

In **Key Count**, select the number of keys you wish to generate. You can generate maximum 10 keys.

Click **Generate**.

The number of keys you selected are generated and saved in a file. Using these keys you can remotely login into the system.

For further assistance for Remote Login, along with this file contact the Matrix Technical Support Team.

- In **Allow Web Server Access**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es. By default, All IP Address/es is selected.

*The option to select **Don't Allow** in the parameter **Allow Web Server Access** will be displayed only when both the LAN and the WAN Ports are configured in the system.*

If you do not want to allow access to the Web Server from any IP Addresses, select **Don't Allow** option.

If you want to allow access to the Web Server from specific IP Addresses only:
- select **Only Trusted IP Address/es** option.
- configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).
- enable the **Allow Web Server** check box in the **Trusted IPv6 Address/es** table.

- In **Allow Auto Configuration of Third Party SIP Phones**, you can select Don't Allow, All IP Address/es or Only Trusted IP Address/es. By default, **Don't Allow** is selected.

If you want to Allow Auto Configuration of Third Party SIP Phones from all IP Addresses, select **All IP Address/es** option.

If you want to Allow Auto Configuration of Third Party SIP Phones from specific IP Addresses only:
- select **Only Trusted IP Address/es** option.
- configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).
- enable the **Allow Auto Configuration of Third Party SIP Phones** check box in the **Trusted IPv6 Address/es** table.

- By default, the **Allow SIP Extensions Registration** check box is disabled. If you want to allow SIP extensions registration from the WAN Port, select the check box.

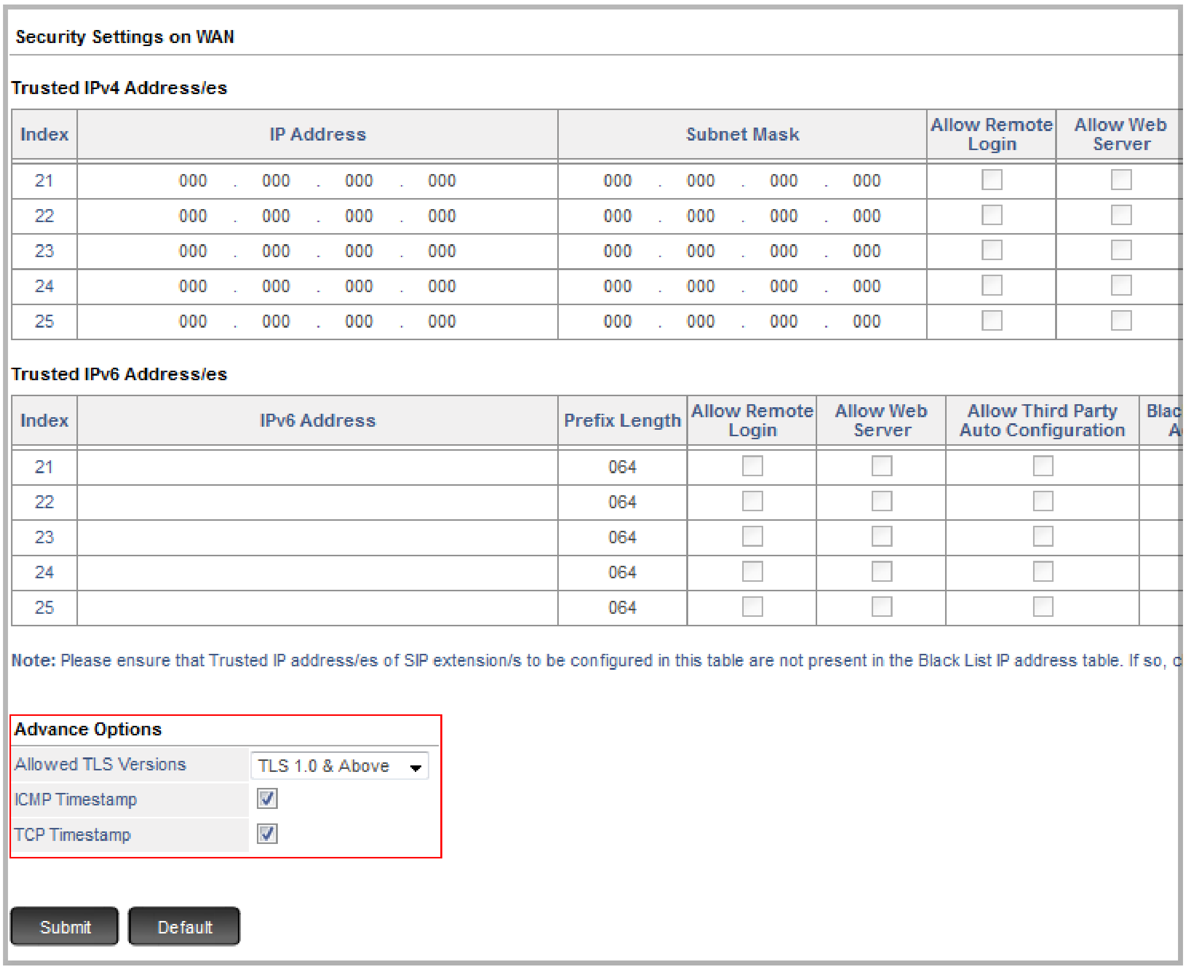
- If you allow SIP Extension Registration, you can **Black List SIP Extension IP Address:Port on multiple Authentication Failure Attempts**.

  By default, **All IP Address/es** option is selected. If you do not want to include the Trusted IP Addresses, select **Except Trusted IP Address/es** option.

  If you select **Except Trusted IP Address/es,**

  - configure the IPv6 Address/es and their respective Prefix Length in the **Trusted IPv6 Address/es** table. The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the address comprise the prefix (the network portion of the address).
  - enable the **Black List SIP Extension IP Address:Port except** check box for those IP Addresses which you do not want to blacklist.

Advance Options



- In **Allowed TLS Versions**, select the TLS Version[112] you want the system to use to establish a secure connection with the clients. You may select — TLS 1.0 & Above, TLS 1.1 & Above or TLS 1.2 as per your requirement. Default: TLS 1.0 & Above.

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112. SPARSH VP330/SPARSH VP210 supports TLS Version 1.0 only.

If the TLS version of the server and the client is not compatible, then secure connection will not be established.

Changing the TLS Version may result in drop of all ongoing TLS connections.

- Select the **ICMP Timestamp** check box if you want to send Date and Time of ANANT UCS in response to the ICMP request received from the remote device. By default, it is enabled.
- Select the **TCP Timestamp** check box if you want to send Date and Time of ANANT UCS in response to the TCP request received from the remote device. By default, it is enabled.

Selective Port Access

The system supports different extension and trunk port types. In the Selective Port Access feature, each port type is assigned a Port Access Code. Extension users can access a particular port by dialing the Port Access Code assigned to the Port and its Port Number.

How it works

- Extension user A wants to access a particular SIP Trunk port, *SIP Trunk Port 1* to make a call. Extension A must dial the Selective Port Access Feature Code, followed by the Port Type Code for SIP Trunk ports and then dial the Port Number.

- The following access codes are assigned to each Port Type:

Port Types	Port Access Code	Port Numbers
SIP Trunk	15	01 to 99
SIP Extension	19	001 to 5000
Virtual Extension	20	01 to 64

Here, Extension A must dial ***69-15-01***, where ***69*** is the feature code for Selective Port Access, **15** is the port access code for the SIP Trunk Port, and ***01*** is the number of the SIP Trunk Port which A wants to access.

How to configure

To be able to use Selective Port Access, extension users must have this feature enabled in their “Class of Service (CoS)”.

How to use

For Extended IP Phone Users

To enable Selective Port Access on an extension,

- Press DSS key assigned to Selective Port Access code
- From the menu select the Port Type
- Enter the Port Number of the selected Port Type
  OR
- Dial 69 / 89 (for users in USA)
- Dial the Port Type - Port Number.

Self Ring Test

You can use Self Ring Test to check the functioning of your own extension phone. Self Ring Test allows you to call your own extension. You can check the ringing volume of your extension phone.

How to use

For Extended IP Phone Users

- Go OFF-Hook.
- Press DSS Key assigned to Self Ring Test.
  OR
- Dial ***1057***.
- Go ON-Hook.
- Your phone rings.
- Go OFF-Hook to stop the ring.
- Go ON-Hook.

Shared Call Appearance

Shared Call Appearance (SCA) allows Standard SIP Phones that are registered with ANANT UCS at different locations with the same address/number, to get notification on call states of the call appearance(s) shared by them.

Whenever a call is made or received from a shared call appearance, ANANT UCS sends each SIP Phone sharing the call appearance(s), a notification on the state of the call appearance. Through these notifications, each user sharing the same address/number can know the current state of the call appearance and act accordingly.

ANANT UCS supports and displays the following call states for a shared call appearance on the SIP phones:

State	Meaning
Idle	When the call appearance is free.
Seized	When the call appearance is been seized from any User binding using the line-seize subscription.
Progressing	When the User has generated a call using the call appearance and the called destination is ringing.
Ringing	When the call is received on the User at the call appearance.
Active	When the call of the User at the call appearance is in matured state.
Held	When the call at the call appearance of the User has been put on public hold from the User binding.
Held-private	When the call at the call appearance of the User has been put on private hold from the User binding.

- *ANANT UCS supports SCA as per the Broadsoft SCA feature Specifications.*
- *ANANT UCS supports this feature only on Standard IP Phones.*
- *Standard IP Phones may differ in the type of indication (LED color and cadence, text message display) they provide for the Call States. Refer to the manufacturer's documentation for SCA indication supported on the phones.*
- *Calls put on 'Consultation Hold' from Location -1 of SIP Extension (binding -1), the SIP Extension registered at Location-2/3 (another binding) can not retrieve that call.*

How it works

ANANT UCS supports up to 10 call appearances on SIP extensions. The number of call appearances that will be shared by the SIP Phones will depend on the number of call appearances you have configured for the SIP extension.

To provide SCA to the Standard IP Phones registered with the same ID, the Shared Call Appearance check box must be enabled in the SIP Extension Settings of ANANT UCS.

On the Standard IP Phones, make sure you have configured as many call appearances as allowed on the SIP extension by ANANT UCS and configure the corresponding number of CA keys.

Here is an example of how Shared Call Appearance works:

- A, B and C are Standard SIP phones registered with ANANT UCS at three different locations with the same SIP ID, 602.

- Two Call Appearances are configured for A, B and C and Shared Call Appearance is enabled for SIP ID 602.
- Two keys are assigned for the two Call Appearances on A, B and C.
- There is an incoming call for SIP ID 602.
- ANANT UCS presents the incoming call on a free call appearance, Call Appearance1, as ‘Ringing’.
- A, B and C get the same alert, ‘Ringing’ simultaneously on the same call appearance, Call Appearance 1.
- A answers the call first and gets connected to it on Call Appearance 1.
- A, B and C get indication of the current call state as ‘Active’ on the same call appearance. B and C will not be able to make or receive any new call from this busy call appearance. However, they can make or receive a new call from the other free call appearance, Call Appearance 2.
- When A makes an outgoing call using Call Appearance 2, ANANT UCS presents the state of the same call appearance on A, B and C as ‘Seized’, then as ‘Progressing’ when the destination number starts ringing, and then as ‘Active’, when the call is answered.
- B and C will not be able to make or receive a call from Call Appearance 2.
- A can put an ‘Active’ call on public Hold or on private Hold. When A puts an ‘Active’ call on public hold, ANANT UCS presents the state of this call as ‘Held’ to A, B and C. Now, B or C can retrieve the call by pressing the corresponding call appearance key.
- When A puts an ‘Active’ call on private Hold, ANANT UCS presents the state of this call as ‘private-Held to A, B and C. Only A can retrieve the call. Thus, if a call is put on private hold (Held-private), it can be retrieved only from the IP Phone that put it on hold.

How to configure

You can provide this feature only to the Standard IP Phones you have registered with ANANT UCS. To provide this feature,

- In ANANT UCS, select the **Shared Call Appearance** check box on the SIP Extension Settings. For instructions, see “Configuring SIP Extensions”.
- On the Standard IP phones,
    - configure as many call appearances as allowed on the SIP extension.
    - For each shared call appearance, configure a corresponding call appearance key on the SIP Phone.

  For instructions refer to the manufacturer’s documentation (Installation Guide/User Guide) for the respective SIP Phones.

SMTP Settings

The Unified Messaging Functionality of ANANT UCS includes using SMTP to send emails for the following functions:

- activities logged in the System Activity Log (SAL)
- faults logged in the System Fault Log (SFL)
- VMS Notification, that is
    - send notification to the extension users about the arrival of new messages (with/without Voice Mail Attachment).
    - send notification to the extension users about the memory usage status of their mailbox.
    - memory usage notification to SE.

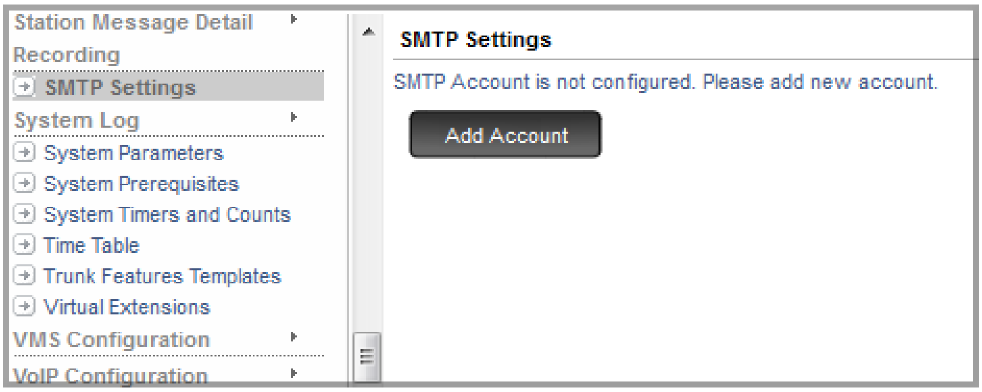
You can configure different account for each application. The systems allows you to configure a maximum of three accounts.

For email transmission, you must:

- configure the parameter *Message Wait Notification via Email*[113] in the VMS settings of the extension. For instructions, see “Email Based Notification”.
- configure the parameters *VMS E-mail Notification* and *Use SMTP Account* in VMS General Parameters. For instructions, see “Configuring VMS General Parameters”.
- configure the parameters in System Log Notification for SAL and SFL. For instructions, see “System Log Notification”.
- configure the SMTP Accounts and its parameters.

Configuring SMTP

- Login as System Engineer.
- Under **Configuration**, click **SMTP Settings**.
- By default no account is configured. You can add a maximum of three accounts. Click **Add Account** and then configure the following parameters.



    - **Account Name:** This will help to identify the application for which the account is being used.

113. *You can also have the new voice message mailed as an attachment with the message wait notification.*

- **SMTP Server Address:** Configure the Server's IP Address that will be used to send the outgoing mails.Both IPv4 and IPv6 addresses are supported.The Server Address can be a maximum of 40 characters.

- **SMTP Server Port:** Configure the desired Port number. If port is not programmed, the default port value will be 25.Valid Port ranges: 25;465;587;1025 to 65535

- **Email ID:** Configure the email ID for the account registered with the Email Server, as provided by your network administrator. This Email ID will appear to the recipient as the originator of the email (that is in the FROM field). The Email ID may consist of a maximum of 64 characters. Default: Blank.

- **Require Authentication?:** If your Email Server uses authentication, enable this check box. Default: Disabled. If you have enabled authentication, you must also configure the *User ID* and the *Password.*

- **User ID:** Configure the User ID as provided by your network administrator. The User ID may consist of a maximum of 40 characters.Default:Blank

- **Password:** Configure Authentication Password as provided by your network administrator.Password can be a maximum of 24 characters. Default: Blank.

- **Enable Secure Socket Layer (SSL/TLS):** To transport all data in a secure manner, select this check box. All the data to the Email Server will be transported over secure layer. Default: Disabled.

- **Display Name:** This name will be displayed to the mail recipient. You can configure a maximum of 24 characters. Only ASCII characters are allowed. Default: Blank.

- **Connection Timeout Interval:** Configure the time duration for which you want the system to wait for a response from the SMTP server. You may change the Connection Timeout Interval timer, if required. The range of Connection Timeout Interval timer is 01 to 99 seconds. By default, it is set to 60 seconds.

- **SMTP Account is used by application:** It displays the name of the application/s — VMS Notification — which is using the account. It will be blank if the account is not being used by any application.

- Click **Submit**.

Test SMTP Settings

- Click the **'Test Account'** button to check if the SMTP Parameters have been configured correctly.

  When you click this button, the alert message will appear: *"Testing SMTP can take up to 99 seconds to complete the process. Would you like to continue?"* Click 'OK' button.

  The Test Result will be displayed in 'Test Status' field.

- **Test Status:** Any one of the results listed below may appear in this field:

Test Status Message	Description
"SMTP Server Connection Not Established"	When connection to SMTP server fails due to any reason.
"Login to SMTP Server Failed"	When connection to SMTP server is established but login to SMTP server fails due to any reason.

Test Status Message	Description
"Sending Test Mail Failed"	When connection to SMTP has been established successfully but there is no acknowledgment for the test mail sent.
"Test Mail Sent Successfully"	When acknowledgment for the test mail sent to SMTP server received successfully.
Already Test is running	When the test is already being processed for another SMTP account.
Timeout	When the test is in process and the Connection Timeout Interval expires.

Email Failure Errors

When message sending fails, the events will be logged into the System Fault Log with a specific code. For more information, see “System Fault Log”.

Simple Network Management Protocol (SNMP)

Simple Network Management Protocol (SNMP) is an application-layer protocol used for exchanging management information between network devices. Using SNMP, you can manage and monitor network elements, audit network usage, detect network faults or inappropriate network access.

The SNMP architecture consists of:

- An **SNMP Agent** is a program that is bundled within the managed device. SNMP agent allows a managed device to collect the Management Information Base from the device and make it available to the SNMP Manager on request. It receives SNMP requests and generates SNMP responses or notifications (traps/informs). The SNMP Agents are SNMP Servers. Here, it is ANANT UCS.

- **SNMP Manager**, usually the Network Management Station. The manager communicates with multiple SNMP Agents implemented in the network. It generates SNMP requests and receives SNMP responses and notifications (traps/informs). The SNMP Manager is an SNMP Client. Here, it is the SNMP Manager installed in the PC.

- **Managed device** or the network element is a part of the network that requires some form of monitoring and management. For example, switch, routers, servers.

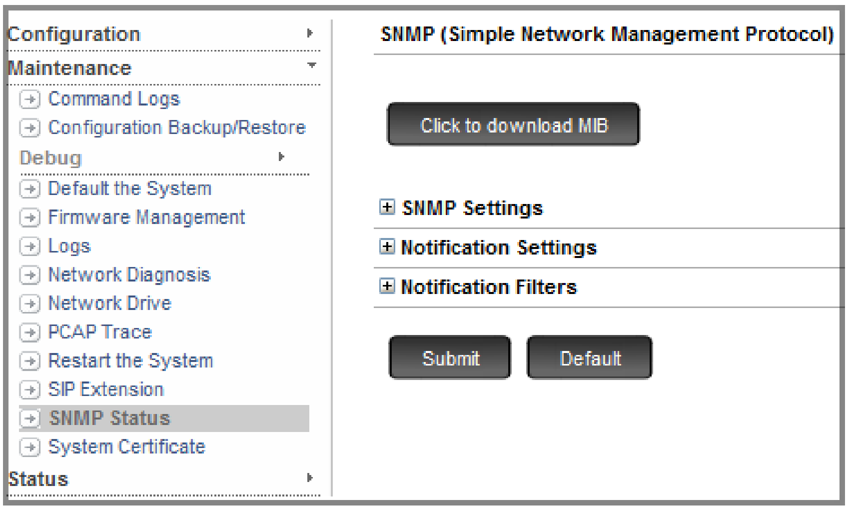
- **Management Information Base** is the commonly shared database between the Agent and the Manager. This information is known as managed objects. These managed objects are defined in MIB (Management Information Base) module.Any sort of status or information that can be accessed by the Manager is defined in a MIB.

SNMP uses UDP (User Datagram Protocol) as the transport protocol for passing information between Managers and Agents. By default, the Agent listens on UDP port 161 for requests from Manager and the Manager listens on UDP port 162 for notification from Agent.

How to Configure

- Login as System Engineer.

- Under **Maintenance**, click SNMP.



MIB Files

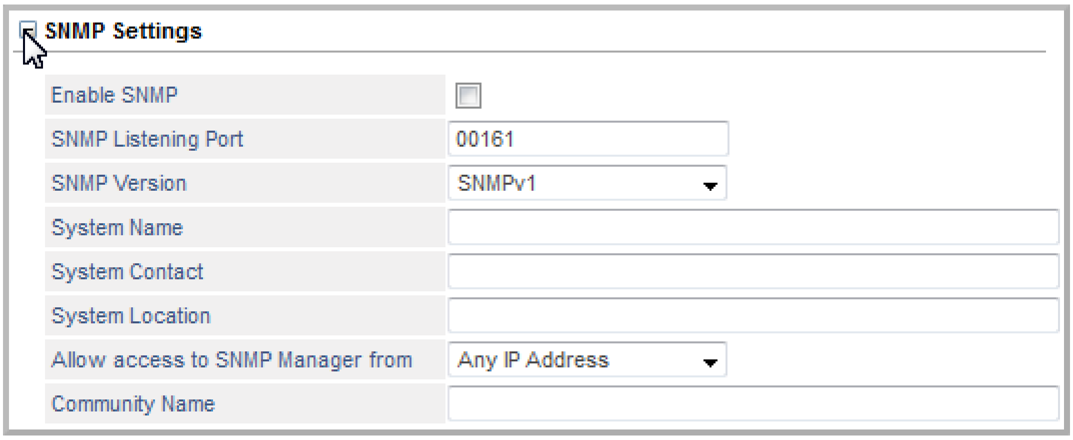
If the Manager needs to access any status or information about the client, the Manager must download and install the MIB files on the local disk.

To download these files,

- Click the **Click to download MIB** button.
- You will get a prompt with the option to open the file or save the file to a location. Save the file on the local disk.

SNMP Settings

- Click **SNMP Settings** to expand.



- Select the **Enable SNMP** check box. Default: Disabled.
- Configure the **SNMP Listening Port**. Valid Range:161, 1025-60000. Default: 161.
- Select the **SNMP Version** as supported by your SNMP Manager. You can select from:
  - SNMPv1
  - SNMPv2c
  - SNMPv3

  For enhanced security, you must select SNMPv3.
- Configure the **System Name**. When there are multiple devices connected in the same network, the name configured helps to identify the ANANT UCS within the network. The System Name can be a maximum of 40 characters. Default: Blank.
- Configure the **System Contact**. It is the name and number of the person to be contacted, in case of notification. The System Contact can be of a maximum of 40 characters. Default: Blank.
- Configure the **System Location**. This is the physical location of ANANT UCS. This information is helpful to the administrator. The System Location may consist of a maximum of 40 characters. Default: Blank.

- Configure **Allow access to SNMP Manager from** to allow/restrict accessibility to the SNMP Manager. You can select **Any IP Address** or **Specific IP Addresses**. Both IPV4 and IPv6 addresses are supported.

  If you want to restrict the accessibility of the SNMP Manager, select **Specific IP Addresses**, configure **IP Addresses 1** to **5**. Default: Blank.

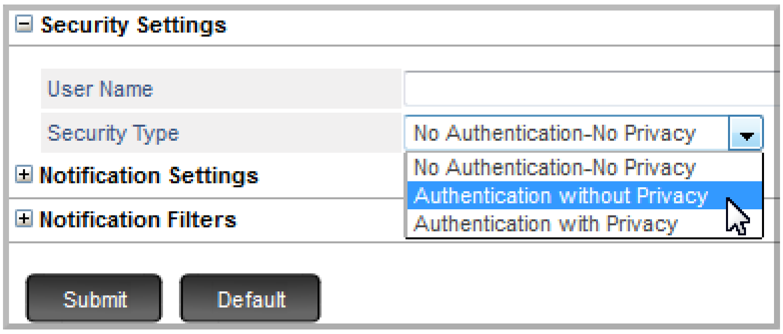
- If SNMP version is set as **SNMPv1** or **SNMPv2c,** configure **Community Name**.

  Community Name identifies the SNMP community in which the sender and recipient of the message are located. It enables communication between ANANT UCS and the Manager. The Community Name can be a maximum of 40 characters. Default: Blank. To avoid unauthorized access, we recommend you to assign a strong Community Name.

- If SNMP version is set as **SNMPv3,** the **System's Engine ID** is displayed. This is a unique identification of the system. It is a hexadecimal field with length of 22 characters. The ID consists of:
  - Enterprise Number (800086df03 which is fixed)
  - MAC Address of the system (MAC address of Ethernet (LAN/WAN) Port)

Security Settings

- If SNMP version is set as **SNMPv3**, click **Security Settings** to expand and configure the following.

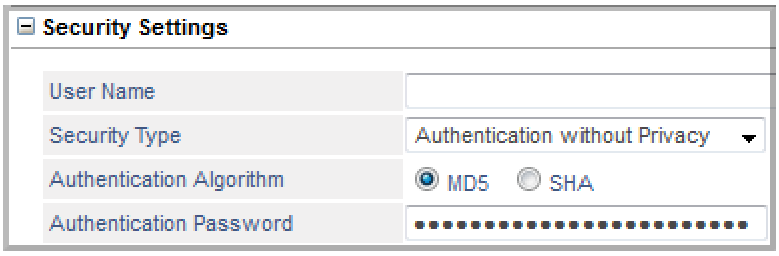


- Enter the **User Name**. The User Name can be a maximum of 40 characters. User Name will be used for authentication and privacy in SNMPV3.

- Select the appropriate **Security Type** as per your requirement. Security Type defines the level of security.

  - When Authentication and Privacy are not required, select **No Authentication-No Privacy**

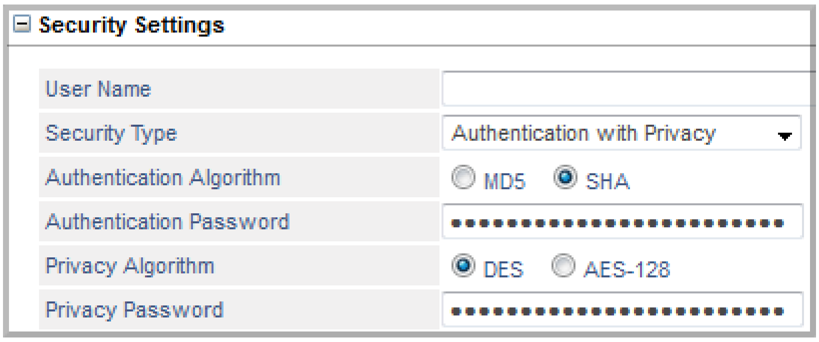
  - When only Authentication is required, select **Authentication without Privacy**. Incoming SNMP Messages will require authentication.



If you select this method, select the **Authentication Algorithm** as **MD5** or **SHA**. Default: MD5.

In

- When both Authentication and Privacy are required, select **Authentication with Privacy**. Incoming SNMP Message will require authentication and these messages will be encrypted, which will be decrypted at the receivers end only.



If you select this method, select the **Authentication Type** as **MD5** or **SHA**. Default: MD5.

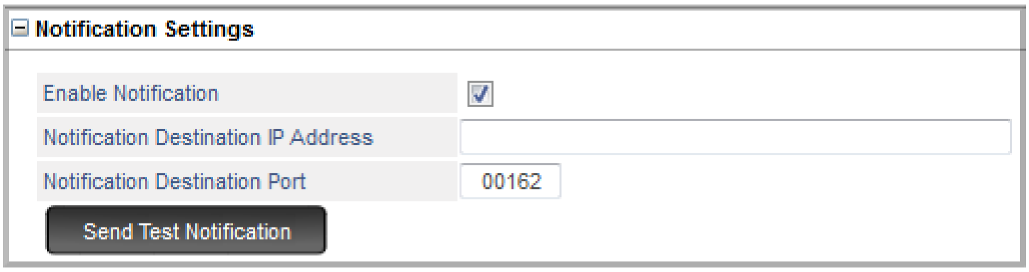
Enter **Authentication Password** for the User Name you have assigned. To avoid unauthorized access, we recommend you to use a strong password and make sure it is kept confidential. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

Select the **Privacy Algorithm** as **DES** or **AES**. Default: DES.

Enter the **Privacy Password** of your choice.We recommend you to use a strong password and make sure it is kept confidential. The Privacy Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

Notification Settings

- Click **Notification Settings** to expand.



If SNMP version is set as **SNMPv1**, configure the following parameters.

- Select **Enable Notification** check box, if you want ANANT UCS to generate Trap message for an error.
  - You must configure the **Notification Destination IP Address** and **Notification Destination Port** of the Manager or of any other device where you want to receive the trap messages.

    ANANT UCS will send the notification (error message) to the destination configured.

Valid range of the Notification Destination Port is 162 or 0-65535. Default port is 162.

- Click **Submit**.

If SNMP version is set as **SNMPv2c or SNMPv3**, configure the following parameters.

- Select **Enable Notification** check box, if you want ANANT UCS to generate Trap or Inform message for an error.
- Select the **Notification Type**. You may select **Trap** or **Inform**.

  If you want the system to send notification message without acknowledgment, select **Trap**.

  If you want the system to send notification message with acknowledgment, select **Inform**.

  - If you select **Inform** as the *Notification Type*, you must configure Retry Attempts and Retry Interval.

    If acknowledgment is not received from the Manager for the notification sent, the system will keep retransmitting the message for the number of attempts you have configured as the **Retry Attempts**. Default: 3.

    The system will retransmit the messages at regular time intervals you have configured as **Retry Interval**. Default: 10 seconds.

  - You must configure the **Notification Destination IP Address** and **Notification Destination Port** of the Manager or of any other device where you want to receive the trap messages.

    ANANT UCS will send the notification (error message) to the destination configured.

    Valid range of the Notification Destination Port is 162 or 0-65535. Default port is 162.

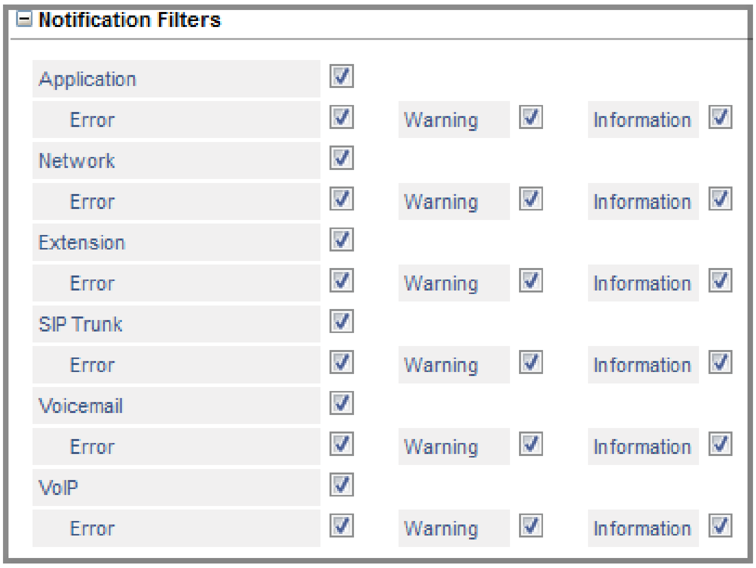
- Click **Submit**.
- Click the **Send Test Notification** button, to send a test notification message.

Notification Filters

By default, you get notifications of errors, information and warnings for events related to the Application, Network, Extension, Trunk Ports — SIP Trunks, Voice Mail System and VoIP. Refer to the table at the end of this topic for the event list. You can choose the type of notification you want by setting the notification filters.

To set the filters,

- Click **Notification Filters** to expand.



- By default, all the filters are enabled. To disable any filter, clear the respective check box.

Simple Network Time Protocol - SNTP

ANANT UCS uses the Real Time Clock (RTC) of the Bare Machine (on which it is installed) to store date and time. When you select the Region, the RTC parameters are set automatically. For this, ANANT UCS supports both, SNTP Client as well as SNTP Server.

SNTP Client

The RTC can drift over a long period. So, you may check and reset the RTC values at regular intervals to correct this drift. You can set the RTC manually (see "Real Time Clock (RTC)") or synchronize it with any SNTP Server in the Public Network. To synchronize time with the SNTP Server, ANANT UCS supports SNTP Client.

ANANT UCS acts as an SNTP Server and the clients can synchronize their time with it. The clients can synchronize their time by configuring the IP address of ANANT UCS in their SNTP settings as the SNTP Server. Thus ANANT UCS acts as Server and will provide GMT time to the clients.

SNTP Server

ANANT UCS supports IP Phones and these phones have SNTP Clients so that they can synchronize their clock with an SNTP Server.

These SNTP Clients in the phones can synchronize their clock with the SNTP Server on the public network only if Internet is available in the network. Hence, to overcome this ANANT UCS also supports an SNTP Server with which the SNTP Clients can synchronize their time without internet connectivity. The clients can synchronize their time by configuring the IP address of ANANT UCS in their SNTP settings as the SNTP Server. Thus, ANANT UCS acts as Server and will provide GMT time to the clients.

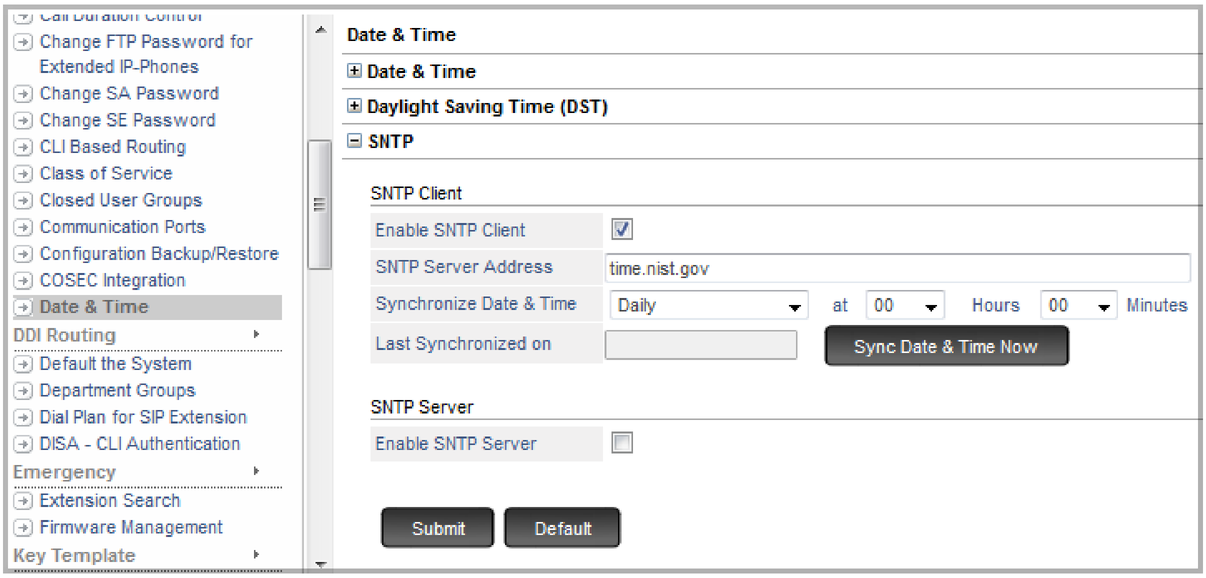
How to Configure

SNTP Client

To use SNTP for synchronizing with the Real Time Clock,

- Login as System Engineer.
- Under **Configuration,** click **Date and Time**.

- Click **SNTP** to expand.



- Configure the following parameters under SNTP Client:

  - **Enable SNTP Client:** Select this check box if want the system to act as a **SNTP Client**. Default: Enabled.

    - **SNTP Server Address:** Enter the Time Server Address. SNTP Server address can be of maximum 64 characters. Both IPv4 and IPv6 addresses are supported. Default: time.nist.gov

    - **Synchronize Date and Time:** You can Synchronize Date and Time with the SNTP Server on daily or weekly basis. If you select weekly, the system will sync the date and time with the SNTP Server on every Monday. Select the desired option as per your requirement. Default: Daily.

    - **Last Synchronized On:** This field displays the time when the system last synchronized with the SNTP Server.

    - **Sync Date and Time Server Now:** Click this button to synchronize date and time of ANANT UCS with the SNTP server, whenever required.

SNTP Server

  - **Enable SNTP Server:** Select this check box if you want the system to act as an SNTP Server and the extensions (IP Phones) registered with it to behave as the clients. Default: Disabled.

- Click **Submit**.

Static Routing Table

Static Routing Table is required when:

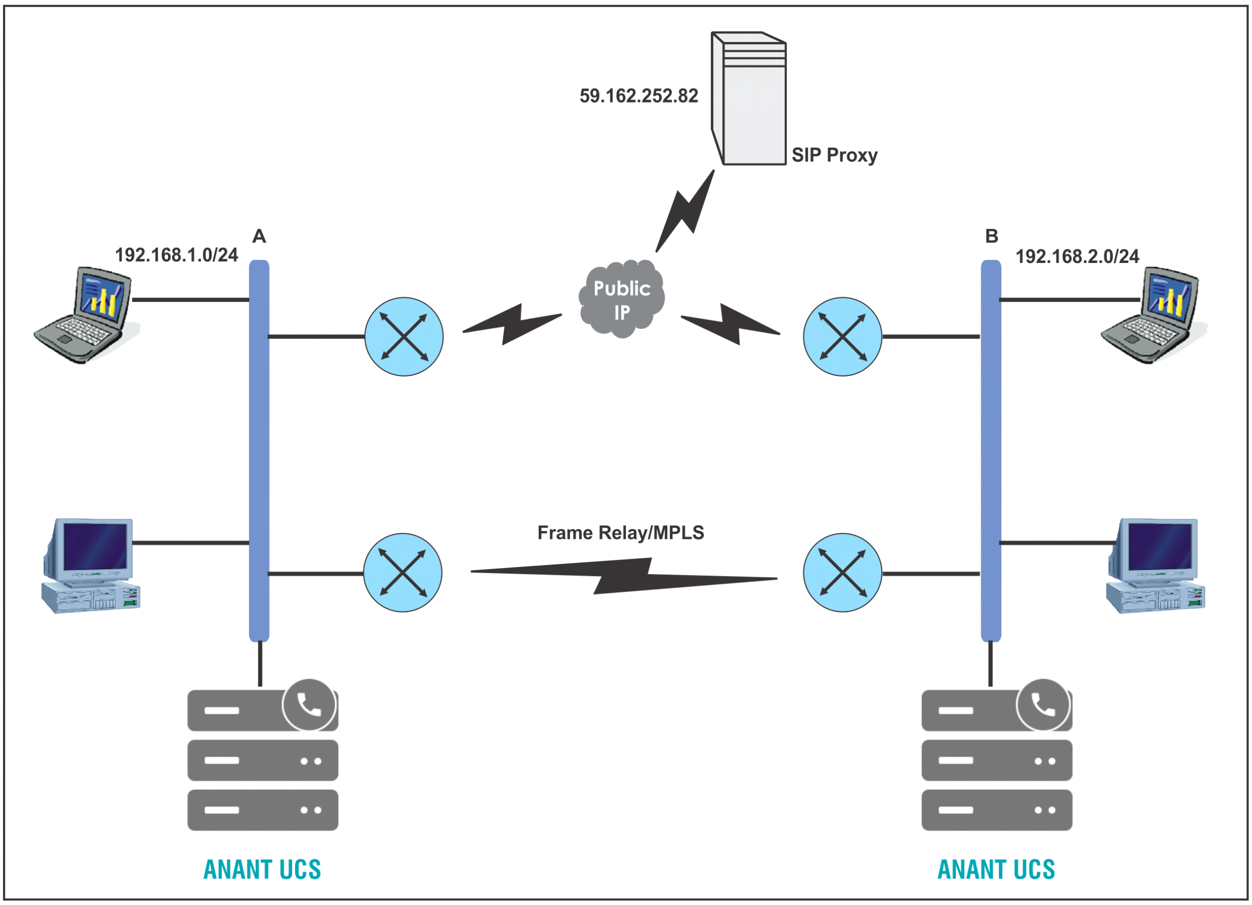
- you have more than one router (gateway) in your network and you want ANANT UCS to send packets to multiple routers/gateways for different types of calls. These packets are routed through the WAN Port.
- you want to route specific packets through the LAN Port.

Static Routing Table helps to route calls between point to point sites (connected through Multi Protocol Label Switching-MPLS, Frame Relay, etc.) and to public internet at the same time.

How it works

Routing the packets through the WAN Port

For example, two Local Area Networks, Network A and Network B, are connected through Frame Relay/ Multi Protocol Label Switching (MPLS) network to give access to local resources and also to make Peer-to-Peer calls.



At both sites ANANT UCS is connected behind a router.

These sites are also connected to public IP network to:

- provide internet access to local hosts.
- access DID service provided by ITSPs to make calls over IP network.

Network A and Network B are in different subnets.

The Static Routing Table makes it possible to route different types of outgoing calls—Peer to Peer or Proxy—made to different subnets through different Gateways.

The Static Routing Table defines the appropriate Gateway Address (or Router's LAN Address) where the IP packets are to be sent.

In the Static Routing Table, you must configure:

- The address of the final Destination where the packets are to be sent.
- The Subnet Mask to be applied on the final destination address.
- The Gateway Address where the IP packets are to be sent.

When ANANT UCS sends packets, if the final destination IP Address and ANANT UCS are not in the same Subnet, the system will check the Static Routing Table.

If a perfect match is found, ANANT UCS will start sending the IP packets to the corresponding Gateway Address configured in the table.

If no match is found, ANANT UCS will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in the Network Parameters. For detailed instructions, see "Configuring Network Parameters".

Routing the packets through the LAN Port

In another scenario:

- SIP Trunk 1 is a proxy trunk
- Source Port IP Address on the SIP Trunk 1 is configured as LAN Port IP Address
- IP Address provided by the ITSP is 192.168.5.10 (Destination IP Address)
- LAN IP Address is 192.168.5.1

In the Static Routing Table, you must configure:

- The address of the final Destination where the packets are to be sent as 192.168.5.10.
- The Subnet Mask to be applied on the final destination address as 255.255.255.0.
- The Gateway Address where the IP packets are to be sent as 192.168.5.1.

When ANANT UCS sends packets, it checks the final destination IP Address in the Static Routing Table.

If a perfect match is found, ANANT UCS will start sending the IP packets to the corresponding Gateway Address configured in the table. In this case the packets will be sent through the LAN Port the Destination IP Address 192.168.5.10

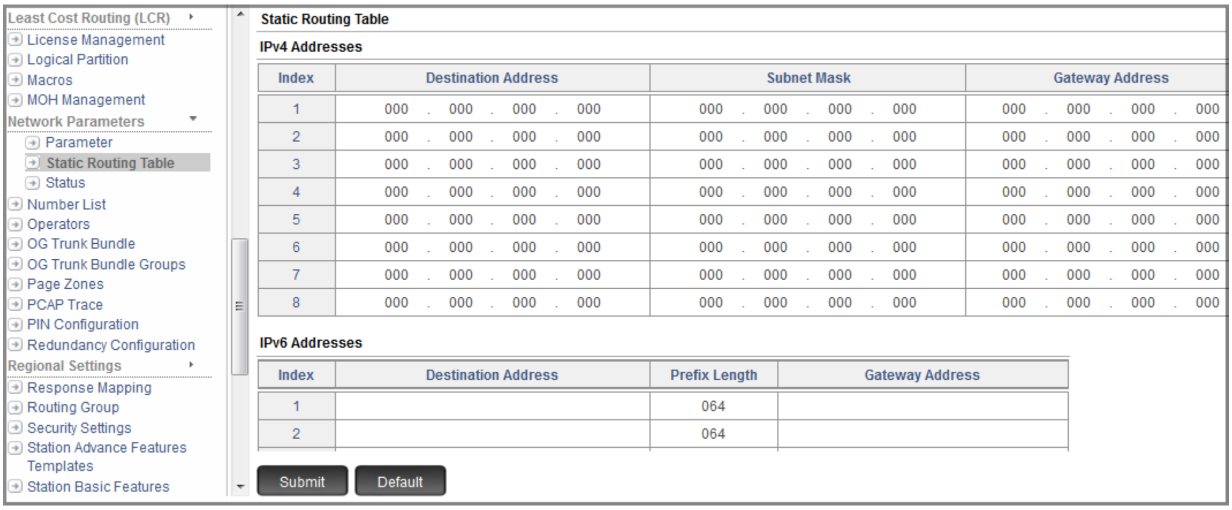
If no match is found, ANANT UCS will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in the Network Parameters. For detailed instructions, see "Configuring Network Parameters".

How to configure

The Static Routing Table must be configured at each location where ANANT UCS is installed. You may configure the Static Routing Table using Jeeves.

Configuring Static Routing Table

- Login as System Engineer.
- Under **Configuration**, click **Network Parameters**.
- Click **Static Routing Table**.



The Static Routing Table allows you to configure upto 8 entries in each table, IPv4 Addresses table and IPv6 Addresses. Each entry is stored against an Index number.

IPv4 Addresses Table

For each entry, you must configure the following fields:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.
- **Subnet Mask**: This is the mask to be applied on the destination address.
- **Gateway Address**: This is the IP address of the node where the IP packets are to be sent. Generally, it is the IP address of the LAN interface of the Router.

  The Gateway Address must be in the same subnet as ANANT UCS.

Click **Submit.**

IPv6 Addresses Table

For each entry, you must configure the following fields:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.

  Valid Range of the IPv6 Address is A to F, a to f, 0 to 9,:(colon). It can be a maximum of 39 characters. Default: Blank.

- **Prefix Length**: The Prefix Length is a decimal value that indicates how many of the high-order contiguous bits of the destination address comprise the prefix (the network portion of the address).

  The Prefix Length range is from 1 to 128 bits. Default: Blank.

- **Gateway Address**: This is the IP address of the node where the IP packets are to be sent. Generally, it is the IP address of the LAN interface of the Router.

  The Gateway Address must be in the same subnet as ANANT UCS.

Click **Submit**.

To take the above example further, the Static Routing Table of ANANT UCS at Location A should be configured as:

Index	Destination Address	Subnet Mask	Gateway Address
1	192.168.2.0	255.255.255.0	192.168.1.1
2			
:			
8			

If you configure the Subnet Mask as 255.255.255.255, then only the Destination Address will be accessible.

If you configure the Destination Address as 192.168.2.1, then only this specific address will be accessible.

- The Destination Address 192.168.2.0 specifies the network address of Location B.
- The Subnet Mask is the mask to be applied on the Destination address.
- The Gateway Address 192.168.1.1 specifies the LAN address of the Router A which connects location A and location B.

  The IP address of the LAN interface of the router which connects Location A to the public internet should be configured as Default Gateway in the Network Parameters of ANANT UCS in location A.

  With the Static Routing Table configured thus, all calls made by ANANT UCS to 192.168.2.0/ 24 will be routed through the router which connects Location A to Location B. Whereas, all calls made by ANANT UCS to addresses other than 192.168.2.0/ 24 will be routed through the Default Gateway.

  Similarly, configure the Static Routing Table in ANANT UCS at location B to enable calling from Location B to Location A.

- Click **Submit.**

Station Message Detail Recording (SMDR)

ANANT UCS can record the details of Internal, Incoming and Outgoing calls made from/to all the extensions. This feature is called Station Message Detail Recording (SMDR).

You can store SMDR, obtain SMDR as a Report whenever you want or obtain it Online, immediately after the call has been made or received. You can also use SMDR to calculate the cost of the calls.

To be able to use SMDR, you must configure:

- **SMDR Storage:** These parameters are configured to enable the storing of the IC, OG and Internal calls. To know more, see “Station Message Detail Recording-Storage”.
- **SMDR Report:** These parameters are configured to assign destination port for getting report of IC, OG and Internal calls and to get offline report. To know more, see “Station Message Detail Recording-Report”.
- **SMDR Online:** These parameters enable you to obtain Online report of Incoming, Outgoing and Internal calls. With Online SMDR you can obtain details of each call immediately after the call has been made or received. You can also set the call record format you want for Incoming calls when ANANT UCS is interfaced with a third party call accounting software (CAS). To know more see,“Station Message Detail Recording-Online”.
- **SMDR Posting:** These parameters enable you to interface third-party call accounting software (CAS) with ANANT UCS for call cost calculation. You can select the protocol supported by the call accounting software and further customize the handshaking parameters and call record formats. To know more see, “Station Message Detail Recording-Posting”.

Refer to the Hospitality Manual for more details about PMS.

Station Message Detail Recording-Storage

ANANT UCS stores SMDR of Incoming calls, Outgoing Calls and Internal Calls. The call records are stored in the SMDR buffer. For this, SMDR storage for these types of calls must be enabled, and if required further filters can be set.

ANANT UCS can store 6000 outgoing calls, 4999 incoming calls and 999 internal calls in the SMDR buffer. Once the SMDR buffer is full, the next call is stored in place of the oldest call in the SMDR buffer, using the First In First Out (FIFO) logic.

The buffer can be cleared[114] at any time from the System Administrator mode.

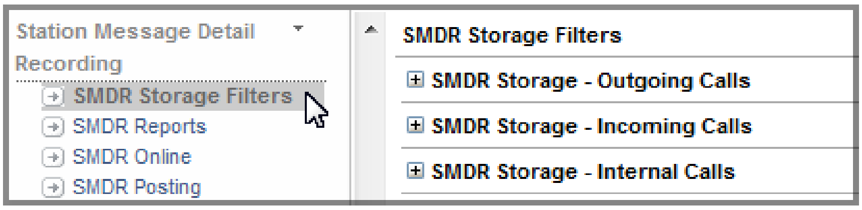
The SMDR buffer data is maintained even during power failures. However it is advisable to take frequent printouts of the calls to avoid accidental loss of the data.

How to configure

To enable storage of SMDR of Outgoing, Incoming and Internal Calls, you must enable this feature in the system and set the storage filters as per your requirement.

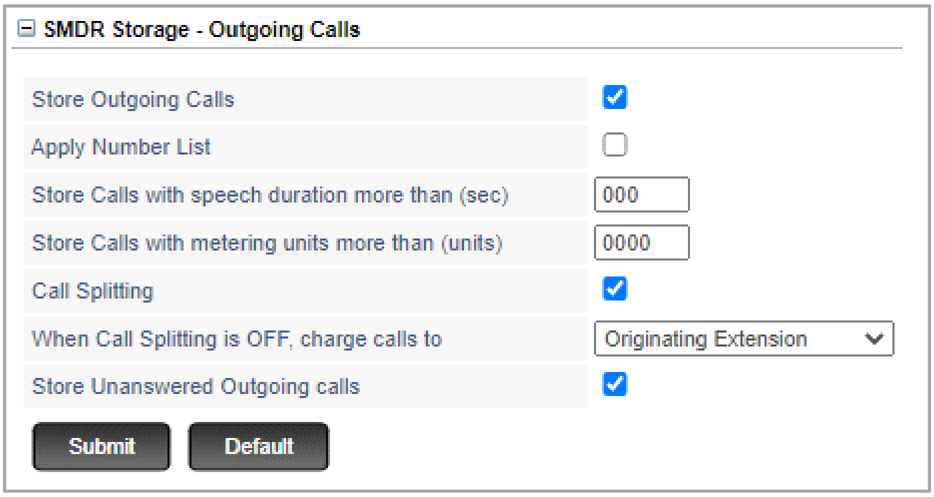
Configuring SMDR-Storage

- Login as System Engineer.
- Under **Configuration**, click **Station Message Detail Recording**.
- Click **SMDR Storage Filters**.



114. To comply with the Indian Government Laws and Regulation, this feature is not provided for India Region.

- To configure storage filters for outgoing calls, click **SMDR Storage - Outgoing Calls** to expand.

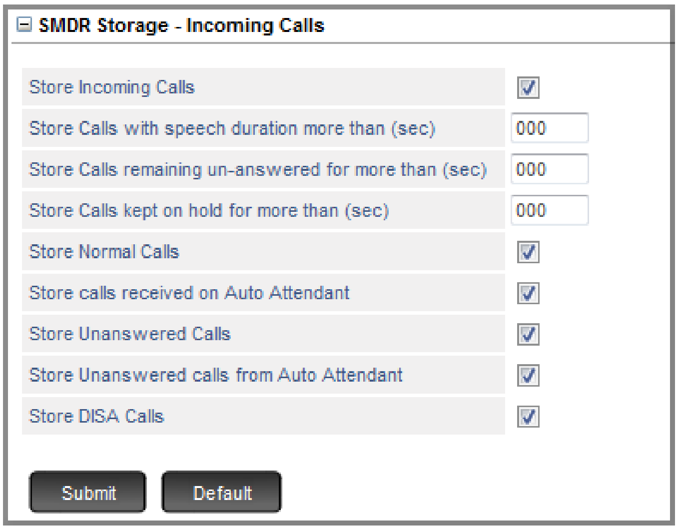


- Select the **Store Outgoing Calls** check box to enable storage of outgoing calls as per the filters you set. If outgoing call storage is disabled, no outgoing call will be stored.
- Select the **Apply Number List** check box and then you can configure **Store Calls of Called Number matching with Number List**.

  You can limit the storage of calls to certain numbers. Select a number list and enter the desired called party numbers in the selected list. In **Store Calls of Called Number matching with Number List**, select the same list number.
- If you want outgoing calls that exceed as certain duration to be stored, set the filter **Store Calls with speech duration more than (sec)** to the desired duration. All outgoing calls with duration greater than this value, will be stored.
- If you want outgoing calls exceeding certain metering units to be stored, set the filter **Store Calls with metering units more than (units)** to the desired value. All outgoing calls that have metering units greater than this value, will be stored.
- Outgoing calls made by an extension user can be transferred to another extension. In such cases, you may enable **Call Splitting** if you want to charge the amount to each extension according to the duration of speech that each extension was involved in the call.
- If Call Splitting is disabled, you have the option of charging the call amount either to the extension that originally made the call, i.e. the Originating Extension, or to the extension that was last in speech on the call, i.e. the Terminating Extension.

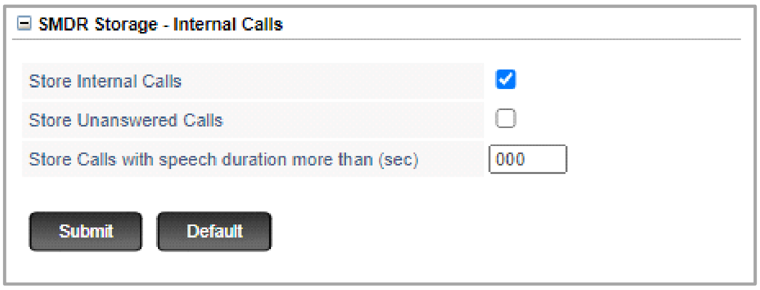
  In the **When Call Splitting is OFF, charge calls to** field, select the desired extension you want to charge the call to as **Originating Extension** or **Terminating Extension**.
- Select the **Store Unanswered Outgoing calls** check box to enable storage of unanswered outgoing calls as per the filters you set. If this check box is disabled, unanswered outgoing calls will not be stored.
- Click **Submit**.

- To configure storage filters for incoming calls, click **SMDR Storage - Incoming Calls** to expand.



- Select the **Store Incoming Calls** check box to enable storage of incoming calls as per the filters you set. If incoming call storage is disabled, no incoming call will be stored.
- If you want incoming calls that exceed as certain duration to be stored, set the filter **Store Calls with speech duration more than (sec)** to the desired duration. All calls with duration greater than this value, will be stored.
- If you want incoming calls that remain unanswered for certain duration to be stored, set the filter **Store Calls remaining un-answered for more than (sec)** to the desired duration. All calls with duration greater than this value, will be stored.
- If you want incoming calls that were kept on hold for a certain duration to be stored, set the filter **Store Calls kept on hold for more than (sec)** to the desired duration.
- If you want all calls, except calls received using Auto Attendant to be stored, select the **Store Normal Calls** check box.
- If you want calls received on Auto Attendant to be stored, select the **Store calls received on Auto Attendant** check box.
- If you want all calls, that remained unanswered to be stored, select the **Store Unanswered Calls** check box.
- If you want all calls received using Auto Attendant that remained unanswered to be stored, select the **Store Unanswered calls from Auto Attendant** check box.
- If you want calls made using DISA, select the **Store DISA Calls** check box.
- Click **Submit.**

- To configure storage filters for internal calls, click **SMDR Storage - Internal Calls** to expand,



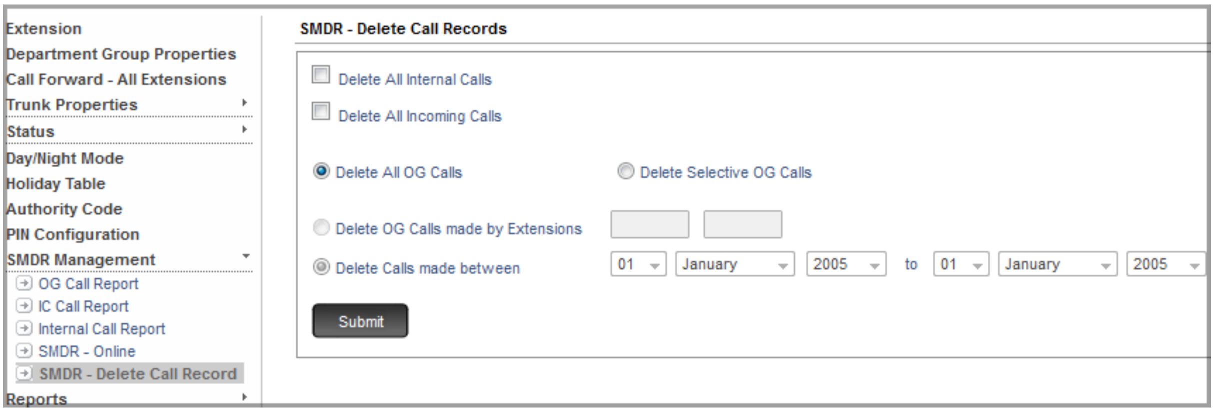
  - Select the **Store Internal Calls** check box to enable storage of internal calls as per the filters you set.
  - To store unanswered/missed internal call, select the **Store Unanswered Calls** check box.
  - To store internal calls that exceed as certain duration, set the filter **Store Calls with speech duration more than (sec)** to the desired duration. All internal calls with duration greater than this value, will be stored.
  - Click **Submit.**

How to use

The SMDR stored in the buffer can be cleared at any time from the System Administrator mode, using Jeeves.

To delete SMDR records,

- Login as System Administrator.
- Under **SMDR Management**, click **SMDR - Delete Call Record**[115].



- To delete all records in the internal SMDR buffer, select **Delete All Internal Calls** check box.
- To delete all records in the Incoming SMDR buffer, select the **Delete All Incoming Calls** check box.
- If you want to delete all records in the Outgoing SMDR buffer, select the **Delete All OG Calls** radio button.

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115. To comply with the Indian Government Laws and Regulation, this feature is not provided for India Region.

- You can also delete records of outgoing calls selectively, i.e. delete only records of outgoing calls made by a particular extension or a range of extensions, or calls made between a certain period.
  - To delete records of outgoing calls selectively, select the **Delete Selective OG Calls** radio button.
  - To delete calls made by a particular extension or a range of extensions, select the **Delete OG Calls made by Extensions** button.
    - In the first edit box, enter the number of the first extension in the range. In the second box, enter the number of the last extension in the range. If you want to delete the records of a particular extension, enter the same extension number in both fields.
  - To delete the records of outgoing calls made on a particular date or during a certain period, select the **Delete calls made between** radio button. Select the start and end Date, Month and Year for this period. If you want to delete the records of a particular date, enter the same date as start and end.
- Click **Submit**.
- The SMDR buffer will be cleared according to the settings you enabled on this page.

Station Message Detail Recording-Report

ANANT UCS can generate SMDR reports in two modes:

- Online: as and when a call is made or received (see “Station Message Detail Recording-Online”)
  Or
- Offline: whenever required, the records of calls stored in the buffer can be printed.

Generation of call record reports offline, is called SMDR - Report.

You can generate SMDR Report, either

- Manually: The report is generated whenever you want.
  Or
- As per Schedule: The report is generated on a preset Day, Date and Time.

ANANT UCS supports SMDR-Reports on TCP/IP Ethernet Port.

How to configure

To be able to generate SMDR -Report, you must do the following:

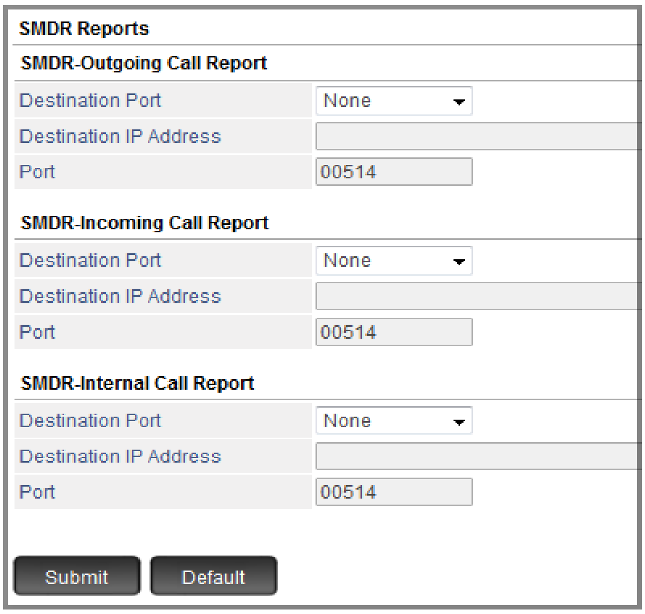
- Enable SMDR Storage in the SMDR buffer. See “Station Message Detail Recording-Storage”.
- Enable and assign the Destination port for Incoming, Outgoing and Internal calls.

Configuring SMDR-Report

- Login as System Engineer.

- Under **Configuration**, click **Station Message Detail Recording**.

- Click **SMDR Reports**



- For **SMDR - Outgoing Call Report**,
    - Select the **Destination Port**. Default: None.
    - If you select Ethernet as the Destination Port,
        - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
        - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.
- For **SMDR - Incoming Call Report**,
    - Select the **Destination Port**. Default: None.
    - If you select Ethernet as the Destination Port,
        - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
        - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.
- For **SMDR - Internal Call Report**,
    - Select the **Destination Port**. Default: None.
    - If you select Ethernet as the Destination Port,
        - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
        - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.
- Click **Submit.**

How to use

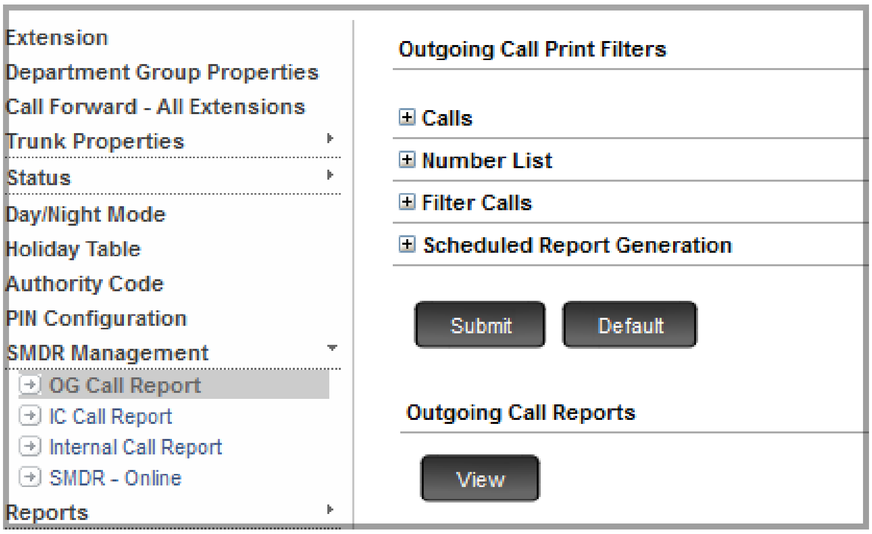
You can print SMDR Report whenever you want or schedule printing of the report from the System Administrator mode using Jeeves.

Printing SMDR-Report

- Login as System Administrator.
- Click **SMDR Management** to expand.

Outgoing Calls

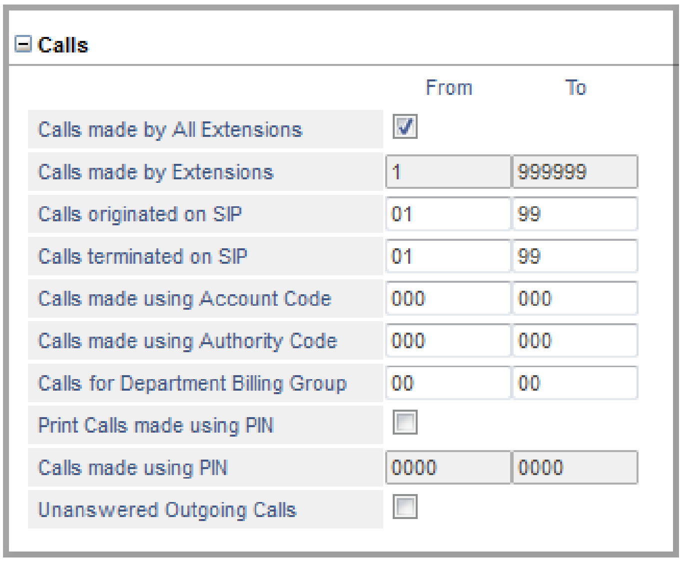
- To print Outgoing Calls with filters, click **OG Call Report**.



Setting Print Filters

To print calls originating and termination on specific trunks/extensions,

- Click **Calls** to expand.



- Set the following filters as desired. You can print records of outgoing calls made by all extensions or by specific extensions: **Calls made by All Extensions**, **Calls made by Extensions.** Default: Calls made by All Extensions

To print records of specific extensions, clear the **Calls made by All Extensions** check box and enter the desired extension range in **From** and **To** in **Calls made by Extension**.

You can also print records of outgoing calls Originating and outgoing calls Terminating on specific SIP Trunks.

You can also print outgoing calls made using **Account Code**, **Authority Codes**, **Calls for Department Billing Groups**, **PIN** and **Unanswered Outgoing Calls**.

*If you have extension numbers beginning with # or *, make sure the range you assign in From and To either have # or *. A mix of both will not work.*

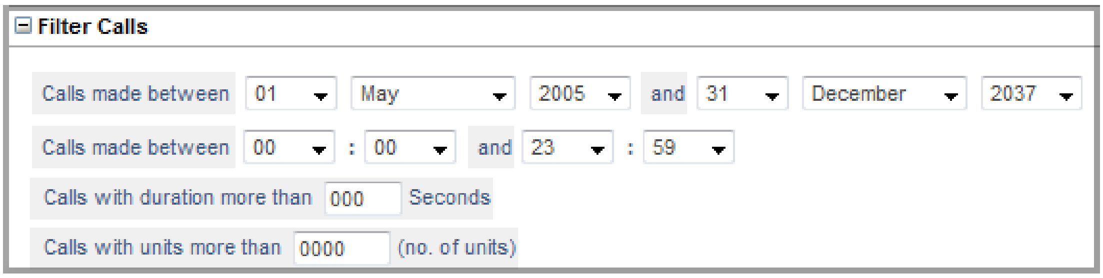
To print outgoing calls made to certain numbers,

- Click **Number List** to expand.

- Enter the desired numbers in the Number List of your choice and select the same number list in **Calls made on called numbers matching with Number List**.

To filter calls according to specific dates, time and duration,

- Click **Filter Calls** to expand,



- To print outgoing calls made on a certain date or between a certain time period, set the filter **Calls made between.** To print calls made on a particular date, select the same Date, Month and Year in both fields.

- To print outgoing calls made at a particular time, set the Hours and Minutes in 24-hour format call in the filter **Calls made between 00: 00 and 23:59**.

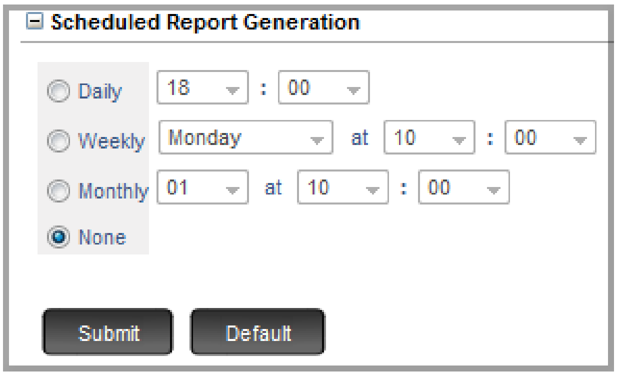
- If you want outgoing calls that exceed as certain duration to be printed, set the filter **Calls with duration more than (sec)** to the desired duration. All outgoing calls with duration greater than this value, will be printed.

- If you want outgoing calls exceeding certain metering units to be printed, set the filter **Calls with units more than (units)** to the desired value. All outgoing calls that have metering units greater than this value will be stored.

- Click **Submit.**

Scheduled Report Generation

- Click **Scheduled Report Generation** to expand.



- To generate SMDR Report of outgoing calls on a particular day, day of the week, or day of the month, set Scheduled Report Generation, as required.

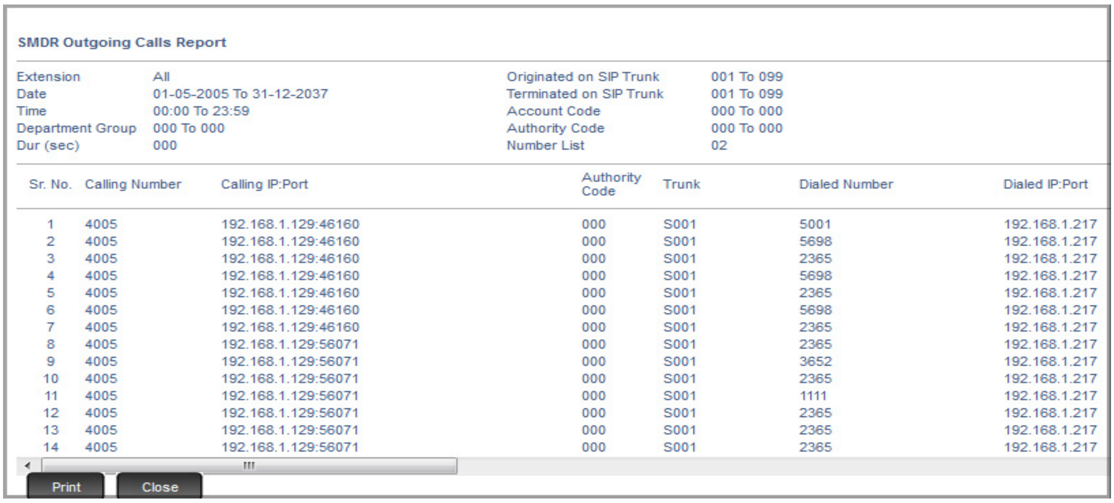
- Click **Submit.**

Manual Report Generation

- You can print the SMDR Report of outgoing calls any time you want. You can print the report on a local printer or the Destination Port of ANANT UCS.

To view/print the report on the local printer,

- Click the **View** button.



SMDR Outgoing Calls Report

Extension	All	Originated on SIP Trunk	001 To 099
Date	01-05-2005 To 31-12-2037	Terminated on SIP Trunk	001 To 099
Time	00:00 To 23:59	Account Code	000 To 000
Department Group	000 To 000	Authority Code	000 To 000
Dur (sec)	000	Number List	02

Sr. No.	Calling Number	Calling IP:Port	Authority Code	Trunk	Dialed Number	Dialed IP:Port
1	4005	192.168.1.129:46160	000	S001	5001	192.168.1.217
2	4005	192.168.1.129:46160	000	S001	5698	192.168.1.217
3	4005	192.168.1.129:46160	000	S001	2365	192.168.1.217
4	4005	192.168.1.129:46160	000	S001	5698	192.168.1.217
5	4005	192.168.1.129:46160	000	S001	2365	192.168.1.217
6	4005	192.168.1.129:46160	000	S001	5698	192.168.1.217
7	4005	192.168.1.129:46160	000	S001	2365	192.168.1.217
8	4005	192.168.1.129:56071	000	S001	2365	192.168.1.217
9	4005	192.168.1.129:56071	000	S001	3652	192.168.1.217
10	4005	192.168.1.129:56071	000	S001	2365	192.168.1.217
11	4005	192.168.1.129:56071	000	S001	1111	192.168.1.217
12	4005	192.168.1.129:56071	000	S001	2365	192.168.1.217
13	4005	192.168.1.129:56071	000	S001	2365	192.168.1.217
14	4005	192.168.1.129:56071	000	S001	2365	192.168.1.217

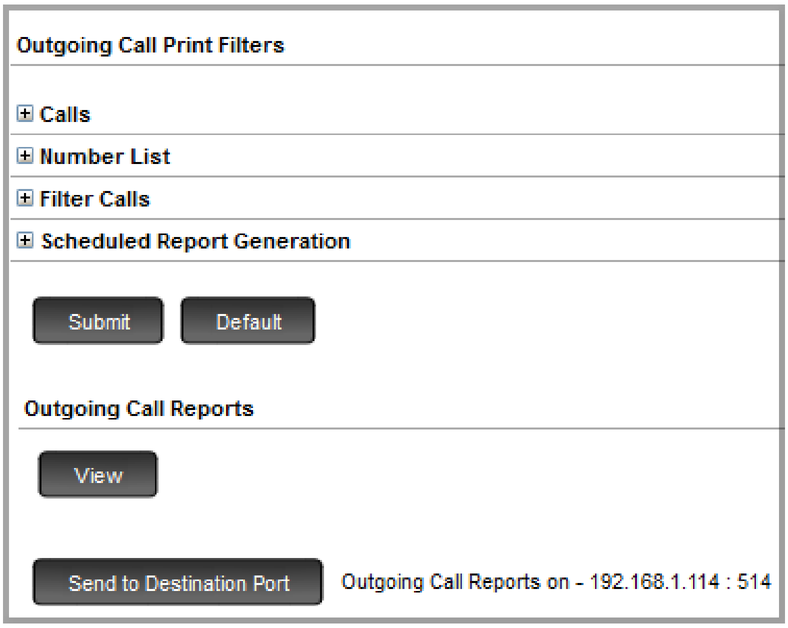
Print Close

- A detailed report is displayed on the screen.

- Click **Print** to print the report.

To print the report on the Destination Port,

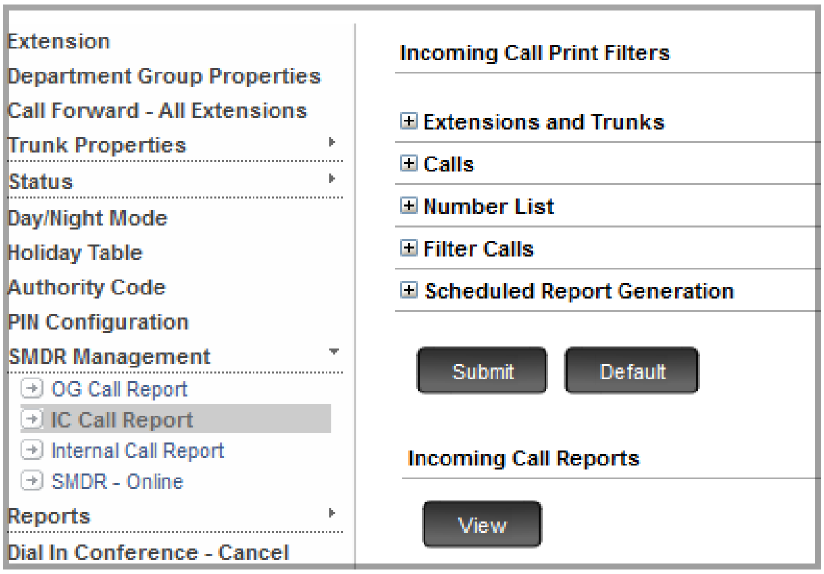
- Click the **Send to Destination Port** button.



- To stop printing, click **Abort** button.

Incoming Calls

- To print Incoming Calls with filters, click **IC Call Report**.



Setting Print Filters

To print calls received on specific trunks/extensions,

- Click **Extension and Trunks** to expand.

- Set the filters as desired. You can print records of incoming calls received on a specific extension or SIP Trunk.

  To print records of specific extensions, clear the **Calls received on All Extensions** check box and enter the desired extension range in **From** and **To**.

- To print calls received only on extensions that have not been assigned Access Codes, select the **Print Calls received on Extensions having blank Access Code** check box. Make sure in **Calls received from Extensions**, you have assigned 0 in the **From** and **To** fields.

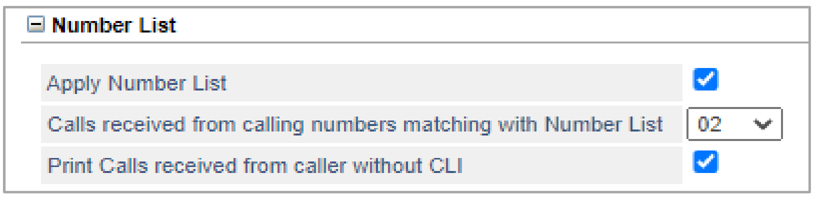
To print incoming calls according to Call Type,

- Click **Calls** to expand.

- You can also print incoming calls of different Call Types: **Normal** calls, calls received using **Auto Attendant**, calls that remained **Unanswered**, **Unanswered calls on Auto Attendant** and calls made using **DISA**.

To print incoming calls received from certain numbers,

- Click **Number List** to expand.



- Select the **Apply Number List** check box and then you can configure **Calls received from calling numbers matching with Number List** and **Print Calls received from caller without CLI.**

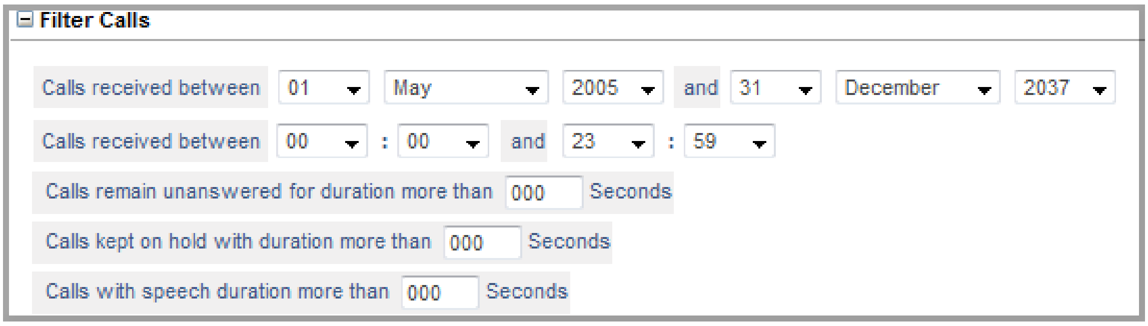
To print outgoing calls received from certain numbers, enter the CLI of these numbers in the Number List of your choice and select the same number list in **Calls received from calling numbers matching with Number List**.

To print calls received without CLI, select the **Print Calls received from caller without CLI** check box.

*If **Accept Anonymous Calls on SIP trunks** is enabled, to view the details in the SMDR Report, make sure you enable the **Print Calls received from caller without CLI** check box.*

To filter calls between specific dates, time or with a specific duration,

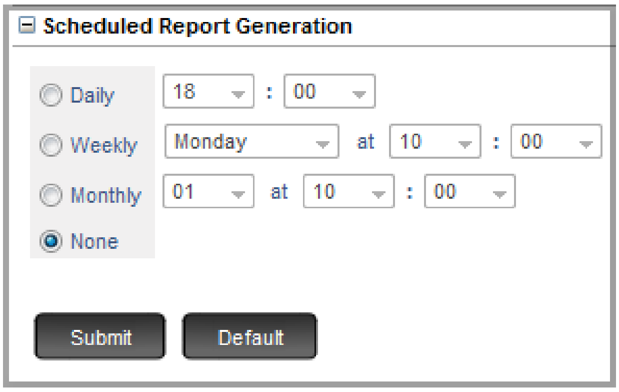
- Click **Filter Calls** to expand.



- To print incoming calls received on a certain date or between a certain time period, set the filter **Calls received between**. To print calls made on a particular date, select the same Date, Month and Year in both fields.
- To print incoming calls received at a particular time, set the Hours and Minutes in 24-hour format call in the filter **Calls received between 00: 00 and 23:59**.
- To print incoming calls that remained unanswered for more than a certain duration, set the filter **Calls remain unanswered for duration more than (sec)**.
- To print calls that were kept on hold for more than a certain duration, set the filter **Calls kept on hold with duration more than (seconds)** to the desired value.
- To print calls with speech duration of a certain duration, set the filter **Calls with speech duration more than (seconds)**
- Click **Submit.**

Scheduled Report Generation

- Click **Scheduled Report Generation** to expand.



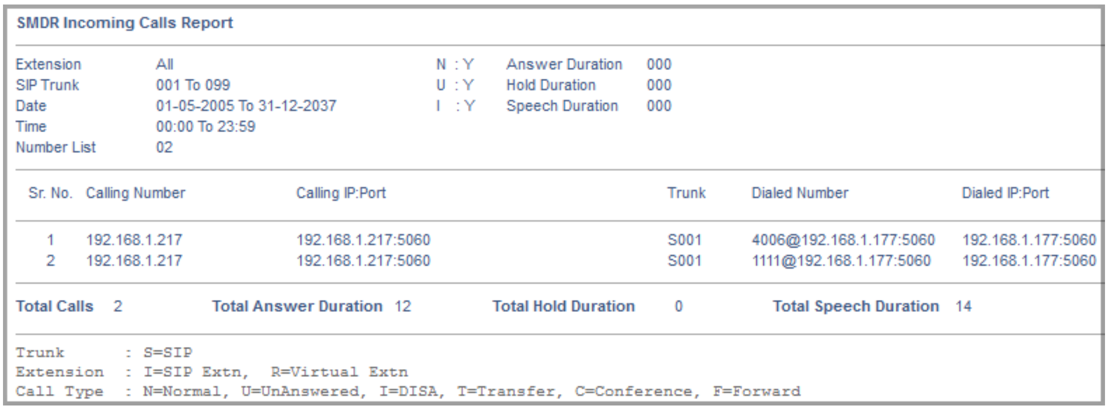
- To generate SMDR Report of incoming calls on a particular day, day of the week, or day of the month, set Scheduled Report Generation, as required.
- Click **Submit**.

Manual Report Generation

- You can print the SMDR Report of incoming calls any time you want. You can print the report on a local printer or the Destination Port of ANANT UCS.

To view/print the report on the local printer,

- Click the **View** button.


SMDR Incoming Calls Report

Extension	All	N : Y	Answer Duration	000
SIP Trunk	001 To 099	U : Y	Hold Duration	000
Date	01-05-2005 To 31-12-2037	I : Y	Speech Duration	000
Time	00:00 To 23:59			
Number List	02			

Sr. No.	Calling Number	Calling IP:Port	Trunk	Dialed Number	Dialed IP:Port
1	192.168.1.217	192.168.1.217:5060	S001	4006@192.168.1.177:5060	192.168.1.177:5060
2	192.168.1.217	192.168.1.217:5060	S001	1111@192.168.1.177:5060	192.168.1.177:5060

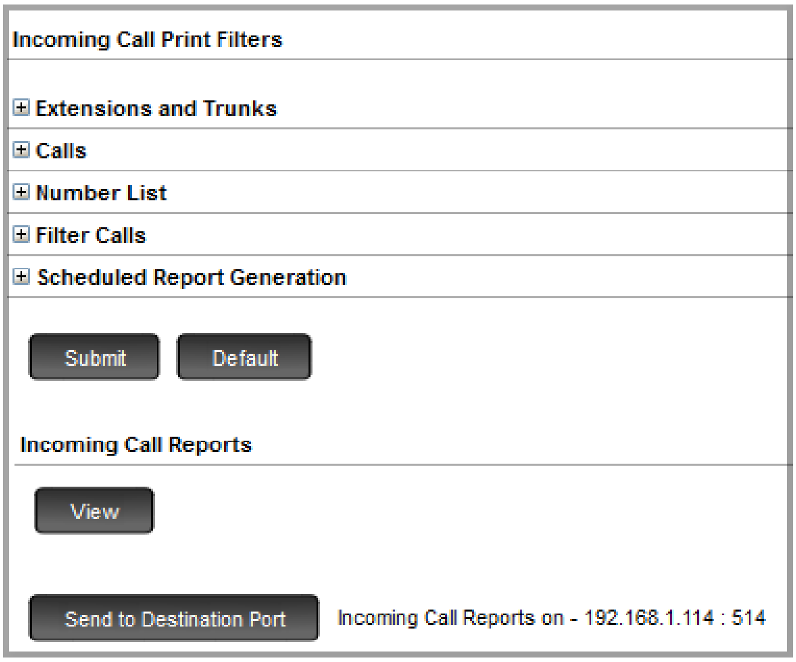
Total Calls 2 **Total Answer Duration** 12 **Total Hold Duration** 0 **Total Speech Duration** 14

```
Trunk      : S=SIP
Extension  : I=SIP Extn,  R=Virtual Extn
Call Type  : N=Normal, U=UnAnswered, I=DISA, T=Transfer, C=Conference, F=Forward
```

- A detailed report is displayed on the screen.
- Click **Print** to print the report.

To print the report on the Destination Port,

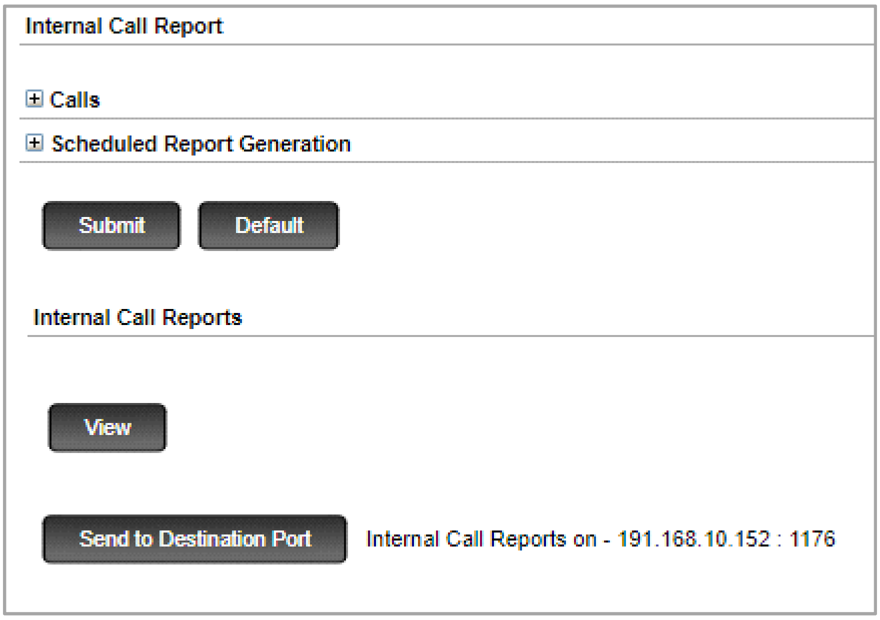
- Click the **Send to Destination Port** button.



- To stop printing, click **Abort** button.

Internal Calls

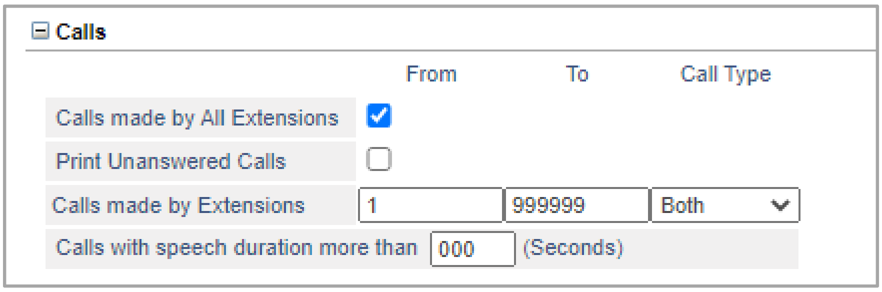
- To print Internal call Report with filters, click **Internal Call Report** link.



Setting Print Filters

To print calls made/received by particular extensions,

- Click **Calls** to expand.



- By default, **Calls made by All Extensions** check box is selected. Hence calls made and calls received by all the extensions will be printed.

- Click **Submit**.

- To print calls made by a particular extension or a range of extensions, clear the **Calls made by All Extension** check box and set the filter **Calls made by Extensions**, by entering the extension numbers in the **From** and **To** fields.

  If you want to print calls made by a particular extension only, enter the same extension number in both **From** and **To** fields.

- Click **Submit**.

  You can also print calls made and calls received by these extensions by selecting the **Call Type**.
  - Select **Both** as Call Type, to print calls made and received by the extensions.
  - Select **Caller** as Call Type, to print only those calls that were made by the extension.
  - Select **Receiver** as Call Type, to print only those calls that were received by the extension.
  - Select **None**, if you do not want to use the Call Type filter.

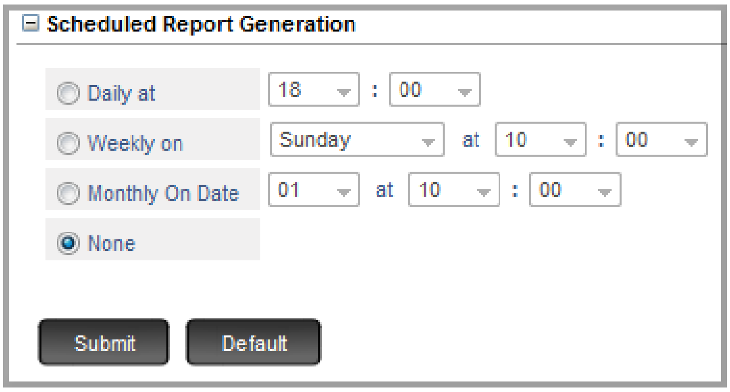
- To print unanswered/missed internal call, select the **Print Unanswered Calls** check box.

- To print calls with speech duration of a certain duration, set the filter **Calls with speech duration more than (Seconds).**

- Click **Submit** to save.

Scheduled Report Generation

- Click **Scheduled Report Generation** to expand.



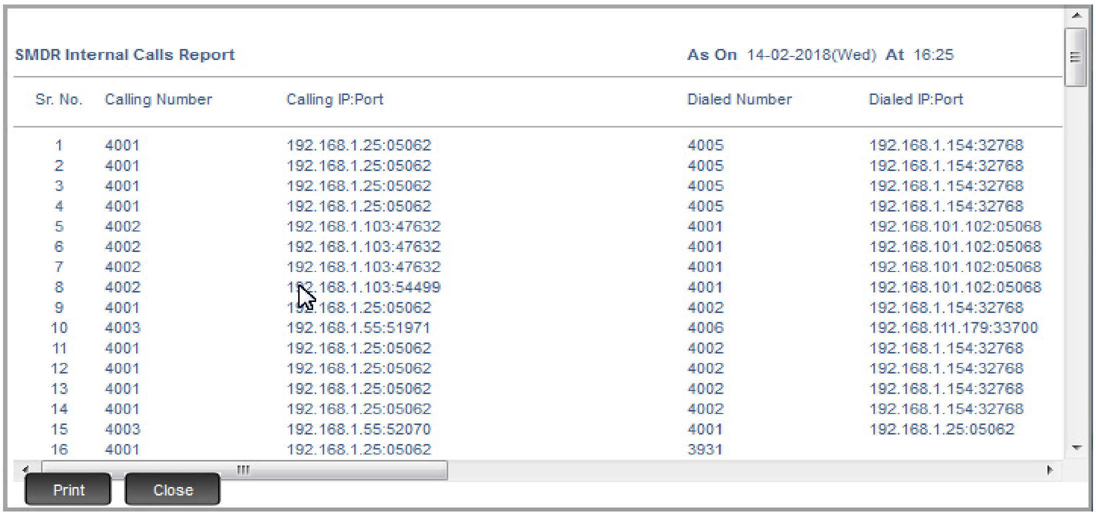
- To generate SMDR Report of internal calls on a particular day, day of the week, or day of the month, set Scheduled Report Generation, as required.
- Click **Submit**.

Manual Report Generation

- You can print the SMDR Report of internal calls any time you want. You can print the report on a local printer or the Destination Port of ANANT UCS.

To view/print the report on the local printer,

- Click the **View** button.



SMDR Internal Calls Report   As On 14-02-2018(Wed) At 16:25

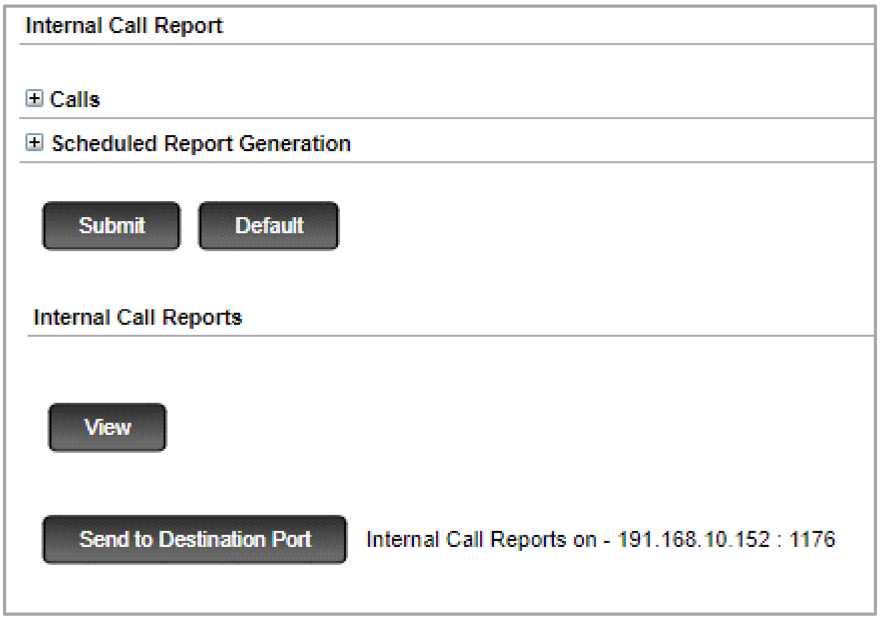
Sr. No.	Calling Number	Calling IP:Port	Dialed Number	Dialed IP:Port
1	4001	192.168.1.25:05062	4005	192.168.1.154:32768
2	4001	192.168.1.25:05062	4005	192.168.1.154:32768
3	4001	192.168.1.25:05062	4005	192.168.1.154:32768
4	4001	192.168.1.25:05062	4005	192.168.1.154:32768
5	4002	192.168.1.103:47632	4001	192.168.101.102:05068
6	4002	192.168.1.103:47632	4001	192.168.101.102:05068
7	4002	192.168.1.103:47632	4001	192.168.101.102:05068
8	4002	192.168.1.103:54499	4001	192.168.101.102:05068
9	4001	192.168.1.25:05062	4002	192.168.1.154:32768
10	4003	192.168.1.55:51971	4006	192.168.111.179:33700
11	4001	192.168.1.25:05062	4002	192.168.1.154:32768
12	4001	192.168.1.25:05062	4002	192.168.1.154:32768
13	4001	192.168.1.25:05062	4002	192.168.1.154:32768
14	4001	192.168.1.25:05062	4002	192.168.1.154:32768
15	4003	192.168.1.55:52070	4001	192.168.1.25:05062
16	4001	192.168.1.25:05062	3931	

- A detailed report is displayed on the screen.

- Click **Print** to print the report.

To print the report on the Destination Port,

- Click the **Send to Destination Port** button.



- To stop printing, click **Abort** button.

Station Message Detail Recording-Online

ANANT UCS can generate report for the calls as and when the call is made and send the report to the computer.

ANANT UCS also supports a Syslog Client for SMDR. The Syslog Client enables the system to send call records in syslog format to the remote 'Syslog Server'. You can view the call records on the remote server. ANANT UCS supports SMDR only on TCP.

SMDR generated as and when calls are made or received is called SMDR **Online** report. In the SMDR **Online** report,

- Each **Internal call** is stored with following fields:
    - Extension who made the call.
    - Extension to which the call was made.
    - Date and time when the call was made.
    - Duration of the call in seconds.
- Each **Outgoing call** is stored with following fields:
    - Extension who made the call.
    - Trunk line port used for the call.
    - Number dialed.
    - Date and time when the call matured.
    - Duration of the call in seconds.
    - Call Units.
    - Call Maturity Type.
    - Call Type (Normal, DISA, ECF etc.).
- Each **Incoming call** is stored with the following fields:
    - The Trunk on which the call is received.
    - Extension number which answered the call.
    - Date and time when the call was received.
    - Calling Number.
    - Hold, speech, ring duration of the call in seconds.

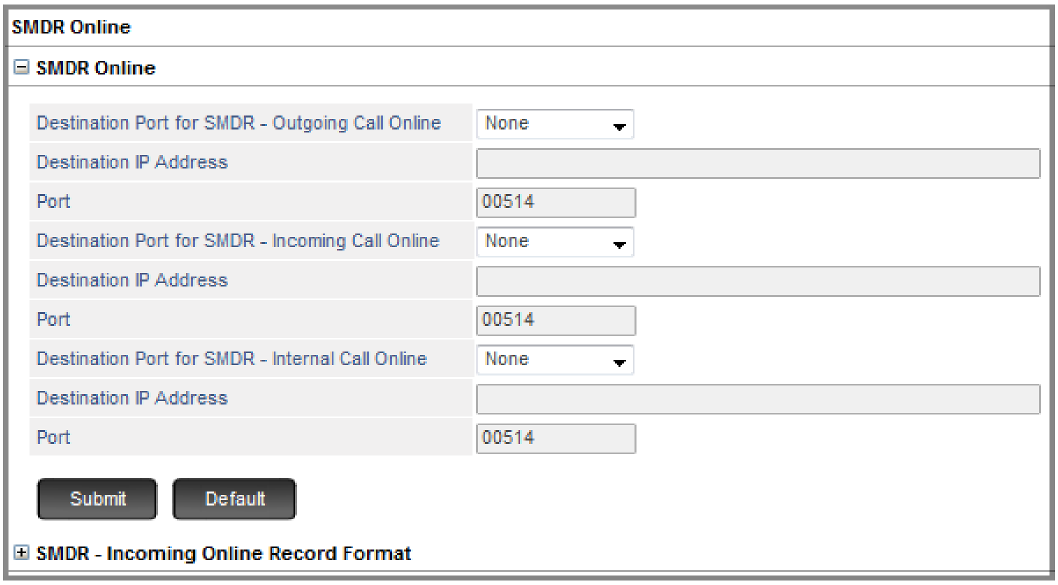
How to configure

To get the **Online** report you must do the following:

- Enable SMDR Storage in the SMDR buffer. See "Station Message Detail Recording-Storage".
- Enable and configure the **Destination IP Address** and **Port** for Incoming, Outgoing and Internal calls.The Online report is sent to this address as soon as the call is completed.
- You may also change the default format for the SMDR Online report for incoming calls, like column position and field length for calling number, speech duration, type of call etc., as required. For this you need to configure the settings of **SMDR Incoming Online Record Format**.

Configuring SMDR-Online

- Login as System Engineer.
- Under **Configuration**, click **Station Message Detail Recording**.
- Click **SMDR Online**.
- Click **SMDR Online** to expand.



For **SMDR - Outgoing Call Online**,

- Select the **Destination Port for SMDR-Outgoing Call Online**. Default: None.
- If you select Ethernet, in **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

For **SMDR - Incoming Call Online**,

- Select the **Destination Port for SMDR-Incoming Call Online**. Default: None.
- If you select Ethernet, in **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

For **SMDR - Internal Call Online**,

- Select the **Destination Port for SMDR-Internal Call Online**. Default: None.
- If you select Ethernet, in **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.

  In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

- To configure Call Record Format for Incoming Calls, click **SMDR Incoming Online Record Format** to expand.

SMDR - Online

Parameter	Start Column No.	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. (Decimal Value)
Serial Number	01	04	Fixed	Right	Yes	032
Increment Counter	00	01	Fixed	N/A	N/A	N/A
Property Code	00	04	Fixed	Left	Yes	032
Connected Number	29	06	Fixed	Right	Yes	032
Trunk Number	23	05	Matrix Format	N/A	Yes	032
Date	36	08	DD-MM-YY	Right	Yes	048
Time	47	08	HH:MM:SS	Right	Yes	048
Answer Duration	56	03	Seconds	Right	Yes	032
Hold Duration	60	03	Seconds	Right	Yes	032
Speech Duration	64	05	Seconds	Right	Yes	032
Dialed Number	00	16	Continuous	Left	N/A	N/A
Calling Number	06	16	Continuous	Left	N/A	N/A
Connected IP:Port	00	48	Fixed	Right	N/A	N/A
Calling IP:Port	00	48	Fixed	Right	N/A	N/A
Dialed IP:Port	00	48	Fixed	Right	N/A	N/A
Remarks	70	02	Fixed	Left	N/A	N/A
Reset Serial Number to 001	Do not Reset					
Reset Increment Counter	Do not Reset					
Property Code	AAA					

- Configure the following Call Record Format parameters as required.

  For each parameter explained briefly below, you can define the column position, field length (i.e. the number of digits), the alignment (whether left aligned or right), and the filler characters, wherever required.

  - **Serial Number:** This is the serial number generated for each call record. Serial numbers are generated from 001 to 999. When serial number '999' is reached, the numbers roll over to 001.

    When this field rolls over, it increments the increment counter.

  - **Increment Counter:** It increments when the serial number counter rolls over. The Increment counter starts from A, ending at Z, and then roll over back to A.

  - **Property Code:** This is the property code, if required. This may be an abbreviation of the property name.

  - **Connected Number:** This is the extension number that answered the call. You can define the column position and the field length for the extension number.

  - **Trunk Number:** This is the number of the trunk on which the call was received.

  - *The Matrix Format occupies 5 character spaces.*
  - *Check-Inn Format occupies 4 character spaces.*
  - *The First Character in the Check-Inn Format is X (Fixed). The remaining three characters show the software port number.*

  - **Date:** The date on which the call was received. The date fill check box is to be enabled.

- *Filler Character field is applicable for Date, Month and Year, i.e. whether the single digit date is to be printed as space-X or 0-X. For example, date = 1 is to be displayed as '1' or '01'.*
- *Where leading zeroes are not required, the date, month and year sub-fields are right aligned and the spaces are filled with character 'space'.*
- *The Date field is not linked to the system set option of Date Format. The system set option of Date format is used, while using features or in configuration reports but not for SMDR Online.*
- **Time:** The time when the call was received. The format of the time field and the time fill check box are to be programmed.
- *Filler Character field is applicable for Hours, Minutes and Seconds i.e. whether the single digit hour is to be printed as space-X or 0-X. For example, hour = 1 is to be displayed as '1' or '01'.*
- *In case leading zeroes are not required, Date, Month and Year sub-fields are right aligned and the spaces are filled with character 'space'.*
- **Answer Duration:** The time after which the call was answered. Program the duration unit and the duration fill check box.
- **Hold Duration:** The time for which the call was put on hold.
- **Speech Duration:** The time for which the call was in speech with the extension.

When Duration Unit = Minutes, the rounding off to the nearest whole number is done. For seconds <= 30, Minute is not incremented. For seconds > 30, minute is incremented.

- **Dialed Number:** This is applicable only for calls received on SIP trunks. The number dialed by the caller is referred to as Dialed Number.
- **Calling Number:** This is the number of the Caller.
- **Connected IP:Port:** This is the IP Address:Port of the called party.
- **Calling IP:Port:** This is the IP Address:Port of the calling party.
- **Dialed IP:Port:** This is the IP Address:Port of the called party.
- **Remarks:** You may use this for indicating the Type of Call.
- **Reset Serial Number to 001:** The Serial number counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'Do not Reset' is selected, which means the serial number counter will not be automatically reset.
- **Reset Increment Counter:** The Increment Counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'Do not Reset' is selected, which means the serial number counter will not be automatically reset.
- **Property Code:** This may be an abbreviation of the property name.

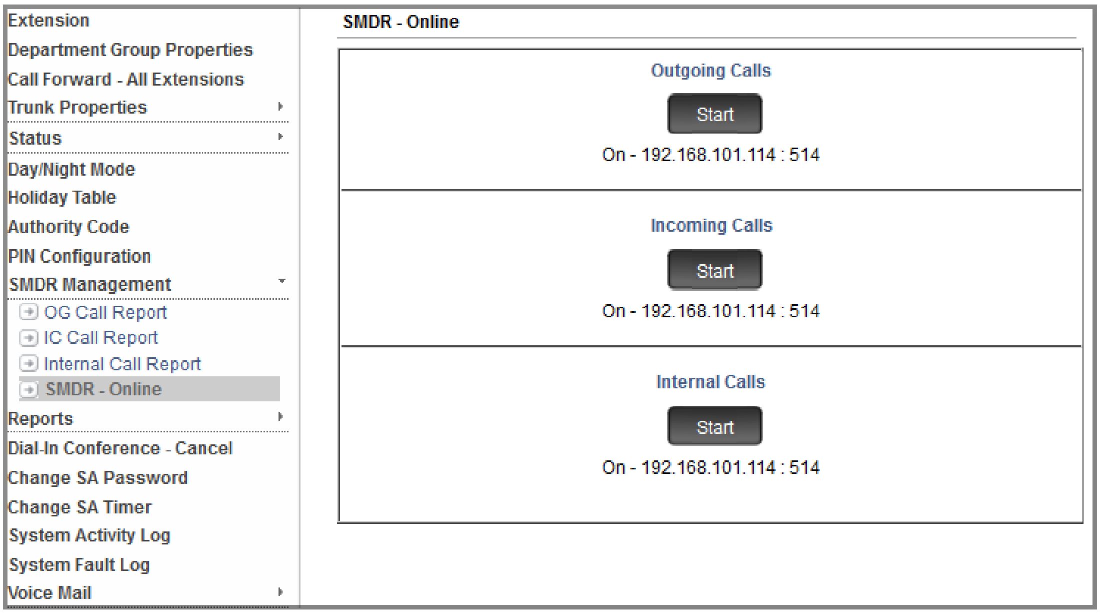
- Click **Submit.**

How to use

You can start and stop SMDR Online report from the System Administrator mode using Jeeves.

To start/stop Online report,

- .Login as System Administrator.
- Under **SMDR Management**, click **SMDR-Online.**



- To start SMDR Online for Outgoing Calls, Incoming Calls and Internal Calls, click the **Start** button.
- To stop SMDR Online for any of these call types, click **Abort** button.

Station Message Detail Recording-Posting

The Station Message Detail Record (SMDR)-Posting feature of ANANT UCS is used for interfacing the system with a third party Call Accounting Software (CAS).

When ANANT UCS is interfaced with a third party Call Accounting Software (CAS) to determine the cost of the calls made by the extension users, the system uses SMDR-Posting to send to CAS call record details, like number to which the call was made by the extension user, number of the extension from which the call was made, the date and time when the call was made, the duration of the call, metering pulses incurred for the call, etc. On receipt of this information, the CAS calculates the cost of the call for billing.

As different CAS interfaces support different protocols, ANANT UCS offers the flexibility to send call detail records using the protocol supported by CAS. ANANT UCS supports as many as 16 different widely-used CAS protocols such as, Holidex, Hobic, Micros A, Micros B, Comm One, Call-Inn, Bell-HOBIC, XIOX, RSI and others.

Each posting protocol has its own handshaking protocol and call record format. You may configure any one of these depending upon the protocol supported by CAS you have interfaced with ANANT UCS. It is also possible to customize the posting protocol to match the settings required by the CAS you have interfaced.

ANANT UCS supports SMDR-Posting on TCP/IP Ethernet Port.

SMDR-Posting sends outgoing call records only.

SMDR-Posting Protocols

ANANT UCS supports different SMDR posting protocols from the system to CAS. The flow of messages between ANANT UCS and the protocols of CAS Interface (Matrix and Blind Send) are described below:

Matrix

- **Positive Response from CAS**

ANANT UCS to CAS	CAS to ANANT UCS
<STX> -Call Record-<ETX> <BCC>	
	ACK

- **Negative Response from CAS**

ANANT UCS to CAS	CAS to ANANT UCS
<STX> -Call Record-<ETX> <BCC> and wait for Response to Data Timeout (sec), default 3 sec.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK

ANANT UCS to CAS	CAS to ANANT UCS
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK

ANANT UCS will make 5 attempts (default value of *Data Transfer Retry Count - on Negative Response*) to send the message after a regular interval of 3 seconds (default value of *Data Transfer Retry Timer - on Negative Response*). If the ACK is still not received from the CAS, ANANT UCS will proceed to the next message.

- **Busy Response from CAS**

ANANT UCS to CAS	CAS to ANANT UCS
<STX> -Call Record-<ETX> <BCC> and wait for Response to Data Timeout (sec), default 3 sec.	
	NAK (CAS responds but cannot accept at this time)
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on Negative Response, default 3 seconds.	
	NAK

ANANT UCS will make 5 attempts (default value of *Data Transfer Retry Count - on Negative Response*) to send the message after a regular interval of 3 seconds (default value of *Data Transfer Retry Timer - on Negative Response*). If the ACK is still not received from the CAS, ANANT UCS will proceed to the next message.

- **No Response from CAS**

ANANT UCS to CAS	CAS to ANANT UCS
<STX> -Call Record-<ETX> <BCC> and wait for Response to Data Timeout (sec), default 3 sec.	
	(no response)
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	
Retransmit <STX> -Call Record-<ETX> <BCC> and wait for Data Transfer Retry Timer (sec) - on No Response, default 3 seconds.	

ANANT UCS will make 5 attempts (default value of *Data Transfer Retry Count - on No Response*) to send the message after a regular interval of 3 seconds (default value of *Data Transfer Retry Timer - on No Response*). If the ACK is still not received from the CAS, ANANT UCS send the new message to CAS.

Blind Send

If you select this protocol as the SMDR-OG Posting Protocol, ANANT UCS sends the call details without waiting for any response from the CAS. Each record is sent with the End of Packet Character.

Customized

If you select this protocol as the SMDR-OG Posting Protocol, ANANT UCS provides you the flexibility to set the values for the OG Handshaking Protocol and the OG Online Call Record Format as per your requirement.

Call Detail Record Formats

The default Call Detail Record formats for Blind Send and Matrix are given below.

Matrix

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Serial Number	01	04	Fixed	Right	Yes	032	Every 6 hours it is cleared to 001. (Starting from mid-night 00:00:00)
Increment Counter	00	01	Fixed	Left	NA	NA	Every 6 hours it is cleared to A. (Starting from mid-night 00:00:00)
Property Code	00	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	05	Fixed	Right	Yes	032	
Authority Code	00	03	Fixed	Left	NA	NA	
Trunk Number	12	05	Matrix Format	Left	Yes	032	
Date	37	10	DD-MM-YYYY	Right	Yes	032	
Time	48	08	HH:MM:SS	Right	Yes	032	
Duration	057	005	Seconds	Right	Yes	032	
Units	063	004	Fixed	Right	Yes	032	
Amount	068	007	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Space	032	Country Specific

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Call Type Indicator	000	001	Fixed	Right	NA	NA	As per the Call Type Indicator table programmed by the SE. The SE should program L = local, F=International and Space shall be used for long distance.
Location	000	005	Fixed	Right	NA	NA	
Called Number	18	19	Continuous	Left	Space	NA	
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	
Remarks	76	02	Fixed	Left	Space	NA	
Reset Serial Number to 001	Do not Reset						
Staring Character - Increment Counter	A						
Reset Increment Counter	Do not Reset						
Prefix String Required	No						
Property Code	000						
Currency Symbol (Enter Decimal Value)	003 000 000 000 000 000 000 000						

AST

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Serial Number	01	04	Fixed	Right	Yes	032	Every 6 hours it is cleared to 001. (Starting from mid-night 00:00:00)
Increment Counter	00	01	Fixed	Left	NA	NA	Every 6 hours it is cleared to A. (Starting from mid-night 00:00:00)
Property Code	00	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	05	Fixed	Right	Yes	032	
Authority Code	00	03	Fixed	Left	NA	NA	
Trunk Number	12	05	Matrix Format	Left	Yes	032	
Date	37	10	DD-MM-YYYY	Right	Yes	032	MM/DD
Time	48	08	HH:MM:SS	Right	Yes	032	HH:MM
Duration	057	005	Seconds	Right	Yes	032	Duration is in Minutes.
Units	063	004	Fixed	Right	Yes	032	
Amount	068	007	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Space	032	$

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Call Type Indicator	000	001	Fixed	Right	NA	NA	As per the Call Type Indicator table programmed by the SE. The SE should program L = local, F=International and Space shall be used for long distance.
Location	000	005	Fixed	Right	NA	NA	
Called Number	18	19	Continuous	Left	NA	NA	Area code, Exchange code and Subscriber Number separated by dash. Space is sent in place of Area Code and first dash if area code is not present.
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	--
Remarks	76	02	Fixed	Left	NA	NA	--
Reset Serial Number to 001	Do not Reset						
Staring Character - Increment Counter	A						
Reset Increment Counter	Do not Reset						
Prefix String Required	No						
Property Code	000						

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Currency Symbol (Enter Decimal Value)	000 000 000 000 000 000 000 000						

Blind Send

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Serial Number	01	04	Fixed	Right	Yes	032	
Increment Counter	00	01	Fixed	Left	NA	NA	
Property Code	00	04	Fixed	Left	Yes	032	As per the Programmed String
Extension Number	06	05	Fixed	Right	Yes	032	
Authority Code	00	03	Fixed	Left	NA	NA	
Trunk Number	12	05	Matrix Format	Left	Yes	032	
Date	37	10	DD-MM-YYYY	Right	Yes	032	
Time	48	08	HH:MM:SS	Right	Yes	032	
Duration	057	005	Seconds	Right	Yes	032	
Units	063	004	Fixed	Right	Yes	032	
Amount	068	007	Currency with Decimal Point	Right	Yes	032	Format is DDD.CC
Currency	000	001	Fixed	Right	Space	032	Country Specific

Parameter	Start Column Number	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. Decimal Value	Remarks
Call Type Indicator	000	001	Fixed	Right	NA	NA	As per the Call Type Indicator table programmed by the SE. The SE should program L = local, F=International and Space shall be used for long distance.
Location	000	005	Fixed	Right	NA	NA	
Called Number	18	19	Continuous	Left	Space	NA	
PIN	00	04	Fixed	Right	Yes	032	--
Account Code	00	04	Fixed	Right	Yes	032	
Remarks	76	02	Fixed	Left	Space	NA	
Reset Serial Number to 001	Do not Reset						
Staring Character - Increment Counter	A						
Reset Increment Counter	Do not Reset						
Prefix String Required	No						
Property Code	000						
Currency Symbol (Enter Decimal Value)	013 010 000 000 000 000 000 000						

Customized SMDR-Posting Protocol

You can use the Customized SMDR Posting Protocol to match the settings required by the CAS you have interfaced. When you use Customized SMDR-Posting Protocol, you can customize the Call Detail Record format to match your requirement.

When the Call Detail Record format is customized, if there is a gap between two fields, these fields will be 'space' (ASCII-32).

It is also possible to customize the posting protocol to match the settings required by the CAS you have interfaced.

Setting up CAS Interface

You can setup the CAS Interface on the Ethernet Port:

To setup the CAS Interface on the Ethernet Port, the following functional components are required to make the interface work:

- A PC, having the CAS server application software running, with a spare LAN/WAN Port Or any free LAN Port of the LAN Switch.
- The CAS Software (not supplied by Matrix).
- ANANT UCS (supplied by Matrix).

  Now, connect the Ethernet (LAN/WAN) Port of ANANT UCS with the Ethernet Port of the PC (on which CAS server application is running) or to one of the Ethernet port of the LAN Switch, if the CAS server is in the same LAN.

How to configure

Configuring the SMDR-Posting feature involves the following steps:

- Enabling storage of Outgoing (OG) SMDR. By default, OG SMDR storage is enabled. Refer "Station Message Detail Recording-Storage".

- Selecting the appropriate SMDR-OG Posting Protocol to be used.

- Selecting the Destination Port for SMDR-Posting.

- Refining the Handshake parameters, if required.

- Refining Call Detail Record format, if required.

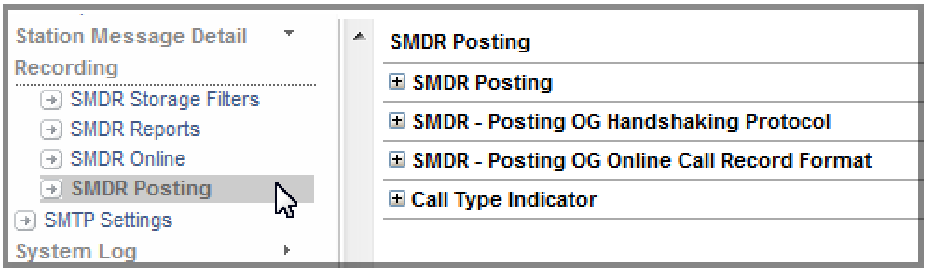
- Starting SMDR-Posting process.

Configuring SMDR Posting

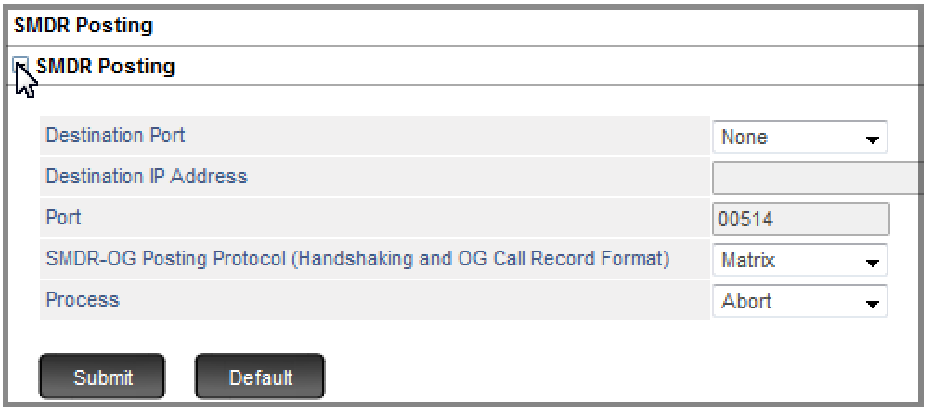
- Log in as System Engineer.
- Under **Configuration**, click **Station Message Detail Recording**.

Selecting SMDR-Posting Protocol for CAS

- Click the **SMDR Posting**. The SMDR-Posting parameters page opens.



- Click **SMDR-Posting** to expand.



- Select the **Destination Port** on which the SMDR Posting is set up. You can select **None, Ethernet.** Default: None.

  If you select **Ethernet**,

  - In **Destination IP Address**, enter the IP Address of the PC on which the CAS server application software is running, that is, where ANANT UCS should post SMDR.
    Both IPv4 and IPv6 addresses are supported.
  - In **Port**, enter the port of the PC on which the CAS server application software is running, that is, where ANANT UCS should post SMDR. Valid port range is: 5000; 514; 1025 to 65535.

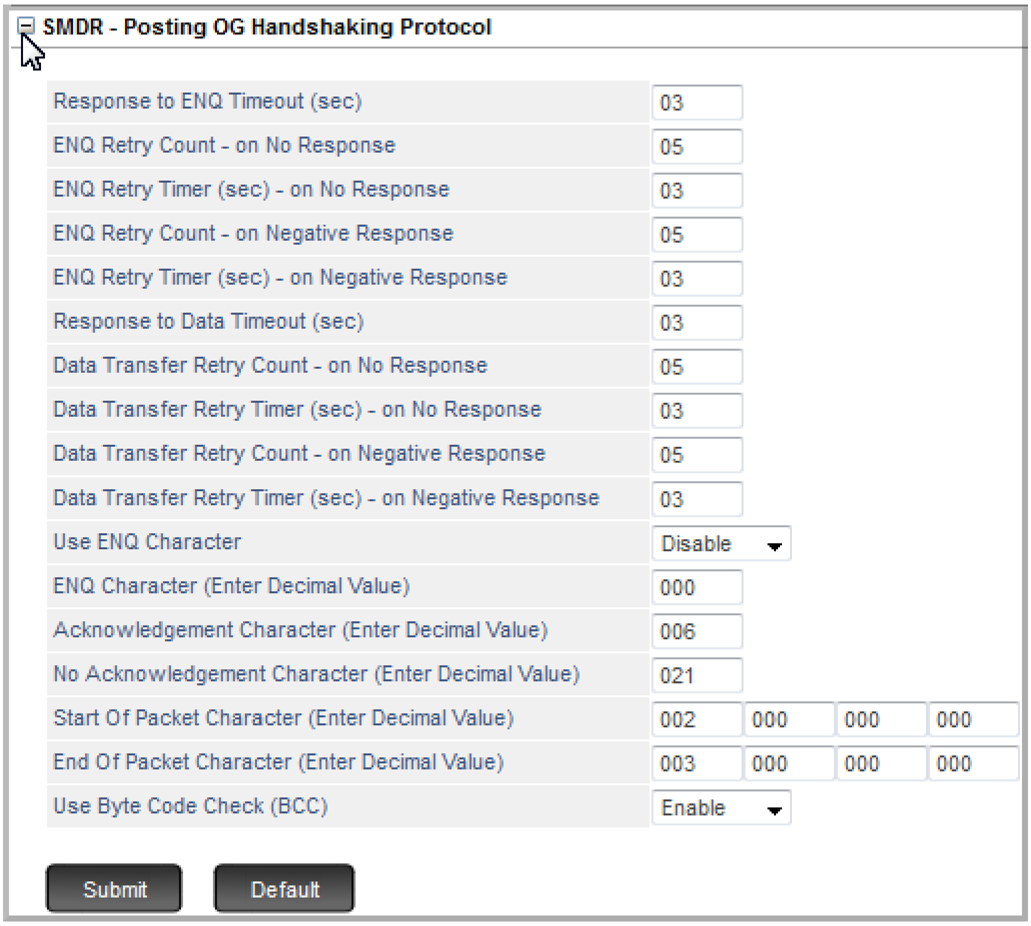
- In the **SMDR-OG Posting Protocol** (Handshaking and OG Call Record Format) drop down list, select the appropriate protocol to be used. Default: Matrix.
- To start SMDR posting, select **Process** as **Start**.

Refining Handshake Parameters

You may need to refine some of the Handshake parameters of the selected SMDR-Posting protocol, that is, change the factory default values of the protocol, to match the software requirements of the CAS being used in the organization. Refer the below table for default values of each protocol supported by ANANT UCS.

To refine Handshake Parameters,

- Click **SMDR-Posting OG Handshaking Protocol**.



Configure the following parameters as required:

- **Response to ENQ Timeout (sec):** The time for which the sender waits for a response to ENQ from the receiver.

- **ENQ Retry Count - on No Response:** The number of times the sender should send ENQ before dropping the process, in case response is not received for the last message sent.

- **ENQ Retry Timer (sec) - on No Response:** The time after which the sender should sent the ENQ again, in case the response is not received for the last message sent.

- **ENQ Retry Count - on Negative Response:** The number of times the sender should send ENQ before dropping the process, in case of a negative response received for the last message sent.

- **ENQ Retry Timer (sec) - on Negative Response:** The time after which the sender should sent the ENQ again.

- **Response to Data Timeout (sec):** The time for which the sender waits for a response to data from the receiver.

- **Data Transfer Retry Count - on No Response:** The number of times the sender should send ENQ before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem while sending the data.

- **Data Transfer Retry Timer (sec) - on No Response:** The time after which the sender should send the ENQ again before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem in sending the data.

- **Data Transfer Retry Count - on Negative Response:** The number of times the sender should send ENQ before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem in sending the data.

- **Data Transfer Retry Time (sec) - on Negative Response:** The time after which the sender should sent the ENQ again before dropping the process. This parameter is used when ACK is received against ENQ and there is some problem in sending the data.

- **Use ENQ Character:** This is enabled if the protocol uses ENQUIRE (ENQ) Signal.

- **ENQ Character:** This parameter signifies the ASCII character (Single Character) used to send ENQUIRE (ENQ) signal to the receiver.

- **Acknowledgement Character:** This parameter signifies the ASCII character (Single Character) used by the receiver to acknowledge the receipt of the Link Control Character/Message Data.

- **No Acknowledgement Character:** This parameter signifies the ASCII character (Single Character) used by the receiver to dis-acknowledge the receipt of the Link Control Character/Message Data.

- **Start of Packet Character:** A string of four ASCII characters used by the receiver to indicate Start of Packet. Each ASCII character is from 000 to 252. Start of Packet may be of one character only, in this case the string should be completed by programming remaining three characters with ASCII Null Character (000).

- **End of Packet Character:** A string of four ASCII characters used by the receiver to indicate End of Packet. Each ASCII character is from 000 to 252. End of Packet may be of one character only, in this case, the string should be completed by programming the remaining three characters should be programmed as ASCII Null (000).

- **Use Byte Code Check (BCC):** This flag is to be enabled when the protocol uses BCC Signal.

- Click **Submit**.

Refining Call Detail Record Format Parameters

The Call Detail Record (CDR) format for the selected SMDR-Posting protocol can also be refined to match the software requirements of the CAS being used by the organization.

This may be required if you have selected a 'customized' protocol. To refine Call Record Format,

- Click **SMDR-Posting OG Online Call Record Format** to expand.

SMDR - Posting OG Online Call Record Format

Parameter	Start Column No.	Field Length	Format	Alignment	Filler Char. Required?	Filler Char. (Decimal Value)
Serial Number	001	004	Fixed	Right	Yes	032
Increment Counter	000	001	Fixed	Left	N/A	N/A
Property Code	000	004	Fixed	Left	Yes	032
Calling Number	006	005	Fixed	Right	Yes	032
Authority Code	000	003	Fixed	Left	N/A	N/A
Trunk Number	012	005	Matrix Format	Left	Yes	032
Date	037	010	DD-MM-YYYY	Right	Yes	032
Time	048	008	HH:MM:SS	Right	Yes	032
Duration	057	005	Seconds	Right	Yes	032
Units	063	004	Fixed	Right	Yes	032
Amount	068	007	Currency with Decimal Point	Right	Yes	032
Currency	000	001	Fixed	Right	Yes	032
Call Type Indicator	000	001	Fixed	Right	N/A	N/A
Location	000	005	Fixed	Right	N/A	N/A
Connected Number	018	019	Continuous	Left	N/A	N/A
PIN	000	004	Fixed	Right	Yes	032
Account Code	000	004	Fixed	Right	Yes	032
Dialed Number	000	019	Continuous	Left	N/A	N/A
Connected IP:Port	000	048	Fixed	Right	N/A	N/A
Calling IP:Port	000	048	Fixed	Right	N/A	N/A
Dialed IP:Port	000	048	Fixed	Right	N/A	N/A
Remarks	076	002	Fixed	Left	N/A	N/A

- **Serial Number:** This is the serial number generated for each call record. Serial numbers are generated from 000 to 999. When serial number '999' is reached, the numbers roll over to 000.

Serial Number starts from 1 and not 0.
When this field rolls over, it increments the increment counter.

- **Increment Counter:** It increments when the serial number counter rolls over. The Increment counter starts from A, ending at Z, and then roll over back to A.

- **Property Code:** This is the property code required by the CAS used in the organization. It is a string of alphanumeric characters and is to be terminated with #*. This field has a maximum of 128 alphanumeric characters.

  - *You must program this string keeping in mind the field length used by the selected/customized posting protocol.*
  - *The default value of the default Property Code String has been set as '000', as at least two known protocols use this field. You can set a different value here and the new value will appear in the CDR record, irrespective of the protocol type selected.*
  - *If XIOX protocol has been selected as SMDR-OG Posting Protocol (Handshaking and OG Call Record Format), you should program Property Code as HTL.*

- **Calling Number:** This is the number from which the call was made. You can define the column position and the field length of the Calling number in the Call Detail Record.

- Authority Code: This is a unique password-protected code which is assigned to the extension user.
- **Trunk Number:** This is the number of the trunk from which the call was made.

- *The Matrix Format occupies 5 character spaces.*
- *Check-Inn Format occupies 4 character spaces.*
- *The First Character in the Check-Inn Format is X (Fixed). The remaining three characters show the software port number.*

- **Date:** The date on which the call was made. The date fill flag is to be enabled.

- *Filler Character field is applicable for Date, Month and Year, that is, whether the single digit date is to be printed as space-X or 0-X. For example, date = 1 is to be displayed as '1' or '01'.*
- *Where leading zeroes are not required, the date, month and year sub-fields are right aligned and the spaces are filled with character 'space'.*
- *The Date field is not linked to the global flag of Date Format. The global Flag of Date format is used, while using features or in configuration reports but not in CAS. This is because the date format used by the CAS is not the same as used by the users of the system.*

- **Time:** The time when the call was made. The format of the time field and the time fill flag are to be programmed.

- *Filler Character field is applicable for Hours, Minutes and Seconds, that is, whether the single digit hour is to be printed as space-X or 0-X. For example, hour = 1 is to be displayed as '1' or '01'.*
- *In case when leading zeroes are not required, Date, Month and Year sub-fields are right aligned and the spaces are filled with character 'space'.*

- **Duration:** The duration of each call. Program the duration unit and the duration fill flag.

When Duration Unit = Minutes, the rounding off to the nearest whole number is done. For seconds <= 30, Minute is not incremented. For seconds > 30, minute is incremented.

- **Units:** The duration of the call interpreted in terms of units. The number of units depends on the Pulse Rate. The number of units is derived from the Call Unit = Call duration in seconds/Pulse rate in seconds.
- **Amount:** This is the Amount of the call. Program the amount format and the fill flag.

- *Filler Character field is applicable for both the sub fields of Amount viz. Rupees/Paisa, that is, whether the single digit Rupee is to be printed as space-X or 0-X. For example, Rupee = 1 is to be displayed as '1' or '01'. Where leading zeroes are not required, the Rupee and Paisa are right aligned and the spaces are filled with character 'space'.*
- *When Amount Format = Higher Currency, rounding to nearest whole number is done. For Lower Currency <= 50, Higher Currency is not incremented and for Lower currency > 50, Higher Currency is incremented.*

- **Currency:** This is the symbol of the currency in which the Amount is charged. A maximum of 8 ASCII Characters are allowed.

- *Generally, Currency Symbol field prefixes to Amount field. Hence, to comply with various CDR formats, it is recommended that the column position of Currency Symbol and Amount field should be programmed properly.*
- *You can change the Currency Symbol used in the OG-SMDR Format.*

- **Call Type Indicator:** This indicates the type of call made, that is, whether local, international, information, etc.

  Your must program the Number String, the Text String and its Meaning as explained in following table:

Number Index	Number String	Text String	Meaning
01	0	LD	Long Distance
02	95	IC	Inter Circle
03	197	INFO	Information
04	0	INTL	International
:	:	:	:
36	2	L	Local

The Text String is a string of Alphanumeric characters. Number String is of a maximum 4-digits.

The Number Index is kept as '36' as one of the SMDR-OG Posting protocols, INN-FORM XL supports 24 different types of calls.

By default, all the entries in this table are blank.

You are advised to program the first 10 entries of this table as below if the selected posting protocol is Bell Hobic or XIOX.

Number Index	*Number String*	*Text String*	*Meaning*
01	*1*	*A*	
02	*2*	*A*	
03	*3*	*A*	
04	*4*	*A*	
05	*5*	*A*	
06	*6*	*A*	
07	*7*	*A*	
08	*8*	*A*	
09	*9*	*A*	
10	*0*	*A*	
:	*:*	*:*	
36	*Blank*	*Blank*	*Blank*

You are advised to program the first 11 entries of this table as below, if the selected posting protocol is Holidex or Hobic.

Number Index	*Number String*	*Text String*	*Meaning*
01	*1*	*L*	
02	*2*	*L*	
03	*3*	*L*	
04	*4*	*L*	
05	*5*	*L*	
06	*6*	*L*	
07	*7*	*L*	
08	*8*	*L*	
09	*9*	*L*	
10	*0*	*L*	
11	*0*	*F*	*International*
:	*:*	*:*	
36	*Blank*	*Blank*	*Blank*

- You are advised to use default (that is, Blank) table, if the selected protocol is Hilton, as Hilton uses blank entries in this field which is 12 bytes long.
- The Text String should preferably be same as Field Length. If not, the remaining spaces will be filled with character 'Space'. If the Field length is less than the Text string characters, then the number of text characters equal to the Field length will be printed.

- **Location:** This column indicates the location of the external number to which the call was made.

- *The system detects the location from the called location programmed in the Area and Country Code Tables.*
- *Called Location is programmed as one of the parameters of the Area Code Table and Country Code Table. Depending upon the prefix dialed, the Location string is picked up from either Country Code table or Area Code table.*
- *Called Location is not displayed for Local Calls.*
- *The Called Location parameter in the Country Code table and Area Code table is of 8 Characters.*
- *If the number of characters in the field Called Location is more than Field length then the remaining characters will not be printed (overlapped by next field).*
- *If the number of characters in the field Called Location is less than Field length then the remaining characters in the field Called Location will be filled by spaces.*

- **Connected Number:** This is the external number to which the call was made.

- *One way to separate the Connected party number is by Area Code, Exchange code and Subscriber Number. This is difficult in an Open numbering system, in which the field size of area code, exchange code are not standard but vary from two digits to four digits (for example, the Area code for 'Mumbai' is of 2 digits, whereas that of 'Vadodara' is 3 digits).*
- *In the Closed numbering system, the Area Code, Exchange Code and the Subscriber number are of fixed length. In such case, including '-' in the called party number is not difficult. Hence, '-' is put in the called party number. The Connected party number is assumed to be of 10 digits. The first '-' is placed after four digits, counting from the right. The second '-'is placed after seven digits, counting from the right. If the dialed number is a local number of 7 digits then the second '-'is not placed. Also, the remaining three digits are not placed, but filled with character 'space'.*
- *In this case, even if the call is made to a geographical area where open numbering system is followed, '-' is placed in the same way.*

- PIN: This is a unique four digit code which is assigned to the Extension user.
- Account Code: This is the Account Code (Refer Note4) using which the call was made.
- **Dialed Number:** This is applicable only for calls received on SIP trunks. The number dialed by the caller is referred to as Dialed Number.
- **Connected IP:Port:** This is the IP Address:Port of the called party.
- **Calling IP:Port:** This is the IP Address:Port of the calling party.
- **Dialed IP:Port:** This is the IP Address:Port of the called party.
- **Remarks:** This column indicates the details of the call; whether it was a DISA call, Auto Redial Call, type of call maturity.

  Fixed Characters are used to indicate the type of call, call details, etc. The notations for the Remarks field are:

D	DISA Call
A	Auto Redial Call
I	Connect

- **Reset Serial Number to 001:** The Serial number counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'No Compulsory Reset' is selected, which means the serial number counter will not be automatically reset.
- **Starting Character - Increment Counter:** Specify the starting character of the increment counter as the serial number rolls over, in this field.
- **Reset Increment Counter:** The Increment Counter can be reset to 001 after 24 hours (from 00:00 HH:MM) or every 6 hours. By default, 'No Compulsory Reset' is selected, which means the serial number counter will not be automatically reset.
- **Prefix String Required:** This flag is to be programmed if the prefix string 0ac1 is to be sent when interfacing with OG-SMDR Posting Protocol.

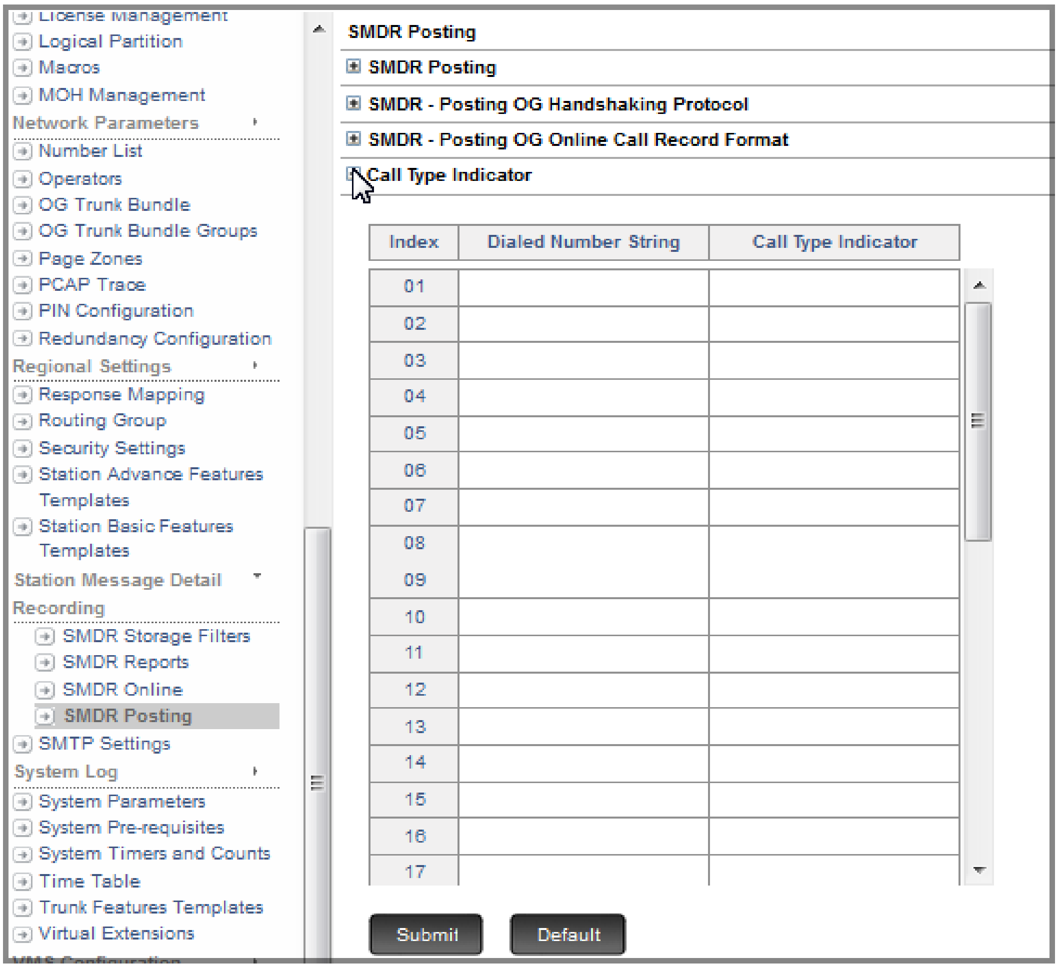
- **Property Code:** Enter the property code required by the CAS.

- **Currency Symbol (Enter Decimal Value):** Enter currency symbol to be used.

Call Type Indicator

In the Call Record, you can also include the Type of Call: local, national, international, by configuring the Call Type Indicator Table.

- Click **Call Type Indicator** to expand.



- In the **Dialed Number String** column, enter the number strings for each Call Type. You can enter the prefix, e.g. 0 for long distance calls, 2 for local numbers, etc.

- For each **Dialed Number String**, define a **Call Type Indicator**, this is an abbreviation of the Call Type, e.g.: LD for long distance, INTL for International, etc.

  Number String is of a maximum 4-digits. The Text String is a string of 4 alphanumeric characters. Your entries may look like these:

Number Index	Dialed Number String	Call Type Indicator (Text String)	Meaning
01	0	LD	Long Distance

Number Index	Dialed Number String	Call Type Indicator (Text String)	Meaning
02	95	IC	Inter Circle
03	197	INFO	Information
04	00	INTL	International
:	:	:	:
36	2	L	Local

You may enter as many Call Types as supported by the Posting Protocol you have selected.

- Click **Submit**.

System Activity Log

ANANT UCS monitors all its activities and maintains record of these activities in the System Activity Log.

The System Activity Log has a buffer capacity of 999 records. The Activity Log stores records using the FIFO method.

This log can be printed on a local printer or downloaded on a computer in form of a report. The activity log can be printed or downloaded in two modes:

- **Online:** The activity report is printed/downloaded as and when an activity occurs.
- **Report (Offline):** The activity report is printed/downloaded whenever desired. In the Offline mode, the last 999 activities recorded by the system are printed/downloaded.

The System Administrator can print/download System Activity Log, online or offline.

ANANT UCS supports Syslog Client for System Activity Logs. The Syslog Client enables the system to send activity logs in syslog format to the remote 'Syslog Server'. You can view the logs on the remote server.

How it works

- A destination port must be assigned for activity logs. The system will send the activity log to this port.
- If the System Administrator has an Extended IP Phone, a DSS Key can be assigned for System Activity Log.
- Each activity is stored in the Activity Log in this format:
  <DD-MM-YYYY> < HH:MM:SS> <Activity Text>
- Whenever an activity is recorded by the system, the DSS key, if assigned for this feature on the System Administrator's IP Phone is turned ON.
- The System Administrator can view the activity log by pressing the DSS key (if assigned). The Extended IP Phone of the System Administrator will display the activity in this format:

  DD-MM HH:MM <Activity Index>

*The format of the Date will be DD-MM or MM-DD as per **Date Format** selected in the **Real Time Clock** settings of the system.*

Index of the Type of Activities recorded in the System Activity Log:

Event Index	Activity	Description
01	System MATRIX ANANT UCS X.Y.Z Started	It displays the System Version.
02	Default Configuration Loaded	
03	SMDR-IC buffer deletion: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
04	SMDR-OG buffer deletion: nnnnnn	nnnnnn = max. 6 digit flexible number of the station

Event Index	Activity	Description
05	SMDR-Internal buffer deletion: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
06	SE Access From: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
07	SA Access From: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
08	Emergency Number Dialed: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
09	Log In nnnnnn	nnnnnn = max. 6 digit flexible number of the station
10	Log Out nnnnnn	nnnnnn = max. 6 digit flexible number of the station
11	Reg Fail, Configuration Parameter Invalid SIPTrunk = xx	x = SIP Trunk Number
	Stack Construct Authenticate Fail	
	Stack Construct IP Addr Invalid	
14	Call Budget exhausted, xx - yyyy, NAME	xx = Port Type yyyy = Port offset NAME = Name of the Trunk
15	Personal Mailbox is full for Extension: nnnnnn	nnnnnn = max. 6 digit flexible number of the station
16	VMS Storage 100% Full	
17	Restart due to change Network Para	Restart due to change Network Parameters
18	Restart due to change SIP Para	Restart due to change SIP Parameters
19	VMS Storage 80% Full	
20	VMS Storage Usage in Limit	
21	LDAP Test completed successfully	
22	LDAP Sync done for Global Directory x	x = Global Directory Part(s) which is synced
25	Message Notification Retry Count over for nnnnnn	nnnnnn = max. 6 digit flexible number of the station
26	Default Password restored for Authority Code:nnn	nnn = 3 digit Authority Code
27	Emergency Call of xxxxxx is Acknowledged by yyyyyy.	xxxxxx = max. 6 digit flexible number of the station from which the Emergency Call was made yyyyyy = max. 6 digit flexible number of the station which had acknowledged the Emergency call.
28	Forward DST Applied at: x	x = Date and Time when DST is applied
29	Backward DST Applied at: x	x = Date and Time when DST is applied

Event Index	Activity	Description
30	DST Config Change: x	x = "Enable" or "Disable" Display "x" as "Disable" when DST Mode is set from "Manual/Scheduled" to "Disable". Display "x" as "Enable" when DST Mode is set from "Disable" to "Manual/Scheduled". Don't display "x" when DST Mode is set from "Manual" to "Scheduled" or vice versa.
33	LOGIN_NAME Login blocked for IP=IP_ADDRESS	LOGIN_NAME = Actual Login ID i.e. SE or SA or FDU IP_ADDRESS = IP Address from which is blocked
34	IP=IP_ADDRESS blocked for Auto Prov. of User SIP_ID	IP_ADDRESS = IP Address from which is blocked SIP_ID = SIP ID of SIP Extension, for which request is made.
35	IP=IP_ADDRESS:PORT blocked	IP_ADDRESS:PORT = IP Address and Port from which is blocked.
36	Send to Network Error for Connection ID XXX	XXX is the connection id for which the call is released by the VoIP
38	System Reset	
39	General Mailbox is Full	
42	Trunk Active, SIP Trunk=xx	xx=SIP Trunk number
43	CPU Usage is High. System Performance may degrade.	
44	CPU Usage is now back to Normal	
45	RAM Usage is High. System Performance may degrade.	
46	RAM Usage is now back to Normal	
47	LAN Ethernet Link Up	
48	WAN Ethernet Link Up	
49	LOGIN_NAME Login IP=IP_ADDRESS	LOGIN_NAME = Actual Login ID i.e. SE or SA or FDU IP_ADDRESS = IP Address from which is login is made
50	System Restart by xx	xx = IP Address when system is restarted from Web JEEVES or Extension Number when system is restarted using command
51	System started as SERVER_MODE	"SERVER_MODE" - Configured "Server Mode" - "Primary" or "Backup" in "Redundancy Configuration".
52	Virtual IP Address Conflict Detected	
53	Isolation Detected	
54	Initial synchronization…	

Event Index	Activity	Description
55	Sync failed	
56	Sync success	
57	Declared as - STATE	"STATE" - "Active" or "Standby" based on startup or heartbeat process or failover process.
58	Redundancy configuration changed - FLAG	"FLAG" - Value of "Enable Redundancy" flag i.e. "Enable" or "Disable"
59	Remote Server connected	
60	Remote Server not available	
61	Redundancy license grace period	
62	License Key in use - SYSTEM	"SYSTEM" - Value can be "Primary" or "Backup" or "Other". Primary or Backup is decided on configured "Server Mode" - "Primary" or "Backup" in "Redundancy Configuration".
63	EULA IP=IP_Address	IP_ADDRESS = IP Address from which is EULA is accepted
64	Redundancy Call Unacknowledged by nnnnnn	nnnnnn = max. 6 digit flexible number of the station that did not acknowledged the Redundancy Call.
65	Redundancy Call is Acknowledged by nnnnnn	nnnnnn = max. 6 digit flexible number of the station that acknowledged the Redundancy Call
66	Validity expires in 5 days - CERTIFICATE_NAME	CERTIFICATE_NAME = Friendly Name of the certificate

When installed in the Hotel mode, ANANT UCS captures Hotel-Motel Activity Log. To know more, see the *ANANT UCS Hospitality System Manual*.

How to configure

To be able to use this feature, you must enable System activity log (online or report), and assign a Destination Port for the Activity Logs.

To view the system activity log, you must assign the IP Address of the remote Syslog server as the Destination Port.

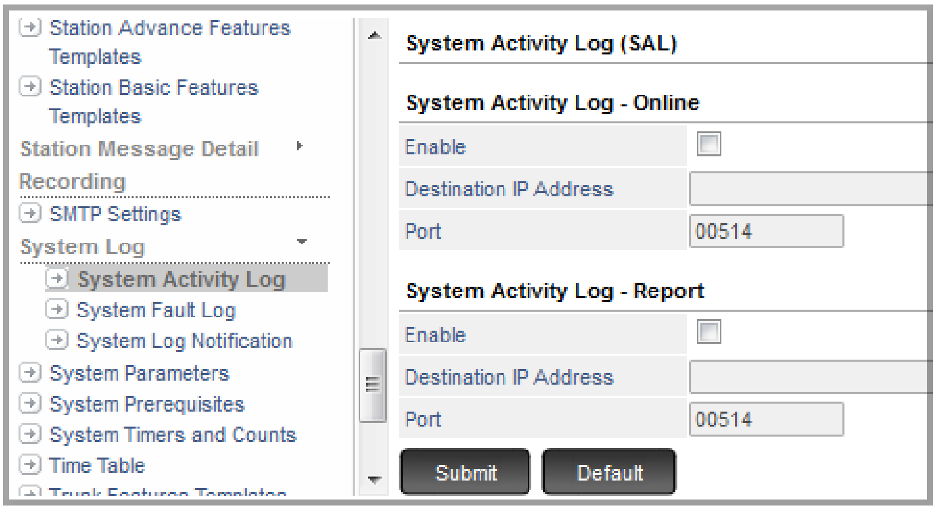
If the System Administrator phone is an Extended IP Phone, you may assign a DSS key for System Activity Log.

For instructions on configuring a DSS key on an Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”.

Configuring System Activity Log

- Login as System Engineer.
- Under **Configuration**, click **System Log**.

- Click **System Activity Log**.



- To generate **System Activity Log - Online**
  - Select the **Enable** check box.
  - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
  - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535. By default, 514 is assigned.
- To generate **System Activity Log - Report**, that is, offline, whenever desired
  - Select the **Enable** check box.
  - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
  - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.
- Click **Submit.**

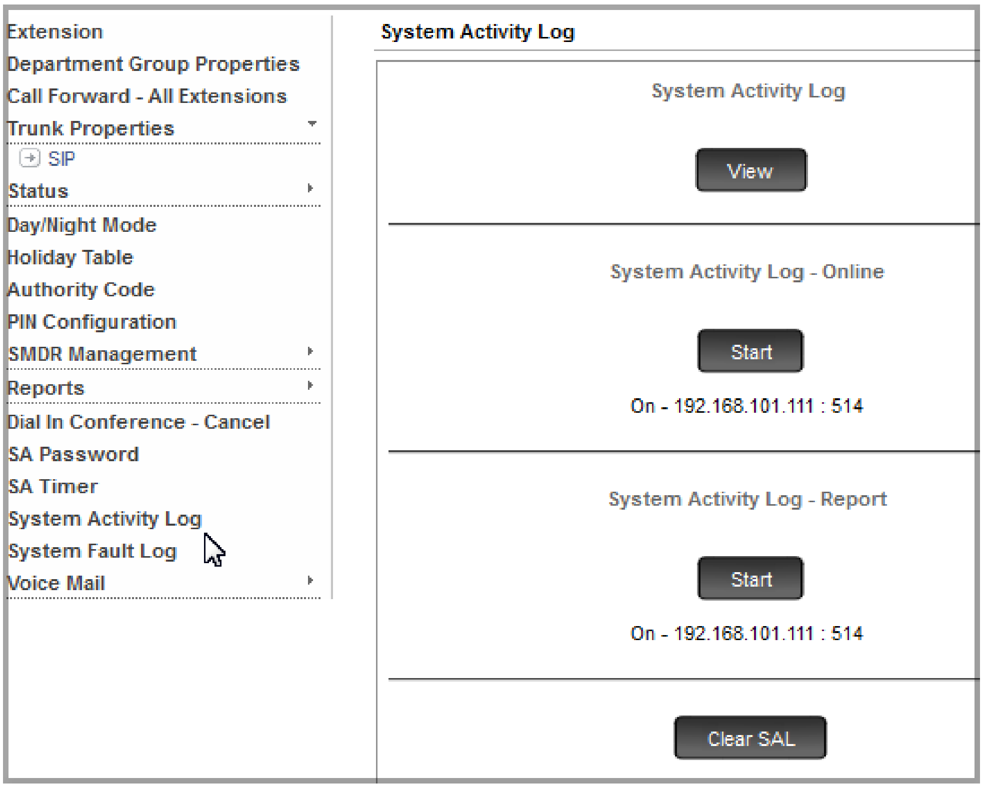
How to use

You can start and stop System Activity Log - Online and Report from the System Administrator mode using Jeeves

To start/stop report generation using Jeeves,

- Login as System Administrator.

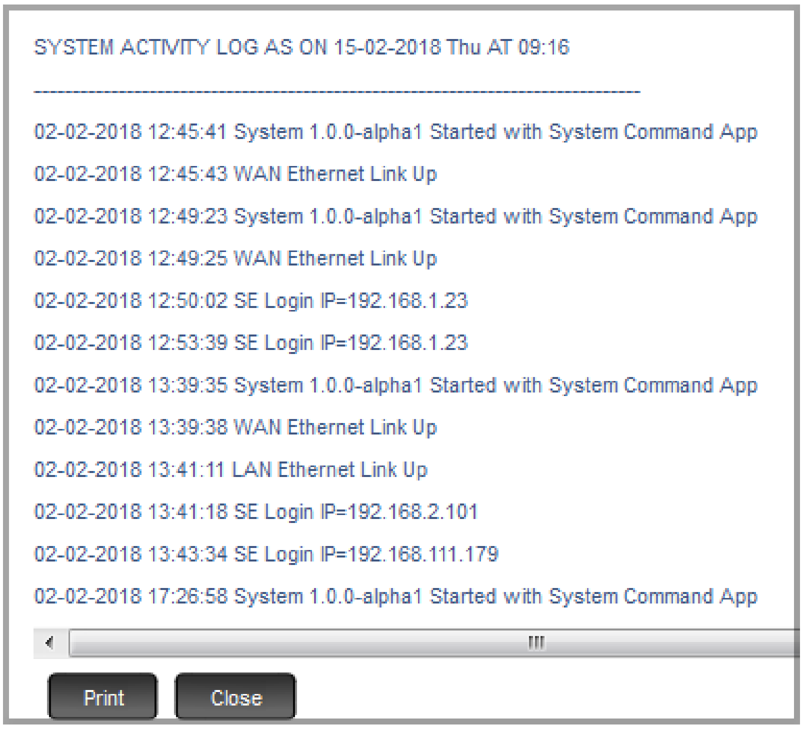
- Click the **System Activity Log**.



- To start **System Activity Log - Online**, click the **Start** button.
- To stop **System Activity Log - Online**, click the **Abort** button.
- To start **System Activity Log - Report**, click the **Start** button.
- To stop **System Activity Log - Report**, click the **Abort** button.
- To clear System Activity Logs from the buffer, click the **Clear SAL** button.

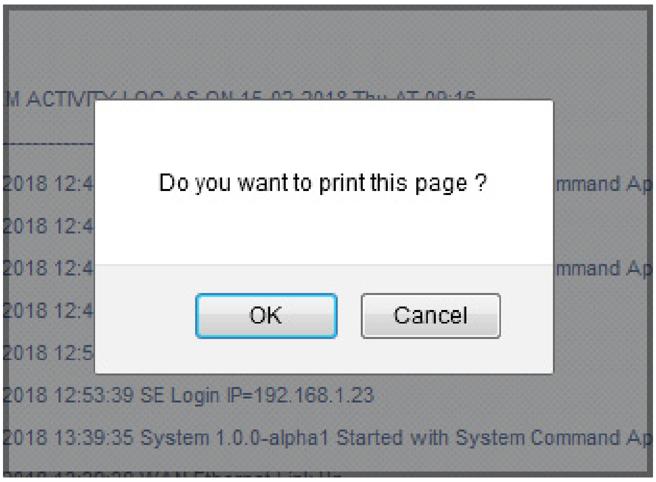
By default, the online and offline reports are printed on the destination port assigned by you.

- To view the **System Activity Log** on your computer screen, click the **View** button.



System Activity Log appears on your screen.

- To print this report on the local printer connected to your computer, click the **Print** button. An alert message will appear.



- Click the **OK** button. The System Activity Log - Report will be printed on the local printer connected to your computer.

System Activity Log Display

ANANT UCS provides a facility to display the last activity monitored by the system on the System Administrator's extension phone. The system also provides you the facility to view all the activities through Jeeves.

To know more, see "How to use" in "System Activity Log".

How to use

To be able to use this feature optimally, the System Administrator extension phone must be an Extended IP Phone, and a DSS Key must be assigned on the phone to System Activity Log Display.

For instructions on configuring DSS Keys on Matrix Extended IP Phone, see "DSS Key Settings" in "Configuring Matrix SPARSH VP330", "DSS Key Settings" in "Configuring Matrix SPARSH VP248", "DSS Key Settings" in "Configuring Matrix SPARSH VP310" and "DSS Key Settings" in "Configuring Matrix SPARSH VP510".

To view the System Activity Log from System Administrator Mode,

- Go Off-hook.
- Press the DSS key assigned to System Activity Log Display.
- The last recorded Activity log appears on your phone's display in this following format: ***Date-Time-Activity Index***
  The Date and Time are in <**DD-MM HH:MM:**> format
  The Activity Index is a two digit number. To know the index details, refer "System Activity Log".

The date and month format will be DD-MM or MM-DD as per date format set in the system. See "Real Time Clock (RTC)" for setting the date format.

System Fault Log

ANANT UCS maintains a log of all system faults. The system Fault Log has a buffer capacity of 999 records. The Fault Log stores records using the FIFO method.

The System Fault log can be printed on a local printer or can be saved on a computer in form of a PDF. The report can be printed or downloaded by the System Administrator in two modes:

- **Online:** The fault report is printed/downloaded as and when a fault occurs.
- **Report (Offline):** The faulty report is printed/downloaded whenever desired. In the Report (Offline) mode, the last 999 faults recorded by the system are printed/downloaded.

The System Administrator can print/download System Fault Log, *Online* or *Report*.

Matrix ANANT UCS supports Syslog Client for System Fault Logs. The Syslog Client enables the system to send fault logs in syslog format to the remote 'Syslog Server'. You can view the logs on the remote server.

How it works

- A destination port for sending the report must be selected to which the system can send the log.
- If the System Administrator has an Extended IP Phone, a DSS Key can be assigned for System Fault Log.
- Whenever a fault is detected, the LED of the Fault Log DSS key, if assigned, is turned ON.
- If more than one Extended IP extension is assigned Fault Log DSS Key, the LED of all keys will be turned ON.
- The System Administrator must acknowledge the Fault indication by pressing the Fault Log key or by dialing the Fault Log access code. The LED of the Fault Log key is turned OFF.

The different fault events that are logged are summarized in this table:

Event Index	Activity	Description
1	Valid MoH File Not Present, MoH will not be played	
4	Trunk Authentication Fail, SIP Trunk = xx	xx = SIP Trunk number
5	Trunk Registration Time Out, SIP Trunk = xx	xx = SIP Trunk number
6	Trunk Disable, SIP Trunk=xx	xx = SIP Trunk number
7	Trunk Registration Fail, SIP Trunk = xx	xx = SIP Trunk number
8	LAN Ethernet Link Down	
9	WAN Ethernet Link Down	
10	LAN and WAN Port IP Addresses are in the same subnet	
11	Redundancy failed - Firmware Mismatch	

Event Index	Activity	Description
12	Remote Server detected with same Server	
13	Remote Server in Active state detected -	
14	Internal failure	
15	Redundancy failed - License Not Activated	
16	LDAP Test Failed - REASON	REASON = Actual reason
17	LDAP Sync Failed - REASON	REASON = Actual reason
18	Application integrity failed	
19	System Restart due to REASON	REASON = Actual reason - either "CM failure" or "SM failure" or "MS failure" or "VMS failure"
20	CERTIFICATE_NAME validity expires	CERTIFICATE_NAME = Friendly Name of the certificate

Registration Timer Fail: *The system may fail to load either the Re-registration Timer or the Registration Retry Timer. In such a case the Proxy SIP trunk will remain un-registered and will not be functional.*
This can happen to one or more SIP trunks, while the other SIP Trunks are functioning normally. You need to restart the system to resolve the problem.

How to configure

To be able to use this feature, you must enable System fault log (online or report), and assign a Destination IP address for the Fault Logs.

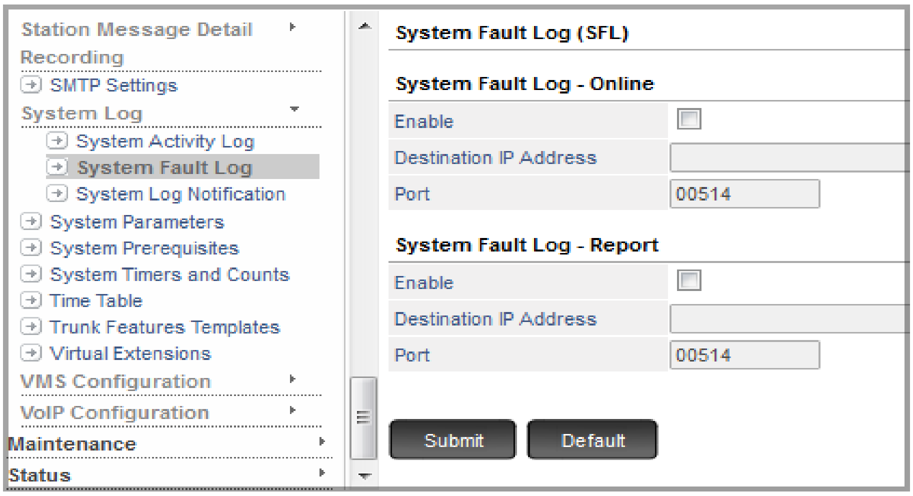
If the System Administrator has an Extended IP Phone, you may assign a DSS key for System Fault Log.

For instructions on configuring DSS Keys on Matrix Extended IP Phone, see “DSS Key Settings” in “Configuring Matrix SPARSH VP330”, “DSS Key Settings” in “Configuring Matrix SPARSH VP248”, “DSS Key Settings” in “Configuring Matrix SPARSH VP310” and “DSS Key Settings” in “Configuring Matrix SPARSH VP510”.

Configuring System Fault Log

- Login as System Engineer.
- Under **Configuration**, click **System Log**.

- Click **System Fault Log**.



- To generate **System Fault Log - Online**
  - Select the **Enable** check box.
  - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
  - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535.

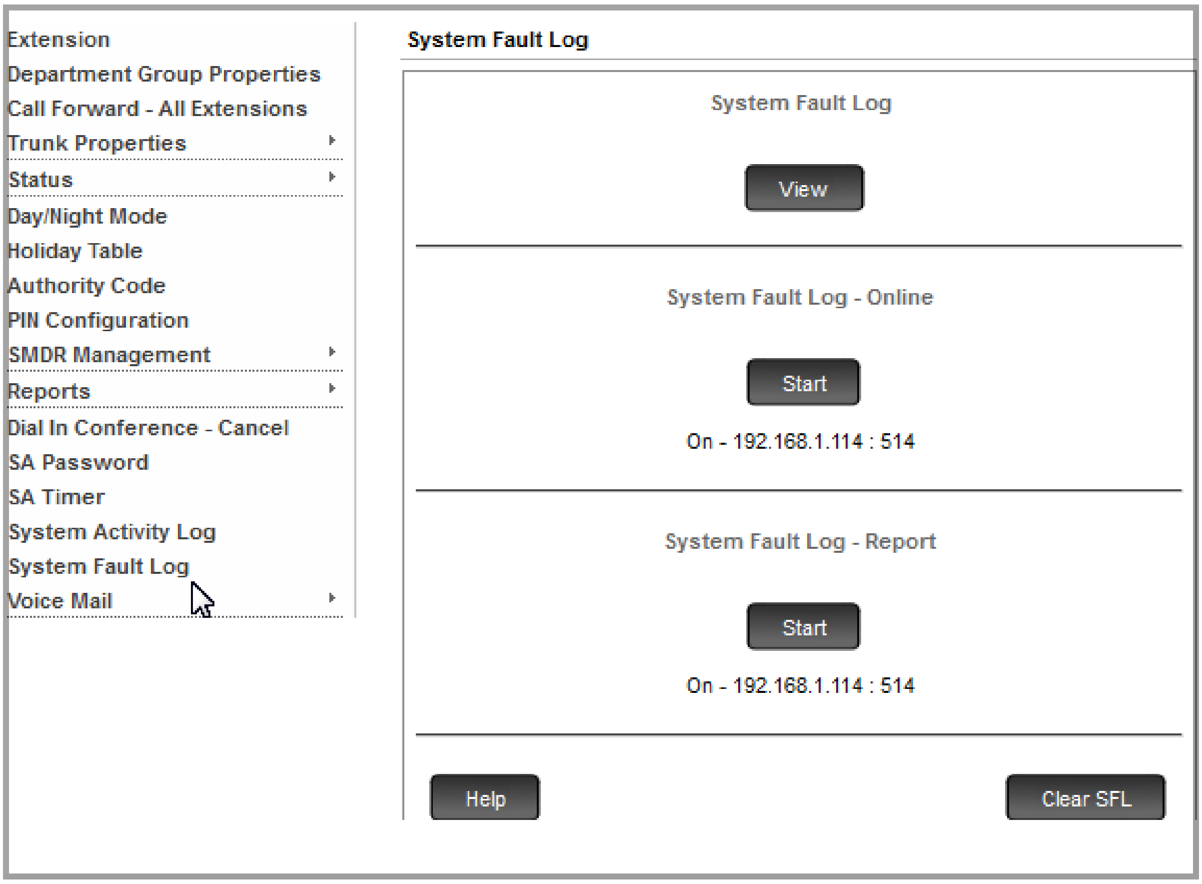
    By default, 514 is assigned.
- To generate **System Fault Log - Report**, that is, offline, whenever desired
  - Select the **Enable** check box.
  - In **Destination IP Address,** enter the IP Address of the remote Syslog Server. Both IPv4 and IPv6 addresses are supported.
  - In **Port**, enter the port of the remote Syslog Server. Valid port range is: 514; 1025 to 65535
- Click **Submit.**

How to use

You can start and stop System Fault Log - Online and Report from the System Administrator mode using Jeeves.

To start/stop fault report generation using Jeeves,

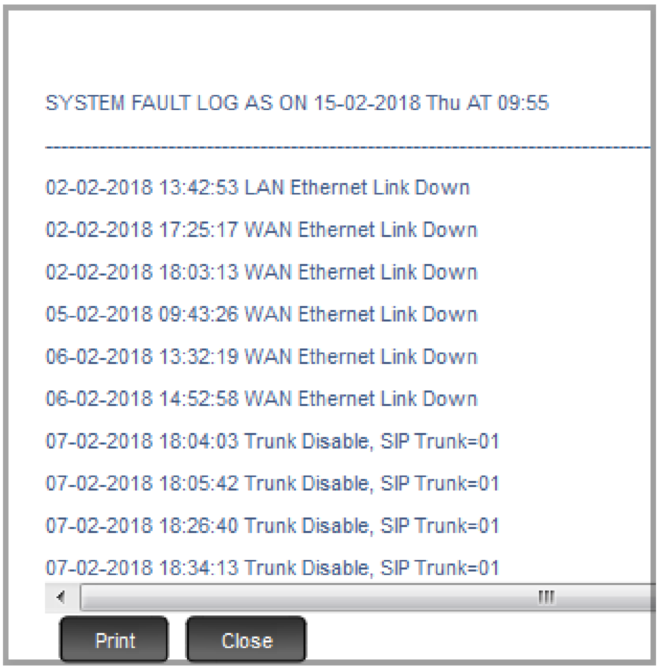
- Login as System Administrator.
- Click the **System Fault** Log.



- To start **System Fault Log - Online**, click the **Start** button.
- To stop **System Fault Log - Online**, click the **Abort** button.
- To start **System Fault Log - Report**, click the **Start** button.
- To stop **System Fault Log - Report**, click the **Abort** button.
- To clear System Fault Logs from the buffer, click the **Clear SFL** button.

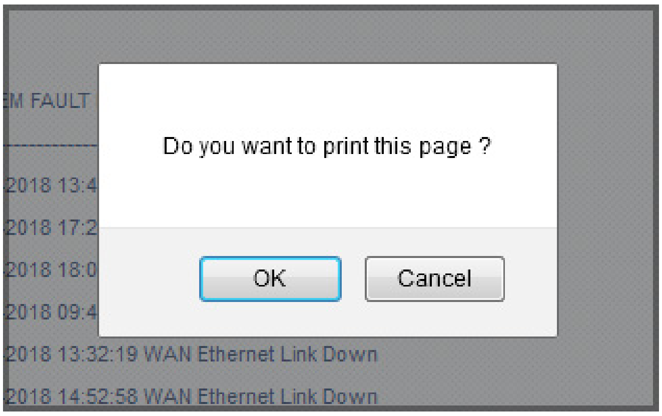
By default, the online and offline reports are printed on the destination port assigned by you.

- To view **System Fault Log** on your computer screen, click the **View** button.



System Fault Log appears on your screen.

- To print this report on the local printer connected to your computer, click the **Print** button. An alert message will appear.



- Click the **OK** button. The System Fault Log - Report will be printed on the local printer connected to your computer.

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System Fault Log Display

ANANT UCS provides a facility to display the last fault monitored on the system on the System Administrator's extension phone. The system also provides you the facility to view all the faults through Jeeves.

To know more, see "How to use" in "System Fault Log".

How it works

To be able to use this feature optimally, the System Administrator extension phone must be an Extended IP Phone, and a DSS Key must be assigned on the phone to System Fault Log.

- When a fault occurs, the LED of the DSS Key assigned for the System Fault Log, glows.
- The System Administrator may press any key or dial the System Fault Log feature access code to acknowledge it.
- On pressing any key or dialing of the acknowledgment command, the LED of the Fault Log key is turned OFF.

How to use

To view the System Fault Log from System Administrator Mode,

- Go Off-hook.
- Press the DSS key assigned to System Fault Log Display.
- The Fault log appears on your phone's display in this format: ***Date-Time-Fault Index***
  The Date and Time are in <**DD-MM- HH:MM:**> format
  The Activity Index is a two digit number.To know the index details, refer "System Fault Log".

System Log Notification

Whenever a fault occurs in the system or an activity takes place, the details of the fault/ activity can be sent to the concerned person to notify him/ her of the same.

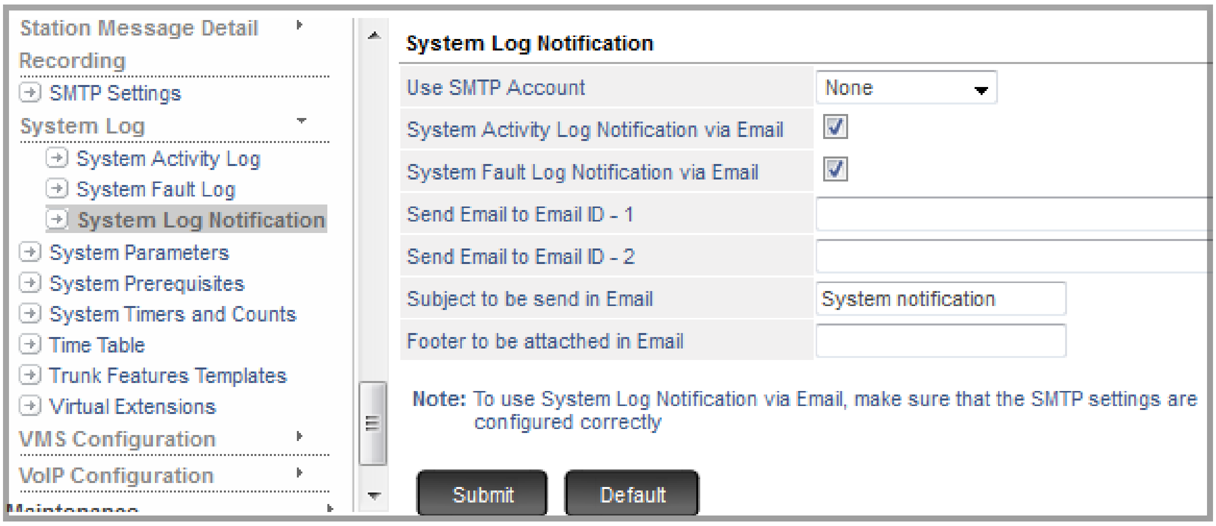
The notification can be sent as an email to the specific email id.

ANANT UCS maintains a log of all the system activities and faults. These logs can be printed on a local printer or downloaded on a computer in the form of a report. To know more, see "System Activity Log" and "System Fault Log".

How to configure

To be able to use this feature, you must configure the following parameters.

- Login as System Engineer.
- Under **Configuration**, click the **System Log**.
- Click **System Log Notification.**



- **Use SMTP Account**[116]**:** Select the desired SMTP account.
- **System Activity Log Notification via Email**: If this check box is enabled, the system will send notifications for the system activities via Email.
- **System Fault Log Notification via Email**: If this check box is enabled, the system will send notifications for the system faults via Email.
- **Send Email to Email ID-1:** Configure the email id to which you want the system to send notifications as an email.
- **Send Email to Email ID-2**: Configure an alternative email id to which you want the system to send notifications as an email.

116. Make sure that the SMTP settings are configured correctly. For more information, refer "SMTP Settings".

- **Subject to be Send in Email**: Configure the text you want to display, as the subject to the receiver of the Email.
- **Footer to be attached in Email**: Configure the text you want to be displayed, as the footer to the receiver of the Email.
- Click **Submit.**

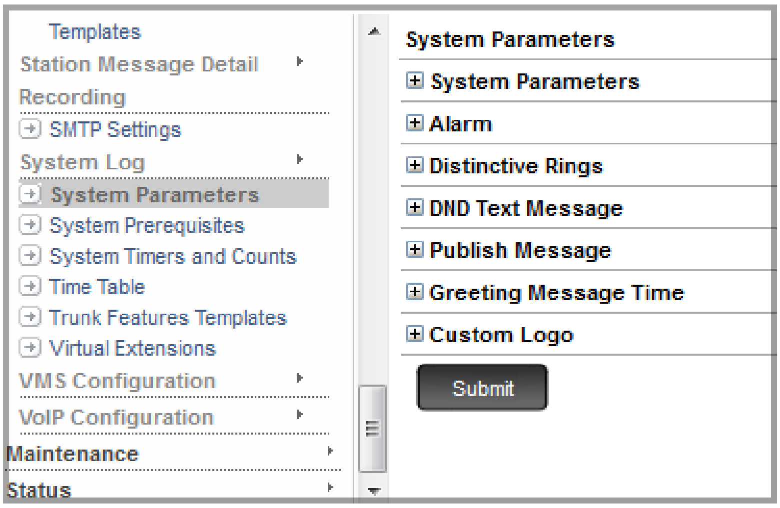
System Parameters

System Parameters are general parameters, related to features and facilities that are applied system-wide, such as customer Profile, Day-Night mode, storage of call logs, alarms, Presence, and DND messages. Each of these is described briefly along with the instructions for configuring them using Jeeves.

How to configure

Configuring System Parameters

- Login as System Engineer.
- Under **Configuration**, click **System Parameters**.



System Parameters

- Click **System Parameters** to expand.

  To view more parameters, use the vertical scroll bar on your right.

- Under *System Parameters*, do the following:

  - **Customer Profile:** ANANT UCS two major applications: Enterprise application to meet requirements of businesses, and Hospitality application to meet the specific requirements of Hotels and Hospitals.

    You must select **Customer Profile** as Enterprise or Hotel according to the application you are using. When you select the Customer Profile, all the features and facilities specific to the application Enterprise/Hotel along with their default settings are loaded. By default, the Customer Profile of ANANT UCS is selected as 'Enterprise'.

  - **Station Name Pattern:** The Station Name Pattern is the format in which the names of extensions will be stored on the extension phones and displayed to other extensions. You can store names by First Names only, First names and Last Names. You can also add Titles indicating gender, designation, rank, social standing, like Mr. Mrs. Ms., Dr., Prof. Cmdr., Rev., to Names of extensions.

    ANANT UCS supports the following Station Name patterns.

Option	Meaning
1	Title<space>First Name<space>Name
2	First Name only
3	Name only
4	First Name<space>Name
5	Title<space> First Name
6	Title<space>Name

Station Name Pattern must be configured for the *Guest Name* and *Title* feature of the ANANT UCS Hospitality module. To know more, refer the feature description in the *ANANT UCS Hospitality System Manual.*

By default, **Name Only** is selected as the Station Name Pattern when ANANT UCS is operated in the Enterprise mode, and Title<space>Name is selected as the Station Name Pattern when ANANT UCS is operated in the Hotel Mode.

- **Default Call Hold Type:** This parameter is related to the "Call Hold" feature of ANANT UCS. You can select Global Hold or Exclusive Hold. Default: Exclusive Hold.

- To have ANANT UCS store also internal calls in the Missed Call Log, select the **Store Internal Calls in Missed Call Log** check box. To know more about this feature, see "Call Logs". Default: Enabled.

- You may have ANANT UCS store internal calls in the Dialed Call Log by selecting the **Store Internal Calls in Dialed Call Log** check box. To know more about this feature, see "Call Logs". Default: Enabled.

- To have the system store internal calls in the Answered Call Log, select the **Store Internal Calls in Answered Call Log** check box. To know more about this feature, see "Call Logs". Default: Enabled.

- To have the system store internal calls in the Redial Call Log, select the **Store Internal Calls in Redial Call Log** check box. To know more about this feature, see "Last Number Redial". Default: Disabled.

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- The SIP Extension users can set multiple call appearances on their phones. If you want the system to play MOH to all the queued internal calls when the user extension is busy, select the **Play MOH to Queued Internal Calls on SIP Extension** check box. To know more, read the feature description for "Music on Hold (MoH)".

- You can set the Time Zone of the system as Working-Hours, Break Hours or Non-Working hours any time you want by setting the **Day/Night Mode**. You can set the system in the **Day Mode** or the **Break Hours Mode** or the **Night Mode**, or let the system **Operate as per the Time Table assignment**[117]. For more details see "Day Night Mode" and "Time Tables". Default: Operate System as per Time Table assignment.

- Select the **Toggle Day/Night mode through 'Set Day/Night Mode' key** check box if you want to switch to Day Mode (Working Hours) or Night Mode (Non-Working Hours) on pressing the DSS key. For more details, see "Day Night Mode". Default: Disabled.

- To have the system detect the extension that has made an emergency call, select the **Emergency Dialing Reporting** check box.

  When this check box is enabled, the system detects the extension that has made the emergency call and reports it to the Operator extension. Thus the Operator can know which extension has made an emergency call. To know more about this feature, see "Emergency Detection and Reporting".

- If you want to remove the '+' prefix in the CLI of the calling party and replace it with another string, select the **Replace '+' from CLI**[118] check box. Default: Disabled.

  If you want to program the number string with which the '+' prefix is to be replaced, in the **Replace '+' from CLI with the number string**[119] field, enter the desired number string.

117. *Certain features of the ANANT UCS require extensions and trunks to behave differently according to the working hours, break hours and non-working hours, which are referred to as Time Zones. The Time Zones, are defined for the entire week in a Time Table. Time Table is assigned to trunks, extensions and other time-zone dependent features.*

118. *As per Government Regulatory Laws, this parameter is not applicable if India is selected as the Region.*

119. *As per Government Regulatory Laws, this parameter is not applicable if India is selected as the Region.*

If you keep the number string field blank, ANANT UCS will remove '+' sign from the CLI of calling party and present the remaining digits on the CLI of the Called Party.

For example:

The number string +919327237228 is received as CLI.
If ‘00’ is configured as the replace string, the CLI number would become 00919327237228
If no replacement string is configured (that is, left blank), the CLI number would be presented as 919327237228.

- **If the Extension creating 3 party conference, disconnects during Conference**, you can select either to Transfer the call or Disconnect other parties.

    - If you select to **Transfer the call**, the 3-party conference is converted into a two-way speech between the other two parties.
    - If you select to **Disconnect other parties**, all the parties involved in the 3-party conference are disconnected.

  See “Conference-3 Party” for more details.

- To play a beep to participants of a conference to indicate inclusion of a new participant, select the **Play Beep when Conference/Dial-in Conference begins** check box selected. Default: Enabled.

  This check box is common for the features “Conference-Multiparty”, “Conference Dial-In” and “Emergency Conference”. When this check box is enabled,

    - the system plays beeps to the other participants in a Dial-In Conference when a new participant joins in (dials into an on-going Dial-In Conference).

    - the system plays beeps to the other participants connected in a Multi-Party Conference and an Emergency Conference, when a new participant is included.

  If you disable this Check box, the existing participants in a Dial-In or Multi-party conference will not hear any beep tone indicating the addition of a new participant.

- **Play Beep when Raid/Call Taping/Conversation Recording Starts** is a common Check box for the features “Raid”, “Call Taping” and “Conversation Recording”. When this check box is enabled,

    - the system plays a warning beep to the extension which is being raided by another extension, before establishing three-way speech.

    - the system plays beeps to the extensions/calling party before it starts taping the call in the common mailbox or recording the conversation in the extension’s mailbox.

  When this check box is disabled, no warning beep will be played in Raid, no indication will be given to the opposite party when the call is being taped/conversation is being recorded. Default: Enabled.

- In **Play Feature Tone in place of Dial Tone when Call Forward is Set**, you can select whether you want the system to play **Feature Tone** instead of **Dial Tone** to the extensions when Call Forward is set on these extensions. When this check box is disabled, the system will play dial tone to the extension on which Call Forward is set, whenever the extension goes Off-hook. Default: Enabled.

- Select the **Ignore call forward set by member extension, when call is routed on Routing/Dept. Group** check box, if you want the system to place calls on member extensions in a Routing Group even if they have set Call Forward.

- Select **Call Proceeding Tone for Multistage Dialing**. Default: Network Tone.
  This parameter is used in Multistage Dialing where you need to configure Pause and Wait for Answer in the Dialed Number column for the number string dialed by the extension users.

  When an extension user makes a call using a Calling Card, and the system dials out the number in stages, the extension user will get Ring Back Tone twice; first after the system has dialed the Calling Card Number, and again after the system has dialed out the destination number (called party number). Thus the extension user will get Ring Back Tone, twice. To avoid this, you may configure the 'Call Proceeding Tone' to be played by the system when using Multi-Stage Dialing.

  You must configure the type of 'Call Proceeding Tone', according to your requirement; whether the extension user should be connected to the speech path when the Calling Card number is out dialed or when the called party number is out dialed. You can select any of the following Call Proceeding Tone options, as per your requirement:

  - **Network Tone:** If this option is selected, the extension user will get Ring Back Tone after dialing the Calling Card number and again, after the system has dialed the called party number (when the system is dialing out the number with Pause and Wait for Answer configured in Dialed number string).

  - **Pseudo Tone:** If this option is selected, the extension user will get Feature Tone when the user has completed dialing all the digits. At the end of the tone, the extension user gets connected to the called party (destination number).

  - **Silent:** If this option is selected, the extension user will get Silence (no tone), after the extension user has completed dialing all digits. After dialing out the called party number in DTMF, the system will connect the caller to the called party number (destination number).

- You can select the **Language of SE, SA and Front Desk User Web Interface** as per your requirement. The GUI of ANANT UCS supports the languages English, Italian, Spanish, French, German, and Portuguese. When you select ‘Region’ for ANANT UCS, one of these languages will be applied as appropriate for the region you selected. For instance, if you selected India, English will be applied. If you selected Spain, Spanish will be applied. If you selected a country where none of these languages are the local language, English will be applied.

  The language set by the system on Region selection will be applied on the pages of the GUI for every login session. You can change the default language set on Region selection, by configuring this parameter.

- To print each system report on a separate page, keep the **Form Feed in Report Printing** check box enabled. Default: Enabled.

- If you want to enable the Operator Console to view the presence status of the extension they are calling, select the **Display Presence Status during call on Extended IP Phone** check box. Default: Disabled.

- Enable **Apply RCOC only if the caller calls back on the same trunk from which the call was made** check box, if you want ANANT UCS to match the Trunk Port Number and Trunk Port Type of the incoming call with the Trunk Parameters of the entry stored in RCOC table before routing the call to the original caller. Default: Disabled.

- Enable the **Stuttered Dial Tone When DND is set** check box, if you want ANANT UCS to play a Stuttered Dial Tone on the extension when DND is set.

- By default, the **Display Guest Station in Directory** check box is enabled. This feature is required when you are using the system for the Hospitality Application.

  The system will display the contact details of the ***Guest - Station*** in the Contact List of all the other extensions.

  In the Hospitality Application, there may be a requirement to keep the guest details confidential. In such cases you must keep this option disabled.That is the guest details will not appear in the contact list of the other extensions.

If a call is made to the guest number that is not visible in the contacts list from any extension, these details will appear in the call logs, even if this check box is disabled.

When disabled, the contact details of the Guest - Station will not appear in the contacts lists of the extensions assigned Station Type as Administration.

- Select **Call Proceeding Tone for 1st caller of a SIP Extension**. Default: Ring Back Tone.

  You must configure the type of 'Call Proceeding Tone for 1st caller of a SIP Extension', according to your requirement. This tone will be played when the system is routing the call. You can select any of the following options:

  - **Ring Back Tone:** If this option is selected, the extension user will get Ring Back Tone while the call is being routed.

  - **Pseudo Tone:** If this option is selected, the extension user will get Feature Tone while the call is being routed.

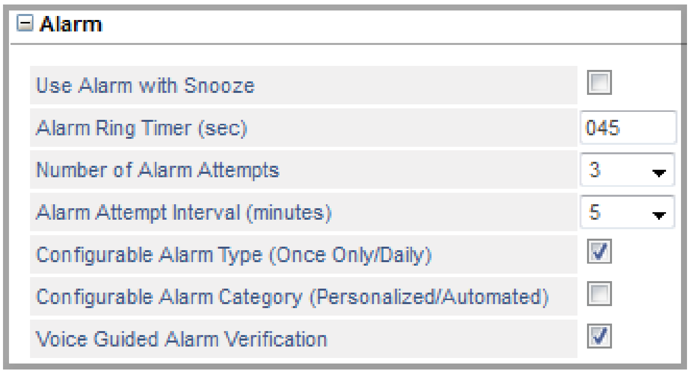
  - **Silent:** If this option is selected, the extension user will get Silence (no tone), while the call is being routed.

For routing the second call the system will always play Ring Back Tone to the second caller.

- Select the **Include “Diversion” for call to SIP Extension** check box to enable. Default: Disabled. If you have any SIP Phone/Entity registered with the system as Open SIP Standard and Call Forward is set on the same, then to include the called number in the forwarded call request, select this check box.

Alarm

- Click **Alarm** to expand.



- Configure the following Alarm and Reminder parameters:

  - **Use Alarm with snooze**: Enable this check box if you want to use the Snooze function for the Alarm Call.
  - **Alarm Ring Timer (Sec)**: You may change the time for which the Alarm Call will ring on the extension phone and the time for which the Operator phone will ring to notify an unanswered Alarm Call.

  - **Number of Alarm Attempts**: You may increase or decrease the number of attempts the system should make to serve an Alarm call.

  - **Alarm Attempt Interval (minutes)**: You may increase or decrease the time gap between each attempt the system makes to serve an Alarm call.

  - **Configurable Alarm Type (Once Only/Daily)**: Disable this check box, if you do not want the system to provide the Operator and the extension users the option of setting 'Once Only' or 'Daily' Alarms. When this check box is disabled, the system will allow only 'Once Only' alarms to be set.

  - **Configurable Alarm Category (Personalized/Automated)**: Disable this check box, if you do not want the system to provide the Operator the option of setting 'Personalized' or 'Automated' Alarm calls.

    When this check box is disabled, the system will follow the 'Automated' Alarm call serving mechanism. The Operator will not be prompted to choose between 'Automated' and 'Personalized' Alarm calls when setting Alarm calls for an extension phone.

  - **Voice Guided Alarm Verification:** Enable this check box if you want to enable extension users to confirm the Time they have set for an alarm or the Date and Time they have set for a reminder.

Distinctive Rings

- Click **Distinctive Rings** to expand.

⊟ Distinctive Rings

Feature	Ring Type	Ring Text
Internal Call	Short, very slow	internal
Trunk Call	Double	external
Auto Call Back	Short, slow	acb
Auto Redial	Long, very slow	autord
Alarm	Long, fast	selfalram
Emergency	Long, fast	emergency
Operator Alarm	Long, fast	operatoralarm
Message Wait	Short, fast	msgwait
Ring Test	Short, slow	test
Priority	Triple	priority
Emergency Conference	Triple	emergencyconf
Conference	Triple	conf

Distinctive Rings are ringing patterns used for distinguishing between different types of call events, like Internal Calls, Trunk Calls, Auto Call Back, Auto Redial, Alarm, Emergency call, Priority, etc. If you want to customize the Ringing pattern, for a call event, select the desired **Ring Type**. For more details see "Distinctive Rings".

Incoming CLI Modification[120]

- Click **Incoming CLI Modification** to expand.

⊟ Incoming CLI Modification

Enable Incoming CLI Modification	☐
Country Code	
Area Code	
International Prefix	
National Prefix	
Area Code required to make Local calls?	Yes
Prefix Area Code	

Incoming CLI Modification is useful in countries where the Calling Line Identification (CLI) received by the System extension users must be suitably modified before it can be used to dial out the number. To know more, see "Incoming CLI Modification".

120. *To comply with the Indian Government Laws and Regulation, this parameter is not provided for India Region.*

If you receive CLI in dialable format, there is no need to use this feature. In such case, keep the check box disabled. You do not need to configure any of the CLI Modification parameters.

*For an incoming call on any trunk, the Incoming CLI Modification is applicable only when both — the **Allow Incoming CLI Modification** check box in SIP trunk and the **Enable Incoming CLI Modification** check box — are enabled. To know more, refer* *“Configuring SIP Trunks”.*

- To apply Incoming CLI Modification, select the **Enable Incoming CLI Modification** check box.
- Configure the following options for CLI modification:
  - **Country Code:** This is the Country Code of the country where ANANT UCS is installed. The Country Code helps ANANT UCS detect whether the Incoming CLI received is a national or an international number. Do not enter any prefix for the Country Code. By default, it is Blank.
  - **Area Code:** This is the Area Code of the place where the ANANT UCS is installed. The Area Code helps ANANT UCS detect whether the Incoming CLI received is a local number. Do not enter any prefix for the Area Code.By default, it is Blank.
  - **International Prefix***:* These are digits required as Prefix for dialing International Numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. By default, it is Blank.
  - **National Prefix:** These are digits required as Prefix for dialing long distance, National (within the country) numbers. The prefix may be up to 5 digits, with numbers from 00000 to 99999. By default, it is Blank.
  - **Area Code required to make local calls?:** Depending on the dialing pattern of your local public telephone network, you may choose:
    - **No (Area Code not required):** Select this option, if your public telephone network does not require the dialing of Area Code for local numbers.
    - **Yes (Area Code is required)***:* Select this option, if your public telephone network requires you to dial the Area Code for local numbers.
    - **Yes, with Prefix Digit**: Select this option, if your public telephone network requires you to dial Area Code with a particular Prefix for local numbers. If you select this option, you must also enter the prefix digits for the area code for local calls in the **Prefix Area Code** field.

DND Text Message

- Click **DND Text Message** to expand.

DND Text Message

Message No.	Message Text
1	Do Not Disturb
2	Unavailable
3	In a Meeting
4	In a Conference
5	Try on Mobile
6	On Vacation
7	On Business Trip
8	Out of Office
9	With a Guest

ANANT UCS allows you to configure DND different Text Messages, which extension users can select when setting DND on their extensions. The DND text message selected by them is displayed to the calling extensions. See "Do Not Disturb (DND)" to know more.

The default DND Text messages appear on your screen. You may customize messages 2 to 9 according to your requirement. Text Message can be of maximum 16 alphanumeric characters. All ASCII characters except <, >, :, ", /, \, |, ?, * are allowed.

Publish Message

- Click **Publish Message** to expand.

Publish Message

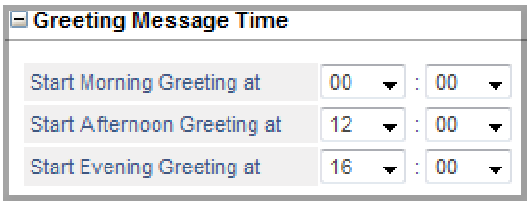
Message No.	Message Text
0	Absent
1	Present
2	Auto Detect
3	Away
4	On the Phone
5	Do Not Disturb
6	I am Mobile
7	In a Meeting
8	Out for a Meal
9	Out of Office

ANANT UCS offers 10 different text Messages to Publish Message, as listed in the table below. You can customize message 6 to 9 to match your requirement. Text Message can be of maximum 16 alphanumeric characters. All ASCII characters except <, >, :, ", /, \, |, ?, * are allowed.

To know more about this feature, see "Presence".

Greeting Message Time

- Click **Greeting Message Time** to expand.



- When **Voice Mail Auto Attendant** is enabled on trunks, the Voice Mail System answers the call and the greeting messages are played to the callers according to the time of the day, morning, afternoon, evening.

  To know more about this feature see "Auto Attendant".

- You can set the desired Start Time for Morning, Afternoon and Evening greetings.

  In **Start Morning Greeting at** set the start time for the Morning Greeting Message. Similarly, in **Start Afternoon Greeting at** and **Start Evening Greeting at** set the start time for the Afternoon and Evening Greeting Message respectively.

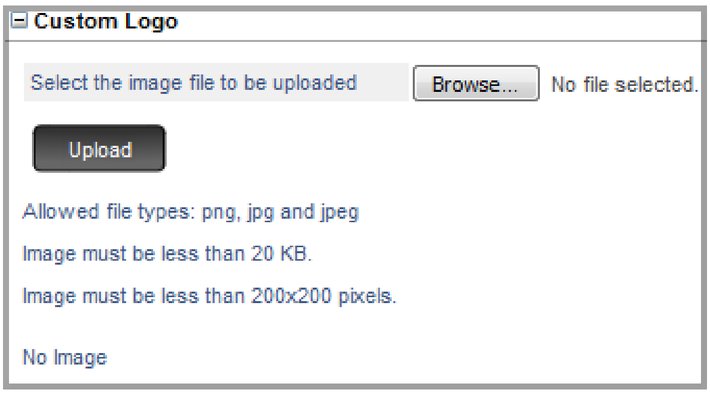
  The time must be in HH:MM format. The valid range for Hours (HH) is 00 to 23 and for Minutes (MM) is 00 to 59. By default, the time in **Start Morning Greeting at** is set to 00:00, **Start Afternoon Greeting at** is set to 12:00 and **Start Evening Greeting at** is set to 16:00.

  The system plays the Morning Greeting Message between the Morning and Afternoon Greeting Start time, the Afternoon Greeting Message between the Afternoon and Evening Greeting Start time and the Evening Greeting Message between the Evening and Morning Greeting Start time.

As the system plays the Evening Greeting Message between the Evening and Morning Greeting Start time, to prevent the Evening Greeting Message from being played after midnight, you are recommended to set the Morning Greeting Start time to 00:00 hrs.

Custom Logo

- Click **Custom Logo** to expand.



For VARTA Mobile UC Clients — Android, iPhone — ANANT UCS supports uploading of any customized logo. This logo shall appear on the Home screen of the VARTA ADR100 and VARTA AMP100.

Make sure the image you want to upload fulfills the following:
- The image must be a JPEG, JPG or PNG file
- Maximum Dimension of the image: 200 x 200 pixels
- Maximum File Size of the image: 20KB

To upload the file,

- Click the **Browse** button to **Select the image file to be uploaded** from the location on the local disk.

- Click the **Upload** button.

  A preview of the image you uploaded appears.

- If you wish to remove the image, click the **Remove** button.

If you upload another image directly without removing the previous image, the system will automatically overwrite the image.

System Timers and Counts

Several features of the ANANT UCS are based on Timers and Counts. For example, how long and how many times an extension should ring when Message Wait is set, or how long the Busy Tone, the Ring Back Tone, or the Error Tone should be played to an extension. ANANT UCS allows you to configure most of these Timers and Counts to suit your requirement.Listed below are the Timers and Counts related to the various features and facilities, along with a brief description and default value of each.

Auto Redial

Name	Description	Range	Default
Auto Redial - Ring Back Tone Wait Timer (sec.)	The time for which system waits to sense the Ring Back Tone from the SIP Trunk service Provider after dialing the requested number.	1 to 255	60 seconds
Auto Redial - Ring Timer (sec.)	The time for which the extension that has requested Auto Redial should ring.	1 to 255	45 seconds
Auto Redial - Normal Timer (sec.)	The time interval between auto redial attempts when Auto Redial ‘Normal’ is set.	1 to 255	45 seconds
Auto Redial - Normal Count	The number of auto redial attempts the system will make when Auto Redial ‘Normal’ is set.	1 to 255	5 Attempts
Auto Redial - Priority Timer (sec.)	The time interval between auto redial attempts when Auto Redial ‘Priority’ is set.	1 to 255	10 seconds
Auto Redial - Priority Count	The number of auto redial attempts the system will make when an extension having the feature Auto Redial Priority in its Class of Service uses Auto Redial ‘Priority’.	1 to 255	20 Attempts

Call Progress Tones

Name	Description	Range	Default
Dial Tone Timer (sec)	The time for which the system plays the Dial tone.	2 to 255	7 seconds
Ring Back Tone Timer (sec)	The time for which the system plays the Ring Back Tone.	2 to 255	45 seconds
Busy Tone Timer (sec)	The time for which the system plays the Busy Tone.	1 to 255	7 seconds
Error Tone Timer (sec)	The time for which the system plays the Error Tone.	1 to 255	30 seconds
Feature Confirmation Tone Timer (sec)	The time for which the system plays the Confirmation Tone when a feature is set or canceled.	1 to 255	7 seconds
Programming Error Tone Timer (sec)	The time for which the system plays the Error Tone when you have entered an invalid command string while configuring a feature from a phone.	1 to 255	3 seconds
Programming Confirmation Tone Timer (sec)	The time for which the system plays the Confirmation Tone when a system command is successfully executed when configuring the system from a phone.	1 to 255	3 seconds

Name	Description	Range	Default
Call Forward - No Reply Timer for Department Group (sec)	The time for which the system will wait for an extension (Department Group member) to answer an incoming call, before forwarding the call to the programmed destination.	1 to 255	30 seconds

Auto Attendant

Name	Description	Range	Default
Auto Attendant Answer Wait Timer (sec.)	The time for which the system waits before answering a Auto Attendant call.	0 to 255	5 seconds
Auto Attendant Beeps Timer (sec.)	The time for which the system plays beeps to the caller to prompt the caller to dial the desired extension number when the call is answered by the Auto Attendant.	0 to 255	10 seconds
Auto Attendant Ring Timer (sec.)	The time for which the system rings on the landing destination extension in an Auto Attendant call.	1 to 255	30 seconds
Auto Attendant Error Tone Timer (sec.)	In an Auto Attendant call, the time for which the system plays Error Tone to the caller, if the caller has dialed an invalid code.	1 to 255	5 seconds

Direct Inward System Access (DISA)

Name	Description	Range	Default
DISA Idle State Timer (sec.)	In a DISA PIN Authentication call, the time for which the system waits for the caller to go Off-hook after entering DISA. If the caller does not go Off-hook within this timer, the system releases the call.	0 to 255	20 seconds
DISA Inactivity Timer (min.)	In a DISA call, the time for which the system waits for the caller to dial digits. If the caller does not dial any digit within this timer, the system disconnects the call.	0 to 255	2 minutes

Message Notification

Name	Description	Range	Default
Message Notification Retry Count	This count defines the number of times the system must make Message Wait Notification Calls to the destination number.	0 to 15	3 Attempts
Message Notification Ring Timer (sec)	The time for which the extension on which the Message Wait Notification Call is made will ring.	1 to 255	45 seconds
Message Notification Interval (min)	This defines time interval between two retries. It is the time after which the system must make another attempt to place the notification call on the destination	1 to 255	5 minutes

Other Features

Name	Description	Range	Default
Auto Call Back Ring Timer (sec.)	The time for which the extension requesting the Auto Call Back and the destination extension will ring.	1 to 255	30 seconds
Interrupt Request Timer (sec.)	The time for which the extension on which the Interrupt Request is made will get the beeps.	1 to 255	45 seconds
Barge-In Timer (sec.)	The time after which the extension that has activated Barge-In gets connected to the extension which is barged in.	1 to 255	10 seconds
Transfer while Ringing Timer (sec.)	When an extension transfers a call to another extension after it starts ringing, this is the time for which the system will wait for the transfer target extension to answer the call. If the transfer target does not answer the call within this timer, the call is returned to the transferor.	1 to 255	30 seconds
Transfer on Busy Timer (sec.)	When a call is transferred to a Busy extension, this is the time for which beeps are played on the transfer target extension.	1 to 255	30 seconds
Call Park Timer (min.)	The time after which the call comes back to the extension that has parked the call.	2 to 255	2 minutes
Call Park Release Timer (min.)	The time after which the parked call gets disconnected.	1 to 255	3 minutes
Conflict Dialing Timer (sec.)	The time for which the system waits for the extension user to dial the next digit to resolve conflicting access codes dialed by the extension user.	1 to 255	2 seconds
Extension - Inter Digit Wait Timer (sec.)	The time for which the system waits for the extension user to dial the next digit.On the expiry of this timer, the system considers it as the end of number dialing.	2 to 255	7 seconds
SA Command - Inter Digit Wait Timer (sec.)	When the user dials SA Commands, the system waits for the SA Command - Inter Digit wait timer for the user to dial the next digit.On the expiry of this timer, the system considers it as end of number dialing and proceeds further with executing the command.	2 to 255	15 seconds
Trunk - First Digit Wait Timer (sec.)	The time for which the system waits for the extension user to dial the first digit, after grabbing the trunk.	1 to 255	25 seconds
Trunk - Inter Digit Wait Timer (sec.)	When an extension user has grabbed the trunk and is dialing a number, the system waits for the Trunk-Inter-Digit wait timer for the extension user to dial the next digit. On the expiry of this timer, the system considers it as end of number dialing and proceeds with the call.	1 to 255	3 seconds
Global Hold Retrieval Timer (sec.)	This is the time for which a call put on Global Hold remains connected in the system. If the call put on Global Hold is not retrieved within this timer, the call is returned to the Extended IP Phone which put it on hold.	1 to 999	60 seconds

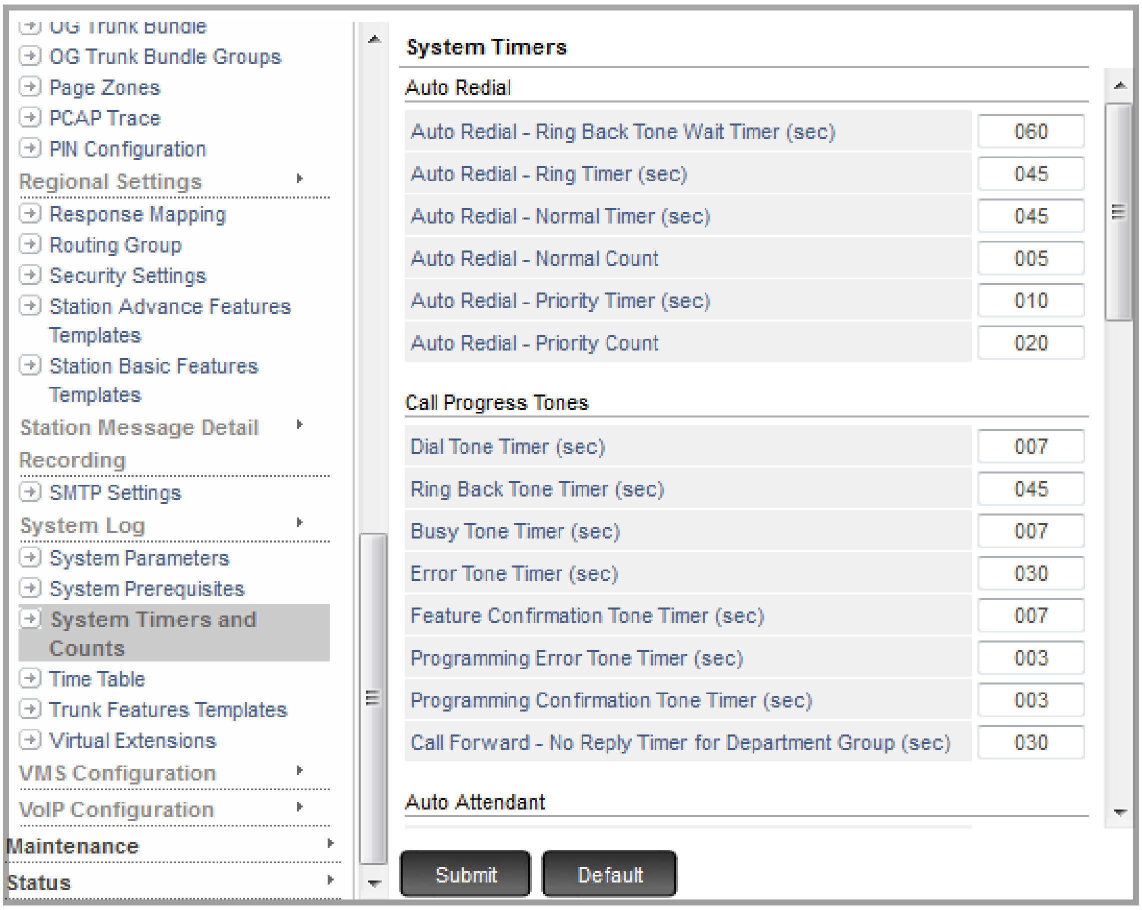
Name	Description	Range	Default
Exclusive Hold Retrieval Timer (min.)	This is the time for which a call put on Exclusive Hold remains connected in the Extended IP Phone. If the call put on Exclusive Hold is not retrieved within this timer, the call is returned to the Extended IP Phone which put it on hold.	1 to 255	2 minutes
RCOC Record Delete Timer (min.)	This is the time for which the record of the outgoing call is stored in the RCOC Table. The timer is activated whenever a record is stored in the RCOC table. At the end of this timer, the system deletes this record from the table.	1 to 999	999 seconds
Release Conference if Idle for more than (min.)	This is the time for which the system will wait for participants of a Dial-In Conference to withdraw or release themselves from the conference, one-by-one. On the expiry of this timer, the system releases the Dial-In Conference and frees the resource occupied by this conference in the conferencing circuit.	1 to 255	2 minutes
Retry Counts for Authority Code	When the user enters a wrong Authority Password, this count defines the number of times the system allows the user to re-enter the password.	0 - 9	3 Attempts
Emergency Reporting Call - Ring Timer (min)	The time for which the Operators extension rings, to inform the operator the number of the extension that dialed out an emergency number.	1 to 255	10 minutes
Held Call Disconnection Timer (min)*	The time after which a held call will be disconnected, if not retrieved.	1 to 255	5 minutes
Web Interface Logout Timer (min)	After user logs into Jeeves and there is no activity, the system waits for the expiry of this timer. Thereafter the user is automatically logged out.	1 to 255	60 minutes
VARTA Client Inactivity Timer (days)	The time for which the server will send the Push Notifications to the VARTA application, when in background. On expiry of this timer, the server will consider the application as unregistered and will stop sending the Push Notifications to the application, even when the application has a persistent internet connection.	1 to 255	10 days

**Applicable for Matrix SPARSH VP330, SPARSH VP210 and Matrix VARTA Mobile UC Clients.*

How to configure

Configuring Timers and Counts

- Login as System Engineer.
- Under **Configuration**, click **System Timers and Counts**.



The Timers and Counts on this page are arranged by the name of the feature or function these are related to.

- Change the value of the Timer or Count by entering the desired duration or count in the respective fields.
- Click **Submit.**

System Security

Access to the ANANT UCS is secured at three levels by way of a password:

- at the System Engineer Level with the System Engineer (SE) password
- at System Administrator Level with the System Administrator (SA) password
- at the User Level with the User Password

The System Engineer and the System Administrator passwords secure the system settings from access and alteration by unauthorized persons thus preventing possible misuse of the features and facilities.

For details, see "System Engineer (SE) Password" and "System Administrator (SA) Password".

System Engineer (SE) Password

For Accessing Jeeves

The SE password is a code used to prevent unauthorized access and alterations or misuse of the features and facilities. As this password is meant for restricting access to the SE mode, we strongly recommend you to:

- Keep the password confidential.
- Select a complex password that cannot be easily guessed.
- Change the password regularly
- Not use the **"Remember Password**" property of your Web Browser.

To provide additional security,

- the password will be valid for 90 days and you will not be able to login with the existing password. You will be prompted to change the password.

- if you enter a wrong password five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

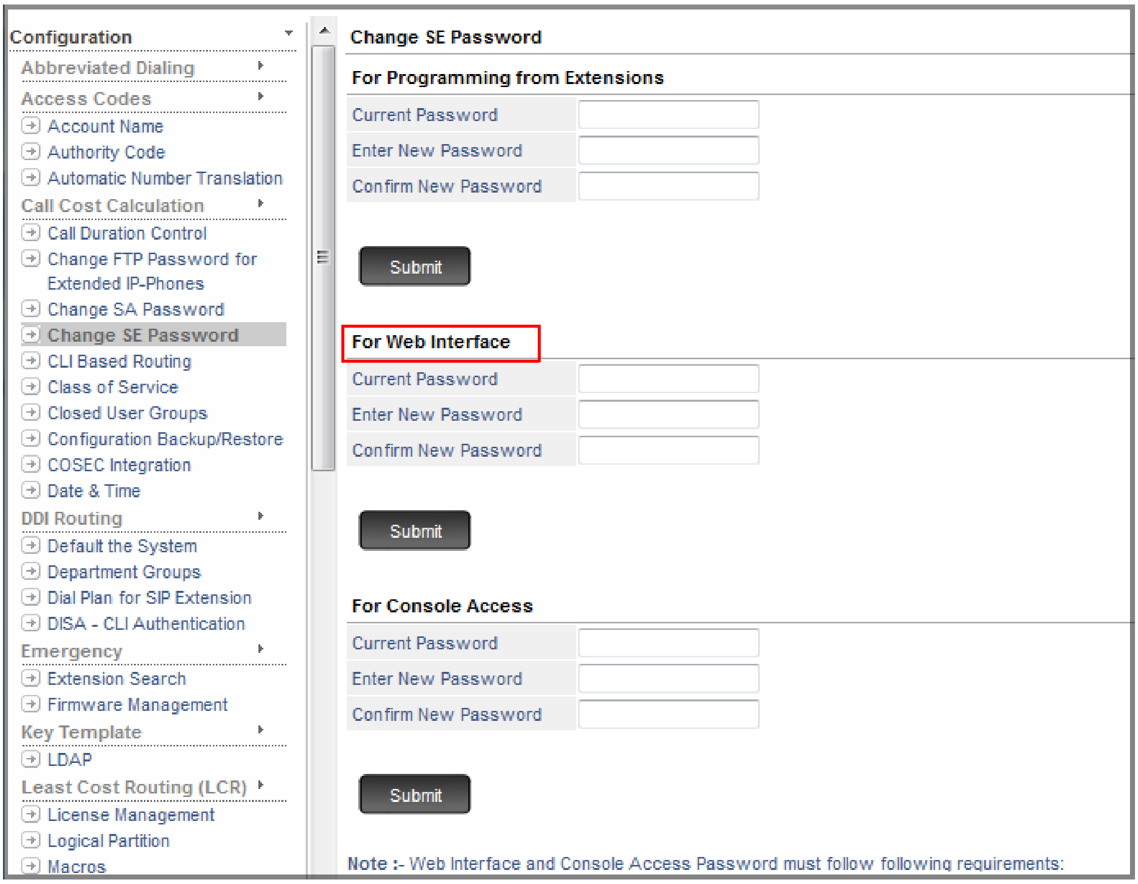
- *In case, you forget the SE Password, contact the Matrix Technical Support Team for assistance.*

- *You are advised to record and store the SE password at a safe place, where it can be accessed by you (the System Engineer) to avoid the inconvenience of restoring the default SE password.*

Changing the SE Password for accessing Jeeves

- Login as System Engineer.

- Under **Configuration**, click the **Change SE Password**.



- Under **For Web Interface**,

  - Enter **Current Password.**

  - **Enter New Password**.

    All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ’ , Double Quote " and Space) are allowed. The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - include atleast one upper-case, one lower-case, one number and one special character.

  - In **Confirm New Password**, re-enter the new password to confirm.

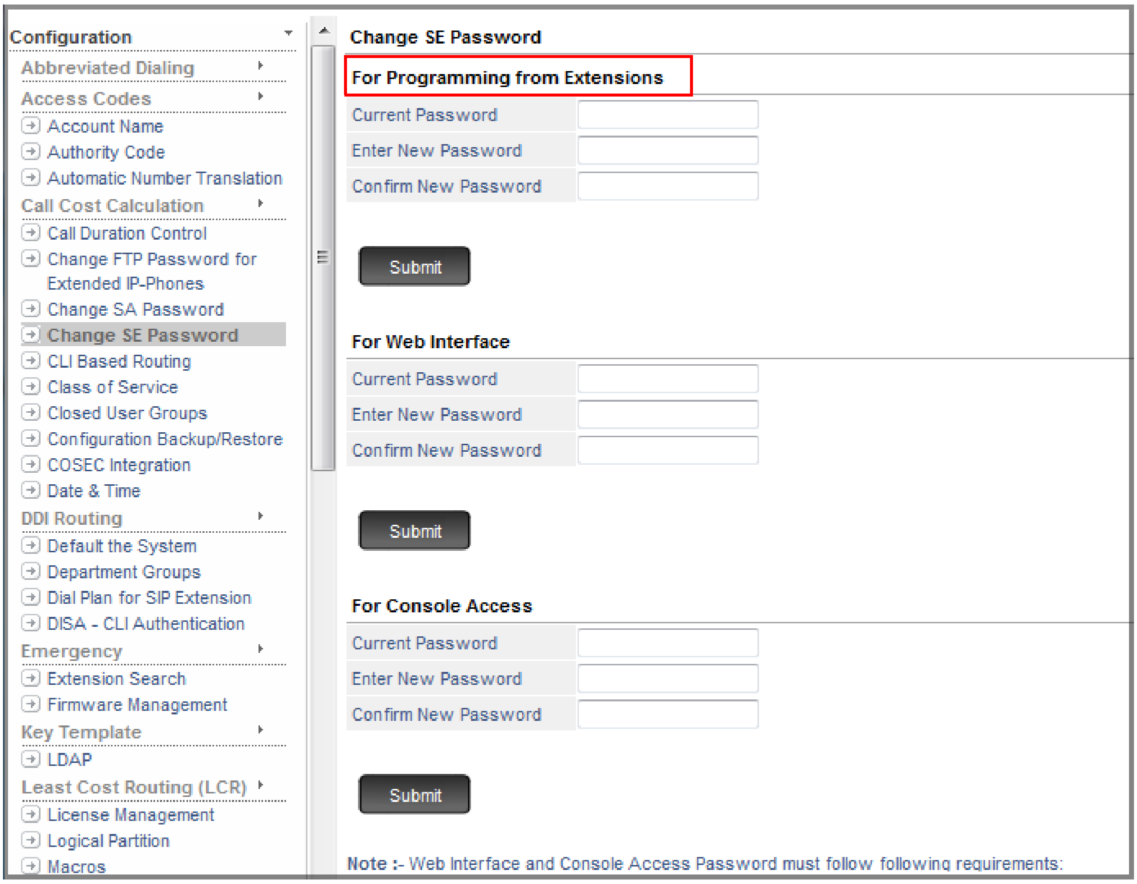
  - Click **Submit**.

For Programming from Extensions

The SE password is a code used to restrict unauthorized access to the SE Mode.The SE Password for programming from extensions can be changed from Jeeves.

Changing the SE Password for programming from Extension

- Login as System Engineer.
- Under **Configuration**, click **Change SE Password.**



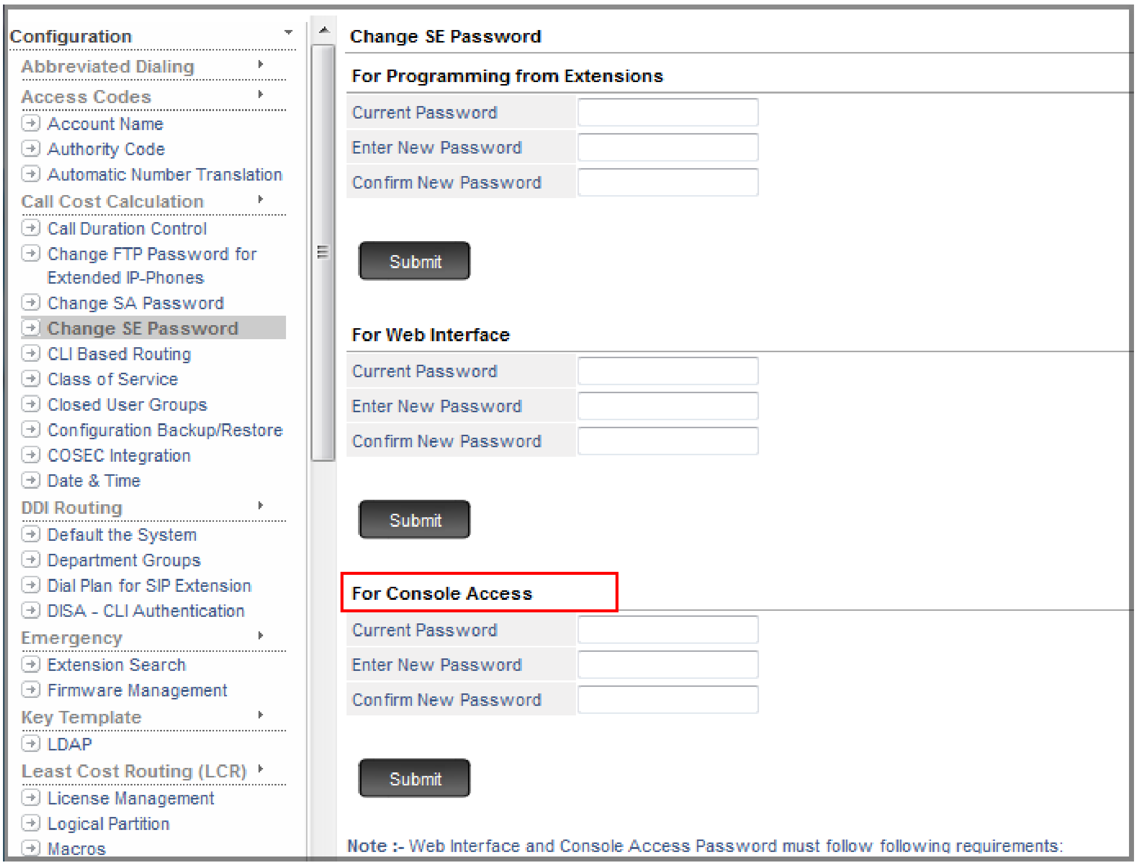
- Under **For Programming from Extensions**,
  - Enter **Current Password.**
  - **Enter New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit**.

For Console Access

The Console Access password is used to restrict unauthorized access to the Console Screen.The Console Access password for accessing the console screen can be changed from Jeeves.

Changing the SE Password for Console Access

- Login as System Engineer.
- Under **Configuration**, click **Change SE Password.**



- Under **For Console Access**,
  - Enter **Current Password.**
  - **Enter New Password**.

    All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ’ , Double Quote " and Space) are allowed. The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - include atleast one upper-case, one lower-case, one number and one special character.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit.**

Forgot the SE Password?

If you have already changed the default SE Password for accessing Jeeves and are unable to recall or locate it, then contact the Matrix Technical Support Team to restore the default SE password.

System Administrator (SA) Password

For Accessing Jeeves

The SA password is a code for preventing unauthorized access to the SA mode. As this password is meant for restricting access to the SA mode, we strongly recommend you to:

- Keep the password confidential.
- Select a complex password that cannot be easily guessed.
- Change the password regularly.
- Not use the **"Remember Password**" property of your Web Browser.

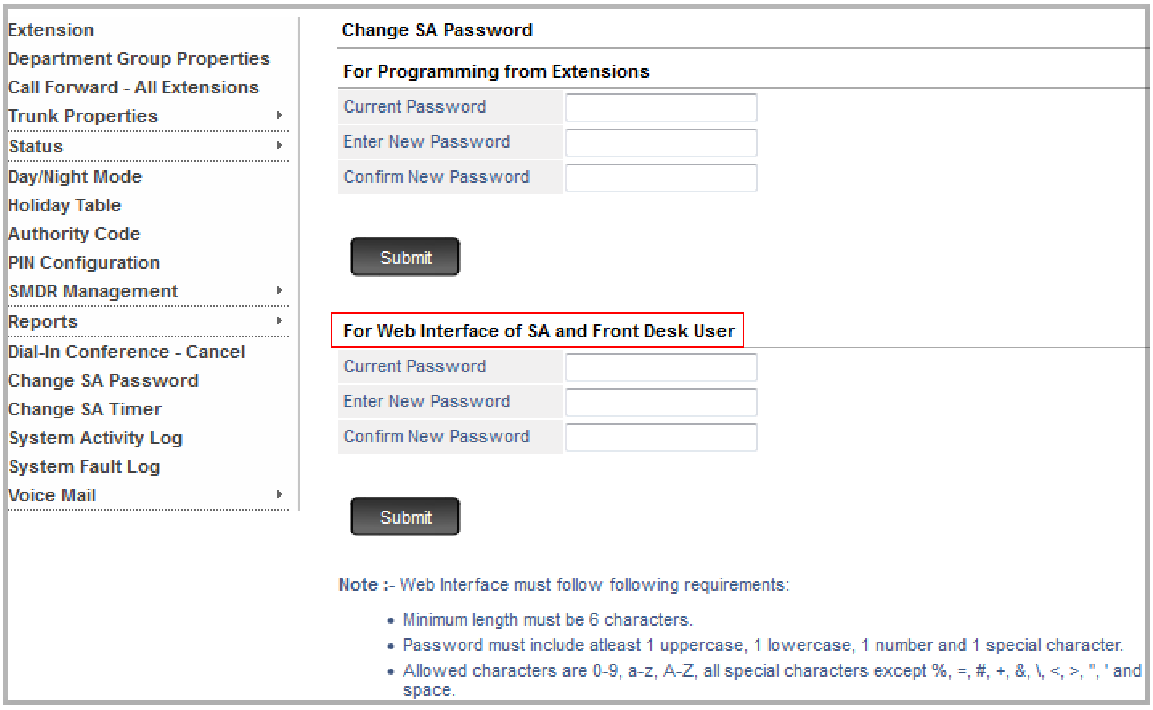
To provide additional security,

- the password will be valid for 90 days and you will not be able to login with the existing password. You will be prompted to change the password.
- if you enter a wrong password five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. This activity will be logged in the "System Activity Log" as well as "Simple Network Management Protocol (SNMP)".

- *In case, you forget the SA password, contact your System Engineer for assistance. The System Engineer can generate the new SA password from Jeeves.*
- *You are advised to record and store the SA password at a safe place, where it can be accessed by you to avoid the inconvenience of restoring the default SA password.*

Changing SA Password for accessing Jeeves

- Login as System Administrator.
- Click **Change SA Password**.



- Under **For Web Interface of SA and Front Desk User,**
  - Enter **Current Password.**
  - **Enter New Password.**

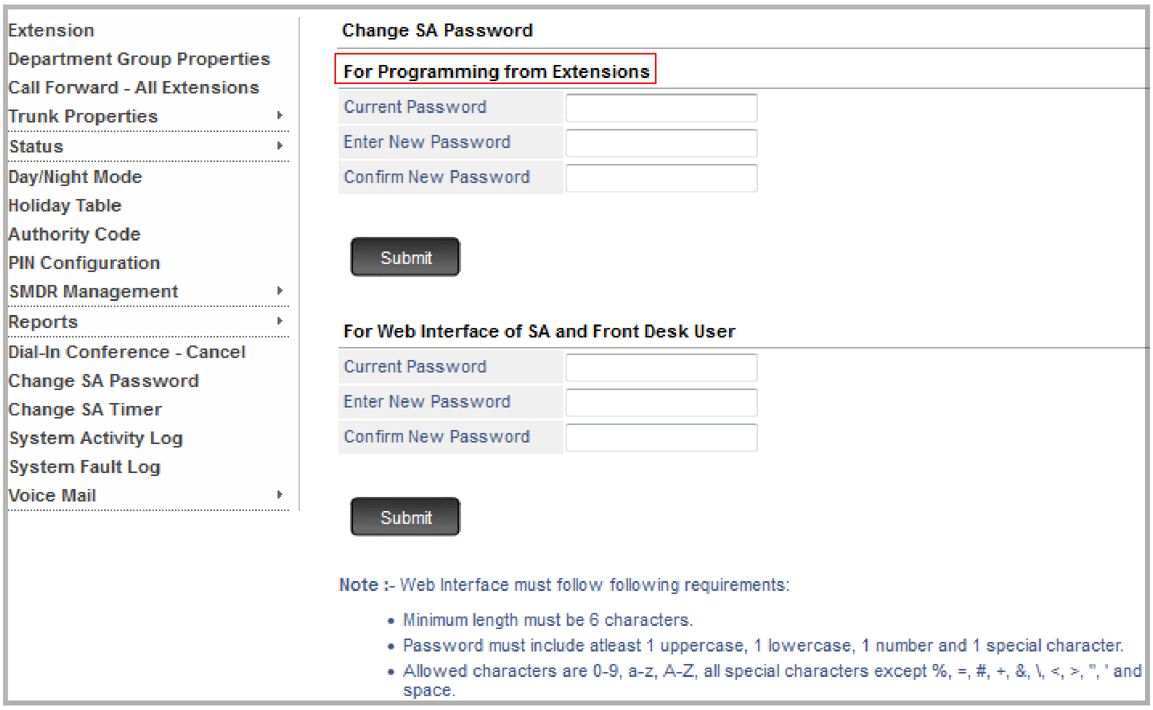
    All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ’ , Double Quote " and Space) are allowed. The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - include atleast one upper-case, one lower-case, one number and one special character.
  - In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit.**

For Programming from Extensions

The SA password is a code used to restrict unauthorized access to the SA Mode.The SA Password for programming from extensions can be changed from Jeeves.

Changing SA Password for programming from Extension

- Login as System Administrator.
- Click **Change SA Password**.



- Under **For Programming from Extensions**,
  - Enter **Current Password.**
  - **Enter New Password**. The new password can be a minimum of 4 digits to a maximum of 12 digits. The valid digits are from 0 to 9.
  - In **Confirm New Password**, re-enter the new password to confirm.
  - Click **Submit.**

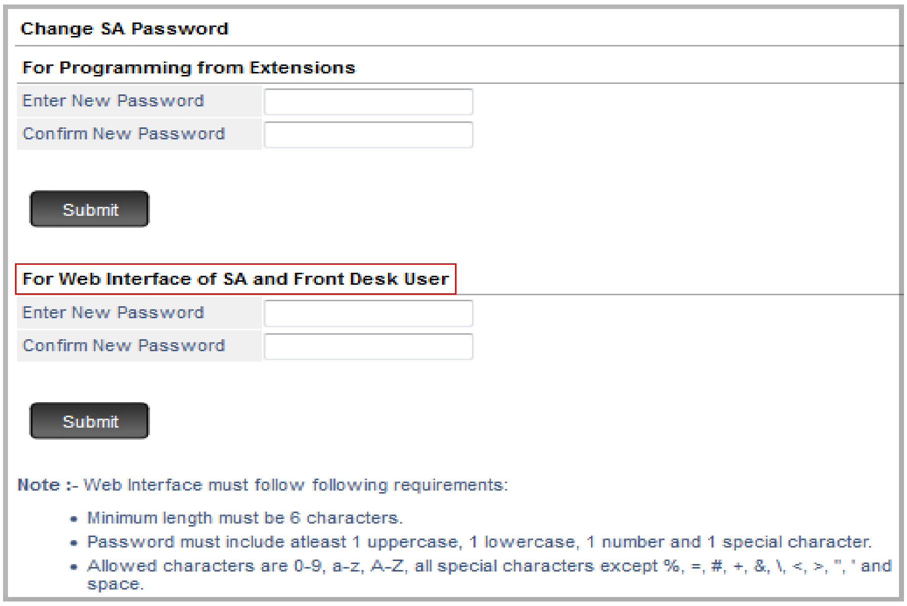
Forgot the SA Password?

In case the System Administrator has forgotten the password, a new password can be issued by the System Engineer only.

The System Engineer can assign a new SA password from Jeeves only.

To issue a new SA Password from Jeeves,

- Login as System Engineer.
- Under **Configuration**, click **Change SA Password**.



- Under **For Web Interface of SA and Front Desk User,**
  - **Enter New Password.**

    All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ' , Double Quote " and Space) are allowed. The new password must be:
    - a minimum of 6 characters to a maximum of 12 characters.
    - include atleast one upper-case, one lower-case, one number and one special character.
  - In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit.**

User Password

Extension Users can secure their respective stations/extensions from unauthorized use with a password unique to each extension. The User password too is a combination of any four digits, from 0 to 9. The default User Password is ***1111***, which each user can change from their respective extensions.

To avoid unauthorized access,

- the extension users must change the default password
- make sure the new password is strong and is kept confidential
- change the password regularly

Refer the topic "User Password" to know more.

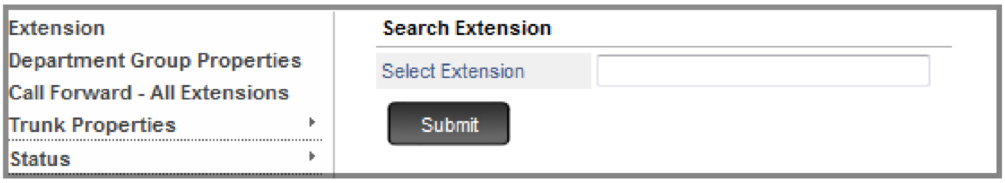
Forgot the User Password?

If an extension user forgets the User Password, a new password can be issued to the extension user by the System Administrator.

To issue a new User password,

- Login as System Administrator.

- Click **Extension**.



- In **Select Extension,** enter the Number or the Name of the extension you want to search.

- Click **Submit.**

  The page of the desired extension will appear.

- Click **Phone Properties** to expand.

- Enter the new User Password in the field **Change User Password to**. The User Password may be a combination of 4 digits. Valid digits: 0 to 9.

- Click **Submit.**

Additional Security to Extension Users

It is also possible to secure unattended extensions. This may be necessary when extension users forget to lock their extensions and are away from their desks. In such situations, the System Administrator can protect the extension from unauthorized access and use by:

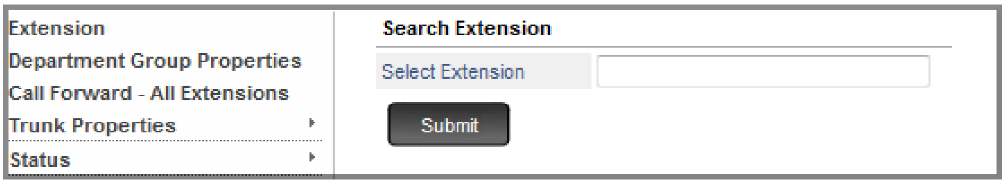
a. locking the Keypad of the extension phone; possible only on Extended IP Phone extensions.

b. setting the User Status for the extension as "Absent"; possible on Extended IP Phones. Read the feature description “User Absent/Present” to know more.

The System Administration can do this, lock the keypad of Extended IP extensions and set Extended IP Phones extension users as ‘Absent’.

To secure an extension,

- Login as System Administrator.
- Click **Extension**.



- **Select Extension:** Enter the Extension Number or the Extension Name of the extension you want to search.
- Click **Submit.**

  The page of the desired extension will appear.

- Click **Phone Properties** to expand.
- Change user status by selecting the option **Absent** for the parameter **Presence**. When you want to change user status again to present, select **Present.**
- Select the option **Lock** for the parameter **Keypad**. When you want to remove keypad lock, select **Unlock**.

When the keypad is locked, the features Call Log, Contact, Call Forward, Dynamic Lock, User Status, DND, Call Cost Display, Hotline, Alarm, Change User Password will not be accessible by the extension user.

- Click **Submit.**

- *Extension users can also set their status as 'Absent' or 'Present' from their respective extension phones. Refer “User Absent/Present”.*
- *Extended IP Phone extension users can also lock the keypad of their phones from the Phone Menu. Refer “Extended IP Phone/VARTA UC Client - Operation” for instructions on navigating the phone menu.*
- *It is also possible to lock/unlock the keypad and set the user extension status as 'Absent'/'Present' from a remote location using “Direct Inward System Access (DISA)”.*

FTP Access for Extended IP Phones

The access to the FTP Server of the Extended IP Phones is secured by a password. The FTP password is a code for preventing unauthorized access. The default FTP Password is 1234. As this password is meant for restricting access, we strongly recommend you to:

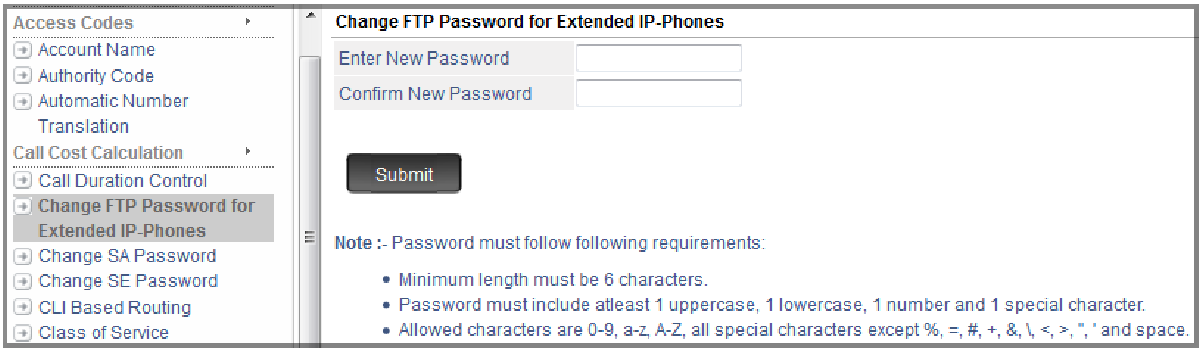
- change the default password.
- make sure the new password is strong and is kept confidential.
- change the password regularly.

- *When you set the System to default, the FTP Password will not be set to default.*
- *The FTP Password will not be set to its default value, when the SE Password is set to default.*

Changing the FTP Password

- Login as System Engineer.

- Under **Configuration**, click **Change FTP Password for Extended IP-Phones**.



- **Enter New Password**.

- In **Confirm New Password**, re-enter the new password to confirm.

- Click **Submit**.

Time Tables

Certain features of the ANANT UCS like Operator, Class of Service, Toll Control, Outgoing Trunk Access, among others, require extensions and trunks to behave differently according to the time of the day, which is referred to as Time Zone.

For example, incoming calls are to be routed to the security personnel extension, instead of the Operator when the office is closed, or certain features in the Class of Service are to be allowed only during working hours, or access to outgoing long distance calls are to be denied during non-working hours, or the station must play a different greeting message to the callers during break hours and holidays.

Time Tables can be assigned to extensions and trunks to define their behavior according to the time of the day, that is, Time Zone.

Time Zones

A day can be divided into three time zones: Working hours, Break hours and Non-working hours. The default Time Zones defined for each day are:

- Working hours: 09:00 to 18:00
- Break hours: 13:00 to 14:00
- Non-working hours: 18:00 to 09:00

Working, Break and Non-Working hours are set to 00:00 for Sunday.

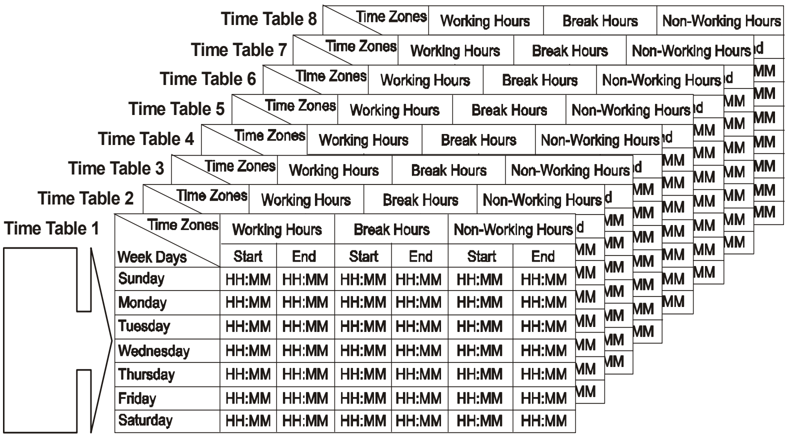
You can define a different Time Zone for your organization. Further, you can program each day of a week with different time zones. For example, you may define the Working hours from Monday to Friday as 09:30 to 18:30, and for Saturday, from 09:30 to 15:00. If you have a 24x7 business, you may set Working Hours also for Sunday.

Time Tables

A Time Table is a schedule of the three Time Zones, namely: Working Hours, Break Hours, Non-Working hours, for the entire week.

A Time Table is assigned to extensions defining the Time Zones for the entire week, so that the system can execute the Time Zone-dependent features and facilities according to the Time Table.

There are 8 different Time Table templates to select from.



By default, the Time Table 1 is assigned to all extensions and trunks in their Station Basic Feature Template and Trunk Feature Template respectively. In Time Table 1, six days of the week - Monday to Saturday have working hours from 9:00-18:00, break hours from 13:00-14:00 hours and non-working hours from 18:00 to 09:00. Sunday is a holiday, with all three Time Zones set to 00:00 hours.

You may also customize the default Time Table 1 OR customize and assign a different Time Table to the extensions and trunks.

ANANT UCS offers the facility to switch the system into Day/Night Mode at any point of time. When you set the system in Day/Night Mode, the system overrides the Time Tables assigned to Trunks, extensions and Operator. According to the mode you selected, it applies Working Hours/Non-Working Hours to run all the Time-Zone dependent features of the system.

Refer the topic "Day Night Mode" to know more.

How to configure

A Trunk can be assigned Time Table in the "Trunk Feature Template" assigned to it. An extension can be assigned a time table in the "Station Basic Feature Template" assigned to it.

The default Time Table 1 is assigned to both extensions and trunks of ANANT UCS. Check if this time table matches the working hours of the organization, and the Time Zone requirements of the individual extensions and trunks.

The following extension parameters can be configured differently for different Time Zones:

- Class of Service.
- Toll Control.
- OG Trunk Bundle Group.

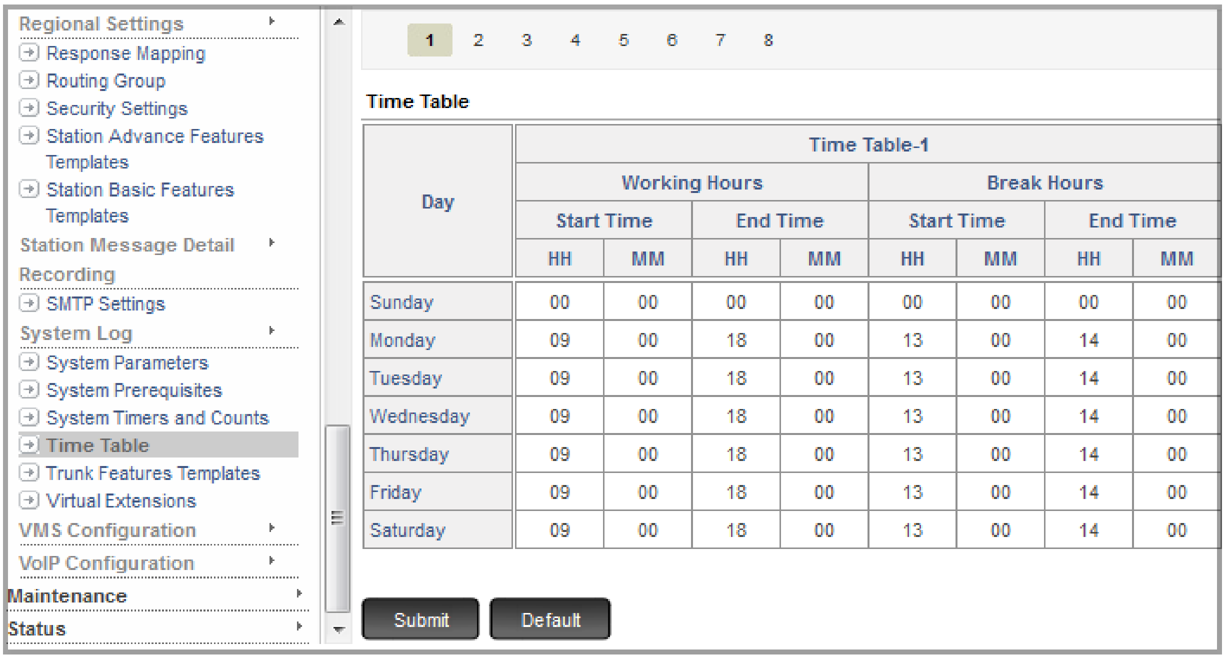
The following trunk parameters can be configured differently for different Time Zones:

- Auto Attendant
- DISA
- Trunk Landing Group

You may retain the default Time Table 1 or customize it to suit your requirements. Or you may customize different Time Tables and assign them to different extensions and trunks.

Configuring Time Table

- Login as System Engineer.
- Under **Configuration**, click **Time Table**.



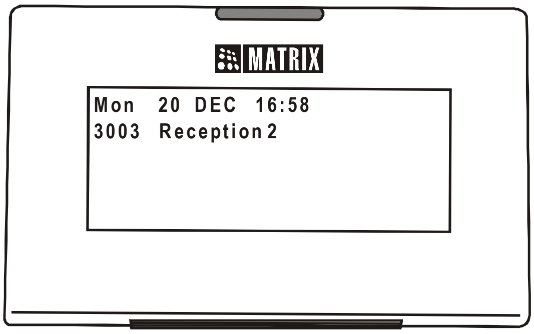
- Select the desired Time Table number and define the Time Zones, that is, working hours, break hours and non-working hours.
- Click **Submit.**
- Now, assign the Time Table you program to the desired Extension/Trunk.
- To assign Time Table to extension, go to Station Basic Feature Template. Refer the topic “Station Basic Feature Template” for instructions.
- To assign Time Table to Trunks, go to Trunk Feature Template. Refer the topic “Trunk Feature Template” for details on how to assign timetable to trunks.

Time Zone Display

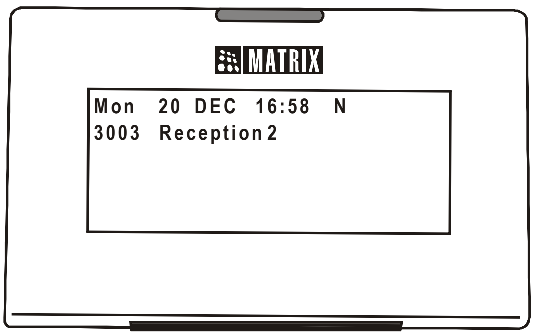
The current time zone—Working Hours, Break Hours and Non-working Hours—is displayed on the LCD of the Extended IP Phone[121].

During Non-working hours the letter ‘N’ is displayed on the LCD of the Extended IP Phone in the idle state, and during Break hours, the letter ‘B’ is displayed.

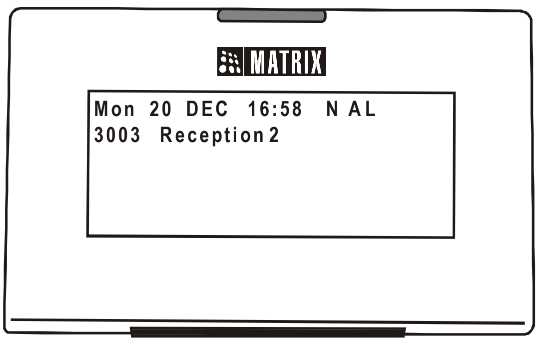
During working hours, in the idle state, the phone display will look like this:



During Non-working hours, in the idle state, the phone display will look like this:



During Non-working hours, in the idle state, if the extension user has set User Absent and activated Keypad Lock on the phone, the phone display will look like this:



121. This feature is explained with reference to SPARSH VP248. The display varies with phone type.

Toll Control

Toll Control (or Toll Restriction) is an expense control feature of ANANT UCS. It enables you to configure the system so that each extension has a designated calling permission referred to as 'Call Privilege'.

Each type of Call Privilege allows the extension to call certain areas and restricts it from calling others.The extension can also be restricted from dialing the specific telephone numbers.

ANANT UCS supports five types of Call Privileges, these are:

- **No Calls:** Dialing of all external numbers is restricted. Only internal (extension-to-extension) calls are allowed.

  By default, only the numbers configured in Global Directory Part 1 will be allowed to be dialed out as it is enabled in the Class of Service of the extension.

  If you have enabled Global Directory Part 2 and 3 in the Class of Service of the extension, the numbers configured in these also will be allowed to be dialed out.

- **Local Calls:** Dialing of outgoing calls to Local area numbers, in addition to internal calls, is allowed. It is possible to restrict calls to certain local numbers. To apply this Call Privilege, you must configure the 'Local Numbers' list.

- **Regional Calls:** Dialing of outgoing calls to regional numbers is allowed, in addition to internal and local calls. It is possible to restrict calls to certain regions. To apply this Call Privilege type, you must configure the 'Regional Numbers' list.

- **National Calls:** Dialing of domestic, long-distance numbers within the country is allowed, in addition to internal and regional calls. You can also restrict calls to certain parts of the country. To apply this Call Privilege type, you must configure the 'National Numbers' list.

- **International Calls:** Dialing of international numbers is allowed, in addition to local area, long distance and internal numbers. You can also restrict calls to certain countries. To apply this Call Privilege type, you must configure the ‘International Numbers’ list.

- **All Calls:** Dialing of all types of numbers - local, regional, national, international - is allowed, without any restriction.

- **Limited Calls:** Dialing of only specific Telephone numbers (local, regional, national or international) is allowed. By applying this Call Privilege type, you can allow and restrict dialing of telephone numbers starting with a particular digit, or a particular area code, or certain telephone numbers only. To apply this Call Privilege type, you must program a list of the ‘Limited Numbers’ that are to be allowed and numbers that are to be restricted. You can configure three such ‘Limited Numbers’ lists.

Toll Control forms the basis of the features Dynamic Lock, Call Budget on Extension and Call Budget on Trunk.

Using “Dynamic Lock”, extension users can change the Toll Control (Call Privilege) of their extensions on their own. The Operator or System Administrator can also change the Toll Control of the extension using Dynamic Lock. To support this feature, ANANT UCS offers fours levels of Toll Control, from 0 to 3.

Call Budget is a cost control feature that allows you to keep a control on the total cost of phone calls made by the extension users. You may also define the calling permission for extensions and trunks can be allotted a budget. For more details, see “Call Budget on Extension” and “Call Budget on Trunk”.

Toll Control Levels

For each Toll Control Level from 0 to 3, a 'Call Privilege' is defined. The system applies the Toll Control Level (that is, the Call Privilege) set by the extension users themselves or set by the System Administrator/Operator for the extensions using Dynamic Lock.

- **Toll Control - Level 0** is Time Zone based, wherein you must define the Call Privilege Type for each Time Zone, that is, Working Hours, Break Hours and Non-Working Hours. For instance, you may define 'All Calls' as Call Privilege for Working Hours, 'Local Calls' as Call Privilege for Break Hours and 'No Calls' as Call Privilege for 'Non-Working' Hours.

  By default, Call Privilege 'No Calls' is selected for all three Time Zones.

- **Calls allowed for Toll Control Level 1** is not based on Time Zones. By default, the Call Privilege Type for this level is 'No Calls'.

- **Calls allowed for Toll Control Level 2** is not based on Time Zones. By default, the Call Privilege type set for this level is 'No Calls'.

- **Calls allowed for Toll Control Level 3** is not based on Time Zones. By default, Call Privilege 'No Calls' is selected for this level.

- **Calls allowed when Call Budget is consumed** is applied only if the “Call Budget on Extension” feature is enabled on the extension.

- If the feature Call Budget on Trunks is enabled and the budget is exhausted, no calls will be routed through the trunk. For details, see “Call Budget on Trunk”.

ANANT UCS offers you the flexibility to redefine the Call Privilege for each of the above Toll Control Levels according to user requirements.

How it works

- When a call is made, the ANANT UCS checks the Toll Control Level assigned to the extension making the call.

- The system checks the 'Call Privilege' programmed in the Toll Control Level of the extension.

- For each call privilege type detected, the system will check the following to determine if call is to be allowed or denied, as summarized in the table below:

Type Call Privilege detected	Logic will check
Local calls	Local Number List configured.
Regional calls	Regional Number List configured
National calls	National Number List configured
Internationals calls	International Number List configured

Type Call Privilege detected	Logic will check
Limited Calls	Limited Number list (1,2 or 3) assigned to the extension

- The Local, Regional, National, International and Limited Calls Number Lists consist of Allowed Numbers and Denied Numbers.
- The system compares the each digit of the dialed number string with the number strings programmed in the Allowed and Denied Number Lists of the Local/Regional/National/International/Limited Number Lists, using the following logic:

Allowed Number List	Denied Number List	Result
match found	match found	Call allowed
match found	no match found	Call allowed
no match found	no match found	Call allowed
no match found	match found	Call denied

- The call is allowed to be made, if the dialed number:
    - matches with Allowed Number list and the Denied Number list.
    - matches with Allowed Number list, but not with the Denied Number list.
    - matches with neither the Allowed List nor the Denied List.
- The call is restricted, if the dialed number matches with the Denied Number list, but not with the Allowed Number list.

How to configure

Decide the type of Call Privilege you wish to assign to each SIP extension.

For Toll Control to work, you must first configure the lists of Local Numbers, Regional Numbers, National Numbers, International and Limited Numbers, according to the type of Call Privilege you want to assign to the extensions. To do this,

Make a two-column tables each for Local, Regional, National, International and Limited Call Numbers on paper or using a computer.

In one column of each list, write down the numbers you want to permit as Allowed Numbers. In the other column write down the numbers you want to restrict as Denied Numbers. Your Table may look like these:

List of Local Numbers for Call Privilege - Local Calls

Sr. No.	Allowed List	Denied List
1		
2		
:		
:		

Sr. No.	Allowed List	Denied List
999		

List of Regional Numbers for Call Privilege - Regional Calls

Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

List of Regional Numbers for Call Privilege - International Calls

Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

List of Limited Numbers for Call Privilege - Limited Calls 1

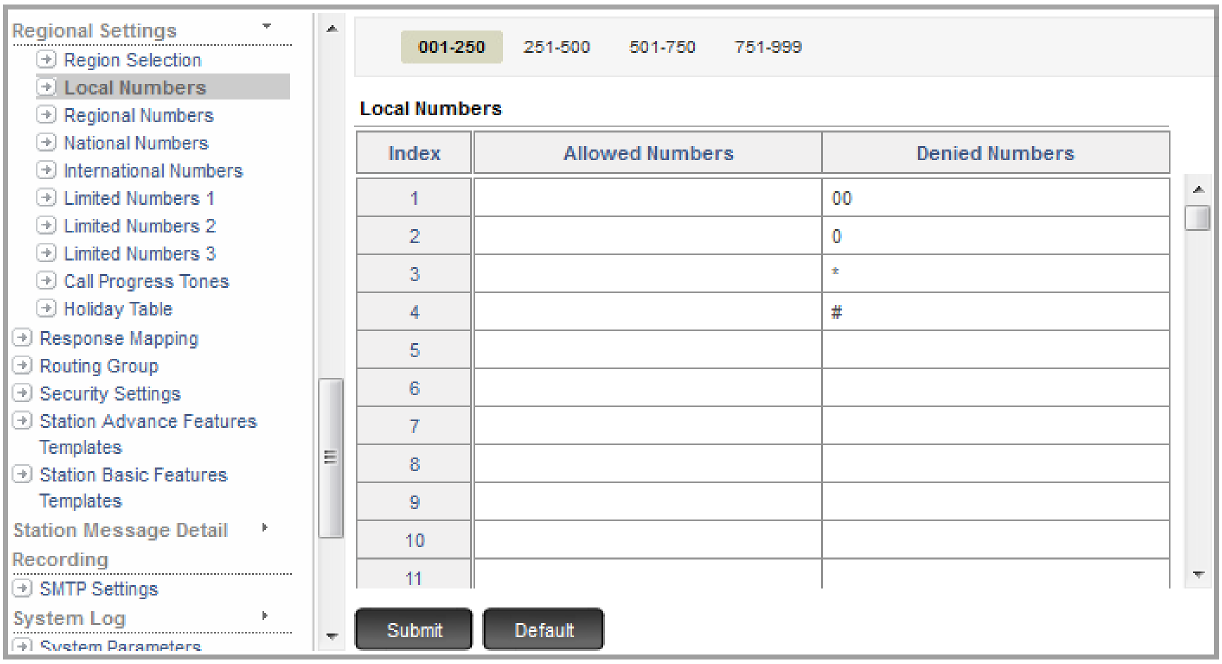
Sr. No.	Allowed List	Denied List
1		
2		
:		
:		
999		

Configuring Toll Control

- Login as System Engineer.
- Under **Configuration**, click **Regional Settings**.

Local Numbers

- Click **Local Numbers**.

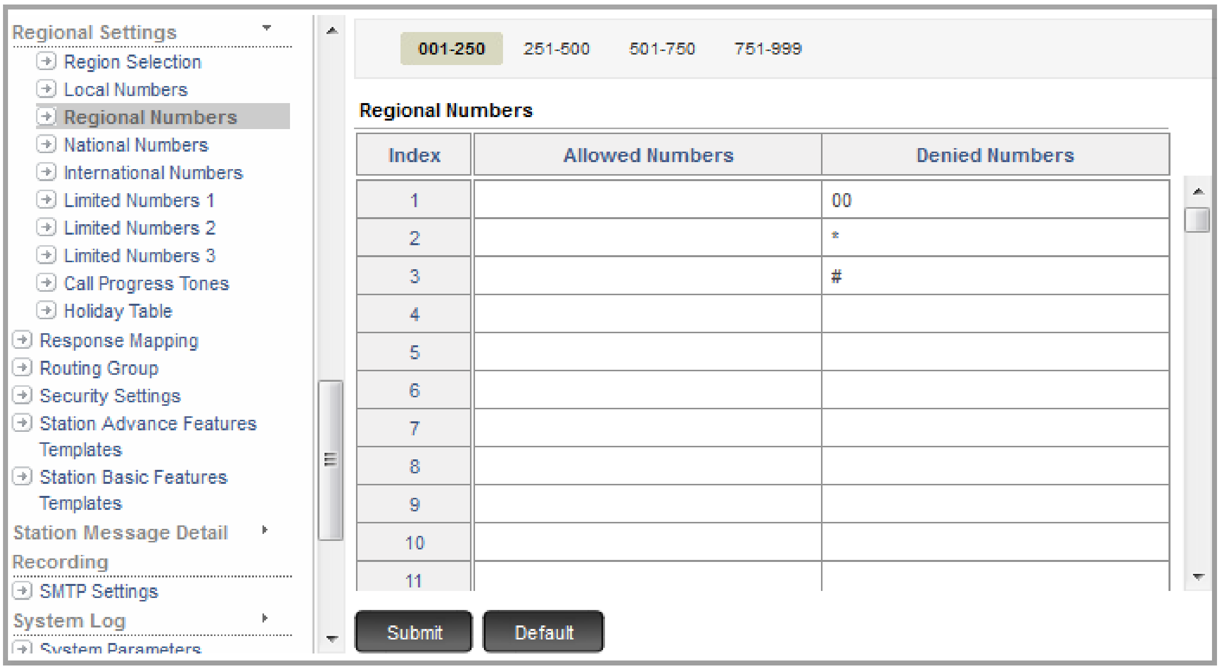


- Enter the local area numbers that are permitted to be dialed in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list. You may enter as many as 999 numbers in each list.

- Click **Submit.**

Regional Numbers

- Click **Regional Numbers.**



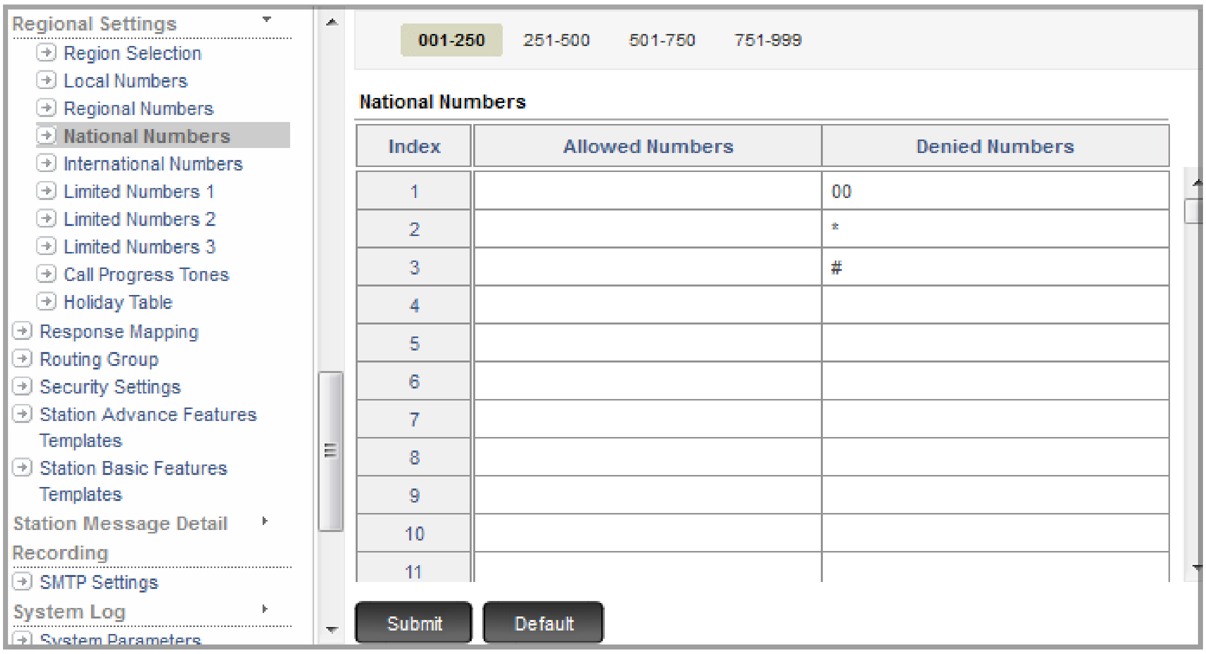
- Enter the regional area numbers that are permitted to be dialed in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list.

- Repeat the entries you made in the **Local Numbers** list also in the **Regional Numbers** list.

- Click **Submit.**

National Numbers

- Click **National Numbers.**



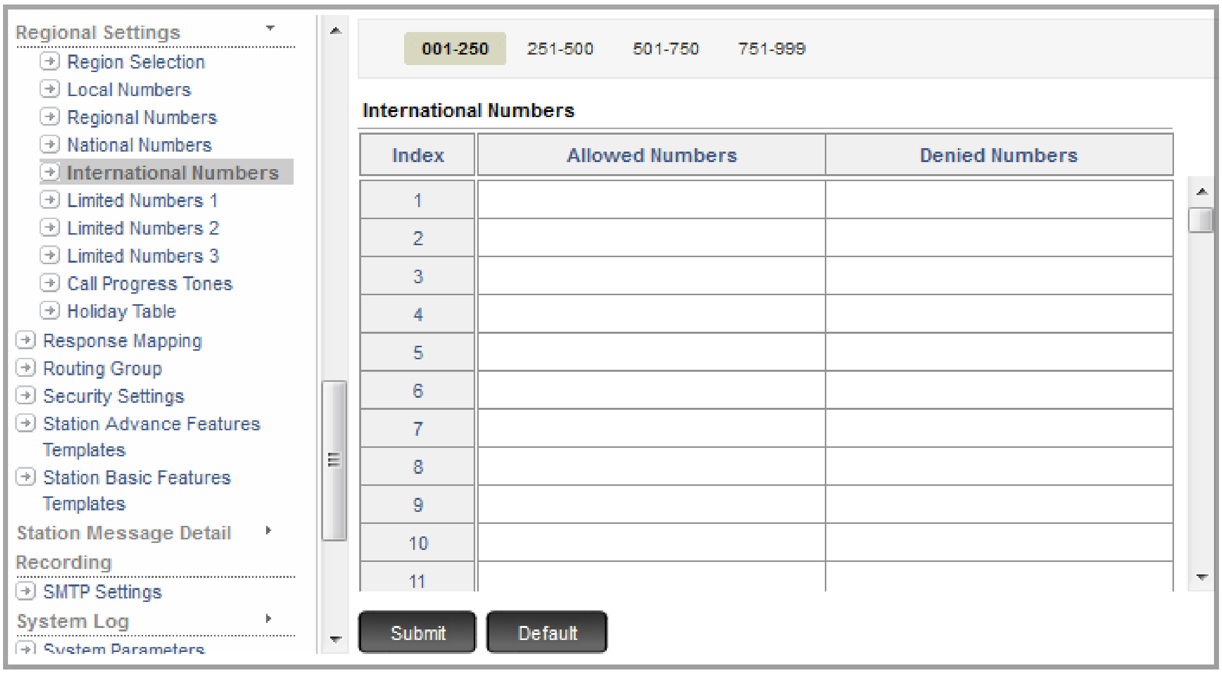
- Enter the long distance numbers within the country that are to be permitted in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list.

- Repeat the entries you made in the **Local Numbers** and **Regional Numbers** lists in this list.

- Click **Submit.**

International Numbers

- Click **International Numbers.**



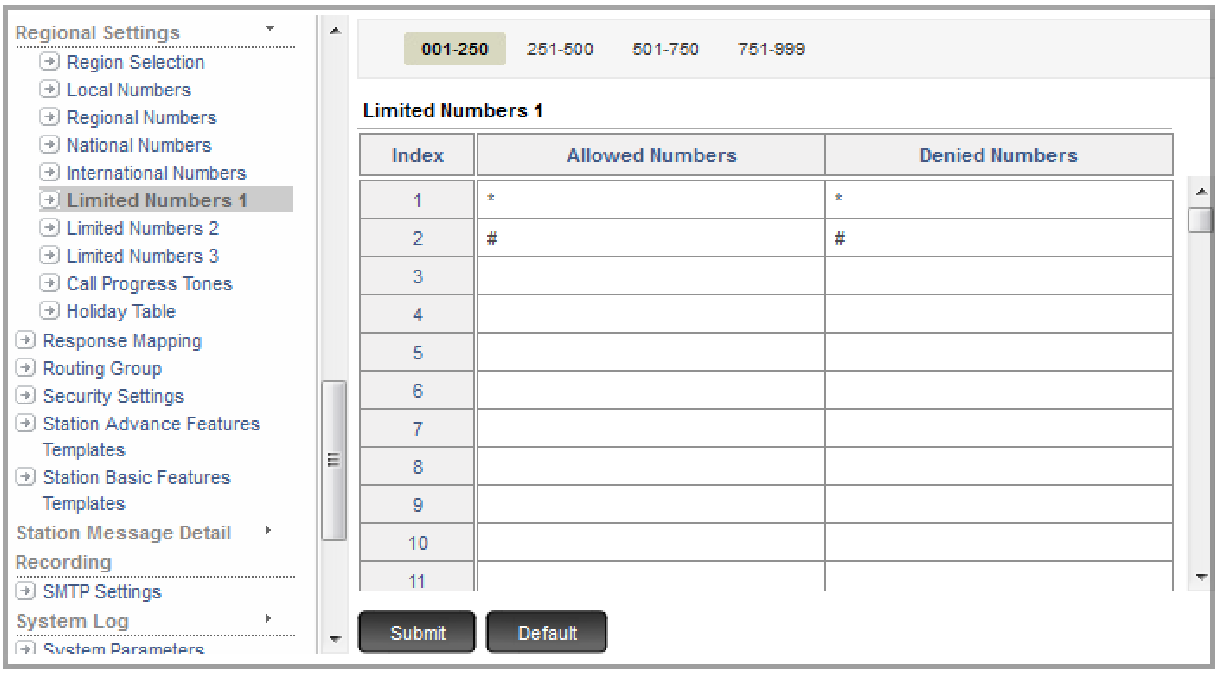
- Enter the overseas numbers that are to be permitted in the **Allowed Numbers** list and the numbers that are to be restricted in the **Denied Numbers** list.

- Repeat the entries you made in the **Local Numbers**, **Regional Numbers** and **National Numbers** lists in this list.

- Click **Submit.**

Limited Numbers

- Click **Limited Numbers 1.**



- Enter the specific numbers or digits that are to be allowed to be dialed in the **Allowed Numbers** list.

- Enter the specific numbers or digits that are to be restricted from being dialed in the **Denied Numbers** this list.

- Click **Submit.**

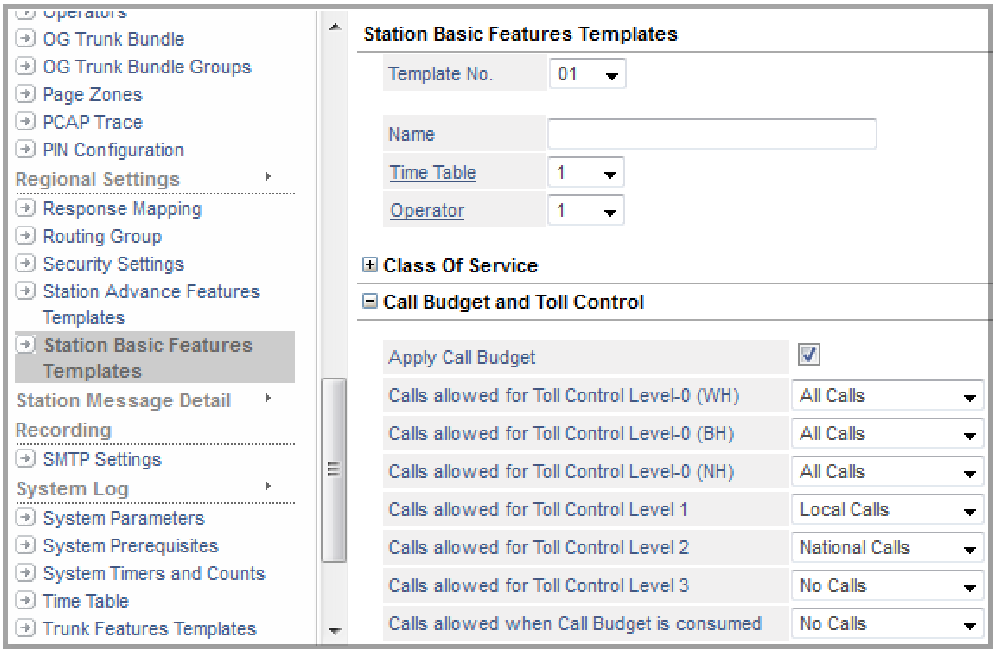
It is not mandatory to assign the same Limited-Calls Allowed-List and Denied-List for all Time Zones of Toll Control Level 0 or to other Toll Control Levels. You can prepare different Allowed and Denied Lists for each Toll Control Level.

Toll Control Levels

- Click **Station Basic Feature Template**.

- By default all extensions of ANANT UCS are assigned Station Basic Feature Template 01. You may customize this template or select another template.

- Click **Call Budget and Toll Control** to expand.



- Select the desired Call Privilege Type for each Time Zone - Working Hours, Break Hours, Non-Working hours.
- Similarly, select the Call Privilege type for other Toll Control Levels 1, 2 and 3.
- For the type of call privilege you select, the respective number list - Local, Regional, National, International or Limited- you configured will be automatically assigned.
- Click **Submit.**

Also refer the topics “Configuring Extensions”,“Station Basic Feature Template”,“Number Lists”.

Trunk Landing Group (TLG)

A Trunk Landing Group is a group of extensions on which incoming calls on a particular trunk are landed.

Trunk Landing Groups are formed for efficient call management. Generally, incoming calls on a trunk are landed on the Operator extensions. However, when several trunks are interfaced with the system, it becomes difficult for the Operator to answer all calls efficiently. Trunk Landing Groups relieve the Operator to a great extent, as the incoming calls get distributed among several extensions.

How it works

- A Trunk Landing Group (TLG) is a "Routing Group".
- You can configure as many as 96 TLGs. Each group is numbered from 01 to 96.
- A maximum of 32 extensions — SIP Extension, Virtual Extensions, OGTB or Voice Mail Auto Attendant— can be included in each Trunk Landing Group.

  To use the Gateway Application of ANANT UCS, select OGTB as the station.
- For each group that you create, you can do the following:
  - set the Sequence in which the extensions in the group should ring, by selecting the member extensions in a sequence from 1 to 32.
  - set the Time for which each extension in the group should ring, by setting the *Ring Timer* (default: 15 seconds).
  - set each extension to ring continuously till the call matures by enabling *Continuous Ring* (default: disabled).

    When Continuous Ring is enabled, once a extension receives a ring, it rings continuously till the call matures. The extension continues to ring even as other extensions in the group are hunted.

    If the call is not answered even after the last station in the group has been hunted, the system will loop back and start hunting from the first station, all over again.
  - have a number of extensions in the group ring simultaneously by enabling *Continuous Ring* on these extensions and setting the *Ring Timer* for these extensions to *'00'* seconds.
  - set equal distribution of incoming calls on all extensions in the group, by enabling *Rotation* for the entire group (default: disabled).

    When Rotation is enabled on a TLG, for each new call on a trunk, the system will land the call on the extension next to the one that received the last call.

    When Rotation is disabled in a TLG, for each new call on a trunk, the system will land the call on the first free extension of the TLG.
- To each Trunk, you must assign a TLG for the Time Zones, working hours, break hours and non-working hours. You may assign the same TLG for all three Time Zones, or a different TLG for each Time Zone.

How to configure

- According to the number of trunks interfaced with your ANANT UCS and the number of extensions you have, identify the trunks to which you want to assign TLG for each Time Zone. This will help you to decide the number of TLGs to be formed, the type and number of extensions in each group, and their sequence.
- Decide the Trunk number to which each TLG is to be assigned.
- Configure each TLG as a Routing Group. See “Routing Group” for instructions.
- Assign the TLGs you formed for each trunk for the three Time Zones in its Trunk Feature Template. for instructions, see “Trunk Feature Template”.

Example:

Two SIP trunks (configured on software ports 001 and 002) are interfaced with ANANT UCS.

- Incoming calls on SIP 1 during working hours should land on extensions 2001, 2003, 2005 (configured on software ports 0008, 0010, 0012 respectively).

  Incoming calls on SIP 1 during break hours and non-working hours should land on extensions 2002, 2004 (configured on software ports 0009, 0011 respectively).
- Incoming calls on SIP 2 should land always on extensions 3001, 3002, 3003, 3009, 3010 (configured on software ports 0013, 0014, 0015, 0016, 0017).
- Incoming call on SIP 1 should ring for 10 seconds on each extension in the TLG.
- Incoming call on SIP 2 should ring for 20 seconds on each extension in the TLG.
- The extensions of the TLGs of SIP 1 and SIP 2 should ring for the set time only.
- “Rotation Method” to be followed in the TLG of SIP 1 and SIP 2.

In this example, for SIP 1, you would need to form two TLGs; one TLG for working hours and one for break hours and non-working hours. So, form two Routing Groups. For example, Routing Group 10 for working hours and Routing Group 11 for break hours and non-working hours.

In Routing Group 10, select the member extensions in this sequence: 2001, 2003 and 2005. Set the Ring Timer for each member extension to 10 seconds. Disable Continuous Ring, as each extension in the group must ring for the set time.

In Routing Group 11, select the member extension in this sequence: 2002 and 2004, and set the Ring Timer for each extension to 10 seconds. Disable Continuous Ring, as each extension in the group is required to ring for the set time only.

Enable Rotation for Routing Group 10 and Routing Group 11.

For SIP 2, you would need to form a common TLG for working hours, break hours and non-working hours. So, configure a single routing group, for example, Routing Group 13 for SIP 2. In Routing Group 13, select the member extensions in this sequence: 3001, 3002, 3003, 3009, 3010. Set the Ring Timer for each member extension to 20 seconds. Disable Continuous ring for each extension in the group. Enable Rotation for Routing Group 13.

Select a Trunk Feature Template number for SIP 1 and SIP 2. For example, Trunk Feature Template 04 for SIP 1 and template 05 for SIP 2.

In the Trunk Feature Template of SIP 1, assign TLG. For working hours, assign Routing Group 10, for break and non-working hours, assign Routing Group 11.

In the Trunk feature Template of SIP 2, assign Routing group 13 as TLG for all three time zones.

User Absent/Present

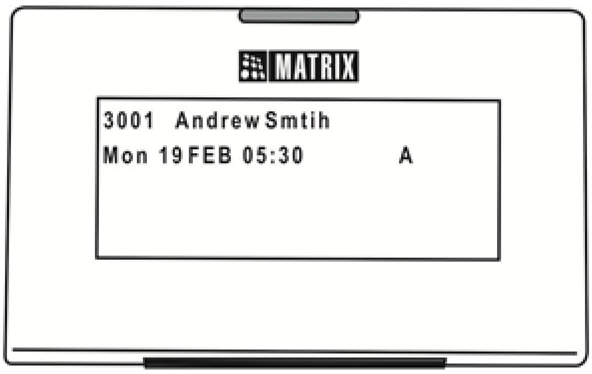
Extension users may sometimes want to leave their desks, and expecting to return soon, they may not have forwarded their calls or set Do Not Disturb on their extensions. In such cases, incoming calls will continue to land on the extension and go unanswered. The callers have no way of knowing that the extension user is not present at the extension and may try the extension number repeatedly.

With the User Absent/Present feature of ANANT UCS, extension users, including the Operator, can set 'User Absent' when they leave their desks. By doing so, they can block all incoming external as well as internal calls from landing on their extension. When they return to their desks, they can set 'User Present' and receive incoming calls again.

There are more options for indicating availability to other extensions. Refer the topic “Presence” to know more.

How it works

When an Extended IP Phone user sets 'User Absent', the letter 'A' appears on the phone's display:



The letter 'A' disappears when the extension user sets 'User Present'.

When an Extended IP Phone user calls the extension which has set 'User Absent', the text message 'User Absent' will appear on the caller's phone display.

External callers who call the extension, on which 'User Absent' is set, will get an error tone only.

- *Outgoing calls can be made from the extension which has set 'User Absent'. Only incoming calls are restricted.*
- *User Absent/Present can be set on an extension from the SA mode.*
- *If more than one extension is configured as “Operator' (routing group), incoming calls will be blocked only on the Operator extension which has set User Absent.*
- *User Password is required for this feature. The default User Password, 1111, will not work. Change the User Password first.*

How to use

For Extension Users

To set User Absent on your extension:

- Dial ***104-User Password-0***

To set User Present on your extension:

- Dial ***104-User Password-1***

The System Administrator can also set an extension as Absent/Present using Jeeves. For instructions, read "Additional Security to Extension Users" under the topic *System Security.*

User Password

The User Password is a 4-digit code for extension users to protect their extension phones from unauthorized use. The default User Password is ***1111***. It must be changed by the extension users from their phones to any desired value, not exceeding 4 digits.

To avoid unauthorized access,

- the extension users must change the default password.
- make sure the new password is strong and is kept confidential.
- change the password regularly.

In case, the extension user forgets the password, it can be cleared and restored to the default value *1111* by the System Engineer (SE) or the System Administrator (SA). Refer the topic “System Security” for instructions.

The User Password is also required to access and use certain features of ANANT UCS, which are listed below.

- Call Follow Me
- Dynamic Lock
- Direct Inward System Access (DISA)
- Walk-In Class of Service
- Presence
- Phone Settings of the Phone
- Mailbox of Voice Mail

The extension user must change the default password for all the above listed features except: Phone Settings, Mailbox of Voice Mail. Both these features allow the extension user to use the default User Password, whereas in the case of others, the system will not allow feature access without changing the User Password.

- *The User Password for an extension can be changed only from that extension phone only.*
- *Since the Mailbox can be accessed using the default User Password, extension users who are assigned a mailbox are recommended to change their User Password to prevent unauthorized access to their mailbox.*

How to use

For Extended IP Phone Users

- Press ‘Enter’ key to enter the menu.
- Scroll to Change User Password.
- Press ‘Enter’ key.
- You get the prompt: Enter Old User Password.
- Dial you current user password.
  If the default password *1111* has not be changed, enter default password.
- You get the prompt: Enter New User Password.
- Dial your new user password, not exceeding 4-digits.

You get confirmation tone and the confirmatory message ‘User Password Changed’ on your phone display.

Video Call

Video calling has become an increasingly important tool in today's business world. It offers people the power of face-to-face communication, at reasonable cost without incurring the expense of traveling. ANANT UCS allows you to make and receive video calls by connecting video capable user terminals.

Among the various UC features offered by ANANT UCS, Video calling is the most important.

Video Calling over SIP Trunks and SIP Extensions

ANANT UCS supports video calling for SIP to SIP calls. ANANT UCS only acts as a relay agent to place video calls between two video capable user terminals with no additional supplementary features.

Video capable user terminals which are communicating through ANANT UCS must confirm to the same type of video codecs[122] as per industry standards. If video codec negotiation is not successful between the end-to-end video terminals, either the call will be converted into an audio call or the call will be dropped.

For enhanced video quality during the call, some of the advanced video attributes including profile/level (H.263+, MPEG4, H.264), bandwidth, standard annexes, framerate, image size must be confirmed between the communicating video terminals. ANANT UCS will not modify or alter any of these attributes which can effect the characteristics of the video stream during communication.

A maximum of 102 simultaneous video calls are supported.

You can convert a normal SIP to SIP audio call into a video call only if both the SIP extensions support video calling.

- *Video calling is supported in VARTA UC Clients.*
- *None of the proprietary Extended SIP Phones of Matrix support video calling.*
- *SRTP is not supported for video calling.*

Feature Interactions

When the below mentioned features are accessed during an ongoing video call, the call will be established into an Audio call:

- Features where-in the system establishes a conference such as 3-Party Conference, Multiparty Conference, Conversation Recording, Call Taping, Raid etc.
- When users access Barge-In or Forced Answer.
- When users unhold, unpark or Blind Transfer the call.
- When system answers the call on a trunk, such as DID, DISA, Callback on Trunk etc.

When the below mentioned features are accessed the call will always be an Audio call:

- When a user access Voicemail, initiate an Emergency Conference or make an Intercom request.
- Features where-in the system initiates the call such as ACB Return call, Auto Redial Return call, Alarm call, etc.

122. *Current industry standard video codecs include H.261, H.263, H.263p, H.264, MPEG4 etc. For more details, refer the documentation of the corresponding video terminals you are using for video calling.*

- *To dial Flash, make sure your IP Phone supports Flash dialing.*
- *You can use #2, if your IP Phone does not support Flash dialing.*

Virtual Extension

The Virtual Extension feature of ANANT UCS enables multiple users to share one telephone instrument as their extension, yet be considered as individual extensions by the system, with distinct extension properties and class of service.

Such shared extensions are called Virtual Extensions, as their users do not have individual phones for their use.

Virtual Extensions are useful in laboratories, common rooms, dormitories, shop floors, and wherever it is not feasible to provide dedicated telephone instruments to individual extension users. Virtual extensions allow you make optimum use of the existing phones without investing in new ones.

How it works

The shared telephone instrument is called the Master Extension.

- Virtual Extensions are assigned to the Master Extension. A Master Extension can have multiple Virtual Extensions, but a Virtual Extension can have only one Master Extension.
- The Virtual Extension functions as any other extension of ANANT UCS. It can be assigned all features and facilities, like Class of Service, Toll Control, Call Forward, just like any other physical extension of ANANT UCS. You can assign Station Basic Feature Template and Station Advanced Feature Template to the Virtual Extension.
- Incoming calls to a Virtual Extension will ring on the Master Extension.
- All incoming, outgoing, internal and external calls of the Virtual Extensions are recorded in the Station Message Detail Records.
- To make outgoing calls, the Virtual Extension user must use the feature "Walk-In Class of Service".
- The Virtual Extension user is logged out of the Master Extension according to the Walk Out mode assigned to it: *Walk out single call* or *Walk out multiple calls.*

How to configure

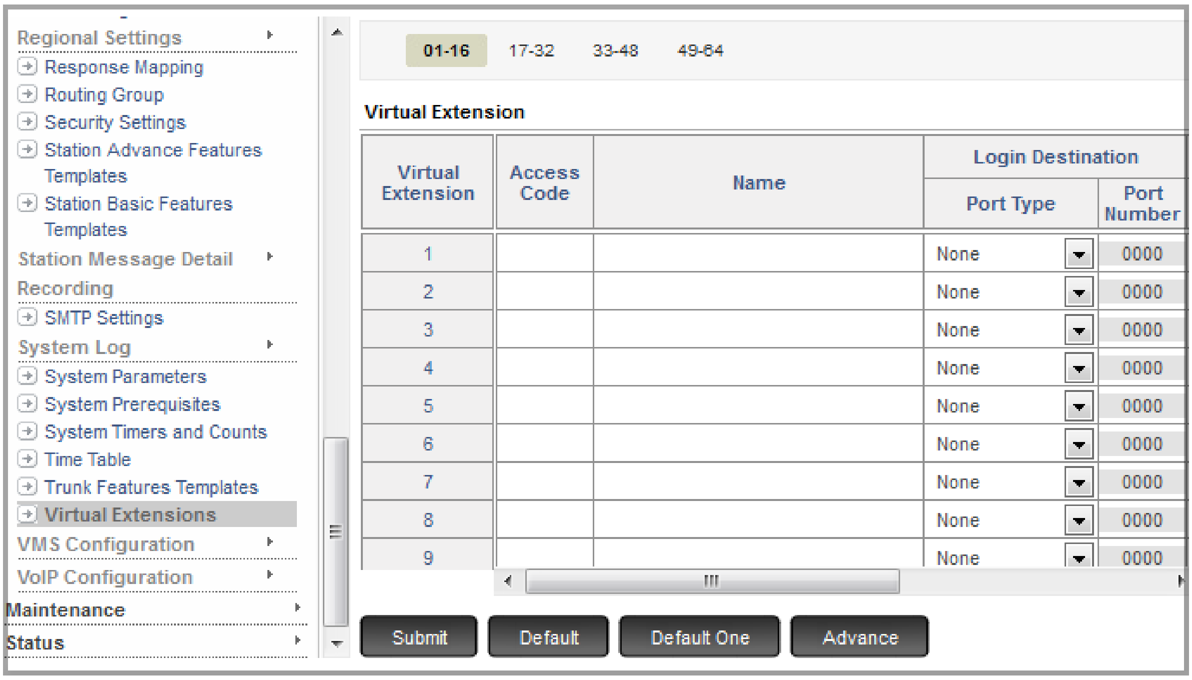
For this feature to work, you must do the following:

- Make a list of the number of Virtual Extensions required by you along with their names and numbers.
- Decide the Master Extension (landing destination) that is, the SIP Extension Number.
- If required, you can customize the Station Basic Feature Template and Station Advanced Feature Template you want to assign to the Virtual Extensions. For making outgoing calls users of Virtual Extensions must have "Walk-In Class of Service" enabled in their CoS.
- Decide the Priority you want to assign to the Virtual Extensions. When an outgoing call is made from any Virtual Extension, the ring type played to the called party will be as per the set priority.

Configuring Virtual Extensions

- Login as System Engineer.

- Under **Configuration**, click **Virtual Extensions**.



Configure the following parameters for each Virtual Extension:

- **Access Code:** Assign Station Access Codes to the Virtual Extensions. Station Access Codes are commonly referred to as Extension Numbers. These may be a combination of 1, 2, 3, 4, 5 and 6 digits, which are dialed to call the Virtual Extension to which they are assigned.

  To assign Station Access Codes according to your preference and requirement to a range of Virtual Extensions, see “Assigning Access Codes to a Range of Extensions”.

If you decide to customize the Station Access Codes, make sure that the numbers do not clash with any other Access Code in the 'Dial' phase. Refer the topics “Access Codes” and “Conflict Dialing” to know more.

- **Name:** Assign a 'Name' to the Virtual Extension. The name may be of the person who will use the extension. This name will be displayed on the LCD of the remote user's phone, if it is equipped with Caller ID.

  You can program a name of a maximum of 18 alphanumeric characters. Valid input characters are All ASCII characters except <, >, :, ", /, \, |, ?, *

- **Login Destination:** Configure the **Port Type** and **Port Number** you want the Virtual Extension user to log into to make outgoing calls.

- **Landing Destination:** Configure the **Port Type** and **Port Number** on which you want the Virtual Extension user to receive incoming calls.

- **Priority:** Select a Priority Level for the Virtual Extension.

Each extension of the ANANT UCS is assigned a Priority Level starting from 1, 2, 3... to 9, with '1' being lowest Priority and '9' being highest Priority. Whenever an extension phone with higher priority calls an extension with lower priority, a triple ring is placed on the called station. To know more, read the feature description “Priority”.

By default, the Priority of all Virtual Extensions is set to '1-None'. So, decide what Priority Level you will assign to each of the Virtual Extensions and set the desired level for each extension.

Advanced Configuration Parameters

- Click the **Advanced** button at the bottom of the page and configure the following parameters:
  - **Mobile Number:** Enter the Mobile Number of the extension user you wish to store. The Number can be a maximum of 16 digits.
  - **Email Id:** Enter the Email Id of the extension user you wish to store. The Email Id can be a maximum of 64 characters.
- If you have completed configuring the parameters, click **Submit** at the bottom of the page to save your settings.

  It is possible to default all the parameters by clicking the **Default** button. You can also restore default values of the parameters of a single Virtual Extension by clicking the **Default One** button and specifying the Virtual Extension Number you want to set to default.
- Click **Submit**.

Walk-In Class of Service

Every extension of ANANT UCS is assigned a Class of Service and Toll Control defining its access to features and its calling permission.

With Walk-In Class of Service, extension users of ANANT UCS can make calls or access features from any other extension of the system as per the Class of Service, Toll Control and other features / facilities assigned to their own extension.

This feature is useful to extension users who frequently move away from their desk, as it allows them the same level of feature access and calling permission as their own extension, from another extension.

Extension users can 'Walk-In' from any SIP extension.

ANANT UCS offers two types of Walk-In:

- **One call:** The extension user is automatically logged out from the extension into which the user has walked-in, after one call.
- **Multiple Calls:** The extension user can make as many calls as desired, and remains 'walked-in' until the user dials the feature code to 'Walk-Out', or until another extension user walks into the same extension.

To allow ‘One call’ or ‘Multiple Calls’ to an extension, you need to set the 'Walk-Out Mode’ in the “Station Advanced Feature Template” of the extensions.

Walk-In Class of Service is a password-protected facility and the default User Password ***1111*** will not be accepted. To be able to walk into another extension, extension users must first change their User Password to another value.

How it works

With the help of this illustration, let us understand how Walk-in Class of Service works.

In this illustration, Extension user A with the number 3001, with long distance calling facility (toll control: All calls). Extension user B with the number 2001, without long distance calling (toll control: local calls).

Here,

- 3001 is the **Source Extension**, whose CoS, Toll Control and other features / facilities (assigned in the template) are used from another extension (2001) by performing Walk-In.
- 2001 is the **Destination Extension** on which Walk-In is performed.

Walking into another extension to make calls

- Now, extension user A is at B's desk and needs to make a long distance call. B’s extension does not have long distance calling.
- Extension user A can 'Walk-In' into B's extension (2001) by dialing
  - the feature code for 'Walk-In Class of Service'.
  - A’s extension number, 3001.
  - A’s User Password (the default password 1111 will not be accepted, it must be changed first).

- On successful Walk-In, ANANT UCS applies the Class of Service, Toll Control and other features / facilities as per the Template of the **Source Extension** 3001 on the **Destination Extension** 2001.

- Extension user A can make external and internal call from extension B.

- If Extension 3001 has 'One Call' selected as the 'Walk-Out Mode' for the extension, A will be 'Walked-Out' when the current call ends or if A goes ON-Hook at any time after walking into extension 2001.

- If Extension 3001 has 'Multiple Calls' selected as the 'Walk-Out Mode' for the extension, extension user A must manually walk out by dialing the feature code for 'Walk-Out'.

- If Extension user A does not 'Walk-Out', the system will perform a walk out for A, if:

  - another extension user walks into extension 2001
  - there is an incoming call on 2001.

- *At a time, only one extension user can walk in to another extension.*

- *Calls made after walk-in will be charged to the Source Extension. Here, calls made by Extension user A from extension 2001 using Walk-In will be calculated and charged to extension 3001 only.*

- *Call record details of calls made after walk-in will be recorded for the Source Extension. Here, call record details of calls made by Extension user A from extension 2001, using Walk-In, will be recorded in the Station Message Detail Record of extension 3001 only.*

Walking into another extension to access a feature

- Extension user A with number 3001 has Call Forward in the Class of Service.

- Extension user B with number 2001 does not have Call Forward in the Class of Service.

- Extension user A is currently at B's desk. A needs to forward calls of own extension to an external number. To do this,
  - A walks into B's extension by dialing
    - the feature code for 'Walk-In Class of Service'.
    - A's extension number (3001)
    - A's User Password[123]

- On successful Walk-In, the system applies the Class of Service, Toll Control and other features / facilities as per the Template of the Source Extension 3001 on the Destination Extension 2001.

- From extension 2001, extension user A sets Call Forward for 3001, to an external number.

- Call Forward can be canceled from the Source Extension or from the Destination Extension. To cancel Call Forward, extension user A can go back to 3001 and cancel Call Forward from 3001, or can cancel Call Forward from 2001, if user A is still walked-in on 2001.

CAUTION! *The Destination Extension user can access their Class of Service or Toll Control only after the Source Extension user has walked out from their extension. For example, user B cannot set or cancel Call forward on extension 2001, until user A has walked out from 2001.*

---

123. The default password 1111 will not be accepted, it must be changed first.

Incoming calls on the Destination Extension (2001) will work according to the setting of the Destination Extension only, whereas outgoing calls on the Destination Extension will work according the settings of the Source Extension (3001).

The following set of features of the Destination Extension will remain unaffected:
- *Key map*
- *Language*
- *Priority*
- *Call Pick-Up group*
- *Personal Directory*

CAUTION! *There is a risk of fraudulent calls being made from your extension, if a third party comes to know the User Password of your extension. The cost of such fraudulent calls will have to be borne by the owner of ANANT UCS.*

So, protect your system from unauthorized access and misuse by putting strong authentication mechanisms in place.
- *Keep Passwords strictly confidential.*
- *Change Passwords regularly.*
- *Choose Passwords that are complex and difficult to guess.*

How to configure

This feature is available to all extensions of ANANT UCS. All you need to do is, select the 'Walk-Out Mode' for the extensions in their "Station Advanced Feature Template".

By default, 'One call' is selected as the 'Walk -Out' mode in the default Station Advanced Feature Template 01 assigned to all extensions.

If you want to allow different walk-out modes to different extensions, use different Templates, but make sure other features on these templates are also configured according to requirement.

For detailed instructions refer the topics "Customizing Station Advanced Feature Template using Jeeves".

How to use

For Extended IP Phone Users

To perform a Walk-In, on the Destination Extension,

- Press DSS Key assigned to Walk-In Class of Service (if programmed).
  OR
- Dial 111
- Select 'Walk-in' and press Enter key.
- You get the prompt 'Walk in from which Station?'
- Dial your extension number.
- You get the prompt 'Enter your User Password'.
- Dial your User Password (default password will not be accepted).
- You get the confirmation message 'Walked In Successfully' on the phone's display and a confirmation tone.
- You can now make your call(s) or access a feature.

To perform a Walk-Out, on the Destination Extension or on the Source Extension,

- Press DSS Key assigned to Walk-In Class of Service.
  OR
- Dial 111.
- Select 'Walk-Out' on your phone's display.
- You get the confirmation message 'Walked Out' on your phone's display and confirmation tone.

You need to perform a Walk-Out only if the Source Extension as Multiple Calls set as Walk-Out mode.
If the Source Extension has 'One Call' set as the Walk-Out mode, you will be walked out of the Destination Extension when you go ON-Hook after making a call or accessing a feature.

If the extension you are walking in has 'One Call' as the Walk-Out mode, and you go ON-Hook before you make the call, you will be 'Walked Out'. You must Walk-In again.

CHAPTER 9

Voice Mail Features

Accessing your Mailbox

You will be able to access your mailbox only if there is a free VMS channel. For more details, see "Configuring VMS General Parameters".

To access your Mailbox from your own extension:

- Dial 3931.
- System prompts you with:

  "You have no new messages", if there are no new messages in your mailbox.

  "You have n new messages", if there are new messages in your mailbox (n = No. of messages).
- Enter your Mailbox password.
- You will enter the "Mailbox Access" menu.

To access your Mailbox from another extension:

- Dial 3932/ 3933/ 3934.
- System prompts you with, "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.
- Dial 8. System prompts you to dial extension number.
- Dial your extension number to access your mailbox.
- System prompts you with:

  "You have no new messages", if there are no new messages in your mailbox.

  "You have n new messages", if there are new messages in your mailbox (n = No. of messages).
- Enter your Mailbox password.
- You will enter the "Mailbox Access" menu.

To access your Mailbox through an external number using DISA Login:

- Log into ANANT UCS using DISA.
- If CLI Based Authentication is enabled, the system prompts you to dial desired extension number.
- If PIN Authentication is enabled, the system prompts you to dial your Extension Number and Password.
- Dial 3931 to access your Mailbox.
- System prompts you with:

  "You have no new messages", if there are no new messages in your mailbox.

  "You have n new messages", if there are new messages in your mailbox (n = No. of messages).
- Enter your Mailbox password.
- You will enter the "Mailbox Access" menu.

To access your Mailbox using VMS Auto Attendant:

- Call the trunk on which Voice Mail Auto Attendant is enabled.
- The incoming call on the trunk is answered by the VMS Auto Attendant.
- The VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.
- Dial 8. System prompts you to dial extension number.
- Dial your extension number to access your mailbox.
- System prompts you with:

  "You have no new messages", if there are no new messages in your mailbox.

  "You have n new messages", if there are new messages in your mailbox (n = No. of messages).
- Enter your Mailbox password.
- You will enter the "Mailbox Access" menu.

Alarms and Reminders

The Voice Mail System of ANANT UCS offers voice-guided Alarms and Reminders, which can be set by the Operator as well as extension users.

How it works

Voice-guided Alarm and Reminder requests are served as per the date and time set by extension users. The different ways in which Alarm or Reminder requests will be served are described in the following:

- **Alarm with Snooze Off (Once Only)**
    - The VMS plays system greeting for the current time zone and the Extension Name followed by the message "This is your Wake up Call. Music of 5 seconds."
- **Alarm with Snooze On (Once Only)**
    - The VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Wake up Call. For Acknowledge, Please Press 0. Music of 5 seconds."
    - When user press '0', VMS prompts: "Your Alarm is Acknowledged."
- **Daily Alarm with Snooze Off**
    - The VMS plays system greeting as per the time zone and the Extension Name followed by the message "This is your Daily Wake up Call. Music of 5 seconds."
- **Daily Alarm with Snooze On**
    - VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Daily Wake up Call. For Acknowledge, Please Press 0. Music of 5 seconds."
    - When user press '0', VMS prompts: "Your Alarm is Acknowledged."
- **Reminder with Snooze Off**
    - VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Reminder call. Music of 5 seconds."
- **Reminder with Snooze On**
    - VMS plays system greeting as per time zone and the Extension Name followed by the message "This is your Reminder Call. For Acknowledge, Please Press 0. Music of 5 seconds."
    - When user press '0', VMS prompts: "Your Reminder is Acknowledged."

How to configure

The VMS allows you to enable/disable the **Alarm Verification** for alarms and reminders, allowing extension users who want to use alarms and reminders to confirm.

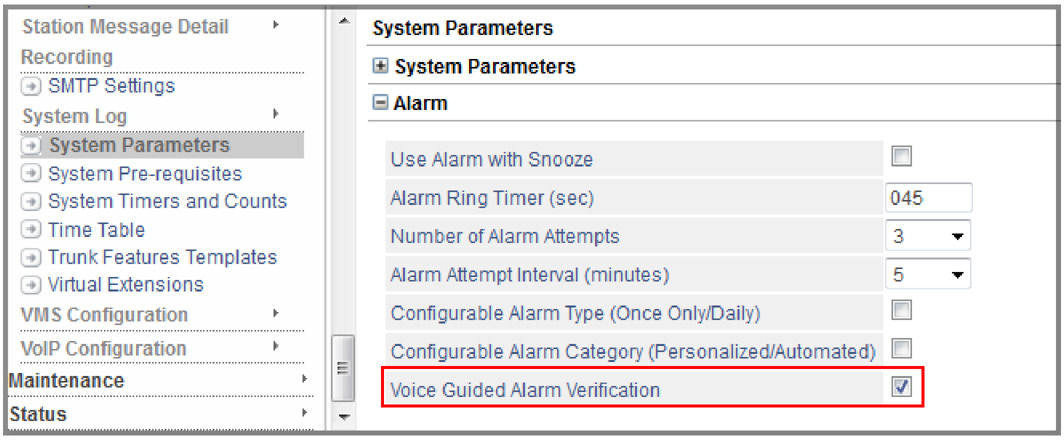
- Time set for an alarm
- Date and time set as a reminder.

When Alarm Verification is disabled, the VMS will not confirm the alarm and reminder set by the extension user.

If you want the VMS prompts to be played when the extension users answer an Alarm Call/Reminder Call and acknowledge it, make sure you have selected *Voice Mail* as *Alarm Notification Type*.

To configure Alarm Verification,

- Login as System Engineer.
- Under **Configuration**, click **System Parameters**.
- On the System Parameters page, click **Alarm** to expand.



- Select the **Voice Guided Alarm Verification** check box to enable extension users to confirm the Time they have set for an alarm or the Date and Time they have set for a reminder. Default: Enabled.
- Click **Submit.**

To configure Alarm Notification Type,

- Under **Configuration**, click **Station Advanced Feature Template.**
- Click Alarm Notification to expand.
- Select the **Alarm Notiifcation Type** as **Voice Mail**.
- Click **Submit**.
- Assign this Station Advanced Feature Template to the desired extensions.

How to use

Alarm set by Extension Users

- Pick up the handset of your telephone and dial **163** → VMS prompts: "Enter the time, HH MM in twenty four hour format[124]. To cancel all alarms, press # (pound/hash)."
    - If no time is entered, VMS prompts: "You have not entered any input"
    - If invalid time is entered, VMS prompts: "You have entered invalid input."
- To set alarm, dial valid time → VMS prompts: "To set once, Press 1, To set Daily Press 2."

Once Only

- Dial 1 → VMS responds with: "You have set Wake up Alarm at …." (WakeupVeri.wav) followed by the prompt: To Confirm[125], Press 1, To Re-enter, Press 2."
- Dial 1 to confirm the time set for alarm → VMS responds with: "Your Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If alarm is not set, the VMS responds with: "Sorry! Your Wake Up Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

Daily Alarm

- Dial 2 → VMS responds with: "You have set Daily Wake up Alarm at …." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
- Dial 1 to confirm the time set for alarm → VMS responds with: "Your Daily Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If no alarm is set, the VMS responds with: "Sorry! Your Wake Up Alarm cannot be set. Please call Operator for further assistance." The VMS further responds with: "Thanks for using this Service."
- Dial # (pound/hash) to cancel all alarms → VMS responds with: "Your all Wake up Alarm are canceled." followed by the prompt: "Thanks for using this Service."
    - If no alarms are set, the VMS responds with: "Sorry! There is no Alarm to cancel." followed by the prompt: "Thanks for using this Service."

Alarm set by Operator

- Enter System Administrator Mode, dialing 1072.
- Dial 034 → the VMS prompts: "Enter the Extension number for which you have to set or cancel Wake Up Alarm."
- Dial 1 to select the extension user for which the Alarm is to be set. VMS responds with: "Enter the time, HH MM in twenty-four hour format. To cancel all alarms, press '#' (pound/hash)'."

---

124. The Date and time format depends on the Region/Country selected for the system.
125. This option will not be played if Alarm Verification is disabled in the System Parameters.

- If no time is entered, the VMS prompts: "You have not entered any input"
- If invalid time is entered, the VMS prompts: "You have entered invalid input."

- To set alarm, dial valid time → VMS prompts: "To set once, Press '1', To set Daily Press '2'."

Once Only

- Dial 1 → the VMS responds with: "To set it as Personal, Press 1. To set it as Automated, Press 2."
  - Dial 1 → VMS responds with: "You have set Personal Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
  - Dial 1 → VMS responds with: "Your Personal Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
  - If alarm is not set, the VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

  OR

  - Dial 2 → the VMS responds with: "You have set Automated Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
  - Dial 1 → the VMS responds with: "Your Automated Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If alarm is not set, VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

Daily Alarm

- Dial 2 → VMS responds with: "To set it as Personal, Press 1. To set it as Automated, Press 2."
  - Dial 1 → VMS responds with: "You have set Daily Personal Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
  - Dial 1 → VMS responds with: "Your Daily Personal Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
    - If the alarm is not set, the VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

  OR

  - Dial 2 → the VMS responds with: "You have set Daily Automated Wake up alarm at…." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
  - Dial 1 → VMS responds with: "Your Daily Automated Wake up Alarm is set." followed by the prompt: "Thanks for using this Service."
  - If alarm is not set, VMS responds with: "Sorry! Your Wakeup Alarm cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."

Reminders set by Extension Users

- Pick up handset of your telephone and dial **164** → VMS prompts: "Enter the Date in DD MM YYYY format[126]. To Cancel all Reminders, Press '#' (pound/hash)'. For example, To enter Date 17th March 2008, Dial One Seven Zero Three Two Zero Zero Eight."
    - If no date is entered then VMS prompts: "You have not entered any input"
    - If invalid date is entered then VMS prompts: "You have entered invalid input."
- Dial valid Date → the VMS prompts: "Enter the time, HH MM in twenty four hour format."
    - If no time is entered, the VMS prompts: "You have not entered any input"
    - If invalid time is entered, the VMS prompts: "You have entered invalid input."
- Dial valid time → VMS prompts: "You have set Reminder for..." followed by the prompt: "To Confirm, Press 1, To Re-enter, Press 2."
- Dial 1 to confirm the date and time set for Reminder → the VMS responds with: "Your Reminder is set." followed by the prompt: "Thanks for using this Service."
    - If Reminder is not set, the VMS responds with: "Sorry! Your Reminder cannot be set. Please call Operator for further assistance." VMS further responds with: "Thanks for using this Service."
- Dial # (pound/hash) to cancel Reminder → the VMS responds with: "Your Reminder is canceled." followed by the prompt: "Thanks for using this Service."
    - If no reminder is set, the VMS responds with: "Sorry! There is no Reminder to cancel." followed by the prompt: "Thanks for using this Service."

126. The date format in the prompt will be MM DD YYYY, if you selected USA as the Region/Country for your system.

Join Conference Dial-In using VMS

This feature allows users to Join a Dial-in Conference using the trunks on which the Voice Mail Auto Attendant is enabled. After Joining the Conference users can also Temporary Leave, Rejoin or Permanently Leave the Conference.

How it works

For this feature to work,

- you must have a free VMS channel available. Refer *“Licenses Supported in ANANT UCS”*.
- you must select **Voice Mail Auto Attendant** option (in the Trunk Feature Template) on the desired SIP trunk.
- you must inform the caller the Dial-In Conference Number and Password along with the time of Conference.

This is how Join Conference Dial-In using VMS works:

- A call lands on a VMS enabled Trunk.
- The incoming call on the trunk is answered by the VMS Auto Attendant. By default, the VMS greets the caller with the Greeting message followed by the Welcome Message: “Welcome! Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.
- While this prompt is being played, the caller must dial the Dial-In Conference Code, *19. The VMS prompts the caller: “Please enter your number and password. Caller must first dial 2 (the Join Dial-In Conference code) and then dial the Dial-In Conference Number and Password.
- System checks if the Conference Number and password is valid or not.
- The Number and Password is valid the caller will be able to join the conference.
- The Caller can temporarily leave from the Dial-In conference, to do so, while in speech, dial Flash-191.
- The Caller can rejoin the Dial-In conference, to do so, go Off-hook (by dialing the Off-hook code #1) and then dial 191.
- The Caller can permanently leave the Dial-In conference, to do so, go Off-hook (by dialing the Off-hook code #1).
-

How to configure

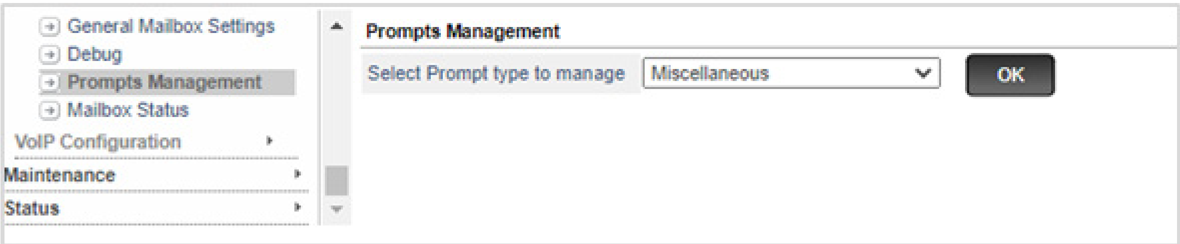
- For instructions to enable Voice Mail Auto Attendant on the desired trunks, see the topic “Trunk Feature Template” in *Configuring Trunks*.

- To customize the DISA prompts as per your requirement, see “Recording Voice Messages”.
- To upload the customized Voice Prompts, see “Prompts Management”.If you are upgrading the firmware with Firmware later that V2.2, you need to manually upload the prompt. Refer to “Dial-In Prompt Upload”.

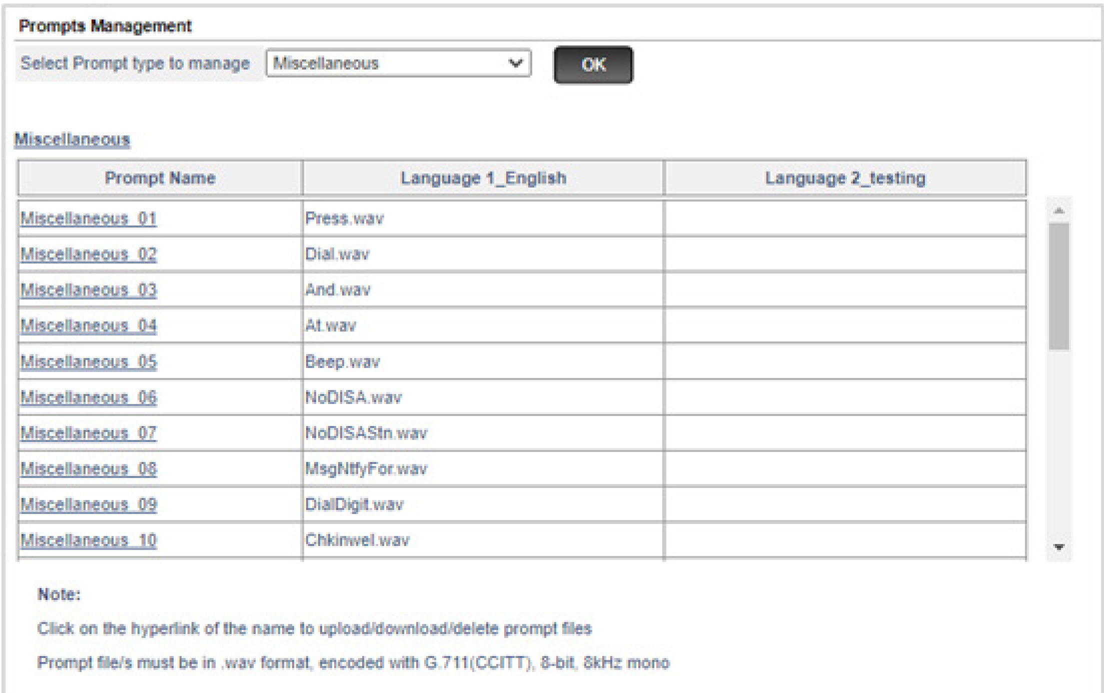
Dial-In Prompt Upload

After upgrading the system, to upload the VMS Dial-In Prompt, follow the steps mentioned below:

- Under **Configuration**, click **VMS Configuration**.
- Click **Prompts Management.**

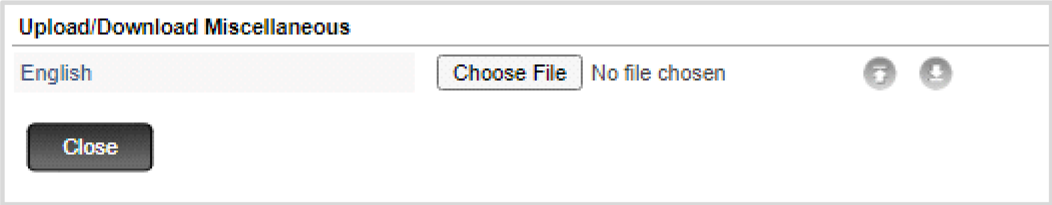


- In **Select Prompt type to manage** select **Miscellaneous**.

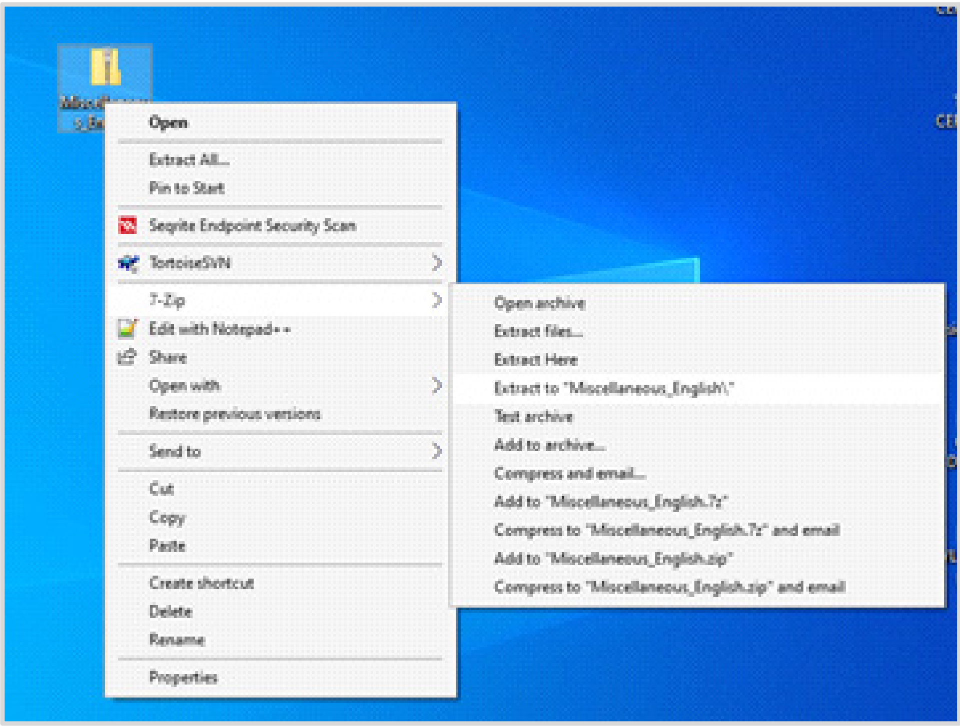


- Click **OK**.
- Click on **Miscellaneous**

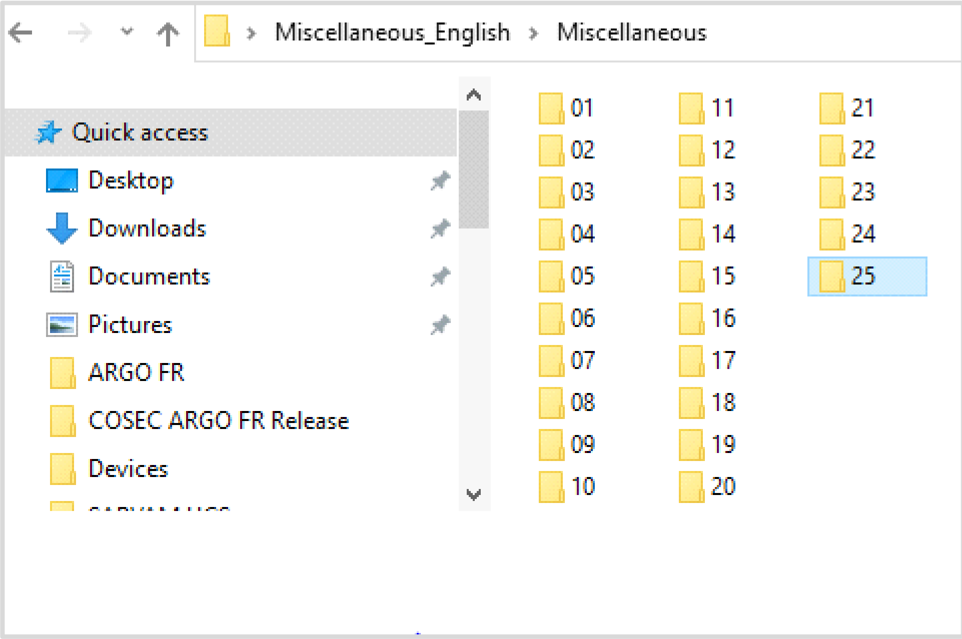
- To download all the prompts of, click .



- The Miscellaneous_English zip folder is downloaded. Extract the folder files.



- Now click the extracted folder and create a new folder 25 in it.



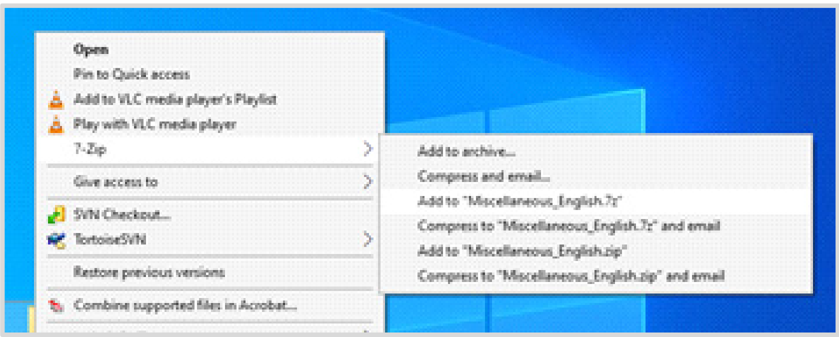
- Click to open Folder number 01, it has two files. Make sure you delete one file which you do not require. This folder must not have two files.

- Copy the new prompt in folder 25 from the USB package (path:Voicemail\database\prompts\language_1\Miscellaneous\25)

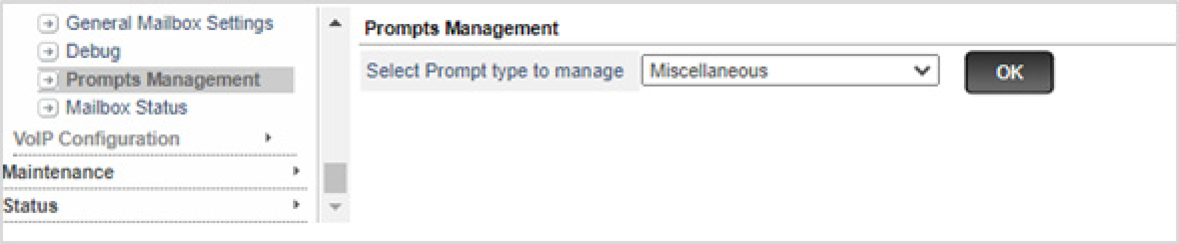
Contact the Dealer/Distributor or Technical Support for the USB package.

- Go back to the extracted Miscellaneous folder and open the folder 25. Paste the prompt here.

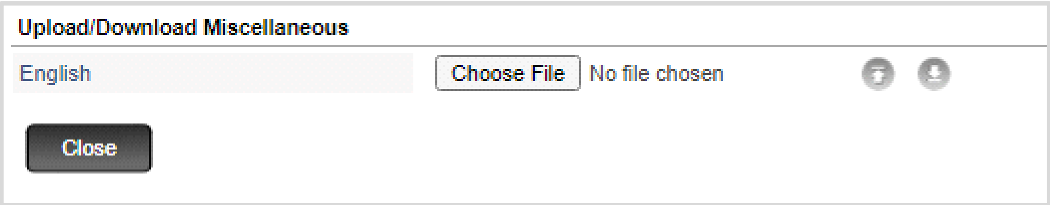
- Zip the Miscellaneous folder.



- Now, click **Prompts Management** again.



- In **Select Prompt type to manage** select **Miscellaneous**.
- Click **OK**.
- Click on **Miscellaneous.**



- Click the **Choose File** button to reach the location on the local disk where the Miscellaneous zip folder is stored in your PC.
- Click to upload.

VMS DISA Login

This feature allows the remote users to log into the DISA mode using the trunks on which the Voice Mail Auto Attendant is enabled. The remote user can access and use the system's features and facilities using the DISA enabled trunks.

The VMS supports DISA-PIN Authentication-Multiple Calls only. For detailed information on the types of DISA Variants, see “Direct Inward System Access (DISA)”.

Using VMS DISA login, remote users can:

- call any extension.
- make external calls.
- use features and facilities of the system.
- configure features and facilities of the system and administer the system.

All these can be done as if being done from a local extension of ANANT UCS.

How it works

For this feature to work,

- you must have a free VMS channel available. Refer *“Licenses Supported in ANANT UCS”*.
- you must enable **DISA-PIN Authentication-Multiple Calls** (in the Trunk Feature Template) on the desired SIP trunk.
- you must select **Voice Mail Auto Attendant** option (in the Trunk Feature Template) on the desired SIP trunk.
- you must enable DISA in the “Class of Service (CoS)” of the extension the caller is allowed to log into (using PIN Authentication).
- you must change the default **User Password** (1111) of the extension the caller is allowed to log into.

This is how VMS DISA Login works:

- A call lands on a DISA enabled Trunk.
- The incoming call on the trunk is answered by the VMS Auto Attendant. By default, the VMS greets the caller with the Greeting message followed by the Welcome Message: “Welcome! Please dial the extension number or To dial by name press ‘6’, To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.
- The caller must dial the DISA Login Code, 1079. The VMS checks if DISA-PIN Authentication-Multiple Calls is enabled on the trunk for the current time zone, that is, working hours, break-hours and non-working hours.
- The VMS finds DISA is enabled on the trunk and prompts the caller: “Please enter the Extension Number.” and starts the First Digit Wait Timer (programmable; default: 25 seconds).
- The caller must dial the DISA Extension Number before the expiry of this timer.

- The VMS checks the CoS of the dialed Extension Number.
- DISA is enabled in the CoS of the dialed Extension Number, the VMS prompts: “Please enter your password.” and waits to receive digits till the expiry of the First Digit Wait Timer.
- The Password must be dialed by the caller before the expiry of this timer.
- The VMS checks if the Password is valid.
- The Password is valid and the DISA Login is successful. The caller is logged into the dialed Extension Number.
- The VMS hands over the DISA call to the ANANT UCS.
- When the caller goes Off-hook by dialing the Off-hook code #1, the system plays the internal dial tone and waits for the caller to dial digits.
- The system collects the digits dialed by the caller and then routes the call.
- The caller can make as many trunk calls and internal calls as the caller wants.
- The caller can terminate the DISA login session either by disconnecting from the remote end or by dialing the Termination Code *#9*.

The VMS plays the default Greeting Message, Welcome Greeting and DISA prompts to the callers. You can customize them as per your requirement, if required.

How to configure

- For instructions to enable DISA-PIN Authentication-Multiple Calls and Voice Mail Auto Attendant on the desired trunks, see the topic “Trunk Feature Template” in *Configuring Trunks*.
- For instructions to enable DISA in the CoS of the extensions which you want to allow callers to access using DISA, see “Class of Service (CoS)”.
- To change the default User Password (1111) of the extensions which you want to allow callers to access using DISA, see “User Password” and “System Security”.
- To set the DISA Timers as per your requirement, see “System Timers and Counts”.
- To customize the DISA prompts as per your requirement, see “Recording Voice Messages”.
- To upload the customized Voice Prompts, see “Prompts Management”.

Sending Messages

The VMS enables extension users to send messages to other extensions that have a mailbox. An extension user can send a message to as many as 10 destinations at a time. The extension user can send the message either to a specific mailbox or to a Distribution List.

VMS also gives facility to the Sender of the message to request a read receipt of the message sent. When the Recipient has read the message, the VMS generates a file containing the first 5 seconds of the message that was sent and delivers it to the Sender's mailbox in the form of a new message with the Date and Time stamp and the prompt: "This message was read by <Extension Name> <5 seconds of message sent>". If the Sender does not request ‘read receipt’, no such message is delivered to the Sender.

How to use

- Call the VMS by dialing **3931**,
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password → VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 3 → VMS prompts: "Enter the destination/s and dial hash to end."
- Dial valid extension numbers/distribution list number → VMS prompts: "Record your message after the beep and press any digit to end".
- Speak to record the message and press any digit to end.

The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see *“Message Send/Forward Settings”*

- VMS prompts "To request read receipt press '1', to ignore read receipt press '2'."
- If Message Verification check box is disabled then VMS will not playback the recorded message.
- Dial 1 to request read receipt of your message, else dial 2.
- VMS responds: "Message sent as normal”.

- *Extension users must be careful in dialing destination numbers.*
- *If an invalid destination is entered then the VMS will discard this entry and will send the message to the remaining users.*
- *If all the entries are invalid, then the VMS will clear all the entries and will ask the mailbox owner to re-enter all the destinations again.*
- *Once a valid destination number is entered and no more extensions are selected, the VMS understands it to be the end of list and sends the message.*

Redirecting Messages

The VMS offers extension users to re-direct the messages in their mailbox to another mailbox. The feature can be used by employees who are out of office or unable to access their mailbox. Using Redirect Messages, they can ensure that important messages are attended to by their colleagues in their absence.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see "Mailbox Access".

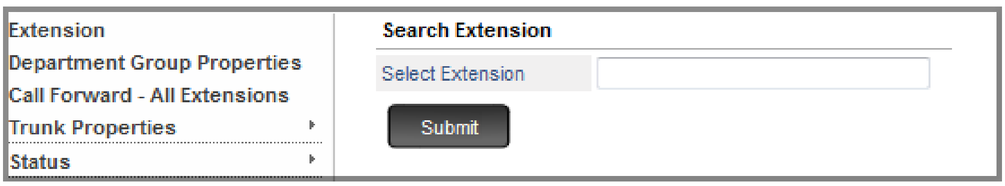
How to use

- Call the VMS by dialing **3931**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password → VMS prompts: "You have n new/no new messages"
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 → The VMS responds with: "For Mailbox Name, press '1', For message redirection, press '2', To delete all old messages of your mailbox, press '3', To delete all messages of your mailbox, press '4', For Mailbox Greetings, press '5', To go to previous menu press '#'."
- Dial 2 → VMS prompts: "To set message redirection press '1', to cancel message redirection press '2', to go to previous menu press '#."
- Dial 1 to set message redirection → VMS prompts: "Enter the destination extension."
  - Dial valid destination extension → VMS responds: "Message redirection set."
- Dial 2 to cancel message redirection → VMS responds: "Message redirection cancelled."

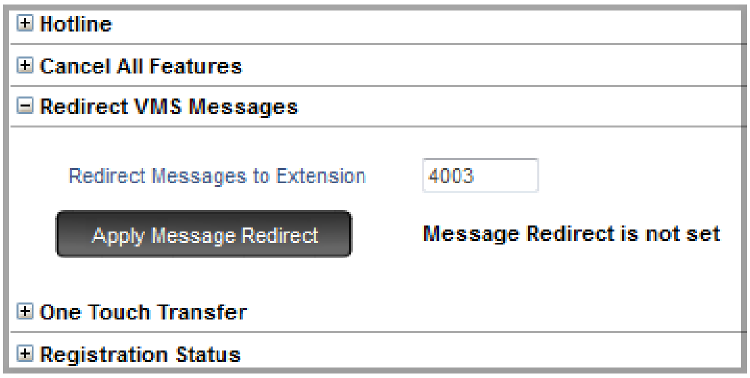
Message Redirection using SA Jeeves

For Extension Users

- Login as System Administrator.
- Click **Extension**.



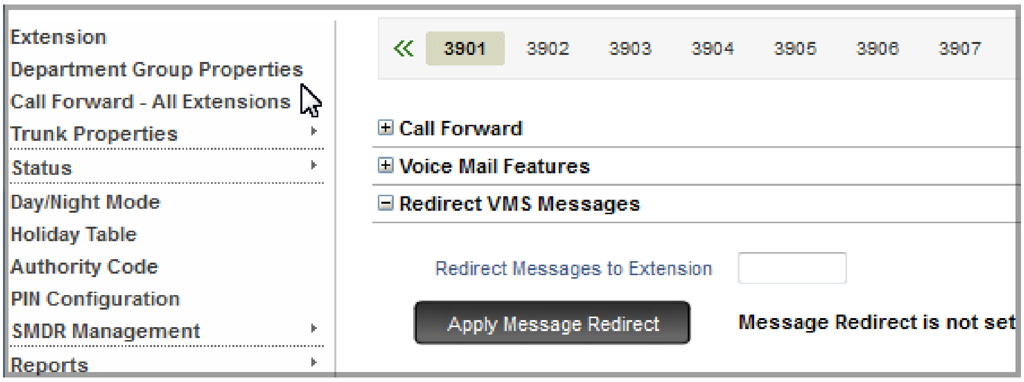
- In **Select Extension,** enter the Number or the Name of the extension on which you want to set this feature.
- Click **Submit.** The searched extension users details appear on your screen.
- Click **Redirect VMS Messages** to expand.



- In **Redirect Messages to Extension**, enter the extension number on which you want the messages to be redirected.
- Click the **Apply Message Redirect** button.

For Department Group

- Login as System Administrator.
- Click **Department Group Properties**.



- Click the desired Department Group Number tab for which you want to set message redirection.
- Click **Redirect VMS Messages** to expand.
- In **Redirect Messages to Extension**, enter the extension number on which you want the messages to be redirected.
- Click the **Apply Message Redirect** button.

Recording Personal Greetings

The VMS offers extension users the facility to customize the greeting messages for their mailbox. Extension users can record a different message for each time zone, namely working hours, break hours and non-working hours.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see "Mailbox Access".

How to use

- Call the VMS by dialing **3931**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password and the VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 → The VMS responds with: "For Mailbox Name, press '1', For message redirection, press '2', To delete all old messages of your mailbox, press '3', To delete all messages of your mailbox, press '4', For Mailbox Greetings, press '5', To go to previous menu press '#'."
- Dial 5 → The VMS prompts: For Personal Greetings, press '1', For Conditional Greetings, press '2', To go to previous menu press '#'.
- Dial 1 → The VMS prompts: "For working hours greeting, press '1', for break hours greetings, press '2', for non-working hours greeting, press '3, to go to the previous menu, press '#".
- Dial the digit of the desired time zone. The VMS prompts: "To record, press '1', to play, press '2', to erase, press '3', to go to the previous menu, press '#' ".
- To record the message, press 1. You can record a greeting message of 120 seconds. If the message duration is less than 60 seconds, press # after you have completed the message.
- The VMS prompts: "To record, press '1', to play, press '2', to erase, press '3', to go to the previous menu, press '#' ".
- To listen to the greeting message you recorded, press '2'.
- On completion of play back of the greeting message, the VMS prompts: "To record, press '1', to play, press '2', to erase, press '3', to go to the previous menu, press '#' ".

Recording Conditional Greetings

The VMS offers extension users the facility to customize the greetings messages played to callers for certain conditions—busy, no reply or unconditional/unregistered Call Forward. Extension users can record a different message for each condition.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see *“Mailbox Access”.*

How to use

- Call the VMS by dialing **3931**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password and the VMS prompts: "You have n new/no new messages."
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."
- Dial 4 → The VMS responds with: “For Mailbox Name, press '1', For message redirection, press '2', To delete all old messages of your mailbox, press '3', To delete all messages of your mailbox, press '4', For Mailbox Greetings, press ‘5’, To go to previous menu press '#'.”
- Dial 5 → The VMS prompts: For Personal Greetings, press ‘1’, For Conditional Greetings, press ‘2’, To go to previous menu press '#'.
- Dial 2 → The VMS prompts: "For Busy press '1', for No Reply press '2', for Unconditional press ‘3’, To go to previous menu press '#'.
- Dial the desired digit. The VMS prompts: "To record press ‘1’, to play press ‘2’, to erase press ’3’, to go to the previous menu press ‘#’ ”.
- To record the message, press 1. You can record a greeting message of 120 seconds. If the message duration is less than 60 seconds, press # after you have completed the message.
- The VMS prompts: "To record press ‘1’, to play press ‘2’, to erase press ’3’, to go to the previous menu press ‘#’ ”.
- To listen to the greeting message you recorded, press ‘2’.
- On completion of play back of the greeting message, the VMS prompts: "To record press ‘1’, to play press ‘2’, to erase press ’3’, to go to the previous menu press ‘#’ ”.

Message Verification

Message Verification enables

- extension users to check the message they have recorded before sending it to someone.
- extension user to check the message they have recorded as a reply to any message.
- external callers to check the message they have recorded in the mailbox of an extension user.

Thus Message Verification is used in the VMS features — Messages left by External Callers, “Sending Messages”, and “Broadcast Message”.

How it works

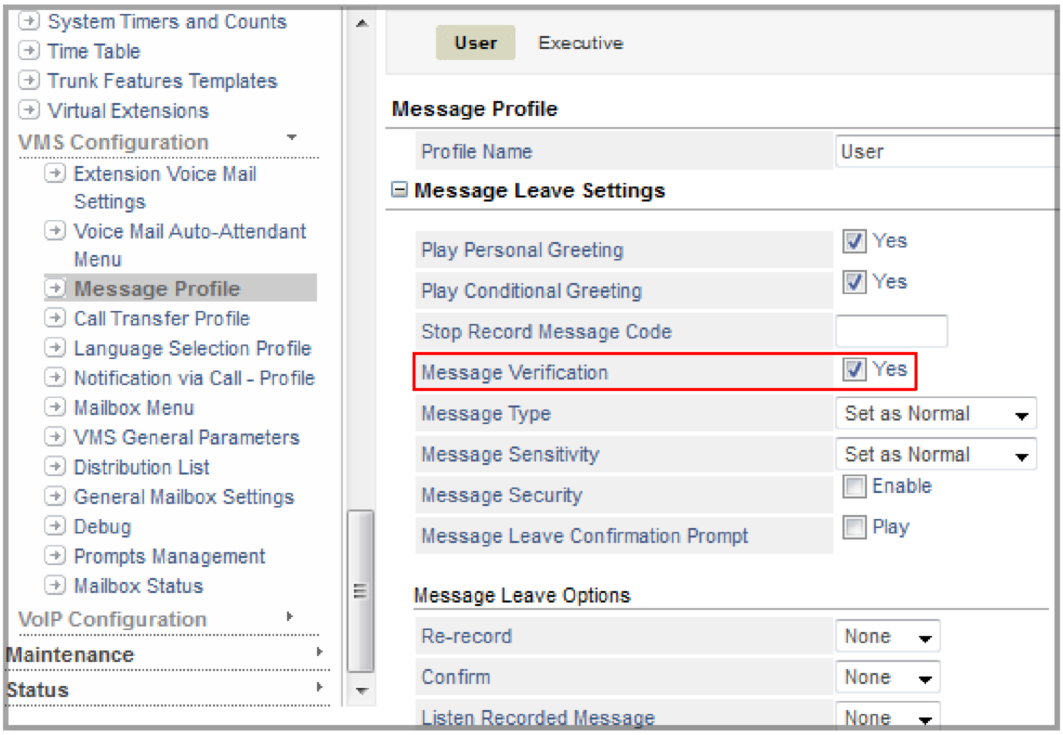
- For Message Verification to work, it must be enabled in the VMS.
- With Message Verification enabled, each time a caller or an extension user records a message, the VMS offers to the caller/extension user the option to verify the recorded message and re-record the message, if they want.
- When the caller/extension user uses the option to verify and re-record the message, the VMS sends the message to the mailbox of the receiver.

How to configure

By default, Message Verification is disabled in the VMS. So, callers and all extension users with voice mail facility cannot verify the message they have recorded. If you want to enable this feature,

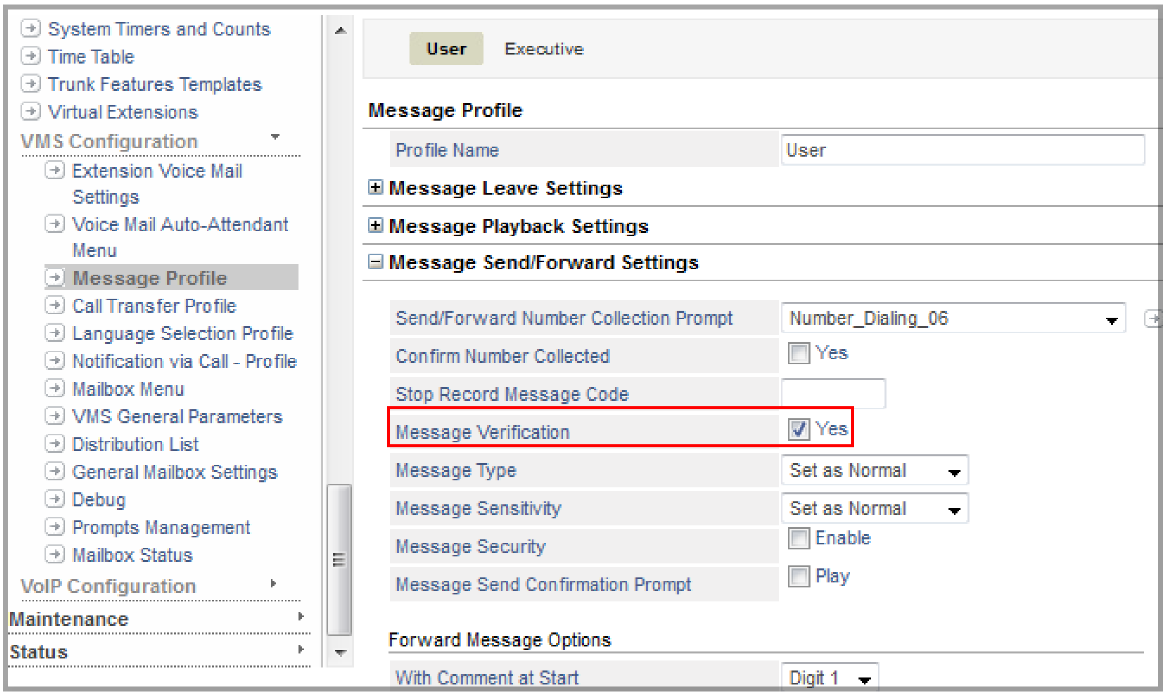
- Login as System Engineer.
- Click **Configuration**, click **VMS Configuration**.
- Click **Message Profile**.
- To allow external callers to check the message they have recorded in the mailbox of an extension user or extension user to check the message they have recorded as a reply to any message.

- Click **Message Leave Settings** to expand.



- Select the **Message Verification** check box, to enable.
- Click **Submit.**

- To allow extension users to check the message they have recorded before sending it to someone,
  - Click **Message Send/Forward Settings** to expand.



- Select the **Message Verification** check box, to enable.
- Click **Submit.**

Message Notification

The VMS sets Message Wait on the extension, whenever a new message arrives in its personal Mailbox of the extension. The VMS indicates the new message to the extension in the form of a Stuttered Dial Tone, if this is set as the Type of *Message Wait Indication* for the extension. See “Message Wait” to know more.

How it works

- Extension A has Message Wait Indication type set as Stuttered Dial Tone.
- Whenever there is a new message in A’s mailbox, the system will play a Stuttered Dial Tone when A goes off-hook.
- A must access the mailbox to listen to the new messages.

How to configure

To provide Message Notification to extensions, you must configure **Message Wait Indication** as **Stuttered Dial Tone** for the extension. For detailed instructions, refer “Message Wait Settings” in “Extension Voice Mail Settings”.

How to use

- Go Off-hook, you will hear a Stuttered Dial Tone as Message Wait Indication.
- Access your mailbox.
- The VMS plays the message: “You have <n> new messages” followed by the prompt: “Enter your mailbox password”.
- Dial your mailbox password, and follow VMS prompts.

Mailbox Settings

The VMS allows extension users to change the settings of the following facilities of their mailbox:

- Record the Extension Name for their mailbox.
- Redirect Messages from their mailbox.
- Delete all old messages.
- Delete all messages.
- Record Personal and Conditional Greetings for their mailbox.
- Configure the Personal (Mobile/Alternate) Number and the Assistance Number.

To be able to use this feature, make sure a digit has been assigned to Mailbox Management. For instructions, see “Mailbox Access”.

How to use

- Call the VMS by dialing **3931**.

- The VMS takes you to your mailbox.

- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".

- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4'."

- Dial 4 to go to Mailbox Settings.

- The VMS responds with: “For Mailbox Name press '1', For message redirection press '2', To delete all old messages of your mailbox press '3', To delete all messages of your mailbox press '4', For Mailbox Greetings press ‘5’, To go to previous menu press '#'.”

By default no digits are assigned for the options - Assistance Number and Personal Number, hence these options will not be played to the caller. For details, see “Mailbox Management”.

To assign Name to your Mailbox:

- Dial 1 → VMS prompts: "To record press '1'. To play press '2'. To erase press '3'. To go to previous menu, press '#'."

- Dial 1 to record name for mailbox → VMS prompts: "Record your name after the beep and press any digit to end."

Names for extensions can also be recorded from the System Administrator mode. See “Record Greetings/ Name”.

To delete old Messages in your Mailbox:

- Dial 3 → VMS prompts: “You are about to delete all old messages of your mailbox. To proceed press 1, to cancel press press any digit.”

- Dial 1 to delete all old messages in your mailbox → VMS responds with: "Your old messages have been deleted".

To delete all Messages in your Mailbox:

- Dial 4 → VMS prompts: "You are about to delete all messages of your mailbox. To proceed press 1, to cancel press press any digit."

- Dial 1 to delete all messages in your mailbox → VMS responds with: "Your messages have been deleted".

To configure Assistance Number for your Mailbox:

- Dial 6 → VMS prompts: "To enter number press '1', to play number press '2', to clear number press '3', to go to previous menu press '#'."

Make sure the Assistance Number is an extension number.

To configure Personal (Mobile/Alternative) Number for your Mailbox:

- Dial 7 → VMS prompts: "To enter number press '1', to play number press '2', to clear number press '3', to go to previous menu press '#'."

Any external number can be configured as the Personal (Mobile/Alternative) Number.

To redirect Messages in your Mailbox, see "Redirecting Messages".

To record personal greetings for your mailbox, see "Recording Personal Greetings".

To record conditional greetings for your mailbox, see "Recording Conditional Greetings".

Listening to Messages

The callers/extension user leave messages in the mailbox of extension users, when they are inaccessible or the user has forwarded his calls to the mailbox. Messages may also be received by the extensions users as notifications for certain events. User should access their mailboxes to listen to the messages.

VMS offers two options:

- To listen to old messages.
- To listen to new messages.

Once the message is heard by the mailbox owner, VMS treats it as an old message and places it in the old message list. The VMS also offers you the option of saving the message you have heard, as a new message.

How to use

- Call the VMS by dialing **3931**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by the prompt: "Enter your mailbox password".
- Enter your mailbox password.
- The VMS checks the utilized mailbox memory,
    - if 80% of the mailbox memory has been consumed, the VMS prompts the caller: “Your Mailbox is 80% Full. Please Delete old messages of your mailbox.”
    - if 100% of the mailbox memory has been consumed, the VMS prompts the caller: ”Your Mailbox is Full. Please Delete old messages of your mailbox.”
- VMS prompts: "To listen to new messages press '1', to listen to old messages press '2', to send a message press '3', to change your mailbox settings press '4’."
- Dial 1 to listen to new messages Or dial 2 to listen to old messages.
- On the completion of a message, the VMS plays the message: ‘To replay the message press '1', for Date and time stamp press '2', to reply to message press ‘3’, to delete the message press '4', to play the next message press '5', to forward the message press '6', to save the message as new press '7', to go to previous menu press '#'.

Leaving a Message

The VMS allows,

- External callers to leave messages in the extension user's mailbox.
- Extension users to leave message in the mailbox of other extensions.

To leave a message, the called extension must have a mailbox. The length of message recorded by the callers/ extension users must not exceed the message length set for the called extension's mailbox. If the message recorded by the callers/extension users exceeds the message length set for the called extension's mailbox, the VMS will stop recording the message after the time set and save the partially recorded message.

How to use

- The incoming call on the trunk is answered by the VMS Auto Attendant.The VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.

The System Greeting may vary depending on the timezone — Working Hours, Break Hours, Non-working Hours. In the above prompt the greeting is as per working hours.

- The caller dials 7 to leave message → VMS prompts: "Enter the Extension number for which you wish to leave message."

- The caller dials the desired extension number → VMS prompts: "Record your message after the beep and press any digit to end."

- VMS prompts: " Thank you for your call."

- *The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see "Message Send/ Forward Settings"*

- *It is mandatory for the caller/extension user to terminate the recording by dialing # (or the digit configured). If recording of the message is terminated simply by going on-hook, the VMS will not terminate the recording and the call will be disconnected only after time-out, that is, the Maximum Message Length configured for the extension.*

If extension users wish to leave a message for another extension, refer "Sending Messages".

Accessing the General Mailbox

A General mailbox is a shared mailbox between extension users. The General Mailbox is used for recording messages when the mailbox of an extension is full.

How it works

The VMS offers the following options to extensions when their mailbox is full:
- Not offer the caller to record a message.
- Overwrite the existing messages in the mail box with the new message.
- Deliver the new message to the General Mailbox.

If you configure Delivery of new messages to General Mailbox on an extension, whenever the mailbox of the extension is full, the VMS will offer the caller to record a message. This message will be recorded in the General Mailbox.

The extension user, whose mail box is full, can listen to the new message by accessing the General Mailbox.

The extension user can access the General Mailbox, if this feature is enabled in the Class of Service of the extension and the Password assigned to the General Mailbox (if configured) is known to the extension user. For details see "General Mailbox Settings".

How to configure

To offer extension users the facility of the General Mailbox when their mailbox is full, you must do the following:

- Configure the option **New Message Delivery Option in Mailbox Full Condition** in the **Extension Voice Mail Settings** of the extension.

  For the option **New Message Delivery Option in Mailbox Full Condition**, select **Deliver to General Mailbox**.

  For instructions on configuring the parameter, see "Extension Voice Mail Settings".

- Enable the feature General Mailbox in the "Class of Service (CoS)" of the extension.

- If required, assign a Password for accessing the General Mailbox, see "General Mailbox Settings".

How to use

For Extended IP Phone Users

- Press DSS Key assigned to General Mailbox.

  OR

- Dial 1176.
- Follow VMS prompts.

Forwarding Messages

The VMS enables extension users to forward messages of their mailbox to other mailboxes.

How it works

The Forwarding Messages feature of the VMS offers to extension users the following options:

- forward messages after adding a comment.
- forward messages without adding comment.
- forwarding messages with Message Read Receipt request.

Before forwarding a message, the VMS asks the Sender, if the Sender needs a confirmation that the message has been read by the Recipient.

If the Sender requests for 'Message Read Receipt', the VMS stores this request. When the Recipient reads the message, the VMS generates a file containing the first 5 seconds of the message that was sent by the Sender and delivers it to the Sender's mailbox in the form of a new message with a prompt: "This message was read by <Extension Name[127]><5 seconds of the message sent>" with the Date and Time at which the message was read.

How to use

- Call the VMS by dialing **3931**.
- The VMS takes you to your mailbox.
- VMS responds with: "You have <n> new messages" followed by "Enter your mailbox password".
- Enter your mailbox password → VMS responds with: "You have no/n new messages".
- The VMS prompts: "To listen to new messages press 1, to listen to old message press 2, to send a message press 3, to change your mailbox settings press 4".
- Dial 1 or Dial 2 and listen to the messages → VMS prompts: "To replay the message press 1' for Date and Time stamp press 2, to reply to message press 3, to delete the message press 4, to play the next message press 5, to forward the message press 6, to save the message as new press 7, to go to previous menu press #."
- Dial 6 → VMS prompts: "To forward the message with comment before the message press 1, to forward the message with comment after the message press 2, to forward the message without comment press 3, to go to previous menu press #."
- To forward message with comment before the message dial '1' or To forward message with comment after the message dial '2'. The VMS prompts: "Enter the Destinations".
  - Dial Destinations/Distribution list number to forward the message.
  - The VMS prompts: "Record your message after the beep and press any digit to end."

127. If the Extension Name is not configured, while playing the prompt the Extension Number will be played instead of the Name.

- Speak to record your comment and press # to end recording.
- The VMS prompts: "To request read receipt press 1, to ignore read receipt press 2."

- To forward message without comment dial '3'. The VMS prompts: "Enter the Destinations".
  - Dial Destination number/Distribution list number to forward the message.

- *The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see "Message Send/ Forward Settings" .*
- *Extension users must be careful in dialing destination numbers. If invalid destination is entered then the VMS will clear all the entries and will ask the mailbox owner to re-enter all the destinations again.*
- *A message can be forwarded to maximum 10 destinations. A destination can be an extension or a distribution list.*

Email Based Notification

VMS supports E-mail Based Notification, the Unified Messaging feature. This is used to inform the extension users, the arrival of new messages in their mailbox and the memory usage status of their mailbox.

Extension users can also receive new messages as attachments to the email.

Extension users will receive the notification for the mailbox memory usage for the following:
- when 80% of their mailbox memory has been consumed.
- when 100% of their mailbox memory has been consumed.

For Email Notification to function, you must configure the "SMTP Settings"

How to configure

To be able to use this feature, you must configure the parameters for **Message Wait Notification via Email** under Message Wait Settings in Extension Voice Mail Settings.

- You can select the desired option as Notification: **Without attachment, With attachment** or **With attachment and mark voicemail as read**.

- Specify the **E-mail Address** of the extension user to which the notifications are to be sent.

  For the General Mailbox you only need to specify the **E-mail Address** to which the notifications are to be sent.

For details see "Message Wait Notification via E-Mail" in "Extension Voice Mail Settings".

The Notification sent for arrival of new messages in the mailbox and the memory usage status users mailbox will be sent to the *Email ID* configured under *Message Wait Notification via Email*.

The messages sent can be customized as per your requirement. For detailed instructions, refer "VMS E-Mail Notification".

Message Wait Notification via Call

The VMS supports Notification via Call to inform the extension users about the arrival of new messages in their mailbox.

Extension users can receive new message notification calls on a phone number of their choice. This number may be another extension number or an external number. You can set the Type of notification calls as

- **Immediate:** Users will receive notifications as soon as a new message arrives in their mail box.
  Or
- **Scheduled:** Extension users will receive notification at specified time intervals.

You can set the preferred time slots in a day during which notification calls should be made to extension users. In addition to the time slot preference, you can also choose to receive notification calls on a Holiday.

Message Notification via Call will not work for Department Group.

How it works

For this feature to work, you must do the following configuration for the extension:

- Select the type of Notification call.
- Define the preferred time slots by configuring Time Zones. You can configure four different Time Zones, defining the Start Time and End Time for each Time Zone.
- Configure the phone number to which the notification call is to be made. If the number is an external number, configure the Trunk Access Code to be used for making the calls.

When **Immediate** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.
- The system checks the preferred start and end time of the time zones configured for the extension. If the message has arrived within the preferred time slot (Start and End Time) it immediately makes the notification call on the number configured for the user.

  If the number is an external number, the system dials out the number using the Trunk Access Code (TAC) assigned for making notification calls.
- When the call is answered, the extension user gets connected to the VMS and can listen to the message.
- If the notification call is not answered, by default, the system makes three attempts (Message Notification Retry Count; configurable) at an interval of 5 minutes (Message Notification Interval; configurable) between each attempt.
- If the notification call remains unanswered after the third attempt, the system will not make any more attempts to place this notification call. The next notification call will be made only when another new message arrives in the mailbox of the user between the start and end time of the configured time zone.

When **Schedule** is selected as the Type of notification,

- A new message arrives in the mailbox of the extension user.

- The system checks the start time of the time zone(s) configured and the notification call will be made on the number at the subsequent start time.

  If the number is an external number the system dials out the number through the Trunk Access Code (TAC) assigned for making notification calls.

- When the call is answered the extension user gets connected to the VMS and can listen to the message.

- If the notification call is not answered, the system makes three attempts (Message Notification Retry Count; configurable) at an interval of 5 minutes (Message Notification Interval; configurable) for each time zone. The system will continue to make attempts to place the notification call till the call is answered.

Thus, when Notification type is Immediate, notification call is made for each message that is received within the start and end time configured in the time zone.

When Notification type is Scheduled, notification call is made for all messages received before the start time configured in the time zone. Where multiple time zones are configured, notification call will be made at the start time of the next time zone.

How to configure

For Message Wait Notification via Call, you need to configure:

- the parameters for **Message Wait Notification via Call** under Message Wait Settings in "Extension Voice Mail Settings".

- select the desired profile in Schedule Profile. For instructions to configure the profile parameters, see "Configuring Notification via Call - Profile".

- Make Message Notification call using TAC for calls to be made to external numbers. See "Configuring VMS General Parameters".

- if required, the Message Notification Retry Count, Message Notification Interval and Message Notification Ring. See "System Timers and Counts".

Dial By Name

The VMS Auto Attendant allows external callers and extension users to reach the desired person in an organization by dialing the name of that person. This feature is useful when caller/extension user cannot recall the extension number of the person they want to speak to.

How to configure

For this feature to work, each extension user's name must be abbreviated and configured on the extension. It is recommended that the extension users' names be abbreviated to the first three letters of the name. As far as possible, abbreviate names such that no two names are the same.

You must configure the **Abbreviated Name** in the *Extension Voice Mail Settings*. For detailed instructions, refer "Extension Voice Mail Settings".

Make sure the names for the desired extension users are recorded. For details, see "Mailbox Settings".

How to use

- The incoming call on the trunk is answered by the VMS Auto Attendant. The VMS greets the caller with the Greeting message followed by the Welcome Message: "Welcome! Please dial the extension number or To dial by name press '6',To leave a message press '7', To access your Personal Mailbox press '8', For further assistance press '9', To disconnect the call press '#'.

The System Greeting may vary depending on the timezone — Working Hours, Break Hours, Non-working Hours. In the above prompt the greeting is as per working hours.

- The caller dial 6 → VMS prompts: "Please enter first three letters of the name."

- If multiple matches are found the VMS prompts: "More than on match found. Matching Names will be played one by one. To Select the name press '1', to Skip the name press '2', To Repeat the last name press '3'."

- Dial 1 → VMS prompts: "To confirm press '1', to Re-enter press '2'."

- The caller dials 1 → VMS transfers the call as per the transfer type assigned to the selected station. Talk.

  - If the caller dials invalid digits, the VMS prompts: "Sorry no match found." followed by the prompt: "Please enter first three letters of the name."

Extension users can use "Dial By Name" to reach another extension.

Dial by Extension Number

The VMS Auto Attendant allows external callers and extension users to reach directly the desired person in an organization by dialing the extension number of that person.

How to use

- The incoming call on the trunk is answered by the VMS Auto Attendant. The VMS greets the caller with the Greeting message followed by the Welcome Message: “Welcome! Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.

The System Greeting may vary depending on the timezone — Working Hours, Break Hours, Non-working Hours. In the above prompt the greeting is as per working hours.

- The caller dials valid extension number.
    - VMS prompts: “Please enter first three letters of the name.”

Make sure you have:
- *selected the **Confirm Name** check box. Refer to “Auto-Attendant Settings”.*
- *configured the **Abbreviated Name** for the respective extension. To know more, refer to “Extension Voice Mail Settings”*

- The caller enters the first three letters of the name.
    - If the caller dials valid digits and if the Abbreviated Name is not configured → VMS transfers the call as per the Call Transfer Profile selected for the extension. Talk.
    - If the caller dials invalid digits → VMS prompts: “No match found.” followed by the prompt: “Please enter first three letters of the name.”
- If the extension number dialed by the caller is invalid, the VMS prompts: "The number is not valid." followed by the prompt: "Please dial the extension number or To dial by name press ‘6’,To leave a message press ‘7’, To access your Personal Mailbox press ‘8’, For further assistance press ‘9’, To disconnect the call press ‘#’.
- Dial valid extension number and talk.

Call Transfer Types

The VMS Auto Attendant answers calls of external callers and extension users (referred to here as 'callers') and transfers the call to the extensions according to the Call Transfer type set for the extension.

You must configure the Call Transfer Type in the Call Transfer Profile assigned to the extension. Different Call Transfer Profiles can be assigned for Working, Break and Non-working hours.

The VMS Auto Attendant offers the following types of Call Transfers:

- **Blind**: When the caller dials the extension number, the VMS Auto Attendant transfers the call on the extension without checking whether it is busy or free.
- **Wait for Ring**: When the caller dials the extension number, the VMS Auto Attendant waits for the extension to start ringing and then transfers the call.
- **Wait for Answer**: When the caller dials the extension number, the VMS Auto Attendant transfers the call when the extension answers (goes OFF-Hook).
- **Screened**: The VMS Auto Attendant prompts the caller to record his/her name. It puts the caller on hold and places the call on the desired extension. If the extension is free and answers the call, the VMS announces the caller's name to the extension user and prompts the extension user to choose whether or not to speak to the caller. If the extension user chooses to talk, the VMS transfers the call.
- **None**: When the caller dials the extension number, the VMS Auto Attendant transfers the call to the desired extension users mailbox directly.

How to configure

Call Transfer Type must be configured in the Call Transfer Profile assigned to each extension. For instructions, refer "Call Transfer Settings" in "Extension Voice Mail Settings" and "Call Transfer Profile".

For calls received on trunks the VMS Auto Attendant transfer the call as per the configuration in "Auto-Attendant Settings" in "Voice Mail Auto-Attendant Menu".

Broadcast Message

Broadcasting Message allows you to send the same message to all extension users having voice mail, at the same time. You can use Broadcast Message to make general announcements like hosting of an event, an unplanned day off, and other such activities or events.

How to use

To Broadcast Message,

- Enter SA Mode.
- Dial 1072-301
- VMS prompts: "Record your message after the beep and press any digit to end".
- Speak to record your message after the beep, and press # (hash/pound) to end the message.

- *The system provides an option to the caller to verify the message before storing the same. Enable the Message Verification check box and configure the digits for the Message Leave Options. The message will be played as per the digits selected for the options. For detailed instructions, see* *"Message Send/ Forward Settings".*
- *The length of the message you want to broadcast must be equal to or less than the minimum of message length configured for the mailboxes, or else your message will be truncated. For instance, if the Maximum message length for a mailbox is configured as 15 seconds, maximum length of the message to be broadcast must be less than or equal to 15 seconds. If the broadcast message exceeds this limit, the system will play the first 15 seconds and truncate the remaining part of the message.*

CHAPTER 10 System Maintenance

Certificate Management

ANANT UCS supports certification for TLS and Web Server. In case, the Clients (Standard SIP phones) require this certificate, you can download the same and install it in them.

ANANT UCS supports the following types of Certificates.

1. Root CA Certificate
2. Self-Signed System Certificate
3. CA-Signed System Certificate

Certificate Authority (CA) is a trusted (third-party) organization which creates and sells TLS Certificates to websites or organizations. CAs issue a TLS Certificate to the organizations/websites after verifying their credentials.

Generally, one TLS Certificate is issued for a particular server/website domain and it is valid for a limited period of time.

ANANT UCS supports upto 5 (1 Default Root CA + 1 default Self-Signed +3 newly generated Self-Signed or CA-Signed) Certificates in the system.

The Default Root Certificate and Default System Certificate cannot be deleted from the system.

Now, let us understand the purpose of each type of certificate.

Root CA Certificate

The Root CA Certificate is an in-built Certificate Authority which signs the System Certificates created by our own server or by other servers. The Root CA Certificate is also used by the clients to validate the Server Certificate that is received during TLS negotiation.

By default, a Root CA Certificate is already provided in the system. You may also regenerate a Root CA Certificate to bind the Certificate with the details of your organization. Refer “Regenerate Root CA” to know more.

Self-Signed System Certificate

When a System Certificate is signed using a Root CA Certificate, it generates a Self-Signed System Certificate. This certificate is generated by the clients themselves or by the Servers and then given to their clients. The Self-Signed System Certificate is faster to create since it is self-issued but it is not as robust as CA-Signed Certificate.

This certificate must be installed in the trusted list of clients that connect over TLS with the Server. Since the certificate is self-signed, it is not likely to be in the clients' trust file, hence, they need to add it. To know more, refer "Generate Self-Signed System Certificate".

If a remote client has a policy of accepting certificates only from trusted CAs, then it is likely that the Self-Signed Certificate sent by the server during TLS negotiation might get rejected. In such cases, you need to install a CA-Signed System Certificate in the system.

CA-Signed System Certificate

CA-Signed System Certificates are the TLS Certificates which are created by trusted (third-party) Certificate Authorities, signed and sold to any applicant. These certificates contains the identity of the owner. It is the responsibility of the CA to verify the owner's (applicant's) credentials.

Since the CA-Signed System Certificate is issued by a trusted CA, it ensures complete protection from security threats.

If you wish to install a CA Signed Certificate in your system, you must do the following.

1. Generate and enroll the Certificate Signing Request (CSR). For more details, refer "Generate Certificate Signing Request".
2. Get the Certificate Signing Request (CSR) verified and signed by the Certified Authority (CA).
3. Upload the CA-signed system certificate in the server. For more details, refer "Upload Certificates".

Enrolling the Certificate Signing Request with CA

Enrollment is a process of obtaining a certificate from any trusted third party (CA). After you have generated and downloaded the Certificate Signing Request (CSR), you must contact any authorized third party that issues TLS Certificates to companies or web owners, such as GoDaddy, DigiCert, Thawte, VeriSign, etc. and enroll the Certificate Signing Request (CSR) with them. These third parties Certificate Authorities (CA) have their charges to sign and validate the Certificate Signing Request (CSR) for a fixed time interval.

Verification and Signing of the Certificate Signing Request by CA

On receiving the Certificate Signing Request (CSR), the CA verifies the Server's / User's credentials. After successful verification, the CA signs and sends the signed certificate to the server. These signed certificates are called as CA-Signed System Certificate.

Upload of CA-Signed System Certificate

After the CA-signed system certificate is received, upload it in your server along with the private key. This certificate will be sent by the server to the clients, if assigned for the service, during TLS negotiation.

How it works

To be able to use this feature,

- Select the **Enable SIP over TLS** check box in "Configuring VoIP Parameters".
- Select **Transport Mode**[128] as TLS in Location-1 of SIP Extension Settings of the respective Extended IP phone.
- Make sure the clients communicating with the server over TLS protocol are compatible with the TLS version configured in the system. To know more, refer **Allowed TLS Versions** in "Security Settings".
- Make sure the Date and Time of the server is synchronized with the SNTP Server.

128. *Not applicable for SPARSH VP248 and Standard SIP Phones.*

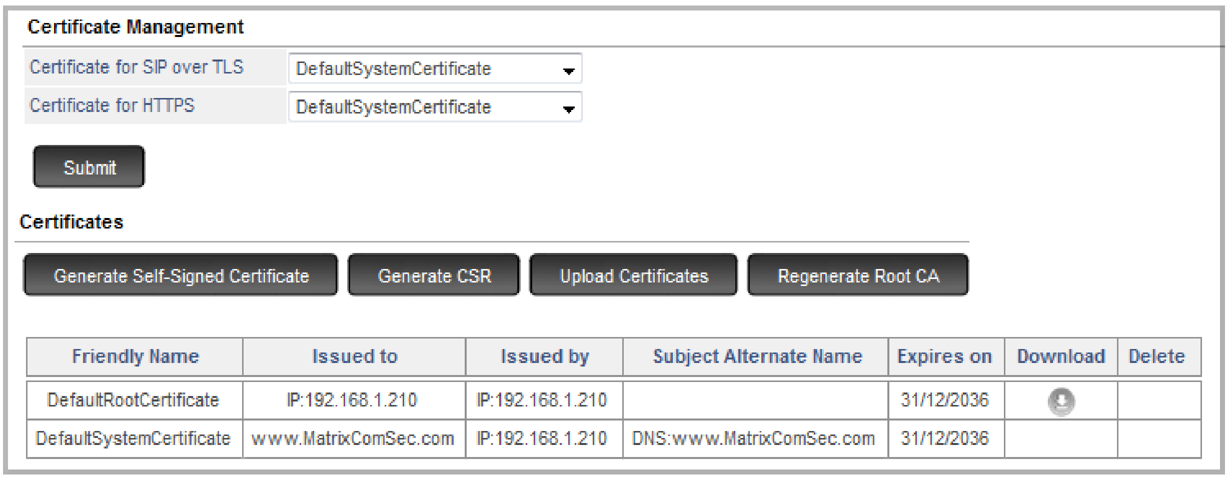
- Make sure that the validity of the Certificate has not expired to ensure secure connection between the clients and the server.

Now, let us understand how it works,

- The system has a default Root CA certificate installed which is generated using your own company related details.
- The Root CA certificate is used to create the System Certificate (self-signed), which is used by the clients for secure communication with the server.
- If you want to use System Certificate signed by some third-party Certificate Authority, then you need to generate a CSR (Certificate signing request). For example, for Standard SIP clients and third-party servers.
- When you generate and download the CSR, two files are saved — the CSR file and the Private key. You may secure the Private key using a Pass-phrase.
- Send the CSR file only to the authorized Certificate Authority (CA), get it signed and upload the CA-Signed System Certificate along with the private key in your system. The same pass phrase, if configured while generating CSR, needs to be entered while uploading the CA-signed certificate in your system.
- After the Certificate is successfully uploaded, it is displayed in the list of certificates.
- The System Certificates (Self-signed or CA-signed) thus ensure secure communication for SIP over TLS and HTTPS between the server and clients.

How to configure

- Log in as System Engineer.
- Click **Maintenance**.
- Under Maintenance, click **Certificate Management**.



Certificate Management

- **Certificate for SIP over TLS:** Select the Certificate that will be used to establish secure SIP connection between server and IP devices. Default: Default System Certificate.
- **Certificate for HTTPS:** Select the Certificate that will be used to establish secure HTTP connection between server and clients. Default: Default System Certificate.

*If you change the Certificate in **Certificate for SIP over TLS** or **Certificate for HTTPS**, then it may result in drop of all ongoing TLS connections and VoIP calls.*

- Click **Submit** to save the changes you made.

Certificates

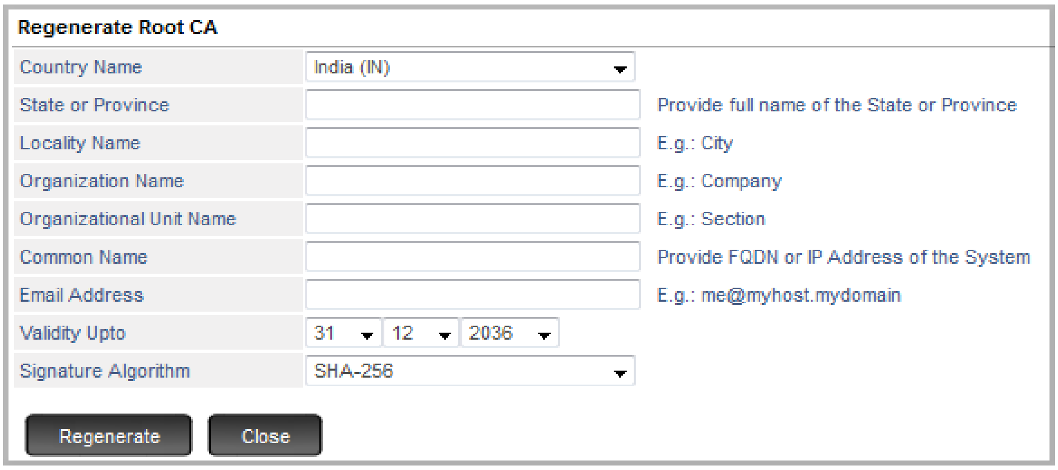
Regenerate Root CA

You may use this facility if you want to generate a Root Certificate depending upon the installation scenario.

On regeneration of a Root CA Certificate, the default Root CA Certificate will get replaced. Additionally, all the Self-Signed System Certificates are regenerated and replace the existing ones.

To regenerate a Root CA Certificate,

- Click the **Regenerate Root CA** button.



A new window opens which displays the parameters to be configured.

- **Country Name:** - Select the name of the country from the list. Default: India (IN).
- **State or Province:** - Enter the full name of the state or province. For example, Gujarat.
- **Locality Name:** Enter the name of the city. For example, Vadodara.
- **Organization Name:** Enter the name of your organization where ANANT UCS is installed. For example, Matrix.
- **Organizational Unit Name:** Enter the name of the unit or section or domain of your organization, where ANANT UCS is installed. For example, Factory.
- **Common Name:** Enter the FQDN or IP Address of the system. This Common Name serves as the distinguishing factor. The Common Name can be a maximum of 64 characters. For example, www.anant.com.

- **Email Address**: Enter the Email address of your host. For example, xyz@matrix.com. The Email Address can be a maximum of 40 characters.

- **Validity Upto:** Select any date from the present date of the system to 31st December 2036, the duration for which the certificate should be valid. Default: 31 December 2036.

- **Signature Algorithm:** Select the algorithm using which the certificate has to be signed. You may select — SHA-1, SHA-256, SHA-512. Default: SHA-256

- Click **Regenerate**.

  A message appears to confirm that the Certificates have been generated successfully.

  Once the Root CA has been successfully generated, the system automatically regenerates the Self-Signed Certificates using the newly generated Root CA Certificate.

- Click **OK** to close the window. You will be redirected to the Certificate Management page.

Generate Self-Signed System Certificate

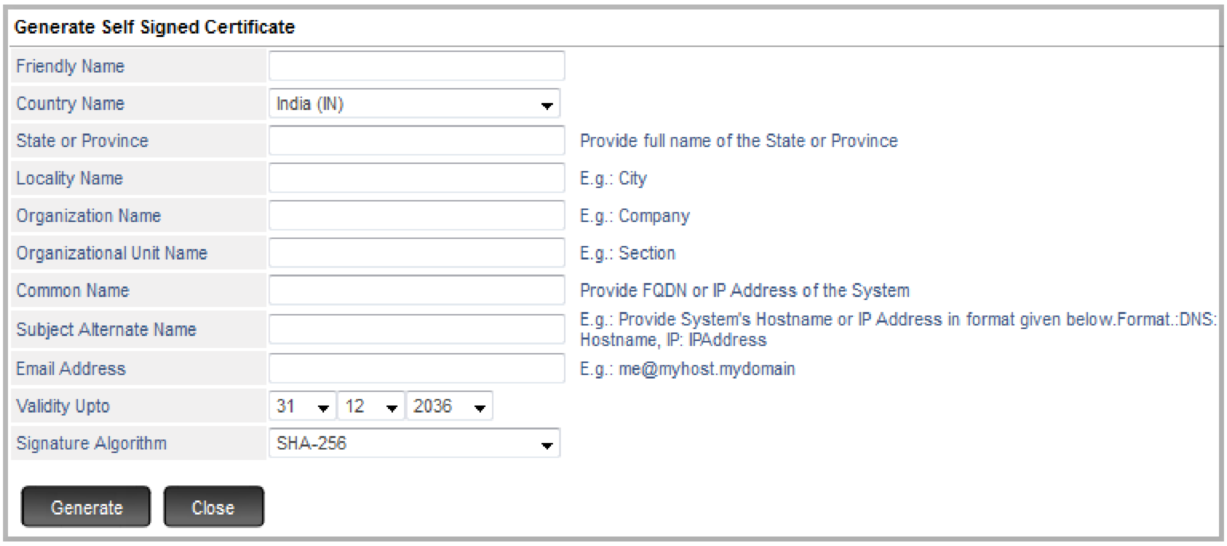
Depending on your installation scenario, you can generate a Self-Signed System Certificate using the details of your own organization.

The Self-Signed System Certificates cannot be downloaded.

You can generate multiple Self-Signed System Certificates using the same Root CA Certificate.

To generate a Self-Signed System Certificate,

- Click the **Generate Self-Signed Certificate** button.



A new window opens which displays the default parameters.

- **Friendly Name:** Enter the name you want to assign to the certificate. Make sure the name you enter is unique and does not match with the name of any other certificate configured in the system. The name can be a maximum of 32 characters. For example, MatrixComsec

- **Country Name:** - Select the name of the country from the list. Default: India (IN).
- **State or Province:** - Enter the full name of the state or province. For example, Gujarat.
- **Locality Name:** Enter the name of the city. For example, Vadodara
- **Organization Name:** Enter the name of your organization where ANANT UCS is installed. For example, Matrix.
- **Organizational Unit Name:** Enter the name of the unit or section or domain of your organization, where ANANT UCS is installed. For example, Factory.
- **Common Name:** Enter the FQDN or IP Address of the system. This Common Name serves as the distinguishing factor. For example, www.ANANT.com.
- **Subject Alternate Name:** Enter the names of the multiple domains separated by comma (if the same certificate is to be issued for multiple domains of an organization). The Subject Alternate Name can be a maximum of 255 characters. For example, DNS:matrixcomsec.com, IP: 192.168.101.123

Make sure you enter "DNS:" before the domain name and "IP:" before the IP address in the Subject Alternate Name.

- **Email Address**: Enter the Email address of your host. For example, xyz@matrix.com. The Email Address can be a maximum of 40 characters.
- **Validity Upto:** Select any date from the present date of the system to 31st December 2036, the duration for which the certificate should be valid. Default: 31 December 2036.
- **Signature Algorithm:** Select the algorithm using which the certificate has to be signed. You may select — SH-1, SHA-256,SHA-512. Default: SHA-256
- Click **Generate** to generate the Self-Signed Certificate.

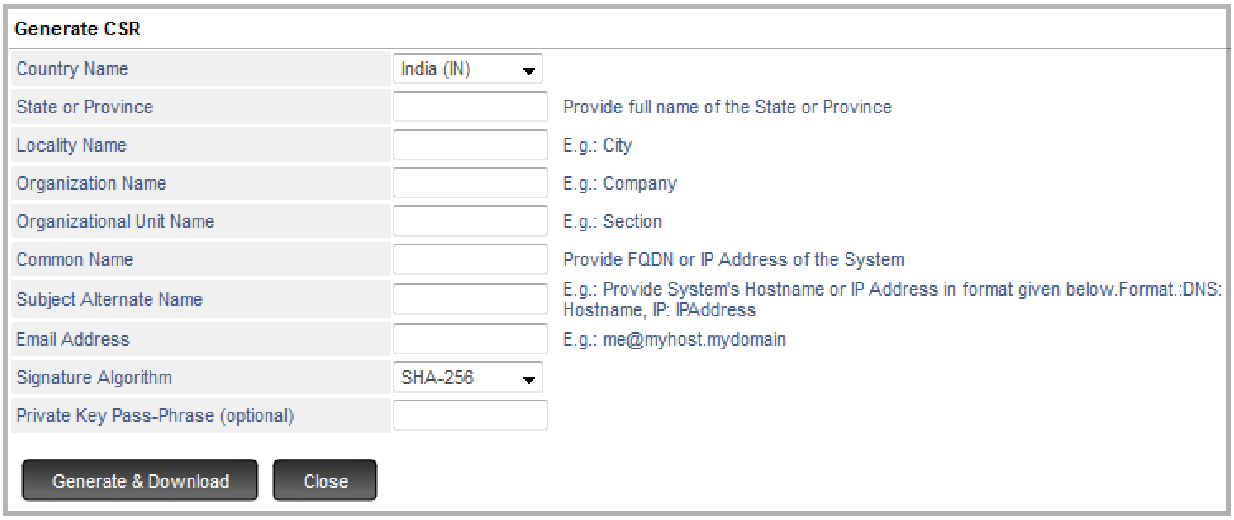
  A message appears to confirm that the Certificates have been generated successfully.
- Click **OK** to close the window. You will be redirected to the Certificate Management page.

  The newly generated Self-Signed System Certificate will appear in the **Certificates** list.

Generate Certificate Signing Request

To generate a Certificate Signing Request (CSR),

- Click the **Generate CSR** button.



A new window opens which displays the parameters to be configured for generating CSR.

- **Country Name:** - Select the name of the country from the list. Default: India (IN).
- **State or Province:** - Enter the full name of the state or province. For example, Gujarat.
- **Locality Name:** Enter the name of the city. For example, Vadodara
- **Organization Name:** Enter the name of the organization where ANANT UCS is installed. For example, Matrix.
- **Organizational Unit Name:** Enter the name of the unit or section or domain of your organization, where ANANT UCS is installed. For example, Factory.
- **Common Name:** Enter the FQDN or IP Address of the system. This Common Name serves as the distinguishing factor. For example, www.anant.com. The Common Name should not exceed 64 characters.
- **Subject Alternate Name:** Enter the names of the multiple domains separated by comma (if the same certificate is to be issued for multiple domains of an organization). For example, DNS:matrixcomsec.com, IP: 192.168.101.123

Make sure you enter "DNS:" before the domain name and "IP:" before the IP address in the Subject Alternate Name.

- **Email Address**: Enter the Email address of your host. For example, xyz@matrix.com. The Email Address should not exceed 40 characters.
- **Signature Algorithm:** Select the algorithm using which the certificate has to be signed. You may select — SH-1, SHA-256,SHA-512. Default: SHA-256
- **Private Key Pass-Phrase (Optional):** Enter the Private Key Pass-Phrase for encrypting the private key. The password must
  - be of minimum 6 characters and can be a maximum of 24 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.

- all ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ’, Double Quote " and Space) are allowed.

  Default: Blank.

- Click **Generate & Download** to generate and download the Certificate Signing Request and the Private key.
  OR
  Click **Close** to close the window.

- *The downloaded file will be a zip file. Extract the files and send the CSR file only to any trusted Certificate Authority.*

- *The zip folder download depends upon the browser you are using. Check the **Download Settings** of your browser and set the Download path accordingly.*
  OR
- *If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.*

- *If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save files**.*

- *The downloaded Private Key should not be sent to the trusted (third-party) Certificate Authority to prevent it from being mishandled by some malicious user.*

- *Make sure that you do not lose the Private Key or forget the Pass-Phrase; otherwise, you will have to generate a new CSR and set the Pass-Phrase again.*

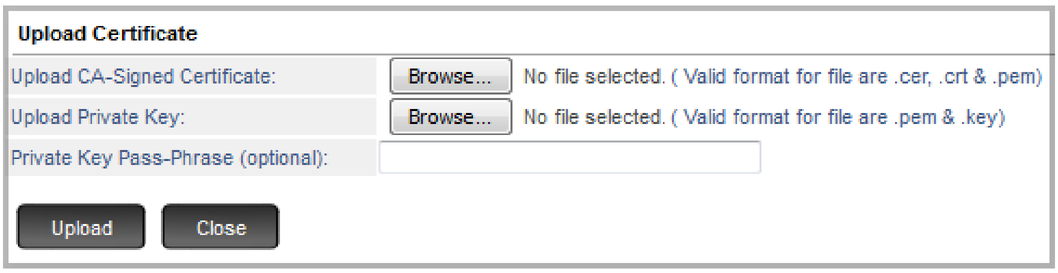
Upload Certificates

The size of the file to be uploaded should not exceed 5MB.

After you download the CSR, you must get it signed from a Trusted Certificate Authority (CA) and then upload it in your system.

To upload the CA-Signed Certificate,

- Click the **Upload Certificates** button.



  A new window opens which provides the option for uploading the CA-signed certificate and Private key.

- **Upload CA-signed Certificate:** Click the **Browse** button to reach the location on the local disk where the CA-Signed Certificate is stored in the PC. The valid formats for certificate are .cer, .crt and .pem.

- **Upload Private Key:** Click the **Browse** button to reach the location on the local disk where the Private key is stored in the PC. The valid formats for key are .pem and .key.
- **Private Key Pass-Phrase (optional):** Enter the Private Key Pass-Phrase for decrypting the private key. Make sure that you enter the same Private Key Pass-Phrase that was configured during generation of the CSR.

  If you have not configured any Pass-Phrase during generation of CSR, that is, Private Key is not Pass-Phrase protected, then you may leave this field blank.
- Click the **Upload** button to upload the CA-Signed Certificate and the Private key.

  A message appears to confirm that the Certificate has been uploaded successfully.
- Click **OK** to close the window. You will be redirected to the Certificate Management page.

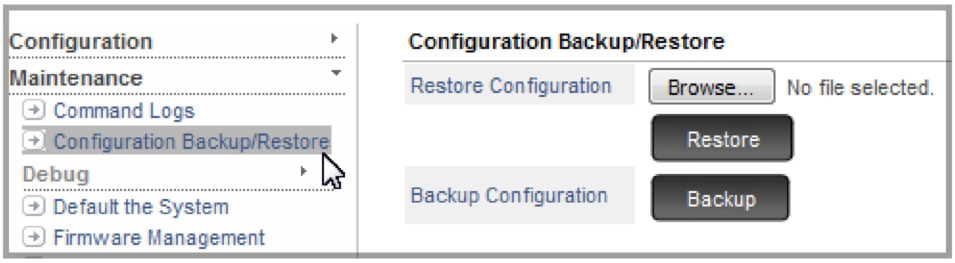
  The CA-Signed System Certificate you uploaded will appear in the **Certificates** list.

  To delete a System Certificate, click .

  To download the Root Certificate or CA-Signed System Certificate, click .

Configuration Backup/Restore

ANANT UCS provides you the facility to Backup the configuration files from the system to your local PC and Restore the configuration files from the local PC to the system at the click of a button.



Restore Configuration

- To restore configuration files from any hard drive to the system, **Restore Configuration** option is provided.

- Click on the **Browse** button to reach the location on the local disk where the configuration files are stored in PC. Make sure that the file is a zip file.

- After selecting the required configuration zip file from PC, click on the **Restore** button.

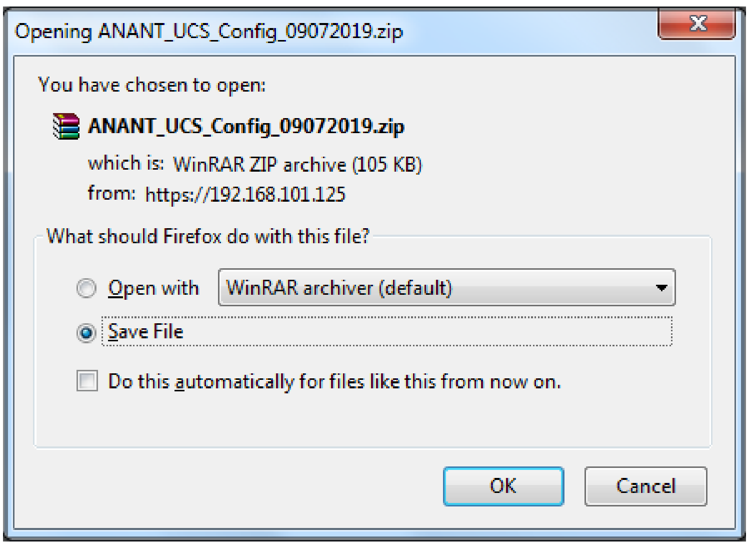
  The system initiates the restoring of configuration zip file. After successfully restoring and validating the file, the system restarts with the new configuration.

If you select a file other than zip file, an error message is displayed when you click on the Restore button.

Backup Configuration

- To save the existing configuration files as backup, **Backup Configuration** option is provided.

  The **ANANT_UCS_Config_ddmmyyyy.zip** window will open; where ddmmyyyy signifies the current date.

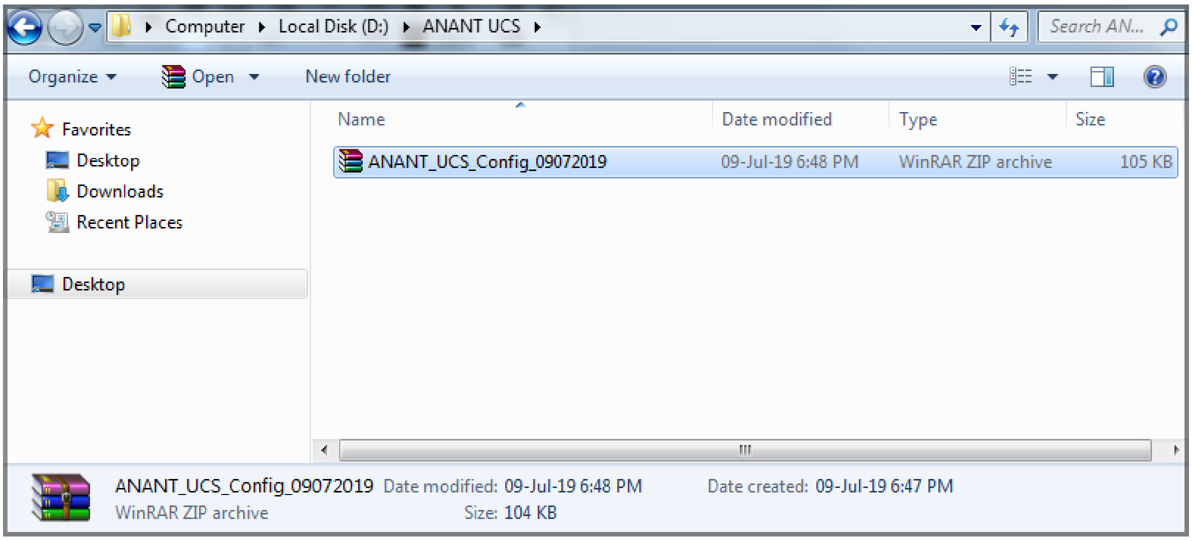


- You can either open the zip file or save the file to a location.

- *The above window display depends upon the browser you are using. Check the* ***Download Settings*** *of your browser and set the Download path accordingly.*
  OR
  If your browser does not ask you for the location you want to save your file, it saves it in the default location according to the download path specified for that browser.

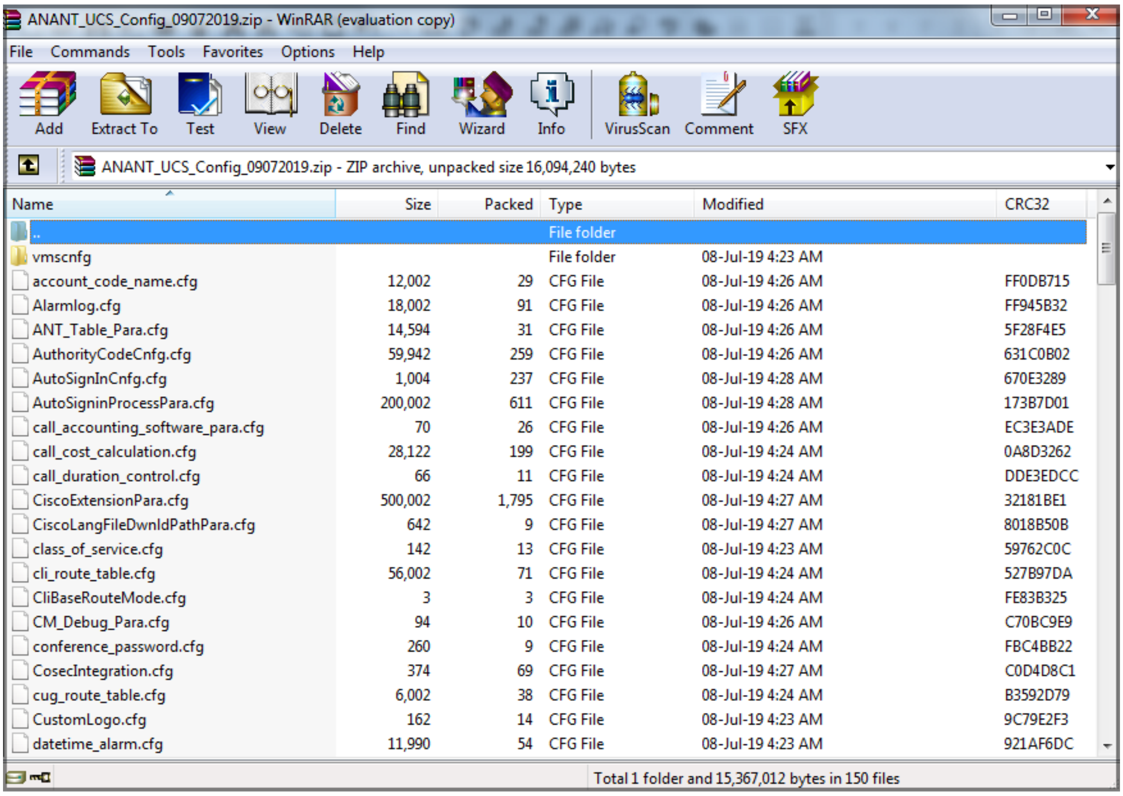
- *If you are using Mozilla Firefox (version 3.5.1recommended), before you save the configuration files, set the* ***Downloads*** *option of your browser as* ***Always ask me where to save files****.*

- Save the file on the local disk.



Save the back up configuration files by tagging the file name with the Version-Revision of the Firmware and tag the name of the backup folder on your computer with the date. This will help you at the time of restoring the back up configuration files.

- Open the ANANT_UCS_Config_09072019 folder to view the configuration files.



- Keep this folder as a backup. In case, there is a problem with the system configuration files these backup files can be restored back in the system.

Firmware Management

ANANT UCS enables you to manage the upgradation of the system firmware from your personal computer. You can upgrade the system firmware stored on your local computer with a click of a button.

Due to security concerns the Default Settings have been changed for systems purchased/upgraded with Firmwares later than V2.2. For new systems purchased these Default Settings will the applicable automatically, refer to "Modified default parameter values for Firmwares later than V2.2".

If you are upgrading the system, refer to "After updating Firmware later than V2.2" and "Modified default parameter values for Firmwares later than V2.2".

How to Configure

- Login as System Engineer.
- Under **Maintenance**, click **Firmware Management**.

To upgrade the system firmware:

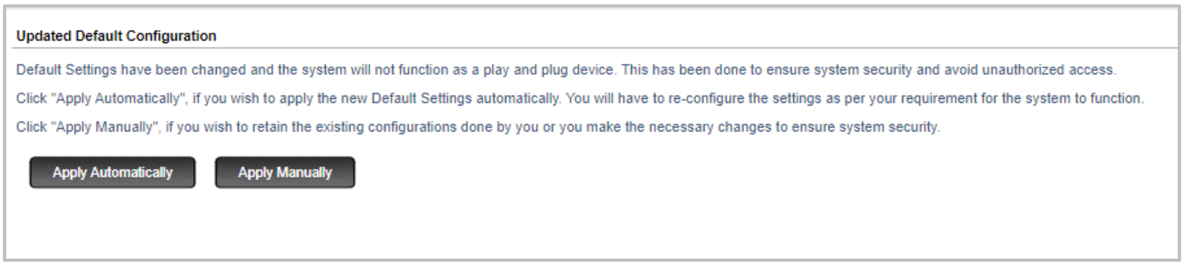
- **Upload Firmware from PC:** Click the **Browse** button to select the firmware file from the local disk on the computer. Make sure that it is an .img file.
- After selecting the required firmware file from the PC, click the **Upload** button.
- Once you upload the selected firmware, the system displays the following information:
  - Current firmware version
  - Upload firmware version
  - The specifications of the uploaded firmware.
- You can now select any of the following options:
  - **Restart:** Click this button, if you want to finally update the firmware.Once you click this button, the system restarts and the new firmware is uploaded.
  - **Schedule Restart:** Click this option if you want to update the firmware at the schedule date and time. A new window opens, select the desired date and time. The system restarts and the new firmware is updated automatically at the set date and time.

- **Discard:** Click this option if you do not want to update the selected firmware.

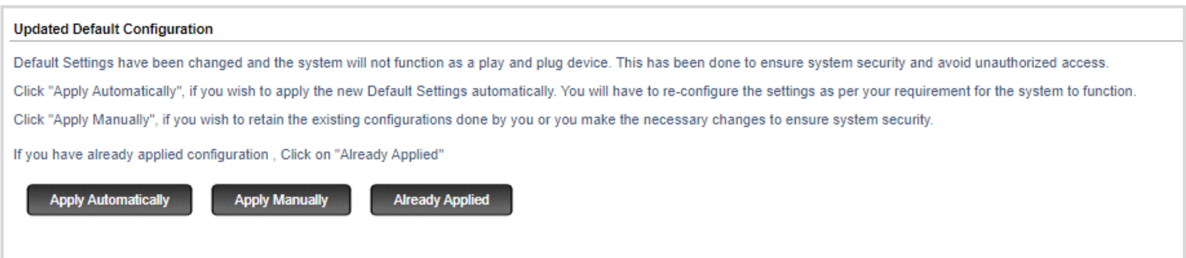
- *If you select a file other than .img file, an error message is displayed when you click on the Upload button.*
- *The specifications for the uploaded firmware will not be displayed, if the version of the uploaded firmware is lower than the current system firmware.*

After updating Firmware later than V2.2

- If you select, Restart, the system restarts and the new firmware is applied
- Re-login into the system. The following screen appears.



- To ensure system security and avoid unauthorized access, the System Default Settings have been changed and the system will not function as a play and plug device.
  - Click **Apply Automatically**, if you wish to apply the new Default Settings automatically. You will have to re-configure the settings as per your requirement for the system to function. Before you click this options make sure you have checked the new default parameters. For details, refer to “Modified default parameter values for Firmwares later than V2.2”.
  - Click **Apply Manually**, if you wish to retain the existing configurations done by you and you will have to modify the settings yourself to ensure system security.
- If you click **Apply Automatically**, the system default settings will be changed and system access will be provided.
- If you click **Apply Manually**, system access will be provided. When you re-login the following screen will appear.



  - To avoid re-appearance of the above screen at every re-login, click **Already Applied**. System access will be provided. If you select this option, make sure you have made appropriate changes in the settings to ensure maximum system security and to avoid unauthorized access. Also refer to “Security Settings” to know more about the login levels and how to set secure passwords.

Matrix will not be responsible for losses/issues arising due to inappropriate configuration.

- If you click **Apply Manually** again, system access will be provided but the above screen will re-appear at every login.
- If you click **Apply Automatically**, the system default settings will be changed and system access will be provided. Before you click this options make sure you have checked the new default parameters. For details, refer to “Modified default parameter values for Firmwares later than V2.2”.

Refer to “Modified default parameter values for Firmwares later than V2.2”, to know the changes in default configurations

Command Logs

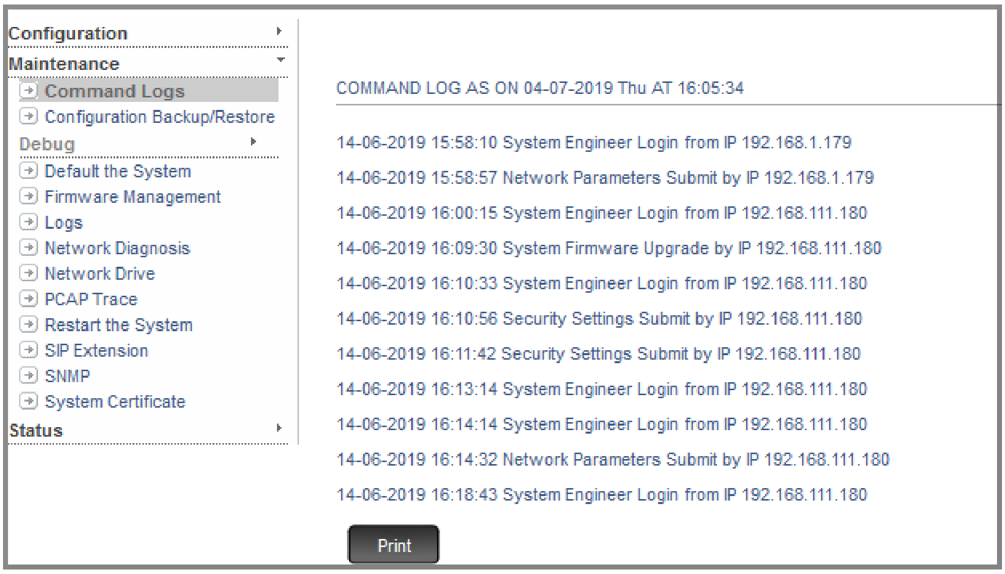
ANANT UCS monitors the important configuration changes and maintains records of these in the Command Logs. This is useful when there are multiple users accessing the system. These logs provide the details of the events along with the IP Address.

A maximum of 999 entries can be logged. Thereafter the entries will be logged on First-in First-out (FIFO) basis.

This log can be printed on a local printer. For the detailed list of events that are logged, refer to “Command Logs List”.

Viewing Command Log

- Login as System Engineer.
- Under **Maintenance**, click **Command Logs**.



Command Logs List

Pages	Events that are logged in the Command Logs
Abbreviated Dialing - Personal and Global Directory	06-09-2017 15:51:05 Abbreviated Dialing Submit by IP 192.168.101.192 06-09-2017 15:51:09 Abbreviated Dialing Default by IP 192.168.101.192 06-09-2017 15:51:19 Abbreviated Dialing Download by IP 192.168.101.192 06-09-2017 15:58:31 Abbreviated Dialing Upload by IP 192.168.101.192
Access Codes	06-09-2017 16:00:59 Access Codes Submit by IP 192.168.101.192 06-09-2017 16:01:54 Access Codes Default by IP 192.168.101.192
Change FTP PW for Extended IP Phones	06-09-2017 16:03:59 FTP Password for Extended IP Phones Submit by IP 192.168.101.192
Change SA Password (For both Phone and Web)	06-09-2017 16:04:51 SA Password Submit by IP 192.168.101.192
Change SE Password (For both Phone and Web)	06-09-2017 16:04:51 SE Password Submit by IP 192.168.101.192
CLI Based Routing	06-09-2017 16:06:10 CLI Based Routing Submit by IP 192.168.101.192 06-09-2017 16:06:20 CLI Based Routing Default by IP 192.168.101.192
Class of Service	06-09-2017 16:07:02 Class of Service Submit by IP 192.168.101.192 06-09-2017 16:07:07 Class of Service Default by IP 192.168.101.192
Date & Time - Date & Time, Daylight Saving SNTP	06-09-2017 16:58:21 Date & Time Submit by IP 192.168.101.192 06-09-2017 16:58:38 Date & Time, SNTP Submit by IP 192.168.101.192 06-09-2017 17:00:31 Date & Time, SNTP Default by IP 192.168.101.192
DDI Routing - DDI Routing Table, Incoming and Outgoing Reference ID table	06-09-2017 17:00:59 DDI Routing Submit by IP 192.168.101.192 06-09-2017 17:01:05 DDI Routing Default by IP 192.168.101.192
Default the System	29-08-2017 15:03:02 System Default from IP 192.168.101.108
Emergency Number	06-09-2017 17:11:30 Emergency Number Submit by IP 192.168.101.192 06-09-2017 17:11:34 Emergency Number Default by IP 192.168.101.192
System Firmware	06-09-2017 17:11:34 System Firmware Upgrade by IP 192.168.101.192
Least Cost Routng (LCR)	30-08-2017 10:10:54 Least Cost Routing Default by IP 192.168.101.72 30-08-2017 10:11:09 Least Cost Routing Submit by IP 192.168.101.72
Licence Management	06-09-2017 17:22:44 License Management Start by IP 192.168.101.192 06-09-2017 17:23:10 License Management Stop by IP 192.168.101.192
Logical Partition	06-09-2017 17:23:44 Logical Partition Submit by IP 192.168.101.192 06-09-2017 17:23:49 Logical Partition Default by IP 192.168.101.192
Network Port - Network Port Parameters, Static Routing, Reiniliatize button on Status page	06-09-2017 17:24:19 Network Parameters Submit by IP 192.168.101.192 06-09-2017 17:24:26 Network Parameters Default by IP 192.168.101.192 06-09-2017 17:58:05 IPv4 Network Reinitialize by IP 192.168.101.192 06-09-2017 17:58:05 IPv6 Network Reinitialize by IP 192.168.101.192
OG Trunk Bundle	06-09-2017 17:25:06 OG Trunk Bundle Submit by IP 192.168.101.192 06-09-2017 17:25:11 OG Trunk Bundle Default by IP 192.168.101.192
OG Trunk bundle Group	06-09-2017 17:25:39 OG Trunk Bundle Group Submit by IP 192.168.101.192 06-09-2017 17:25:44 OG Trunk Bundle Group Default by IP 192.168.101.192
Operator	06-09-2017 17:26:10 Operator Submit by IP 192.168.101.192 06-09-2017 17:26:18 Operator Default by IP 192.168.101.192

Routing Group	06-09-2017 17:26:40 Routing Group Submit by IP 192.168.101.192 06-09-2017 17:26:45 Routing Group Default by IP 192.168.101.192
Security Settings	06-09-2017 17:27:07 Security Settings Submit by IP 192.168.101.192 06-09-2017 17:27:23 Security Settings Default by IP 192.168.101.192
SMDR - SMDR Reports, Filters, Posting, Online	06-09-2017 17:39:01 Station Message Detail Recording Submit by IP 192.168.101.192 06-09-2017 17:39:08 Station Message Detail Recording Default by IP 192.168.101.192
System Activity Log (SAL)	06-09-2017 17:47:23 System Activity Log Submit by IP 192.168.101.192 06-09-2017 17:47:30 System Activity Log Default by IP 192.168.101.192
System Fault Log (SFL)	06-09-2017 17:47:34 System Fault Log Submit by IP 192.168.101.192 06-09-2017 17:47:41 System Fault Log Default by IP 192.168.101.192
System Parameters Custom Logo Upload and Delete	06-09-2017 17:44:26 System Parameters Submit by IP 192.168.101.192 06-09-2017 17:53:59 System Parameters, Custom Logo Upload by IP 192.168.101.192 06-09-2017 17:54:05 System Parameters, Custom Logo Remove by IP 192.168.101.192
System Pre-requisites	06-09-2017 17:55:10 System Pre-requisites Submit by IP 192.168.101.192
System Timers and Counters	06-09-2017 17:55:41 System Timers and Counters Submit by IP 192.168.101.192 06-09-2017 17:55:45 System Timers and Counters Default by IP 192.168.101.192
VoIP Configuration- VoIP Parameters, SIP Extension, SIP Trunk, Voice Mail Settings	06-09-2017 17:59:37 VoIP Configuration Submit by IP 192.168.101.192 06-09-2017 17:59:43 VoIP Configuration Default by IP 192.168.101.192
Ssytem Restart	29-08-2017 16:03:14 System Restart from IP 192.168.101.108
Login-Logout	30-08-2017 10:08:42 System Engineer Logout from IP 192.168.101.72 30-08-2017 10:08:44 System Engineer Login from IP 192.168.101.72 31-08-2017 19:16:01 System Engineer Auto Logout from IP 192.168.101.240

Logs

ANANT UCS monitors the boot-up process activities and maintains records of these in the Logs. This is useful source of information for troubleshooting.

You can download these files and use the same for analysis.

Viewing Logs

- Login as System Engineer.
- Under **Maintenance**, click **Logs**.

- Click on the desired file to download it

Default Settings

ANANT UCS is supplied with preset values for system and feature settings, as which may be altered and customized by users to match their requirements and preferences. The factory-set values for system and feature settings that are automatically assigned by the system are referred to as Default Settings or standard settings.

Every configurable parameter in the system has factory-set default values, which may be changed or customized to match user requirements and preferences.

If you have purchased / upgrading the system with Firmware Version later than V2.2, due to security concerns the default values of certain parameters have been changed. For new systems purchased these Default Settings will the applicable automatically. For details refer to “Modified default parameter values for Firmwares later than V2.2”.

If you are upgrading the system, refer to “After updating Firmware later than V2.2”.

How it works

The default settings are to be loaded or restored in the following situations:

1. ***Installing the ANANT UCS in a country other than India.***

   ANANT UCS provides default settings to match country/region-specific requirements of users worldwide.

   The default settings are set to match user requirements of India.

   So, you must select the appropriate “Configuring Region” for the country/region in which the system is installed.

   The system will load the default settings for the country/geographical region where the system is installed.

   The system is designed to work efficiently with the default settings. So, if the country/region-specific default settings match their requirements, you may not even need to alter or customize the values of various parameters.

   They may work with default settings for the most part, customizing only some of the parameters to match their specific requirements.

   The country-specific default settings of various parameters that will be loaded on changing the 'Region' are presented in the table below. For default values of Trunk Access Codes, Configuring Emergency Number Dialing, Distinctive Rings, for various countries refer the respective topics.

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Phone Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
001	Afghanistan	GMT+04:30				English				
002	Algeria	GMT+01:00				English				
003	Antigua and Barbuda	GMT-04:00				English				
004	Argentina	GMT-03:00			04	Spanish				
005	Australia (Perth)	GMT+08:00	Scheduled	2	05	English		9	0	

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Phone Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
006	Australia (Note2) (Adelaide)	GMT+09:30	Scheduled	2	05	English		9	0	
007	Australia (Brisbane, Canberra, Melbourne, Sydney)	GMT+10:00			05	English		9	0	
008	Austria	GMT+01:00	Scheduled	1		German		9	0	
009	Bahamas	GMT-05:00				English		9	0	
010	Bahrain	GMT+03:00	Scheduled	3		English		9	0	
011	Bangladesh	GMT+06:00				English				
012	Belarus	GMT+02:00				English				
013	Belgium	GMT+01:00	Scheduled	2		French		0	0	
014	Bhutan	GMT+06:00				English				
015	Bolivia	GMT-04:00				Spanish				
016	Bosnia and Herzegovina	GMT+01:00				English				
017	Botswana	GMT+02:00				English				
018	Brunei	GMT+08:00				English				
019	Brazil (Fernando De Noronha)	GMT-02:00			06	Portuguese		0	9	
020	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	GMT-03:00	Scheduled	4	06	Portuguese		0	9	
021	Brazil (Manaus)	GMT-04:00			06	Portuguese		0	9	
022	Brazil (Acre)	GMT-05:00			06	Portuguese		0	9	
023	Bulgaria	GMT+02:00				English				
024	Cambodia	GMT+07:00				English				
025	Cameroon	GMT+01:00				English				
026	Canada (St. John's)	GMT-03:30	Scheduled	5	03	English	T3	0	9	6
027	Canada (Halifax)	GMT-04:00	Scheduled	5	03	English	T3	0	9	6
028	Canada (Montreal, Ottawa, Toronto)	GMT-05:00	Scheduled	5	03	English	T3	0	9	6
029	Canada (Winnipeg)	GMT-06:00	Scheduled	5	03	English	T3	0	9	6
030	Canada (Calgary)	GMT-07:00	Scheduled	5	03	English	T3	0	9	6
031	Canada (Vancouver)	GMT-08:00	Scheduled	5	03	English	T3	0	9	6
032	Chile	GMT-04:00	Scheduled	6		Spanish		0	9	
033	China	GMT+08:00			08	English		0	9	
034	Colombia	GMT-05:00				Spanish				
035	Costa Rica	GMT-06:00				Spanish				
036	Croatia	GMT+01:00				English				
037	Cuba	GMT-05:00	Scheduled	18		Spanish		0	9	
038	Cyprus	GMT+02:00				English				
039	Czech Republic	GMT+01:00				English				
040	Denmark	GMT+01:00	Scheduled	7		English		0	9	
041	Egypt	GMT+02:00	Scheduled	11	09	English		9	0	

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Phone Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
042	Fiji	GMT+12:00				English				
043	Finland	GMT+02:00	Scheduled	8		English		9	0	
044	France	GMT+01:00	Scheduled	2	10	French		9	0	
045	Germany	GMT+01:00	Scheduled	2	11	German		9	0	
046	Greece	GMT+02:00	Scheduled	2	12	English		9	0	
047	Guyana	GMT-04:00				English				
048	Hong Kong	GMT+08:00				English		9	0	
049	Hungary	GMT+02:00	Scheduled	2		English		9	0	
050	India	GMT+05:30			01	English	T1	9	0	8
051	Indonesia	GMT+07:00			14	English		0	0	
052	Iran	GMT+03:30			15	English		9	0	
053	Iraq	GMT+03:00	Scheduled	9	16	English		9	0	
054	Ireland	GMT	Scheduled	7		English		0	9	
055	Israel	GMT+02:00			17	English				
056	Italy	GMT+01:00	Scheduled	2	18	Italian		9	0	
057	Japan	GMT+09:00			19	English				
058	Jordan	GMT+02:00				English	T1	0	9	
059	Kazakhstan	GMT+06:00				English				
060	Kenya	GMT+03:00			20	English				
061	Korea - North	GMT+09:00			21	English				
062	Korea - South	GMT+09:00				English				
063	Kuwait	GMT+03:00				English				
064	Kyrgyzstan	GMT+06:00	Scheduled	10		English		9	0	
065	Lebanon	GMT+02:00	Scheduled	12		English		9	0	
066	Libya	GMT+02:00				English				
067	Malaysia (Note1)	GMT+08:00			22	English		0	9	6
068	Maldives	GMT+05:00				English				
069	Mauritius	GMT+04:00				English				
070	Mexico (Mexico City)	GMT-06:00	Scheduled	3	03	Spanish	T3	0	9	6
071	Mexico (Chihuahua)	GMT-07:00	Scheduled	3	03	Spanish	T3	0	9	6
072	Mexico (Tijuana)	GMT-08:00	Scheduled	3	03	Spanish	T3	0	9	6
073	Mongolia	GMT+08:00				English				
074	Mozambique	GMT+02:00				Portuguese				
075	Myanmar	GMT+06:30				English				
076	Namibia	GMT+01:00	Scheduled	13	03	English	T3	9	0	6
077	Nepal	GMT+05:45				English				
078	Netherlands	GMT+01:00				English				
079	New Zealand	GMT+12:00	Scheduled	14	24	English		0	1	
080	Nigeria	GMT+01:00				English				
081	Norway	GMT+01:00	Scheduled	15		English		9	0	
082	Oman	GMT+04:00				English				
083	Pakistan	GMT+05:00				English				
084	Paraguay	GMT-04:00	Scheduled	16		Spanish		9	0	
085	Peru	GMT-05:00				Spanish				
086	Philippines	GMT+08:00			25	English				
087	Poland	GMT+01:00	Scheduled	1	26	English		9	0	
088	Portugal	GMT	Scheduled	7	27	Portuguese		9	0	
089	Qatar	GMT+03:00				English				
090	Romania	GMT+02:00				English				

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Phone Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
091	Russia (Moscow, St. Petersburg)	GMT+03:00	Scheduled	1	28	English		9	0	
092	Russia (Novosibirsk)	GMT+06:00	Scheduled	1	28	English		9	0	
093	Russia (Vladivostok)	GMT+10:00	Scheduled	1	28	English		9	0	
094	Singapore	GMT+08:00			30	English		9	0	
095	Slovakia	GMT+01:00				English				
096	South Africa	GMT+02:00			31	English				
097	Spain	GMT+01:00	Scheduled	1	32	Spanish		9	0	
098	Sri Lanka	GMT+05:30				English				
099	Sudan	GMT+03:00				English				
100	Sweden	GMT+01:00	Scheduled	2		English		9	0	
101	Switzerland	GMT+01:00	Scheduled	2		German		9	0	
102	Syria	GMT+02:00	Scheduled	17		English		9	0	
103	Taiwan	GMT+08:00				English		0	0	
104	Tajikistan	GMT+05:00				English				
105	Thailand	GMT+07:00			33	English				
106	Turkey	GMT+02:00			34	English		9	0	
107	Uganda	GMT+03:00				English				
108	Ukraine	GMT+02:00				English				
109	United Arab Emirates	GMT+04:00			35	English		9	0	
110	United Kingdom	GMT	Scheduled	7	02	English	T2	0	9	8
111	United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	GMT-05:00	Scheduled	3	03	English	T3	0	9	6
112	United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	GMT-06:00	Scheduled	3	03	English	T3	0	9	6
113	United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	GMT-07:00	Scheduled	3	03	English	T3	0	9	6
114	United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	GMT-08:00	Scheduled	3	03	English	T3	0	9	6

Country Code	Country Name	Default Time Zone	Default DST Mode	Default DST Schedule Type	CPTG[a]	Default Phone Language	Distinctive Ring[b]	Opr	TAC	Abbr. Dialing
115	United States (Juneau)	GMT-09:00	Scheduled	3	03	English	T3	0	9	6
116	United States (Hawaii)	GMT-10:00			03	English	T3	0	9	6
117	Uzbekistan	GMT+05:00				English				
118	Venezuela	GMT-04:30				Spanish				
119	Vietnam	GMT+07:00				English				
120	Yemen	GMT+03:00				English				
121	Yugoslavia	GMT+02:00				English				
122	Zambia	GMT+02:00				English				
123	Zimbabwe	GMT+02:00				English				
124	Saudi Arabia	GMT +3.00				English	T1	9	0	
125	Cote d'Ivoire	GMT+01:00	Scheduled	2	10	French		9	0	
126	American Samoa	GMT-11:00			03	English	T3	0	1	8
127	Australia (Eucla)	GMT+08:45			05	English	T1	0	9	6
128	Australia (Lord Howe Island)	GMT+10:30			05	English	T1	0	9	6
129	Cape Verde (Cabo Verde)	GMT-01:00			03	English	T3	9	0	8
130	French Polynesia	GMT-09:30			10	French	T1	9	0	8
131	Kiribati	GMT+14:00			05	English	T1	9	0	8
132	New Zealand (Chatham Islands)	GMT+12:45			24	English	T1	0	1	8
133	Samoa	GMT+13:00			05	English	T1	9	0	8
134	Solomon Island	GMT+11:00			02	English	T2	9	0	8
135	US Minor Outlying Islands(Baker Island, Howland Island)	GMT-12:00			03	English	T3	0	9	6

a. See “Call Progress Tones” for more information on CPTG.
b. See “Distinctive Rings” for more information on Ring Type and Cadence.

2. ***Installing ANANT UCS in a Hospitality Application.***

The two main applications of ANANT UCS are:

- Enterprise application to meet the communication requirements of businesses.
- Hospitality application to meet the specific requirements of Hotels and Hospitals.

In addition to the common set of System features, there is a distinct set of features for each of these applications. When ANANT UCS is to be installed in any of the two application scenarios, the 'Customer Profile' - whether the user is an Enterprise or a Hotel - is to be defined at the time of installation.

When the Customer Profile is defined, all features specific to the application Enterprise/Hotel, along with their default settings are loaded. By default, the Customer Profile of ANANT UCS is defined as 'Enterprise'.

Refer *ANANT UCS Hospitality System Manual* to know more.

3. ***Malfunctioning of the System.***

When there is a system malfunction, possibly caused by a programming error that you are unable to diagnose, you may restore default settings.

Restoring Default Settings

Default settings can be loaded or restored from Jeeves. To be able to do this, you must have the “SE password”.

Whenever you restore the default settings in the system, all the configurable parameters except Network Port Parameters[129] *and the “Configuring Region” will be set back to their default values.*

All parameters will be assigned default values except the following, when the system is set to default:

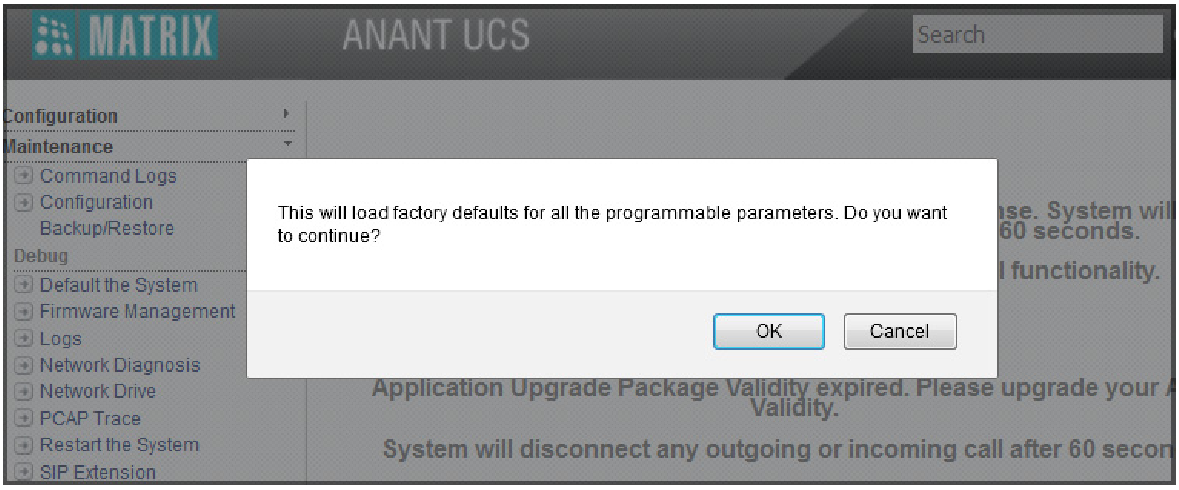
- System Password
- Region (Regional setting should get default but not region)
- Date Format
- RTC Parameters
- OG SMDR Records
- IC SMDR Records
- INT SMDR Records
- Custom Logo
- Network Drive backup
- IP Phone FTP Password
- VMS Configuration[130]
- System Activity Log
- System Fault Log
- Network Configuration
- DynDNS Configuration
- SNTP Server
- Call Logs
- VMS Access Codes

To Restoring Default Settings,

- Login as System Engineer.
- Under **Maintenance**, click **Default the System**.

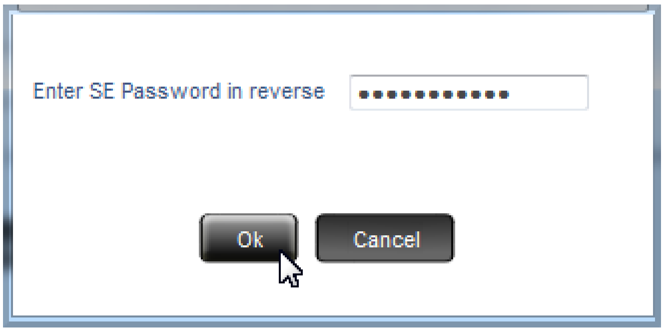
129. The IP Address, Subnet Mask, Primary DNS, Secondary DNS, Host Name, Domain Name, DHCP Server Address.
130. Notification via Call parameters, VMAA Access codes are set to default.

- You will be prompted whether you want to default the system.



- Click **OK**.

- Enter Reverse SE Password on the prompt.



- The SE password you enter must be the current password. For Example: if it is 1234, enter 4321 and click OK.

- The system starts loading the default settings.

- You are returned to the Login page of Jeeves.

Modified default parameter values for Firmwares later than V2.2

Page Name	Feature/Parameter	Old Default Value	New Default Value	Impact on behavior after update
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-0 (WH)	All Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-0 (BH)	All Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-0 (NH)	All Calls	No Calls	Extension Users will not be able to make external calls.

Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-1	Local Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-2	National Calls	No Calls	Extension Users will not be able to make external calls.
Station Basic Feature Template [1-50]	Call Privilege ->Toll Control Level-3	No Calls	No Calls	Extension Users will not be able to make external calls.
Class of Service [1-20]	Closed User Group (CUG)	Enable	Disable	Extension Users will not be able to access this feature by default.
Class of Service [1-20]	Global Directory Part-1	Enable	Disable	Extension Users will not be able to access this feature by default.
Class of Service [1-20]	Trunk-Trunk Transfer	Enable	Disable	Extension Users will not be able to access this feature by default.
OG Trunk Bundle Groups	OG Trunk Bundle Members	1,2,3,4	0	Extension Users will not be able to make eternal calls from their extensions.
OG Trunk Bundle	Trunk Port -Type	SIP Trunk	None	Extension Users will not be able to make eternal calls from their extensions.
SIP Trunk Parameter	Accept anonymous calls?	Enable	Disable	Anonymous Calls will not be accepted.
Call Duration Control for Table-1	Apply CDC for incoming calls received from trunk	Disable	Enable	CDC will be applied on All trunks and Extensions for external calls.
Call Duration Control for Table-1	Apply CDC for outgoing calls made from trunk	Disable	Enable	CDC will be applied on All trunks and Extensions for external calls.
Call Duration Control for Table-1	CDC Timer (sec)	160	300	CDC will be applied on All trunks and Extensions for external calls.
Call Duration Control for Table-1	Disconnect Call after CDC Timer	Disable	Enable	CDC will be applied on All trunks and Extensions for external calls.
Call budget	For All Trunk	None	Minutes (300 minutes)	Budget will be applied on all the Trunks.

PCAP Trace

PCAP or packet capture consists of intercepting and logging the traffic passing over a network. PCAP intercepts each packet in the data streams that flow across the network, and can decode and analyze its contents.

PCAP can be used, among others, to monitor the network, detect and analyze network problems, debug client/ server communications, debug network protocol implementations.

ANANT UCS supports PCAP Trace for the LAN Port, WAN Port, Internal Interface. PCAP Trace is supported for both IPv4 and IPv6 addresses.

The PCAP Trace of the Extended IP Phones can be accessed using the FTP of the phone. The access to the FTP is secured by a password and it can be changed. For detailed instructions, see “FTP Access for Extended IP Phones”.

Packets traveling over a network are captured and saved locally in the system or at a remote location. You can save these trace files (packets captured by the system) on a PC and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

ANANT UCS also supports Filters and 'Promiscuous' mode for capturing packets, which you can use to specify the types of data packets to be captured.

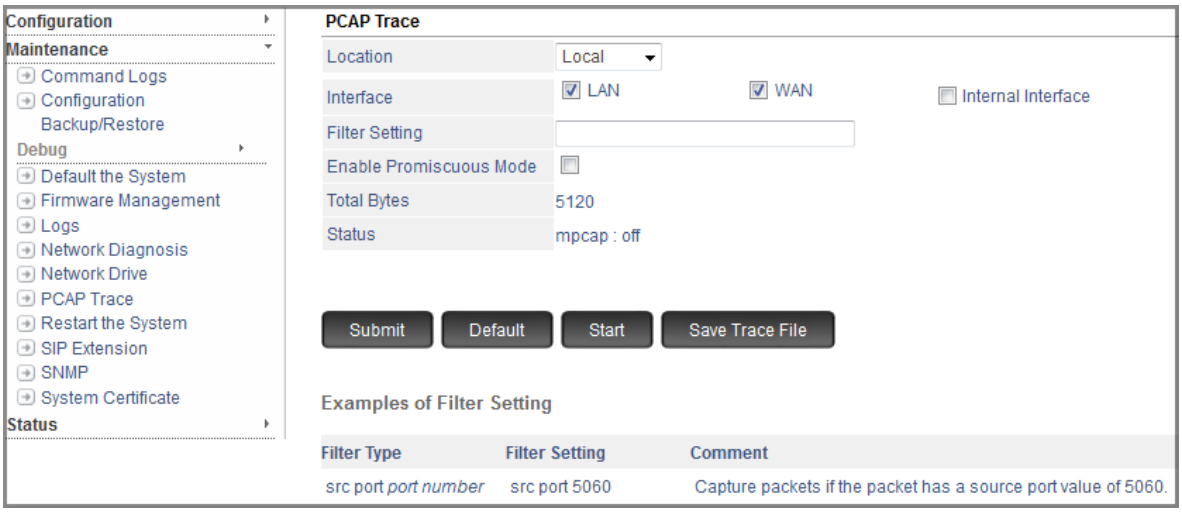
How to use

Using PCAP Trace of ANANT UCS

When the PCAP Trace data are stored locally a maximum of 50 MB of packets can be captured and stored in ANANT UCS.

To use PCAP Trace of ANANT UCS,

- Login as System Engineer.
- Under **Maintenance**, click **PCAP Trace**.



- **Location:** Select the location — Local or Remote — on which you want to store the PCAP data.

  If you select **Local** as the Location:

  - **Interface:** Select the interfaces for which you want to capture the packets — LAN, WAN, Internal. Select the check box of the desired interfaces. By default, LAN and WAN options are enabled.

  - **Filter Settings**: Decide the type of packets to be captured and set the Filter accordingly. The Filter Settings parameter should be maximum 60 characters in length; all ASCII characters are allowed. By default, this field is blank. So all packets will be captured.

    Refer the following examples to know how to set the Filters.

    Examples of Filter settings:

    To capture only SNMP traces:
    - Filter Settings = port 161
      Where, 161 is the SNMP Port number for which the traces are to be captured.

    To capture packets which are transmitted from the system, *from* IP address 192.168.1.191:
    - Filter Settings = src 192.168.1.191

    To capture packets which are received for the system, *to* IP address 192.168.1.191:
    - Filter Settings = dst 192.168.1.191

    To capture only packets which are transmitted from the system and received to the system, IP address 192.168.1.191:
    - Filter Settings = src 192.168.1.191 or dst 192.168.1.191

    To capture packets which are transmitted from the system for particular port number only, *from* IP address 192.168.1.191 and port number 161
    - Filter Settings = src 192.168.1.191 and port 161

    It is not mandatory to set Filters. When the Filter Settings field is left blank, the system will capture all packets.

  - **Enable Promiscuous Mode:** Select the **Enable Promiscuous Mode** check box, if required.

    When you enable Promiscuous mode, ANANT UCS will capture all network traffic. However, this will work only in a non-switched environment.

    When Promiscuous Mode check box is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through ANANT UCS will be picked up by the PCAP Trace.

'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

  - Click the **Start** button to begin the capturing of the packets.

  - Click the **Stop** button to stop packet capture.
    OR
    Wait for the system to stop packet capturing. The system stops packet capturing once the maximum allotted memory of 50 MB (RAM) is utilized.

- **Total Bytes:** Bytes captured as per the filter setting will be displayed as **Total Bytes**.

Capturing of packets will not stop if you open any other page of Jeeves. So, you may continue using Jeeves for any other purpose while PCAP Trace is being used.

- When the packet capturing is stopped (by you or the system), click the **Save Trace File** button to save the files on the PC.

The current packets captured will not be deleted after you have saved the trace file. The current packets will be deleted when you start the PCAP capture again.

- Now, you can open the downloaded trace file using Wireshark or Ethereal or any other similar software which supports opening of trace files.

If you select **Remote** as the Location:

Configure the following:

- **Listening Interface:** Select WAN or LAN as per your requirement.
- **Listening Port:** Enter the port number using which you want ANANT UCS to send the packets to the remote device. Make sure this port number is also configured in the remote device.
- **Remote Device IP Address:** This is IP address of the Remote Device on which ANANT UCS will send the PCAP data.

  The PCAP packets will be captured on the remote device as per the filters set in Wireshark or Ethereal or any other similar software at the remote end.
- Click the **Start** button to begin the capturing of the packets.
- Click the **Stop** button to stop packet capture.

- *If you are not changing any configuration related to the remote location, we recommend you not to press the Start-Stop-Start buttons frequently within a 1 minute, as it may lead to improper functioning.*
- *If remote capture is started from multiple locations concurrently, it can lead to data loss in PCAP capture. We recommend Remote PCAP from one location only.*

Using PCAP Trace for Matrix SPARSH VP248 Extended IP Phone

PCAP Trace supported by the Matrix Extended IP Phone can capture up to 1 MB of packets and store them in the embedded FTP server of the phone. The access to the FTP is secured by a password, for details see "FTP Access for Extended IP Phones".

You are not required to set any filters; the phone captures all packets that it sends and receives.

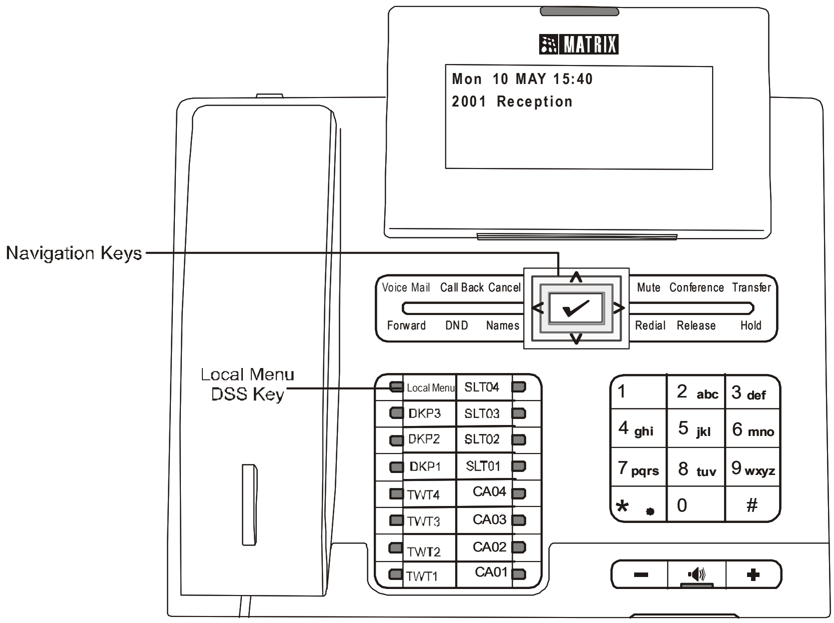
To be able to use PCAP Trace for a Matrix Extended IP Phone, there must be a DSS Key for accessing the Local Menu on the phone. If you have not already configured the DSS Key for Local Menu, you may do so now.

For instructions on configuring DSS keys of the Extended IP Phone, see "Configuring Matrix SPARSH VP248".

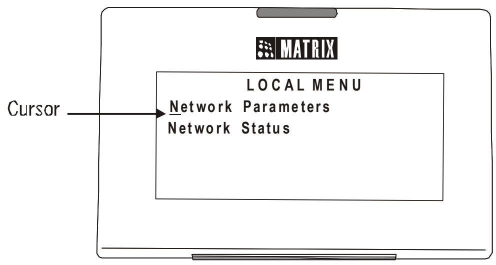
To use PCAP Trace from an Extended IP Phone extension,

- Press the DSS Key assigned to ‘Local Menu’.

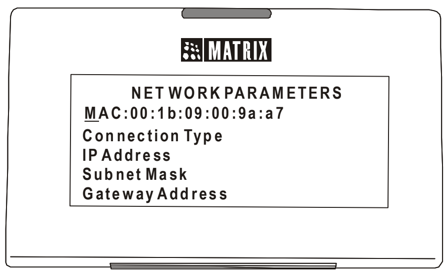
  You can access the Local Menu only when your phone is in idle state.



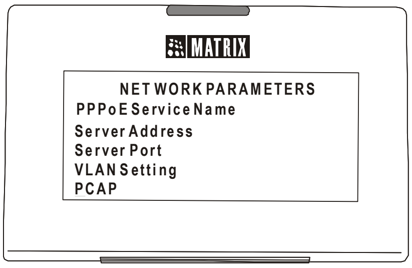
- The Local Menu appears on your phone display.



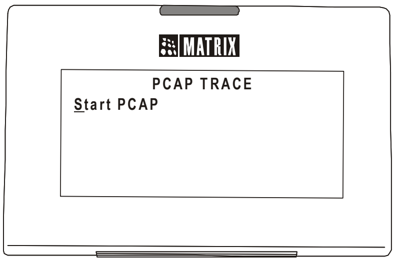
- The cursor appears under ‘Network Parameters’.
- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.



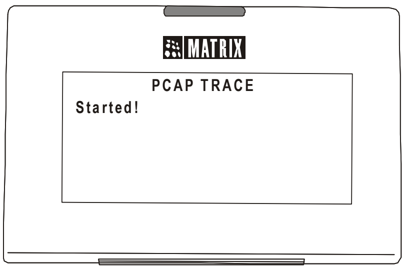
- Scroll with the DOWN navigation key to PCAP.



- Press Enter key.
- 'Start PCAP' appears on your phone display.

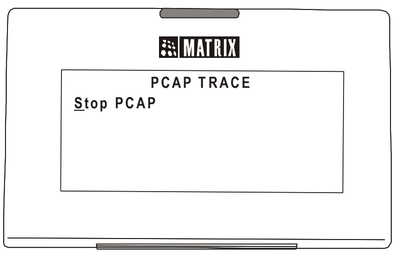


- Press Enter key. You get a confirmation message 'Started!'

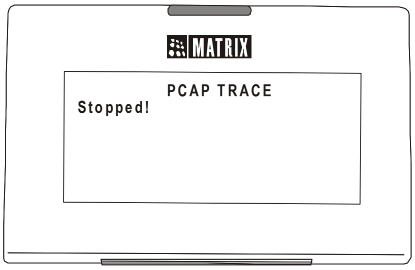


To stop PCAP,

- Enter the Local Menu of the phone again by pressing the DSS key.
- Enter the Network Parameters submenu.
- Select PCAP and press Enter.
- The message Stop PCAP appears on your phone display.



- Press Enter key. You get the confirmation message: ‘Stopped!’



- Go idle.

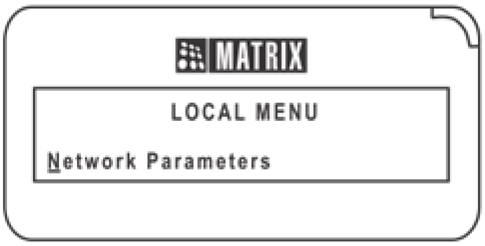
Using PCAP Trace for Matrix SPARSH VP310 Matrix Extended IP Phone

PCAP Trace supported by the Matrix Extended IP Phone can capture up to 1 MB of packets and store them in the embedded FTP server of the phone.

You are not required to set any filters; the phone captures all packets that it sends and receives.

To use PCAP Trace,

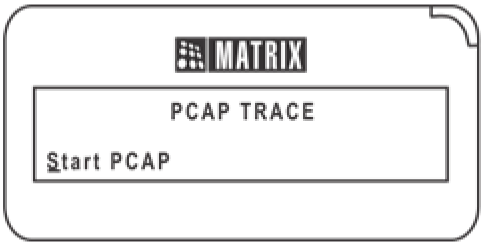
- When the phone is in idle state, press the DOWN key V to access the Network Settings.
- The cursor appears under ‘Network Parameters’.



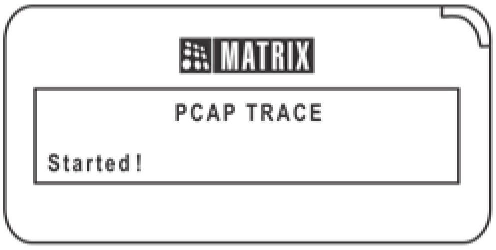
- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.
- Scroll with the DOWN navigation key to PCAP.

- Press Enter key.

- ‘Start PCAP’ appears on your phone display.

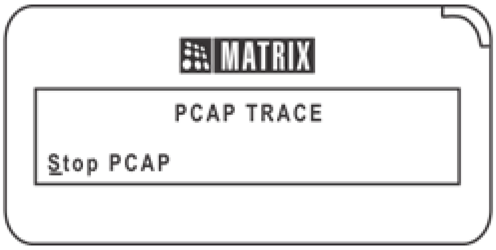


- Press Enter key. You get a confirmation message ‘Started!’



To stop PCAP,

- Press the DOWN key V to access the Network Settings.
- The cursor appears under ‘Network Parameters’.
- Press Enter key to select ‘Network Parameters’.
- The Network Parameters submenu appears.
- Scroll with the DOWN navigation key to PCAP and press Enter key.
- The message Stop PCAP appears on your phone display.



- Press Enter key. You get the confirmation message: ‘Stopped!’

- Go idle.

You can download the Trace file from the embedded FTP server of the Extended IP Phone. To access the FTP server using Windows FTP, do the following:

- Go to **My Computer**.
- Type the current IP Address of the Extended IP Phone in the Address bar. For example: **ftp://192.168.201.134**
- Click **Go** or Press Enter key on your keyboard**.**
- The **Log on as** window of the FTP server opens.
- In **User Name**, type **se** (lower case).
- In **Password**, enter the FTP Password for the phone.
- Click **Log on**.
- On successful login, the FTP window will open. You will see the different Configuration folders in this window.
- Click the folder **ramdisk.**
- In the **ramdisk** folder, right click the file **trace.pcap** and copy it on to your local disk.
- Open the **trace.pcap file** using Wireshark or Ethereal or any other similar software which supports opening of trace files.

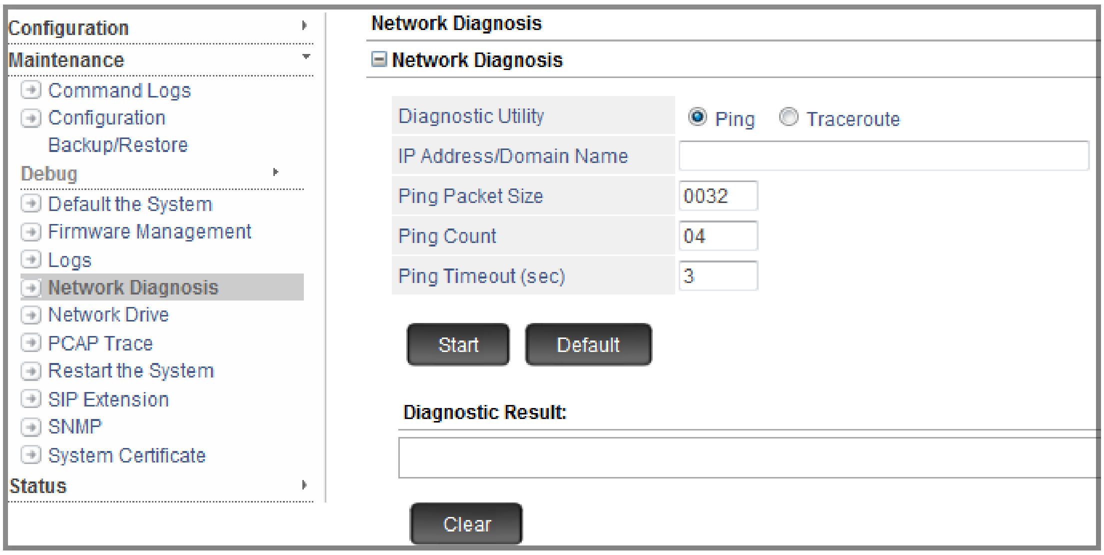
- *You may also use FireFTP, if you are using Mozilla Fire Fox. Make sure your browser has the **FireFTP** Add-on installed.*
- *To download the Trace file for SPARSH VP330, SPARSH VP210 and SPARSH VP510, refer to their respective User Guides.*

Network Diagnosis

ANANT UCS provides you an option to check the Internet/WAN connectivity using Ping and Traceroute as the diagnostic tools.

How to Configure

- Login as System Engineer.
- Under **Maintenance**, click **Network Diagnosis**.



- In **Diagnostic Utility**, select the diagnostic tool — Ping or Traceroute — to check the Internet/WAN connectivity.
- In **IP Address/Domain Name**, enter the IPV4 or IPV6 Address or the Domain Name of the system whose connectivity you wish to test. Default: Blank

  If you have selected *Ping* as the *Diagnostic Utility* option, configure the following parameters:

  - In **Ping Packet Size**, enter the number of bytes you want the system to send for Ping test. Valid Range: 4 to 1024. Default: 32 bytes.
  - In **Ping Count**, enter the number of times you want system to send the request message for Ping test. Valid Range: 1 to 50. Default: 4 times.
  - In **Ping Timeout (sec)**, enter the time for which you want the system to wait to get the response for each request message sent. Valid Range: 1 to 9. Default: 3 sec.

  If you have selected *Traceroute* as the *Diagnostic Utility* option, configure the following parameters:

  - In **Traceroute Max TTL**, enter the maximum number of hops (Time-To-Live value) you want the system to take in the path to find the IP Address configured. Valid Range: 1 to 255. Default: 30.

- In **Traceroute Protocol**, select the protocol — ICMP or UDP — which you want the system to use for traceroute functionality.

- To start the Network Diagnosis, click **Start** button.

  The Diagnostic result will appear on the screen.

- To clear the Diagnostic result, click **Clear** button.

To know about the Connectivity between the third party server and ANANT UCS system, refer to “Apple Push Notification Service Support” and “Firebase Cloud Messaging (FCM) Support”.

Network Drive Settings

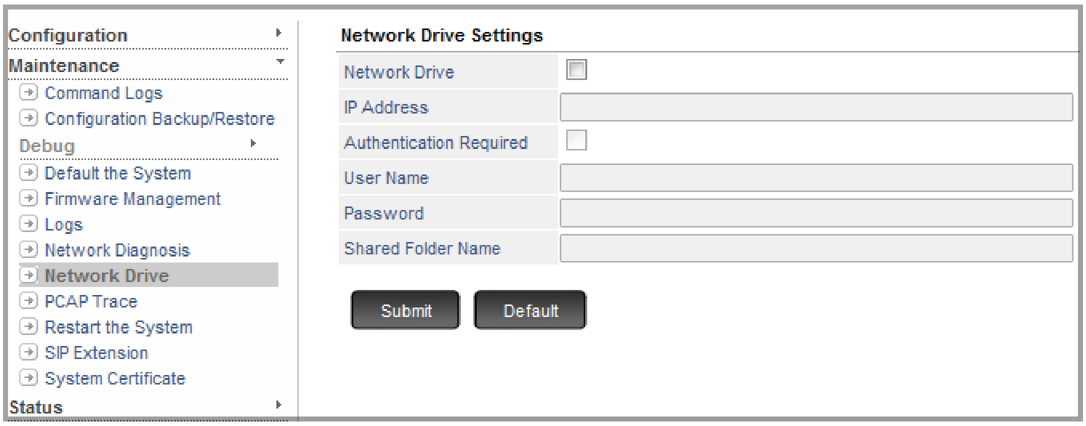
The Network Drive Settings allows you to test the connectivity between the system and the Network Drive where you wish to save the mailbox backup files. The authenticity to access the shared folder is also verified by the Network drive for security purpose.

For information regarding the voicemail backup, see "Voicemail Backup".

Make sure the System and the Network Drive are in the same Local Network.

How to Configure

- Login as System Engineer.
- Under **Maintenance**, click **Network Drive**.



- Select the **Network Drive** check box to enable the Network Drive.

  Clear the check box to disable the Network Drive.

  By default, it is disabled.

  - In **IP Address**, enter the IP Address of the Network Drive where you wish to store the mailbox backup files.
  - Select the **Authentication Required** check box to enable the authentication process. Enable this check box if the Network Drive folder requires authentication for accessibility.
    - In **User Name**, enter the authentication ID set for the Shared Folder of the Network Drive.
    - In **Password**, enter the authentication password set for the Shared Folder of the Network Drive.
  - In **Shared Folder Name**, enter the name of the Shared Folder where you wish to store the mailbox backup files.
- Click **Submit**.

  You may now test the connection between the system and the Network Drive. To do so,

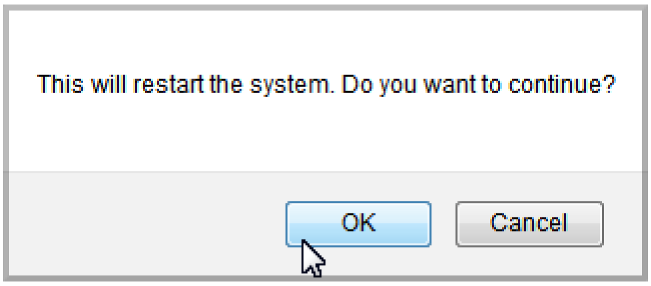
- Click **Test**.

  A message will be displayed showing whether the connection is successful or not. In case of an error, the system will display the error message.

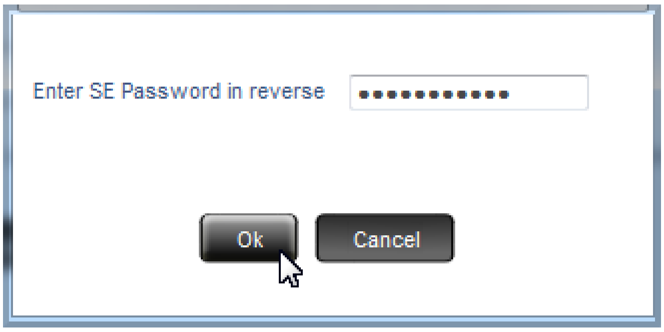
Restart the System

To restart the System,

- Login as System Engineer.
- Under **Maintenance**, click **Restart the System**.



- You will get the message, “This will restart the system. Do you want to continue?”.
- Click **OK**.
- Enter Reverse SE Password on the prompt.



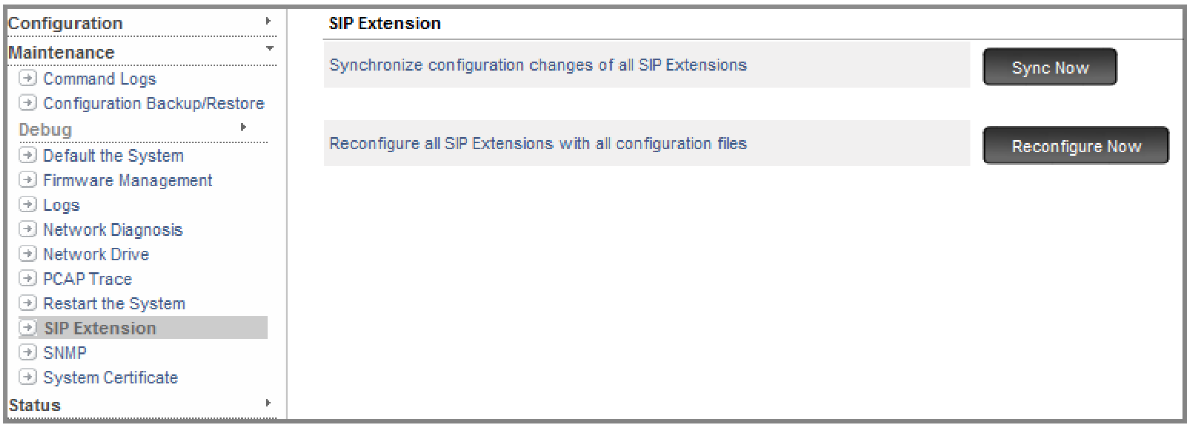
- The SE password you enter must be the current password. For Example: if it is Matrix@1234, enter 4321@xirtaM and click OK.
- The system will restart.

SIP Extension

ANANT UCS provides you an option to synchronize configuration changes of all SIP Extensions and reconfigure all SIP Extensions with all configuration files with a click of a button.

How to Configure

- Login as System Engineer.
- Under **Maintenance**, click **SIP Extension**.



- In **Synchronize configuration changes of all SIP Extensions**, click the **Sync Now** button to synchronize the configuration of Extended Clients, SPARSH VP110, SPARSH VP710 and Third Party IP-Phones.

  Only those configuration files in which there are changes will be synchronized.

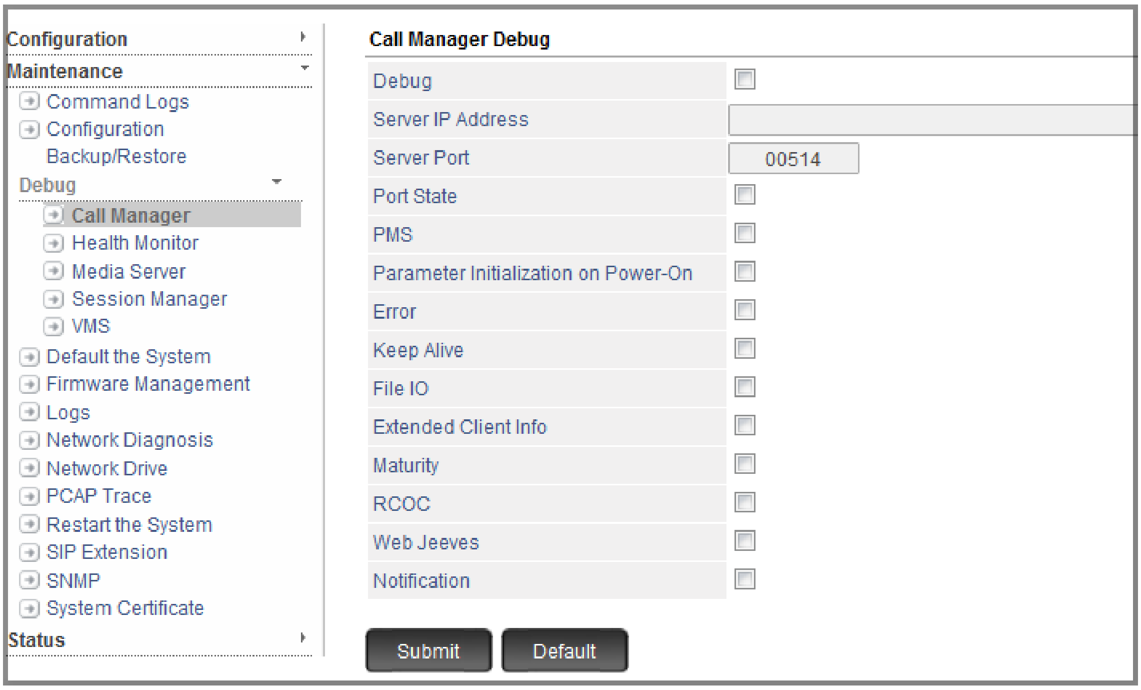
- In **Reconfigure all SIP Extensions with all configuration files**, click the **Reconfigure Now** button to reconfigure all the configuration files of Extended Clients, SPARSH VP110, SPARSH VP710 and Third Party IP-Phones.

Call Manager Debug

ANANT UCS supports a feature by which SE can view the Call Manager debugs of the server over IP network. This is done by 'Syslog Client' in the ANANT UCS which supports multiple debug levels.

Configuring Call Manager Debug

- Login as System Engineer.
- Under **Maintenance**, click **Debug**.
- Click **Call Manager**.



- Configure the following:
  - Select the **Debug** check box to enable.
  - Configure **Syslog Server IP Address** and **Port**.
    Both IPv4 and IPv6 addresses are supported. By default, the Port is 514. Valid range of the port is from 1025 to 65535 and 514.
  - Now, select the respective Debug Level check box to enable:
    - Port State
    - PMS
    - Parameter Initialization on Power-On
    - Error
    - Keep Alive
    - File IO
    - Extended Client Info
    - Maturity

- RCOC
- Web Jeeves
- Notification

Applications
- Media Server
- Session Manager
- VMS

Redundancy
- Redundancy Process
- Configuration File
- Communication Process
- Call
- Feature
- Phone
- RTC

As per the level selected, debug log will be generated.

- If required you can also enable the **Call Manager Layer**.

  - Select the **Debug** check box to enable.

  - Configure **Syslog Server IP Address** and **Port**.
    Both IPv4 and IPv6 addresses are supported. By default, the Port is 514. Valid range of the port is from 1025 to 65535 and 514.

- Click **Submit**.

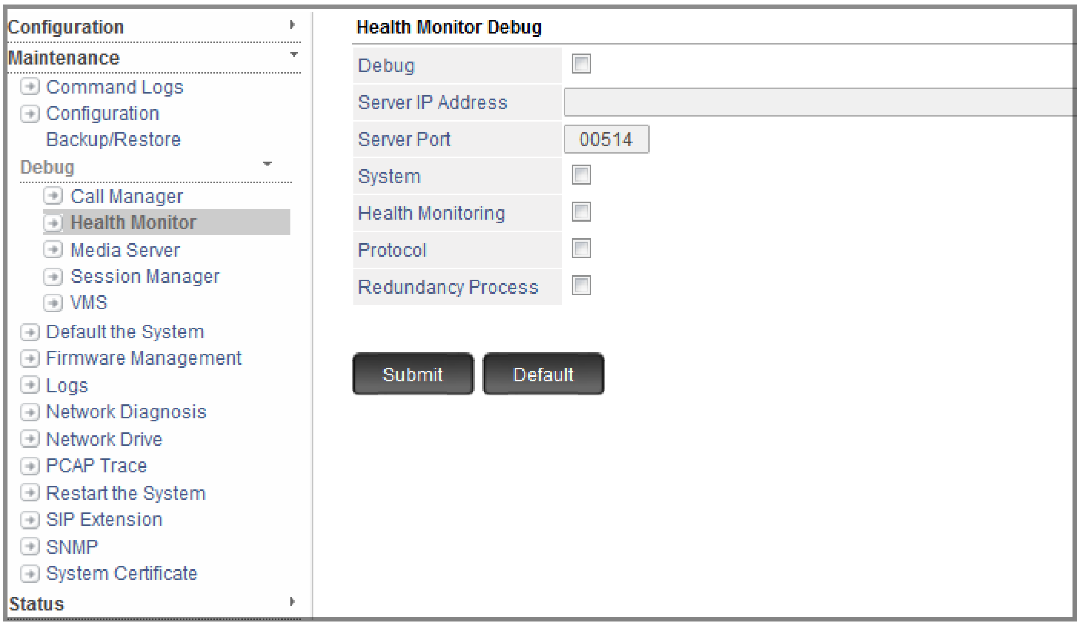
We recommend you to consult the Matrix support team before you enable debugs. These debugs lead to debug data streaming in huge quantities which may affect the system performance.

Health Monitor Debug

ANANT UCS supports a feature by which SE can view the Health Monitor debugs on the server over IP network. This is done by 'Syslog Client' in ANANT UCS which supports multiple debug levels.

Configuring Health Monitor Debug

- Login as System Engineer.
- Under **Maintenance**, click **Debug**.
- Click **Health Monitor**.



- Configure the following:
  - Select the **Debug** check box to enable.
  - Configure **Syslog Server IP Address** and **Port**.
    Both IPv4 and IPv6 addresses are supported. By default, the Port is 514. Valid range of the port is from 1025 to 65535 and 514.
  - Now, select the respective Debug Level check box to enable:
    - System
    - Health Monitoring
    - Protocol
    - Redundancy Process

    As per the level selected, debug log will be generated.
  - Click **Submit**.

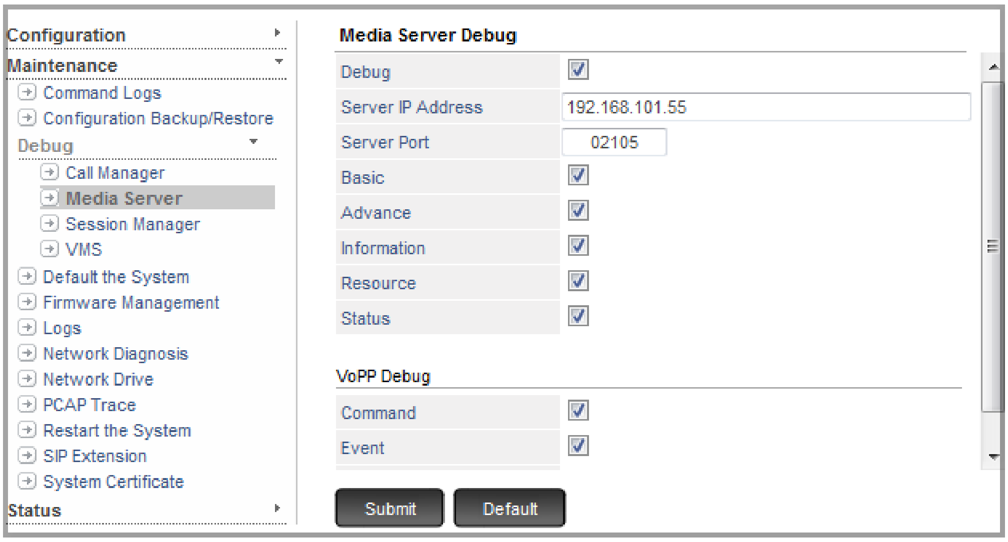
We recommend you to consult the Matrix support team before you enable debugs. These debugs lead to debug data streaming in huge quantities which may affect the system performance.

Media Server Debug

ANANT UCS supports a feature by which SE can view the Media Server debugs on the server over IP network. This is done by 'Syslog Client' in ANANT UCS which supports multiple debug levels.

Configuring Media Server Debug

- Login as System Engineer.
- Under **Maintenance**, click **Debug**.
- Click **Media Server**.



- Configure the following:
  - Select the **Debug** check box to enable.
  - Configure **Syslog Server IP Address** and **Port**.
    Both IPv4 and IPv6 addresses are supported. By default, the Port is 514. Valid range of the port is from 1025 to 65535 and 514.
  - Now, select the respective Debug Level check box to enable:
    - Basic
    - Advance
    - Information
    - Resource
    - Status

    VoPP
    - Command
    - Event
    - Advance

    As per the level selected, debug log will be generated.

- Click **Submit**.

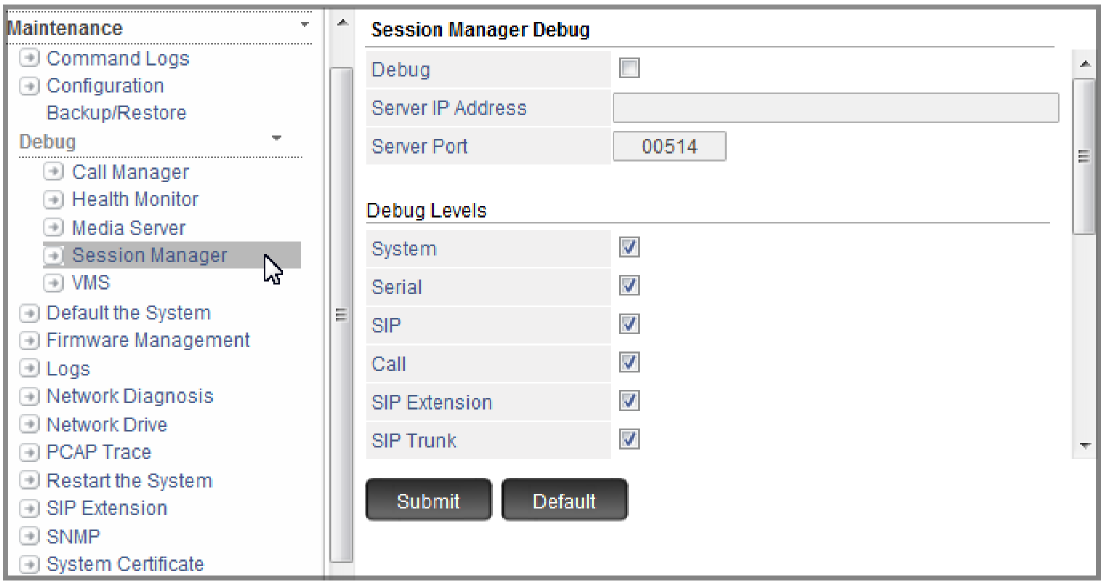
We recommend you to consult the Matrix support team before you enable debugs. These debugs lead to debug data streaming in huge quantities which may affect the system performance.

Session Manager Debug

ANANT UCS supports a feature by which SE can view the Session Manager debugs on the server over IP network. This is done by 'Syslog Client' in ANANT UCS which supports multiple debug levels.

Configuring Session Manager Debug

- Login as System Engineer.
- Under **Maintenance**, click **Debug**.
- Click **Session Manager**.



- Configure the following:
  - Select the **Debug** check box to enable.
  - Configure **Syslog Server IP Address** and **Port**.
    Both IPv4 and IPv6 addresses are supported. By default, the Port is 514. Valid range of the port is from 1025 to 65535 and 514.
  - Now, select the respective Debug Level check box to enable:
    - System
    - Serial
    - SIP
    - Call
    - SIP Extension
    - SIP Trunk
    - BLF/MWI
    - Presence
    - IM
    - Redundancy

Advance Debug Levels

- SIP
- Call
- SIP Extension
- SIP Trunk
- BLF/MWI

As per the level selected, debug log will be generated.

- Click **Submit**.

We recommend you to consult the Matrix support team before you enable debugs. These debugs lead to debug data streaming in huge quantities which may affect the system performance.

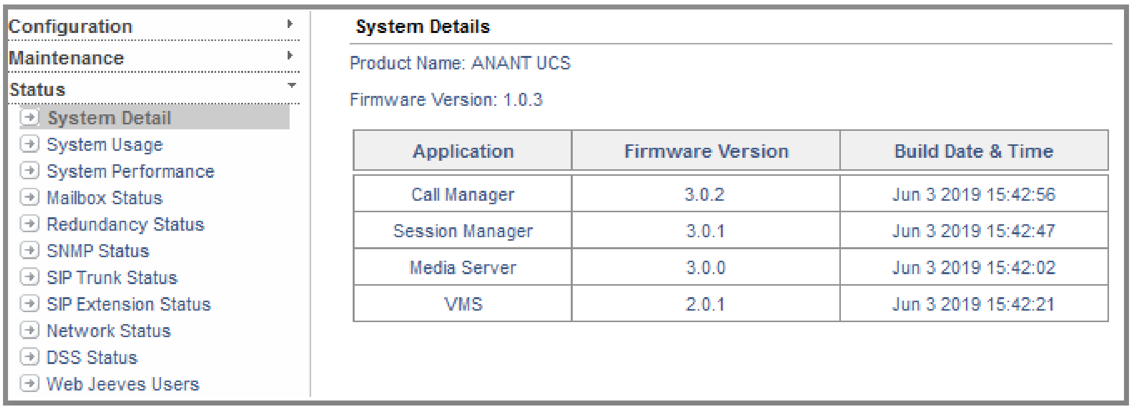
System Status

System Details

System Details displays the Product Name, Application Name along with the respective Firmware Version, Build Date and Time.

To view the System Details,

- Login as System Engineer.
- Under **Status**, click **System Details**.



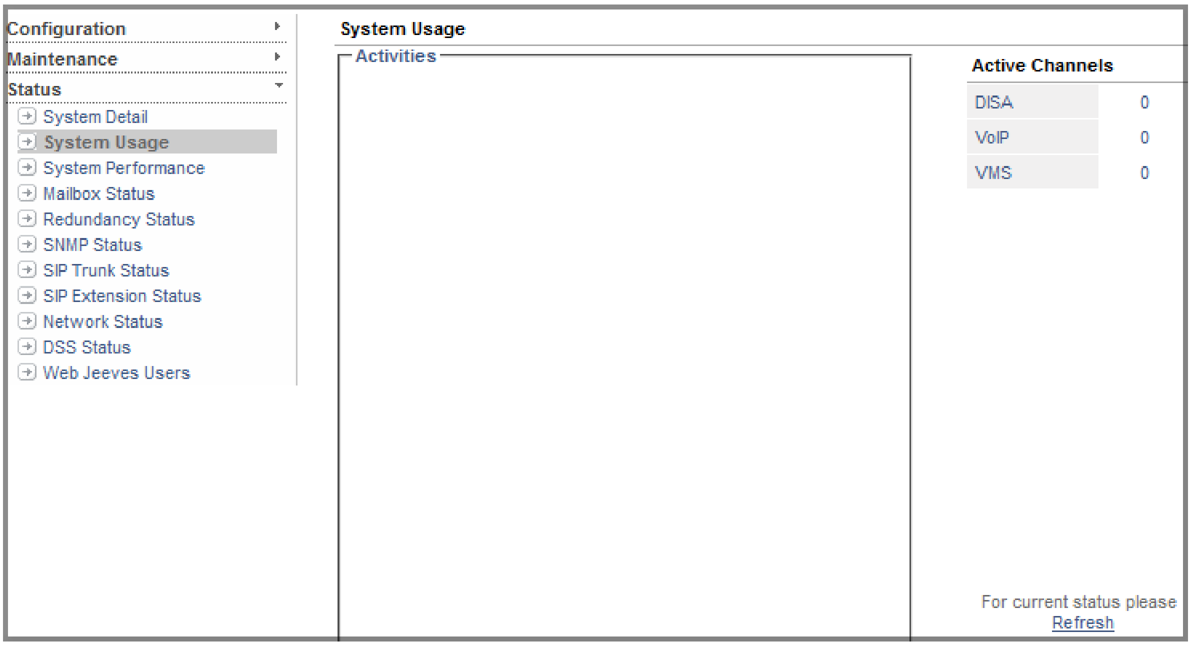
Application	Firmware Version	Build Date & Time
Call Manager	3.0.2	Jun 3 2019 15:42:56
Session Manager	3.0.1	Jun 3 2019 15:42:47
Media Server	3.0.0	Jun 3 2019 15:42:02
VMS	2.0.1	Jun 3 2019 15:42:21

System Usage

To view the active channels and their activities,

- Login as System Engineer.
- Under **Status**, click **System Usage**.

  The page displays the number of active channels. These channels may be busy via DISA, VoIP or VMS.



System Performance

ANANT UCS provides the facility to view the performance of the system online. You can check — the CPU usage, Memory (RAM) usage, speed and status of the Transmitted and Received data from the LAN port as well as the WAN port and Disk Storage. You can also view the system uptime, that is, the time for which the system has been functioning after the last restart.

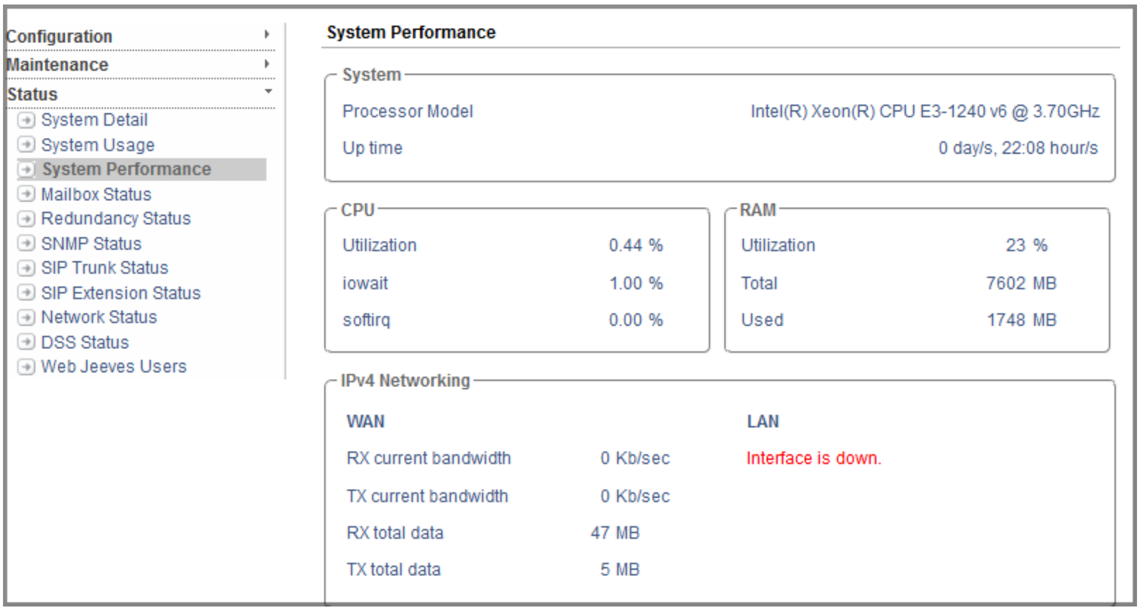
When the CPU and/or Memory (RAM) usage exceeds 80%, it is considered as high usage and the system sends notifications through Email, if configured to do so. These events are also logged in the System Activity Log.

Similarly, when the CPU and/or Memory (RAM) usage is back to normal the system sends notifications through Email, if configured to do so. These events are also logged in the System Activity Log.

For details, refer "System Log Notification" and "System Activity Log".

To view the System Performance,

- Login as System Engineer.
- Under **Status**, click **System Performance**.

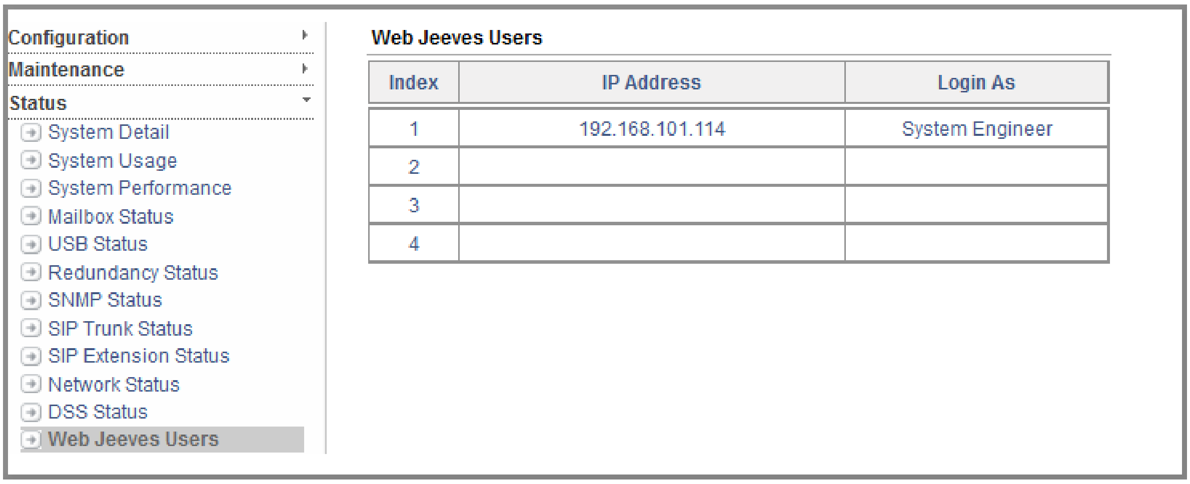


Web Jeeves Users

The Web Jeeves Users Page displays the IP Address and Login Mode of the users accessing the Jeeves. It also displays the type of protocol used for accessing Jeeves.

To view the status of Web Jeeves User,

- Login as System Engineer.
- Under **Status**, click **Web Jeeves Users**.



Appendix

Technical Specifications of Supported Terminals

Supported Terminals

UC Clients

VARTA ADR100	Unified Communication Client for Android
VARTA AMP100	Unified Communication Client for iOS
VARTA WIN200	Unified Communication Client for Windows

Desk Phones

SPARSH VP248	The High-Definition Edge to your IP Communication
SPARSH VP310	The Executive IP Phone
SPARSH VP330	Intuitive Touch-Screen IP Phone
SPARSH VP510	Premium IP Phone
SPARSH VP110	The Business IP Phone
SPARSH VP710	The Smart Video IP Deskphone
SPARSH VP210	The Entry Level IP Phone

SPARSH VP248

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, STUN, PPPoE
NAT	STUN and NAT Keep Alive
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723-5.3, G.723-6.3, G.726-16, G.726-24, G.726-32, G.726-40, G.729AB
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone

Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Quality of Service	Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
LCD Display	2 Lines and 6 Lines Display
Security	Password Protected Configuration
Power Supply	
Input	5VDC @2A through External Adaptor (90 - 265 VAC, 47 - 63Hz) and Power-over-Ethernet (PoE)
Power Consumption	5W (Typical)
Mechanical	
Dimensions (WxHxD)	20.7 x 23.2 x 4.5 cm
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental	
Operating Temperature	0°C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	0 to 95% RH, Non-Condensing
Unit Weight	1.18 Kgs (2.6 lbs) Approx.

SPARSH VP248 (Standard)

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, STUN, PPPoE
SIP	9 Multiple SIP Accounts Out Bound Proxy Support Main and Secondary DNS Server Support
NAT	STUN and NAT Keep Alive
Voice CODECS	G.722 Wideband, G.711 A/µ-Law, G.723-5.3, G.723-6.3, G.726-16, G.726-24, G.726-32, G.726-40, G.729AB
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Quality of Service	Layer 3 DIFFServ and TOS

Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
Security	Password Protected Administration
Power Supply	
Input	5VDC @2A through External Adaptor (90-265VAC, 47-63Hz) Power-over-Ethernet (PoE)
Power Consumption	5W (Typical)
Mechanical	
Dimensions (WxHxD)	24.0 x 20.0 x 9.9 cm (9.4”x7.9”x3.9”)
Material	ABS Plastic
Installation Mounting	Wall Mount and Table-Top
Environmental	
Operating Temperature	0°C to 45°C

SPARSH VP310

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729, iLBC - 20/30 msec
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
LCD Display	2 x 24 Character Display
Security	TLS, SRTP
Power Supply	
Input	5VDC @2A through External Adaptor (100-240 VAC, 50 - 60 Hz)
Power Consumption	4W (Typical)
Mechanical	
Dimensions (WxHxD)	20.7 x 23.2 x 4.5 cm
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental	

Operating Temperature	0°C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 - 95% RH, Non-Condensing
Weight (Without Foot Stand)	830 gms Approx.

SPARSH VP330

LCD Display	4.3" Colour TFT Touch Screen Display
VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
LCD Display	2 x 24 Character Display
Security	TLS, SRTP
Power Supply	
Input	5VDC @2A through External Adaptor (100-240 VAC, 50 - 60 Hz)
Power Consumption	5.50 W (Typical)
Mechanical	
Dimensions (WxHxD)	20.7 x 23.2 x 4.5 cm
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing
Weight (Without Foot Stand)	840 gms Approx.

SPARSH VP510

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/µ-Law, G.723, G.729
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100 Base T (PoE Optional) PC Port (RJ45), 10/100 Base T
Security	TLS, SRTP
LCD	240*64 Pixel Graphic LCD Display
Power Supply	
Input	5VDC @2A through External Adaptor (100-240 VAC, 50 - 60 Hz)
Power Consumption	2.25W (stand alone)
Mechanical	
Weight	805 gm
Dimension in mm [L*B*H]	247*183*43
Material	ABS Plastic
Installation Mounting	Wall Mount and Table - Top
Environmental Conditions	
Operating Temperature Range	0°C to 45°C
Storage Temperature	-20°C to +70°C
Operating and Storage Humidity	5 to 95%, RH, Non-Condensing

DSS532

Terminals Supported	SPARSH VP510
Source of Power	Powered from the Host Phone
Programmable Keys	32
Stackable	upto 4 modules
LED Indicator	Dual Colour, Red/Blue
Signaling	Proprietary Digital (2B+D)
Interface	Single Pair for Speech, Signaling and Power
Physical Connector	RJ11

Physical Port	IN Port: Connects with the AUX port of the phone or OUT Port of the preceding DSS532 attached.
	OUT Port: Connects with IN Port of the succeeding DSS532 attached.
Installation Option	Table Mount
Mechanical	
Weight (Product + Stand)	235g
Product Weight	218g
Stand Weight	17g
Dimension [L*B*H]	178.7mm * 100mm * 40.5mm
Environmental Conditions	
Operating Temperature Range	0°C to 45°C
Operating Humidity	5 to 95%, RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	0 to 95%, RH, Non-Condensing

SPARSH VP210(Extended)

LCD Display	128 x 64 Graphical LCD
VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, UDP, DHCP, PPPoE
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	Dynamic Jitter Buffer (Adaptive)
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100/1000 Base T (PoE Optional) PC Port (RJ45), 10/100/1000 Base T
Security	TLS, SRTP
Power Supply	
Input	5VDC @2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	1.0 W (Typical)
Mechanical	
Dimensions (WxHxD)	163 x 210 x 101 mm without stand and with receiver placed on phone
Material	ABS Plastic
Installation Mounting	Table - Top

Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing
Weight (Without Foot Stand)	650 gms Approx.

SPARSH VP210 (Standard)

LCD Display	128 x 64 Graphical LCD
VoIP	
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833), SRTP
Network Protocol	IPv4, TCP, UDP, DHCP, SNTP, NAT, STUN, HTTP, TLS
Voice CODECS	G.722, G.711 A/µ-Law, G.723, G.729
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Quality of Service	Layer 2 CoS, Layer 3 DIFFServ and TOS
Data Network	LAN Port (RJ45), 10/100/1000 Base T (PoE Optional) PC Port (RJ45), 10/100/1000 Base T
Security	Password Protected Administration
Power Supply	
Input	5VDC(+/-0.25V)@2A through External Adapter (100-240 VAC, 50 - 60 Hz, Optional) and Power-over-Ethernet (PoE)
Power Consumption	1.0 W (Typical)
Mechanical	
Dimensions (WxHxD)	163 x 210 x 101 (mm) without stand and with receiver placed on the phone
Material	ABS Plastic
Installation Mounting	Table - Top
Environmental	
Operating Temperature	0° C to 45°C
Operating Humidity	5 to 95% RH, Non-Condensing
Storage Temperature	-20°C to +70°C
Storage Humidity	5 to 95% RH, Non-Condensing
Weight (Without Foot Stand)	650 gms Approx.

Physical Features of SPARSH VP110

This section lists the available physical features of SPARSH VP110 IP phones.

- 132 x 64 Graphic LCD
- Single VoIP account
- 29 keys including 4 soft keys
- 1 x RJ9 (4P4C) Handset port
- 1 x RJ9 (4P4C) Headset port
- 2 x RJ45 10/100Mbps Ethernet ports
- 1 LED: 1 x Power
- Power Adapter (Optional): AC 100~240V input and DC 5V/600mA output
- Power over Ethernet (IEEE 802.3af)

Technical Specifications

Audio Features

- Full-duplex Hands-free Speakerphone with AEC
- Codecs: G.711(A/μ), G.722, G.723, G.729, G.726, iLBC
- DTMF: In-band, Out-of-band (RFC 2833) and SIP INFO
- VAD, CNG, AEC, PLC, AJB, AGC

Phone Book

- Local Phone book up to 1000 entries Black List
- XML Remote Phone book
- Intelligent Search Method
- Phone book Search/Import/Export
- Call History: Dialed/Received/Missed/Forwarded

Phone Features

- Single VoIP Account
- Call Hold, Mute, DND
- One-touch Speed Dial, Hotline
- Redial, Call Return, Auto Answer
- Call Forward, Call Waiting, Call Transfer
- Local 3-way Conference
- Direct IP Call without SIP Proxy
- Ringtone Selection/Import/Delete
- Keypad Lock, Emergency Call

- Set Date & Time Manually or Automatically
- Dial Plan, XML Browser, Action URL/URI
- Instant Messaging (Web UI and Phone)

Call Management

- Anonymous Call (CLIR)
- Anonymous Call Rejection
- Message Waiting Indicator (MWI)
- Voicemail, Call Pickup
- Intercom, Music on Hold
- Call Completion, Hot-desking
- Dial out Number from Web UI

Display

- 132x64-pixel Graphical LCD
- LED for Indicating Incoming calls, Voice/Text Messages, Mute, Call Hold/Held, On call
- Intuitive User Interface with Icons and Soft keys
- Multiple Language Options
- Caller ID with Name, Number

Networking and Security

- SIP v1 (RFC2543), v2 (RFC3261)
- IPv6
- NAT Transverse: STUN Mode
- Proxy Mode and Peer-to-Peer SIP Link Mode
- IP Assignment: Static/DHCP/PPPoE
- HTTPS Web Server
- Time and Date Synchronization using SNTP
- UDP/TCP/DNS-SRV (RFC 3263)
- QoS: 802.1p/Q Tagging (VLAN), Layer 3 ToS, DSCP
- SRTP for Voice
- Transport Layer Security (TLS)
- HTTPS Certificate Manager
- AES Encryption for Configuration File
- Digest Authentication
- IEEE802.1X
- SNMP v1/v2

Management

- Configuration: Browser/Phone/Auto-Provision
- Auto Provision via FTP/TFTP/HTTP/HTTPS for Mass Deploy
- Server Redundancy
- Factory Reset
- Soft Reboot
- Packet Tracing Export
- System Log

Physical Features

- 2 x 10/100 Mbps LAN & PC Ports
- 29 keys including 4 Soft Keys
- 1 x RJ9 Handset Port
- 1 x RJ9 Headset Port
- Dimension (W x D x H): 185 x 188 x 143 mm

Power Supply

- Power Adapter (Optional): 5VDC/600mA
- Power over Ethernet (IEEE 802.3af)
- Power Consumption: 5W (Typical)
- Connector: DC Power Jack

Mechanical

- Packaging: 10 Qty/CTN
- Net Weight: 9.8 Kg
- Gross Weight: 10.8 Kg
- Gift Box: 215 x 200 x 121 mm
- Installation: Wall Mount, Table-top
- Color: Gray

Environmental

- Operating Temperature: -10° C to 50°C (14° F to 122° F)
- Operating Humidity: 10 - 95% (Non-Condensing)

Certifications

CE, FCC-15 (Class-B), RCM, RoHS

Physical Features of SPARSH VP710

This section lists the available physical features of SPARSH VP710 IP phone.

- 7” 1024 x 600 pixel color touch screen with backlight
- Operating System: Android™ 5.1.1
- 16 VoIP accounts
- HD Voice: HD Codec, HD Handset, HD Speaker
- 20 dedicated hard keys, 3 dedicated soft Android keys for BACK, HOME and RECENT
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100/1000Mbps Ethernet ports
- 4 LEDs: 1*power, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/2A output
- 1*USB2.0 port (on the top of the phone), support, USB camera.
- 1*USB2.0 port (on the rear of the phone), USB flash drive or USB headset
- Built-in Wi-Fi, support 802.11b/g/n
- Built-in Bluetooth 4.0, support Bluetooth headset
- Power over Ethernet (IEEE 802.3af)
- Wall Mountable

Key Features of the IP Phone

In addition to physical features introduced above, IP phone also supports the following key features when running the latest firmware:

Phone Features

- **Call Options:** emergency call, call waiting, call hold, call mute, call forward, call transfer, call pickup, five-way audio-only conference, five-way audio-only and video mixed conference (up to three-way video conference.
- **Basic Features:** DND, auto redial, live dialpad, dial plan, hotline, caller identity, auto answer.
- **Advanced Features:** BLF, server redundancy, distinctive ring tones, remote phone book, LDAP.

Codecs and Voice Features

- Wideband codec: G.722, Opus
- Narrowband codec: G.711, G.726, G.729, iLBC, G.723
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

Video Features

- Video codec: H264HP, H264, VP8
- Image codec: JPEG, PNG, BMP
- Adaptive bandwidth adjustment

Network Features

- SIP v1 (RFC 2543), v2 (RFC 3261)
- NAT Traversal: STUN mode
- DTMF: INBAND, RFC 2833, SIP INFO
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: Static/DHCP/PPPoE
- VLAN assignment: LLDP/Static/DHCP/CDP

- Bridge mode for PC port
- HTTP/HTTPS server
- DNS client
- NAT/DHCP server
- IPv6 support
- Wi-Fi

Management

- FTP/TFTP/HTTP/PnP auto-provision
- Configuration: browser/phone/auto-provision
- Direct IP call without SIP proxy
- Dial number via SIP server
- Dial URL via SIP server
- TR-069

Security

- HTTPS (server/client)
- SRTP (RFC 3711)
- Transport Layer Security (TLS)
- VLAN (802.1q), QoS
- Digest authentication using MD5/MD5-sess
- Secure configuration file via AES encryption
- Phone lock for personal privacy protection
- Admin/User configuration mode
- 802.1X authentication

Packing List

Verify contents of the package shipped to you with the contents listed below. If any of the items is missing or damaged, contact your Dealer/Reseller.

You can view or download the documentation of the following products by scanning the QR code printed on the Product Label/Packaging Label of the respective product.

SPARSH VP248 (Extended/Standard)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP310

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP330

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP510

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1

Sr. No.	Item	Quantity
2.	Ethernet Cable	1
3.	Foot Stand	1
4.	Wall Mounting Template	1
5.	Self tapping screws and screw grips for wall mounting	2

SPARSH VP210 (Extended/Standard)

Sr. No.	Item	Quantity
1.	Phone, Handset and Spring Cord	1
2.	Ethernet Cable	1
3.	Foot Stand	1

DSS532

Sr. No.	Item	Quantity
1.	DSS532 Console Unit	1
2.	DSS Extender	1
3.	Foot Stand	1
4.	RJ11 Cable	1
5.	Clamps (2 DSS-Phone Clamps and 2 DSS-DSS Clamps)	4

SPARSH VP110

Sr. No.	Item	Quantity
1.	Matrix SPARSH VP110 IP Phone Unit	1
2.	Phone Stand	1
3.	Handset & Handset Cord	1
4.	Ethernet Cable	1
5.	Power Adapter	1

SPARSH VP710

Sr. No.	Item	Quantity
1.	Matrix SPARSH VP710 IP Phone Unit	1
2.	Phone Stand	1

Sr. No.	Item	Quantity
3.	Handset & Handset Cord	1
4.	Ethernet Cable	1
5.	Camera	1
6.	Power Adapter (Optional)	1
7.	Wall Mount Bracket (Optional)	1

VMS Prompts

Given below are the prompts which are already recorded in English language and are provided to you by default. Along with the prompts, the Prompt Names and File Names are also listed below. You must use these prompts as a reference while recording and uploading the prompts for other languages. For further information, refer “Prompts Management”.

Greetings		
Prompt Name	**File Name**	**Prompts**
Greeting 01	Morning.wav	Good Morning!
Greeting 02	Afternoon.wav	Good Afternoon!
Greeting 03	Evening.wav	Good Evening!

Auto Attendant Prompts		
Prompt Name	**File Name**	**Prompts**
Auto Attendant 01	Working.wav	Welcome! Please dial the extension number or To dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press # (HASH).
Auto Attendant 02	Break.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press # (HASH).
Auto Attendant 03	Nonworking.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press # (HASH).
Auto Attendant 04	WH_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect **the call** press # (HASH).

Auto Attendant 05	BH_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect **the call** press # (HASH).
Auto Attendant 06	NWH_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect **the call** press #. (HASH)
Auto Attendant 07	WH_USA.wav	Welcome! Please dial the extension number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. To connect to the operator press 0. To disconnect the call press £. (POUND)
Auto Attendant 08	BH_USA.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press £. (POUND)
Auto Attendant 09	NWH_USA.wav	Thank you for calling! We are closed at this time. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the call press £. (POUND)
Auto Attendant 10	WH_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the call press £. (POUND)
Auto Attendant 11	BH_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the call press £. (POUND)

Auto Attendant 12	NWH_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the call press £. (POUND)
Auto Attendant 13	Holiday.wav	Thank you for calling! We are closed due to holiday. To leave a message press 7. For further assistance press 9. To disconnect the Call press #. (HASH)
Auto Attendant 14	Holiday_Hotel.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 9. To disconnect the Call press #. (HASH)
Auto Attendant 15	Holiday_USA.wav	Thank you for calling! We are closed due to holiday. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the Call press £. (POUND)
Auto Attendant 16	Holiday_Hotel_USA.wav	Welcome! Please dial the room number or to dial by name press 6. To leave a message press 7. To access your Personal Mailbox press 8. For further assistance press 0. To disconnect the Call press £. (POUND)

Number Dialing		
Prompt Name	**File Name**	**Prompts**
Number Dialing 01	Entextn.wav	Please enter the extension number.
Number Dialing 02	DialName.wav	Please enter the first three letters of the name.
Number Dialing 03	MBAccess.wav	Please enter your extension number.
Number Dialing 04	Leavemsg.wav	Please enter the extension number for which you wish to leave message.
Number Dialing 05	LeavemsgH.wav	Please enter the room number or the extension number for which you wish to leave message.
Number Dialing 06	MsgdestI.wav	Please enter the destination and press hash [#] to end.
Number Dialing 07	MsgdestU.wav	Please enter the destination and press Pound [£] to end.

No Digit Dialed		
Prompt Name	**File Name**	**Prompts**
No Digit Dialed 01	NoDigitDialed.wav	Sorry, You have not dialed any digit.

No Digit Dialed 02	NoOptionSelected.wav	Sorry, You have not selected any option.
No Digit Dialed 03	NoInput.wav	Sorry, You have not entered any input.
No Digit Dialed 04	NoDestination.wav	Sorry, this is an invalid input.

Invalid Digit Dialed		
Prompt Name	**File Name**	**Prompts**
Invalid Digit Dialed 01	InvalidInput.wav	Sorry, this is an invalid input.
Invalid Digit Dialed 02	InvalidDigit.wav	Sorry, this is an Invalid digit.
Invalid Digit Dialed 03	Invalidno.wav	Sorry, this is an Invalid Number.
Invalid Digit Dialed 04	NoMatchFound.wav	No match found.
Invalid Digit Dialed 05	InvalidDest.wav	Invalid recipient. Please enter all recipients again.

Expiry of Count		
Prompt Name	**File Name**	**Prompts**
Expiry Of Count 01	RetryCountOver.wav	Sorry! Maximum attempts exceeded.

Call Transfer Type		
Prompt Name	**File Name**	**Prompts**
Call Transfer Type 01	PlsHold.wav	Please hold.
Call Transfer Type 02	XfrCall.wav	Please hold, transferring the call.
Call Transfer Type 03	Xfrcallto.wav	Please hold, transferring the call to
Call Transfer Type 04	XfrcalltoOpr.wav	Please hold, transferring the call to Operator.
Call Transfer Type 05	XferMailbox.wav	Please hold, transferring the call to mailbox.
Call Transfer Type 06	XferMailboxof.wav	Please hold, transferring the call to mailbox of

Call Transfer Unsuccessful		
Prompt Name	**File Name**	**Prompts**
Call Transfer Unsuccessful 01	Noreply.wav	Sorry! The person is unavailable to take your call right now.
Call Transfer Unsuccessful 02	Extbusy.wav	Sorry! The person you are trying to call is busy.
Call Transfer Unsuccessful 03	Noattnd.wav	Sorry! The call could not be attended.

MoH		
Prompt Name	**File Name**	**Prompts**
MoH 1	Holdmusic.wav	<Music>

Disconnect		
Prompt Name	**File Name**	**Prompts**
Disconnect 01	Thankyou.wav	Thank you for your call.

Language		
Prompt Name	**File Name**	**Prompts**
Language 01	english.wav	For English,

Dial by Name		
Prompt Name	**File Name**	**Prompts**
Dial by Name 01	PreName.wav	The following matches are found
Dial by Name 02	selectname.wav	To select the name,
Dial by Name 03	nextname.wav	To skip to the next name,
Dial by Name 04	repeatlastname.wav	To repeat the previous name,
Dial by Name 05	repeatnames.wav	To repeat all the names,
Dial by Name 06	househenternameagain.wav	To re-enter the name,
Dial by Name 07	previousmenu.wav	To return to the previous menu,
Dial by Name 08	disconnect.wav	To disconnect,
Dial by Name 09	Selected.wav	Name is selected,
Dial by Name 10	NameSelected.wav	Selected name is,
Dial by Name 11	ConfirmName.wav	You have selected,
Dial by Name 12	confirm.wav	To confirm,
Dial by Name 13	reenter.wav	To re-enter,

Call Transfer		
Prompt Name	**File Name**	**Prompts**
Call Transfer 01	RecName.wav	Please Record your name after the beep and press any digit to end
Call Transfer 02	Callfrom.wav	Hello! There is a call from,
Call Transfer 03	acceptcall.wav	To accept the call,
Call Transfer 04	reject.wav	To reject the call,
Call Transfer 05	rejectbusy.wav	To reject the call as busy,
Call Transfer 06	rejectnoreply.wav	To reject the call as no reply,
Call Transfer 07	Leavemsg1.wav	To leave a message,
Call Transfer 08	XfertoOperator.wav	To transfer the call to operator,
Call Transfer 09	XfertoAssitance.wav	To transfer the call to assistant,
Call Transfer 10	XfertoAlternate.wav	To transfer the call to alternate number,
Call Transfer 11	DialExtn.wav	To dial an extension,
Call Transfer 12	XferMainMenu.wav	To return to Main Menu,

Call Transfer 13	XferPreviousMenu.wav	To return to Previous Menu,
Call Transfer 14	StayonHold.wav	To stay on hold,
Call Transfer 15	disconnect.wav	To disconnect,
Call Transfer 16	nonumber.wav	Number not programmed.

Message Record		
Prompt Name	**File Name**	**Prompts**
Message Record 01	Recmsg.wav	Please Record your Message after the beep and press any digit to end.
Message Record 02	RecMsgStopCode.wav	Please Record your message after the beep and press
Message Record 03	RecMsgEnd.wav	To end,
Message Record 04	MsgLeaveConfirm.wav	Message sent as,
Message Record 05	Rerecord.wav	To re-record the message,
Message Record 06	Confirm.wav	To confirm the message as normal,
Message Record 07	urgent.wav	To confirm the message as urgent,
Message Record 08	private.wav	To confirm the message as private,
Message Record 09	listenmsg.wav	To play the recorded message,
Message Record 10	appendmsg.wav	To append to the recorded message,
Message Record 11	disconnect.wav	To disconnect,
Message Record 12	normalset.wav	Normal
Message Record 13	urgentset.wav	Urgent
Message Record 14	privateset.wav	Private
Message Record 15	securedset.wav	Secured
Message Record 16	NoMailbox.wav	Mailbox not assigned.
Message Record 17	Mailboxfull.wav	Sorry! your message cannot be delivered as Mailbox is full.
Message Record 18	Memoryfull.wav	Sorry! your message cannot be delivered as System Storage is full.
Message Record 19	appendfail.wav	Sorry! message cannot be appended as System Storage is full.
Message Record 20	urgentandprivate.wav	To confirm the message as urgent and private,

Message Send Forward		
Prompt Name	**File Name**	**Prompts**
Message Send Forward 01	Recmsg.wav	Please Record your Message after the beep and press any digit to end
Message Send Forward 02	RecMsgStopCode.wav	Please Record your message after the beep and press
Message Send Forward 03	RecMsgEnd.wav	to end
Message Send Forward 04	MsgSendConfirm.wav	Message sent as
Message Send Forward 05	reenter.wav	To re-enter
Message Send Forward 06	Confirm.wav	To confirm
Message Send Forward 07	commentatstart.wav	To forward the message with comment before the message

Message Send Forward 08	commentatend.wav	To forward the message with comment after the message
Message Send Forward 09	nocomment.wav	To forward the message without comment
Message Send Forward 10	Rerecord.wav	To re-record the message
Message Send Forward 11	Confirm.wav	To confirm the message as normal
Message Send Forward 12	urgent.wav	To confirm the message as urgent
Message Send Forward 13	private.wav	To confirm the message as private
Message Send Forward 14	listenmsg.wav	To play the recorded message
Message Send Forward 15	appendmsg.wav	To append to the recorded message
Message Send Forward 16	Requestreceipt.wav	To request read receipt
Message Send Forward 17	ignorereceipt.wav	To ignore read receipt
Message Send Forward 18	previousmenu.wav	To return to previous menu
Message Send Forward 19	normalset.wav	Normal
Message Send Forward 20	urgentset.wav	Urgent
Message Send Forward 21	privateset.wav	Private
Message Send Forward 22	securedset.wav	Secured
Message Send Forward 23	NumberCollected.wav	Recipient number entered is
Message Send Forward 24	FwdMsgFail.wav	Sorry! Message cannot be forwarded as message length has exceeded
Message Send Forward 25	Pending.wav	Sorry! the message cannot be sent
Message Send Forward 26	NoDigitDialed.wav	You have not dialed any digit
Message Send Forward 27	nextnumber.wav	Enter the next recipient number
Message Send Forward 28	NoMailbox.wav	Sorry! Mailbox is not assigned
Message Send Forward 29	appendfail.wav	Sorry, message cannot be appended as System Storage is full
Message Send Forward 30	Mailboxfull.wav	Sorry, your message cannot be delivered as Mailbox is full
Message Send Forward 31	enterDDMMYY24.wav	Enter Delivery Date in DD MM YYYY format and time in Twenty four hour format
Message Send Forward 32	enterDDMMYY12.wav	Enter Delivery Date in DD MM YYYY format and time in Twelve hour format. For AM, press 1, for PM, press 2.
Message Send Forward 33	enterMMDDYY24.wav	Enter Delivery Date in MM DD YYYY format and time in Twenty four hour format
Message Send Forward 34	enterMMDDYY12.wav	Enter Delivery Date in MM DD YYYY format and time in Twelve hour format. For AM, press 1, for PM, press 2.
Message Send Forward 35	checkfuturedelivery.wav	To check the last entered Future Delivery
Message Send Forward 36	cancelfuturedelivery.wav	To cancel the last entered Future Delivery
Message Send Forward 37	futuredelconf.wav	To confirm press 1. To cancel press any digit
Message Send Forward 38	DiscardMsg.wav	To discard the message and return to previous menu

Mailbox Access		
Prompt Name	**File Name**	**Prompts**
Mailbox Access 01	Enterpwd.wav	Please Enter your mailbox password.
Mailbox Access 02	Youhave.wav	You have,
Mailbox Access 03	Newmsg.wav	New message.
Mailbox Access 04	Newmsgs.wav	New messages.
Mailbox Access 05	Nonewmsg.wav	You have no new messages.
Mailbox Access 06	Urgentmsg.wav	Urgent message
Mailbox Access 07	Urgentmsgs.wav	Urgent messages
Mailbox Access 08	Nooldmsg.wav	You have no old messages.
Mailbox Access 09	PerMB.wav	In your personal mailbox
Mailbox Access 10	DeptMB.wav	In your department group mailbox
Mailbox Access 11	MessageCount.wav	Message
Mailbox Access 12	Msgrecon.wav	This message was recorded on,
Mailbox Access 13	ConvBetn.wav	This conversation was between,
Mailbox Access 14	Callnum.wav	And calling number was,
Mailbox Access 15	MsgreadBy.wav	This message was read by,
Mailbox Access 16	DeptOrStn.wav	Press '1' to access personal mailbox, Press '2' to access department group mailbox
Mailbox Access 17	RecdCncl.wav	Sorry! Your message cannot be delivered.
Mailbox Access 18	MB80Full.wav	Your Mailbox is 80% Full. Please Delete old messages of your mailbox.
Mailbox Access 19	MBFull.wav	Your Mailbox is Full. Please Delete old messages of your Mailbox.
Mailbox Access 20	Invalpwd.wav	Sorry! This is an invalid password
Mailbox Access 21	MBblocked.wav	Sorry! Mailbox is currently in use. Please try again later.
Mailbox Access 22	NoMailbox.wav	Sorry! Mailbox is not assigned
Mailbox Access 23	SystemMemFull.wav	System Storage is Full. Please Delete some stored messages.
Mailbox Access 24	privatemsg.wav	Private Message
Mailbox Access 25	urgentprivatemsg.wav	Urgent and Private message
Mailbox Access 26	Disconnect.wav	Thank you for your call.
Mailbox Access 27	Messagefor.wav	Message for extension,
Mailbox Access 28	MsgDelete.wav	Message deleted.
Mailbox Access 29	MsgSaveCnf.wav	Your message has been saved as new.
Mailbox Access 30	recorddel.wav	Recording erased.
Mailbox Access 31	Numdel.wav	Number erased.

Mailbox Access Menu		
Prompt Name	**File Name**	**Prompts**
Mailbox Access Menu 01	listennewmsg.wav	To listen to a new message,
Mailbox Access Menu 02	listenoldmsg.wav	To listen to an old message,
Mailbox Access Menu 03	sendmsg.wav	To send a message,

Mailbox Access Menu 04	Mbmgnt.wav	For mailbox management,
Mailbox Access Menu 05	replaymsg.wav	To re-play the message,
Mailbox Access Menu 06	playmsgdetails.wav	For Date and Time of the message,
Mailbox Access Menu 07	replymsg.wav	To reply the message,
Mailbox Access Menu 08	deletemsg.wav	To delete the message,
Mailbox Access Menu 09	listennextmsg.wav	To Play the next message,
Mailbox Access Menu 10	forwardmsg.wav	To forward the message,
Mailbox Access Menu 11	msgasnew.wav	To mark the message as unread,
Mailbox Access Menu 12	Mbname.wav	For mailbox name,
Mailbox Access Menu 13	MsgRedirection.wav	For message redirection,
Mailbox Access Menu 14	DeleteAllOldmsgs.wav	To delete all old messages of your mailbox,
Mailbox Access Menu 15	RecMBgrt.wav	For mailbox greetings,
Mailbox Access Menu 16	Assistance.wav	For assistant number,
Mailbox Access Menu 17	Personal.wav	For personal number,
Mailbox Access Menu 18	SetMsgRedirection.wav	To set message redirection,
Mailbox Access Menu 19	CancelMsgRedirection.wav	To cancel message redirection,
Mailbox Access Menu 20	MsgRedirectNo.wav	Enter the recipient number,
Mailbox Access Menu 21	PersonalGrt.wav	For personal greetings,
Mailbox Access Menu 22	WHGrt.wav	For working hours greeting,
Mailbox Access Menu 23	BHGrt.wav	For break hours greeting,
Mailbox Access Menu 24	NWHGrt.wav	For non-working hours greeting,
Mailbox Access Menu 25	ConditionalGrt.wav	For conditional greetings,
Mailbox Access Menu 26	BusyGrt.wav	For busy,
Mailbox Access Menu 27	NoReplyGrt.wav	For no-reply,
Mailbox Access Menu 28	UnconditionalGrt.wav	For unconditional,
Mailbox Access Menu 29	Record.wav	To record,
Mailbox Access Menu 30	Play.wav	To play,
Mailbox Access Menu 31	Erase.wav	To erase,
Mailbox Access Menu 32	EnterNum.wav	To enter number,
Mailbox Access Menu 33	PlayNum.wav	To play number,
Mailbox Access Menu 34	ClearNum.wav	To clear number,
Mailbox Access Menu 35	EnterAssistanceNum.wav	Enter the assistant extension number.
Mailbox Access Menu 36	EnterPersonalNum.wav	Enter the personal number and Press Hash [#] to end
Mailbox Access Menu 37	previousmenu.wav	To return to Previous Menu
Mailbox Access Menu 38	Recname.wav	Please Record your name after the beep and press any digit to end.
Mailbox Access Menu 39	Recgreeting.wav	Please Record the greeting after the beep and press any digit to end.
Mailbox Access Menu 40	DelAllOldCnf.wav	You are about to delete all read messages of your mailbox. To proceed, Press 1, to Cancel, Press any digit.
Mailbox Access Menu 41	DelAllOldDone.wav	All old messages are deleted.
Mailbox Access Menu 42	MsgRedirectSet.wav	Message redirection set.
Mailbox Access Menu 43	MsgRedirectCancel.wav	Message redirection canceled.

Mailbox Access Menu 44	Numbersaved.wav	Number saved.
Mailbox Access Menu 45	Numberdelete.wav	Number deleted.
Mailbox Access Menu 46	Invalidno.wav	Sorry, this is an invalid number.
Mailbox Access Menu 47	NoDigitDialed.wav	Sorry, you have not dialed any digit.
Mailbox Access Menu 48	DelAllMsgsCnf.wav	You are about to delete all messages of your mailbox. To proceed press 1. To cancel, press any digit.
Mailbox Access Menu 49	DelAllMsgsDone.wav	All messages are deleted.
Mailbox Access Menu 50	DelAllMsgs.wav	To delete all messages of your mailbox,
Mailbox Access Menu 51	Recordingdelete.wav	Recorded file deleted.

Number and Month		
Prompt Name	**File Name**	**Prompts**
Number and Month 01	Num0.wav	Zero
Number and Month 02	Num1.wav	One
Number and Month 03	Num2.wav	Two
Number and Month 04	Num3.wav	Three
Number and Month 05	Num4.wav	Four
Number and Month 06	Num5.wav	Five
Number and Month 07	Num6.wav	Six
Number and Month 08	Num7.wav	Seven
Number and Month 09	Num8.wav	Eight
Number and Month 10	Num9.wav	Nine
Number and Month 11	Num10.wav	Ten
Number and Month 12	Num11.wav	Eleven
Number and Month 13	Num12.wav	Twelve
Number and Month 14	Num13.wav	Thirteen
Number and Month 15	Num14.wav	Fourteen
Number and Month 16	Num15.wav	Fifteen
Number and Month 17	Num16.wav	Sixteen
Number and Month 18	Num17.wav	Seventeen
Number and Month 19	Num18.wav	Eighteen
Number and Month 20	Num19.wav	Nineteen
Number and Month 21	Num20.wav	Twenty
Number and Month 22	Num30.wav	Thirty
Number and Month 23	Num40.wav	Forty
Number and Month 24	Num50.wav	Fifty
Number and Month 25	Num60.wav	Sixty

Number and Month 26	Num70.wav	Seventy
Number and Month 27	Num80.wav	Eighty
Number and Month 28	Num90.wav	Ninety
Number and Month 29	Num100.wav	Hundred
Number and Month 30	Num1000.wav	Thousand
Number and Month 31	Month1.wav	January
Number and Month 32	Month2.wav	February
Number and Month 33	Month3.wav	March
Number and Month 34	Month4.wav	April
Number and Month 35	Month5.wav	May
Number and Month 36	Month6.wav	June
Number and Month 37	Month7.wav	July
Number and Month 38	Month8.wav	August
Number and Month 39	Month9.wav	September
Number and Month 40	Month10.wav	October
Number and Month 41	Month11.wav	November
Number and Month 42	Month12.wav	December
Number and Month 43	Hash.wav	Hash
Number and Month 44	Pound.wav	Pound
Number and Month 45	Star.wav	Star
Number and Month 46	asterisk.wav	Asterisk

Alarm		
Prompt Name	**File Name**	**Prompts**
Alarm 01	EntertimeI.wav	Enter the time, HH:MM in Twenty four hour format. To cancel all alarms, press Pound [£]
Alarm 02	EntertimeU.wav	Enter the time, HH:MM in Twelve hour format. For AM, press 1. For PM, press 2 To cancel all alarms, press Pound [£]
Alarm 03	WakeupCancel.wav	Your all wake up alarms are canceled.
Alarm 04	SetOnceDaily.wav	To set once press '1'. To set daily press '2'
Alarm 05	WakeupVeri.wav	You have set wake up alarm for
Alarm 06	DailyWakeupVeri.wav	You have set daily wake up alarm for
Alarm 07	WakeupSet.wav	Your wake up alarm is set.
Alarm 08	DailyWakeupSet.wav	Your daily wake up alarm is set.
Alarm 09	Am.wav	A.M.
Alarm 10	Pm.wav	P.M.
Alarm 11	AlarmConf.wav	To confirm press 1, To re-enter press 2
Alarm 12	AlarmDateI.wav	Enter the Date in DD MM YYYY format. To cancel all reminders, press Pound [£]
Alarm 13	AlarmDateU.wav	Enter the Date in MM DD YYYY format. To cancel all reminders, press Pound [£]

Alarm 14	ReminderCancel.wav	Your all reminders are canceled.
Alarm 15	ReminderVeri.wav	You have set reminder for
Alarm 16	ReminderSet.wav	Your reminder is set
Alarm 17	Alarmnoset.wav	Sorry! Your wake up alarm cannot be set. Please call operator for further assistance.
Alarm 18	Remindernoset.wav	Sorry! Your reminder cannot be set. Please call operator for further assistance.
Alarm 19	RemoteExt.wav	Please Enter the extension number for which you wish to set or cancel wake up alarm.
Alarm 20	RemoteExtH.wav	Please Enter the room number for which you wish to set or cancel wake up alarm.
Alarm 21	PerWakeupVeri.wav	You have set personal wake up alarm for,
Alarm 22	AutoWakeupVeri.wav	You have set automated wake up alarm for,
Alarm 23	PerWakeupSet.wav	Your personal wake up alarm is set.
Alarm 24	AutoWakeupSet.wav	Your automated wake up alarm is set.
Alarm 25	DailyPerWakeupSet.wav	Your daily personal wake up alarm is set.
Alarm 26	DailyAutoWakeupVeri.wav	You have set daily automated wake up alarm for,
Alarm 27	DailyPerWakeupVeri.wav	You have set daily personal wake up alarm for,
Alarm 28	DailyAutoWakeupSet.wav	Your daily automated wake up alarm is set.
Alarm 29	PerReminderVeri.wav	You have set personal reminder for,
Alarm 30	AutoReminderVeri.wav	You have set automated reminder for,
Alarm 31	PerReminderSet.wav	Your personal reminder is set.
Alarm 32	AutoReminderSet.wav	Your automated reminder is set.
Alarm 33	Alarmmode.wav	To set it as Personal, Press 1. To set it as Automated, Press 2.
Alarm 34	Alarmnocancel.wav	Sorry! There is no alarm set to cancel.
Alarm 35	Remindernocancel.wav	Sorry! There is no reminder set to cancel.
Alarm 36	WakeUpgreeting.wav	This is your wake up call.
Alarm 37	DailyWakeUpgreeting.wav	This is your daily wake up call.
Alarm 38	Remindergreeting.wav	This is your reminder call.
Alarm 39	SWakeUpgreeting.wav	This is your wake up call. To acknowledge press '0'.
Alarm 40	SDailyWakeUpgreeting.wav	This is your daily wake up call. To acknowledge press '0'.
Alarm 41	SRemindergreeting.wav	This is your reminder call. To acknowledge press '0'.
Alarm 42	Acknowledge.wav	Your alarm is acknowledged.
Alarm 43	Entertime2I.wav	Enter the time, HH MM in Twenty four hour format
Alarm 44	Entertime2U.wav	Enter the time, HH MM in Twelve hour format. For AM, press 1. For PM, press 2.
Alarm 45	RemoteExtRem.wav	Please Enter the extension number for which you wish to set or cancel reminder.
Alarm 46	RemoteExtHRem.wav	Please Enter the room number for which you wish to set or cancel reminder.
Alarm 47	RemAcknowledge.wav	Your reminder is acknowledged.
Alarm 48	Thankservice.wav	Thank You for using this service.
Alarm 49	Morning.wav	Good Morning!
Alarm 50	Afternoon.wav	Good Afternoon!
Alarm 51	Evening.wav	Good Evening!
Alarm 52	NoInput.wav	Sorry, You have not entered any input.
Alarm 53	InvalidInput.wav	Sorry, this is an invalid input.

Miscellaneous		
Prompt Name	**File Name**	**Prompts**
Miscellaneous 01	Press.wav	Press,
Miscellaneous 02	Dial.wav	Dial,
Miscellaneous 03	And.wav	and,
Miscellaneous 04	At.wav	At,
Miscellaneous 05	Beep.wav	<Beep>
Miscellaneous 06	NoDISA.wav	Sorry! DISA feature is not allowed.
Miscellaneous 07	NoDISAStn.wav	Sorry! DISA feature is not allowed for dialed extension.
Miscellaneous 08	MsgNtfyFor.wav	Message notification for extension number,
Miscellaneous 09	DialDigit.wav	Press any digit to proceed.
Miscellaneous 10	Chkinwel.wav	Welcome! It is our pleasure to serve you. We will do our best to make your stay comfortable.
Miscellaneous 11	Checkout.wav	Sorry! The Guest has checked out.
Miscellaneous 12	to.wav	to,
Miscellaneous 13	Holdmusic.wav	<Music>
Miscellaneous 14	Entextn.wav	Please enter the extension number.
Miscellaneous 15	EntPwd.wav	Please enter your password.
Miscellaneous 16	Invalpwd.wav	Sorry, this is an invalid password.
Miscellaneous 17	enterfoldernum.wav	Enter folder number.
Miscellaneous 18	enterfilenum.wav	Enter file number.
Miscellaneous 19	RecPrompt.wav	Please Record your prompt after the beep and press any digit to end.
Miscellaneous 20	InvalidInput.wav	Sorry, this is an invalid input
Miscellaneous 21	NoInput.wav	Sorry, you have not entered any digit
Miscellaneous 22	Record.wav	To record
Miscellaneous 23	Play.wav	To play
Miscellaneous 24	Erase.wav	To erase
Miscellaneous 25	EntDialInnum.wav	Please enter your number and password

ANANT UCS Features Supported in Terminals

Sr. No.	Features	Supported In					
		VP330	VP210	VP248/310/510	VARTA ADR100	VARTA iOS100	VARTA WIN200
1	Abbreviated Dialing	Yes	Yes	Yes	Yes	Yes	Yes
2	Access Codes	Yes	Yes	Yes	Yes	Yes	Yes
3	Account Codes	Yes	Yes	Yes	Yes	Yes	Yes
4	Alarms	Yes	Yes	Yes	No	No	Yes
5	Authority Codes	No	No	Yes	No	No	No
6	Auto Answer	Yes	Yes	Yes	No	No	No
7	Auto Call Back	Yes	Yes	Yes	Yes	Yes	Yes
8	Auto Redial	Yes	Yes	Yes	Yes	Yes	Yes
9	Background Music	No	No	No	No	No	No
10	Barge-In	Yes	Yes	Yes	Yes	Yes	Yes
11	BLF for Trunks	Yes	Yes	Yes	Yes	Yes	Yes
12	Call Chaining	Yes	Yes	Yes	No	No	Yes
13	Call Cost Display	No	No	Yes	No	No	No
14	Call Duration Display	Yes	Yes	Yes	Yes	Yes	Yes
15	Call Forward	Yes	Yes	Yes	Yes	Yes	Yes
16	Call Forward - Remote	No	No	Yes	No	No	No
17	Call Forward - Scheduled	Yes	Yes	Yes	Yes	Yes	Yes
18	Call Forward - Not	Yes	Yes	Yes	Yes	Yes	Yes
19	Call Hold	Yes	Yes	Yes	Yes	Yes	Yes
20	Call Park	Yes	Yes	Yes	No	No	Yes
21	Call Logs	Yes	Yes	Yes	Yes	Yes	Yes
22	Call Pickup	Yes	Yes	Yes	Yes	Yes	Yes
23	Call Toggle	Yes	Yes	Yes	Yes	Yes	Yes
24	Call Transfer	Yes	Yes	Yes	Yes	Yes	Yes
25	CLIP	Yes	Yes	Yes	Yes	Yes	Yes
26	CLIR	Yes	Yes	Yes	No	No	Yes
27	Cancel All Station Features	No	No	Yes	No	No	No
28	CUG	Yes	Yes	Yes	Yes	Yes	Yes
29	CUG with Exchange ID	Yes	Yes	Yes	Yes	Yes	Yes
30	Conference 3-Party	Yes	Yes	Yes	Yes	Yes	Yes
31	Conference Multiparty	Yes	Yes	Yes	Yes	Yes	Yes
32	Conference Dial In	Yes	Yes	Yes	Yes	Yes	Yes
33	Conversation Recording	Yes	Yes	Yes	Yes	Yes	Yes
34	COSEC Integration	No	No	No	No	No	No
35	Daylight Saving Time (DST)	Yes	Yes	Yes	NA	NA	NA
36	Department Call	Yes	Yes	Yes	Yes	Yes	Yes
37	Digital Key Phone-	Yes	Yes	Yes	NA	NA	NA
38	Distinctive Rings	Yes	Yes	Yes	No	No	Yes
39	Do Not Disturb (DND)	Yes	Yes	Yes	Yes	Yes	Yes
40	Door Phone	Yes	Yes	Yes	Yes	Yes	Yes
41	DSS Call Pick-Up	Yes	Yes	Yes	Yes	Yes	Yes
42	Dynamic Lock	Yes	Yes	Yes	No	No	Yes
43	Emergency Conference	Yes	Yes	Yes	Yes	Yes	Yes
44	Emergency Dialing	Yes	Yes	Yes	Yes	Yes	Yes

Sr. No.	Features	Supported In					
		VP330	VP210	VP248/310/510	VARTA ADR100	VARTA iOS100	VARTA WIN200
45	Extended IP Phone/Mobile Softphone Client -	Yes	Yes	Yes	Yes	Yes	Yes
46	Flashing on Trunks	Yes	Yes	Yes	No	No	Yes
47	Flexible Numbers	Yes	Yes	Yes	Yes	Yes	Yes
48	Follow Me	Yes	Yes	Yes	No	No	No
49	Forced Answer	Yes	Yes	Yes	Yes	Yes	Yes
50	Forced Call Disconnection	Yes	Yes	Yes	No	No	Yes
51	Handover and Handoff	Yes	Yes	No	Yes	Yes	Yes
52	Hot Desking	No	No	No	No	No	No
53	Hotline	Yes	Yes	Yes	No	No	No
54	Intercom	Yes	Yes	Yes	Yes	Yes	Yes
55	Interrupt Request (IR)	Yes	Yes	Yes	Yes	Yes	Yes
56	Last Caller Recall	Yes	Yes	Yes	Yes	Yes	Yes
57	Last Number Redial	Yes	Yes	Yes	Yes	Yes	Yes
58	Live Call Screening	No	No	No	No	No	No
59	Live Call Supervision	Yes	Yes	Yes	Yes	Yes	Yes
60	Macros	No	No	Yes	No	No	No
61	Meet Me Paging	Yes	Yes	Yes	No	No	No
62	Message Wait	Yes	Yes	Yes	No	No	Yes
63	Mute	Yes	Yes	Yes	Yes	Yes	Yes
64	OFF-Hook Alert	No	No	No	No	No	No
65	One Touch Transfer	Yes	Yes	Yes	Yes	Yes	Yes
66	Paging	Yes	Yes	Yes	Yes	Yes	Yes
67	PIN Dialing	Yes	Yes	Yes	No	No	No
68	Presence	Yes	Yes	Yes	Yes	Yes	Yes
69	Quick Dial	Yes	Yes	Yes	No	No	No
70	Raid	No	No	Yes	No	No	No
71	Reminder	Yes	Yes	Yes	No	No	Yes
72	Room Monitor	Yes	Yes	Yes	Yes	Yes	Yes
73	Selective Port Access	Yes	Yes	Yes	No	No	No
74	Self Ring Test	No	No	Yes	No	No	No
75	System Activity Log Display	No	No	Yes	No	No	No
76	System Fault Log Display	No	No	Yes	No	No	No
77	Time Zone Display	No	No	Yes	No	No	No
78	Trunk Call Waiting	No	No	No	No	No	No
79	Trunk Reservation	Yes	Yes	Yes	No	No	Yes
80	User Absent/Present	Yes	Yes	Yes	Yes	Yes	Yes
81	User Password	Yes	Yes	Yes	Yes	Yes	Yes
82	Video Call	No	No	No	Yes	Yes	Yes
83	Virtual Extension	Yes	Yes	Yes	Yes	Yes	Yes
84	Voice Message Applications	Yes	Yes	Yes	Yes	Yes	Yes
85	Walk-In Class of Service	Yes	Yes	Yes	No	No	No
86	Voice Mail Features	Yes	Yes	Yes	Yes	Yes	Yes

Sr. No.	Features	Supported In					
		VP330	VP210	VP248/310/510	VARTA ADR100	VARTA iOS100	VARTA WIN200
	Others*						
87	IM	No	No	No	Yes	Yes	Yes
88	SMS	No	No	No	Yes	Yes	Yes
89	BLF for Extensions	Yes	No	No	Yes	Yes	Yes
90	Soft Keys	Yes	No	No	Yes	Yes	Yes
91	Contact Grouping	No	No	No	No	No	Yes
92	Favorites	Yes	No	No	Yes	Yes	Yes

* Refer to the respective User Guide for details

Feature List	Supported In SPARSH VP110/VP710	Remarks
Abbreviated Dialing	Yes	Using Access Code
Access Codes	Yes	Using Access Code
		Note : Except Access Code having "F" - Flash
Account Codes	Partially Functional	Using Access Code
		Note: To apply Account Code during call, dial: Transfer Access Code > Account Code Access Code > Account Code Number. You need to change Transfer Access Code from "F" to any of "0-9 digit" or "*" or "#"
Alarms	Yes	Using Access Code
Alternate Number Dialing	Yes	Using Access Code
Authority Codes	Yes	Using Access Code
Auto Answer	Yes	Auto Answer will work in both ways:
		1. Local Auto Answer of the Phone
		2. Using Call-Info or Alert-Info in SIP message from ANANT [Features Like Intercom, Paging]
Auto Callback	Yes	Using Access Code
Auto Redial	Yes	Using Access Code
Barge - In	Yes	Using Access Code
BLF for trunks	NA	
Call Chaining	Partially Functional	Using Access Code
		Note: To apply Call Chaining during call, dial: Transfer Access Code > Call Chaining Access Code. You need to change Transfer Access Code from "F" to any of "0-9 digit" or "*" or "#"
Call Forward	Yes	Call Forward will work in 3 ways:
		1. Using Access Code - It will work
		2. Using Local Phone Feature - It will set Locally
		3. Using Local Phone Feature + On/Off Code - It will work. However if for some reason ANANT rejects the Call Forward set request from phone then user won't get any indication of rejection and the Phone will still set Call Forward Locally even after re
		In above 2 cases - Case 2 and Case 3, if Call Forward is not allowed through CoS of ANANT, then the call won't be forwarded even if the Call Forward can be seen locally set
Call Hold	Partially Functional	Only Exclusive hold will work.
		Global hold will not work.
		As Phone supports only 2 calls, You can hold only one call at a time.

Feature List	**Supported In SPARSH VP110/VP710**	**Remarks**
Call Park	Partially Functional	Using Access Code
		Note: To park call, dial: Transfer Access Code > Call Park
Call Retrieve	Yes	Using Access Code
Call Pick Up	Yes	Using Access Code
Call Progress Tones	Yes	
Call Taping	Yes	
Call Toggle	Partially Functional	Using Access Code
		Note: To apply Account Code during call, dial: Transfer Access Code > Account Code Access Code > Account Code Number. You need to change Transfer Access Code from "F" to any of "0-9 digit" or "*" or "#"
Call Transfer	Yes	Blind, Attended and Semi Attended Transfer will work
		Using Access Code you need to change the Transfer Access
CLIP	Yes	
CLIR	Yes	CLIR will work in 3 ways:
		1. Using Access Code - It will work
		2. Using Local Phone Feature - It will set Locally
		3. Using Local Phone Feature + On/Off Code - It will work. However if for some reason ANANT rejects the CLIR set request from phone then user won't get any indication of rejection and the Phone will still set CLIR Locally even after rejection.
		In above 2 cases - Case 2 and Case 3, if CLIR is not set in ANANT , then your identity will be visible to the remote user even if CLIR can be seen locally set
Cancel All Station Features	Yes	Using Access Code
Conference - 3 Party	Yes	Conference 3-Party will work locally. Refer User Guide.
		Using Access Code you need to change the Transfer Access Code from "F" to any of "0-9 digit" or "*" or "#"
Conference - Multiparty	Yes	Using Access Code
Conference Dial-In	Yes	Using Access Code
Conversation Recording	Yes	Using Access Code
COSEC Integration	Yes	Using Access Code
DST	Yes	
Distinctive Rings	Yes	

Feature List	Supported In SPARSH VP110/VP710	Remarks
DND	Yes	DND will work in 3 ways:
		1. Using Access Code - It will work
		2. Using Local Phone Feature - It will set Locally
		3. Using Local Phone Feature + On/Off Code - It will work. However if for some reason ANANT rejects the DND set request from phone then user won't get any indication of rejection and the Phone will still set DND Locally even after rejection.
		In above 2 cases - Case 2 and Case 3, phone will reject the call locally
Dynamic Lock	Yes	Using Access Code
Emergency Call	Yes	
Emergency Conference	Yes	Using Access Code
Follow Me	Yes	Using Access Code
Forced Answer	Yes	Using Access Code
Forced Call Disconnection	Yes	Using Access Code
Handover (Manual) and Handoff	Yes	Using Access Code
Handover (Automatic)	No	Using Access Code
Hotline	Yes	Using Access Code
Intercom	Yes	Using Access Code
Interrupt Request	Yes	Using Access Code
Last Caller Recall	Yes	Using Access Code
Last Number Redial	Yes	Using Access Code
Lightweight Directory Access Protocol (LDAP)	Yes	
Live Call Supervision	No	
Meet Me Paging	No	
Message Wait	Yes	Using Access Code
MOH	Yes	
Mute	Yes	
One Touch Transfer	Yes	Using Access Code
Paging	Yes	Using Access Code
Pin Dialing	Yes	Using Access Code
Presence	Yes	Using Access Code
Raid	Yes	Using Access Code
Reminder	Yes	Using Access Code
Room Monitor	Yes	Using Access Code
Selective Port Access	No	

Feature List	Supported In SPARSH VP110/VP710	Remarks
User Absent/Present	Yes	Using Access Code
User Password	Yes	Using Access Code
Voice Mail	Yes	Using Access Code
Walk-In Class of Service	Yes	Using Access Code
IM	Yes	Refer Respective User Guide
Early media	Yes	Refer Respective User Guide
Rport	Yes	Refer Respective User Guide
SRTP	Yes	Refer Respective User Guide
Transport Mode - UDP, TCP	Yes	Refer Respective User Guide

ANANT UCS Features tested on IP Phones of different Brands

Features and Supportive Phones		
S.NO	**Feature**	**Phones Supported**
1	Intercom	1. GrandStream GXP2020
		2. GrandStream GXV3140
		3. GrandStream GXP2120
		4. Yealink T28P
		5. Yealink T26P
		6. Yealink T22P
		7. Yealink T20P
		8. Snom 300
		9. Cisco SPA504G
2	Distinctive Ring	1. Snom 300
		2. Yealink T28P
		3. Yealink T26P
		4. Yealink T22P
		5. Yealink T20P
		6. Polycom VVX1500D
3	Last Caller Recall	1. Polycom VVX1500D
		2. Cisco SPA504G
5	Conversation Recording	1. Yealink T28P/T26P/T22P/T20P
		2. Snom 300
6	Call Park and Retrieve	1. Cisco SPA504G
		2. Polycom WX1500D
		3. Snom 300
7	Group Call Pickup and Selective Call Pickup	1. Cisco SPA504G
		2. Polycom WX1500D
8	SCA and Line Seize	1. Cisco 504G
		2. Polycom WX1500D
		3. Snom 300
		4. Yealink T28P/T26P/TT22P/T20P
		5. Grandstream GXP2120
9	Resume Call Transfer	1. Yealink T28P and T26P
10	Semi-Attend Transfer	1. Grandstream GXP2020
		2. Grandstream GXP2120
		3. Yealink T28P/T26P/TT22P/T20P
		4. Polycom WX1500D
11	Busy Lamp Field	1. Snom 300
		2. Cisco SPA504G
		3. Yealink T28P/T26P/TT22P/T20P
		4. Grandstream GXP2020
		5. Grandstream GXP2120
		6. Polycom WX1500D
12	Support of Call Hold	1. Snom 300 (only when a Key is configured as Extension)

ANANT UCS Features supported with RTP/Direct RTP

ANANT UCS features that use Transcoding Channels

If RTP mode is set as RTP Relay or Direct RTP, the system will use Transcoding channels for the following features:

- Call Taping
- DISA Call
- Voice Mail
- Voice Mail Auto-Attendant

ANANT UCS features that use Transcoding channels when accessed from Extended IP Phones

If Extended IP Phones are connected as SIP Extensions and the RTP mode is set as RTP Relay or Direct RTP, system will use Transcoding channels for the following features:

- Conference
- Raid
- Interrupt Request
- Barge-In
- Conversation Recording

ANANT UCS features supported on Standard SIP Clients

If Standard SIP Phones are connected as SIP Extensions and the RTP mode is set as RTP Relay or Direct RTP, users can make internal calls, external calls as well as access the following features and facilities of ANANT UCS:

- Abbreviated Dialing (both Personal and Global)
- Call Taping
- CUG Calling
- Operator
- Voice Mail
- Emergency Number

Features that need to be handled locally by Standard SIP Clients:

- Hold
- Transfer
- Conference
- Call Toggle

Features at a Glance

Abbreviated Dialing

Personal/Global Abbreviated Dialing	***8-Location Code***
Program Personal memory	***1071-Location Code-Number-#*-TAC***

Account Code

Account Code by Number	***1058-Account Code***
Account Code by Name	***1059-Account Name***

Alarms

Once Only Alarm	***161-Hours-Minutes-1***
Daily Alarms	***161-Hours-Minutes-2***
Cancel Once Only/Daily Alarm	***161-#***
Set/Cancel Voice Guided Alarm	***163-Follow VMS Prompts***

Auto Call Back

Auto Call Back-On Busy	***2***
Auto Call Back-On No Reply	***2***
Cancel Auto Call Back	***102***

Auto Redial

Auto Redial	***17***
Cancel Auto Redial	***1070***

Barge-In

Barge-In	***4***

Call Cost Display

Last Ten dialed numbers Cost display	***1075***

Call Chaining

Call Chaining	***1050***

Call Forward

Call Forward-All Calls to Another Station	***131-Station/Department Group/VMS***
Call Forward-All Calls to External Station	***131-Trunk Access Code-Dest. Number-#****
Call Forward-If Busy	***132-Station/Department Group/VMS***
Call Forward-If Busy-All Calls to External Number	***132-Trunk Access Code-Dest. Number-#****
Call Forward-If No Reply	***133-Station/Department Group/VMS***
Call Forward-If No Reply-All Calls to External Number	***133-Trunk Access Code-Dest. Number-#****
Call Forward-If Busy or No Reply	***134-Station/Department Group/VMS***
Call Forward-If Busy or No Reply-All to External Number	***134-Trunk Access Code-Dest. Number-#****
Call Forward-Dual Ring	***1361***
Disable Call Forward-Dual Ring	***1360***
Cancel Call Forward	***130***

Call Forward - Scheduled

For Working Hours	
Call Forward-Scheduled - Unconditional	***1175-1-1-Destination Number***
Call Forward - Scheduled - Busy	***1175-1-2-Destination Number***
Call Forward - Scheduled - No Reply	***1175-1-3-Destination Number***
Call Forward - Scheduled - Busy/No Reply	***1175-1-4-Destination Number***
Call Forward - Scheduled - Set Dual Ring	***1175-1-5-1***
Call Forward - Scheduled - Cancel Dual Ring	***1175-1-5-0***
Call Forward - Scheduled - Cancel for Working Hours	***1175-1-0***
For Break Hours	
Call Forward-Scheduled - Unconditional	***1175-2-1-Destination Number***
Call Forward - Scheduled - Busy	***1175-2-2-Destination Number***
Call Forward - Scheduled - No Reply	***1175-2-3-Destination Number***
Call Forward - Scheduled - Busy/No Reply	***1175-2-4-Destination Number***
Call Forward - Scheduled - Set Dual Ring	***1175-2-5-1***
Call Forward - Scheduled - Cancel Dual Ring	***1175-2-5-0***
Call Forward - Scheduled - Cancel for Break Hours	***1175-2-0***
For Non-Working Hours	
Call Forward-Scheduled - Unconditional	***1175-3-1-Destination Number***
Call Forward - Scheduled - Busy	***1175-3-2-Destination Number***
Call Forward - Scheduled - No Reply	***1175-3-3-Destination Number***

Call Forward - Scheduled - Busy/No Reply	***1175-3-4-Destination Number***
Call Forward - Scheduled - Set Dual Ring	***1175-3-5-1***
Call Forward - Scheduled - Cancel Dual Ring	***1175-3-5-0***
Call Forward - Scheduled - Cancel for Non-Working Hours	***1175-3-0***
Cancel Call Forward - Scheduled for all Time Zones	***1175-0***

Call Forward - Department Group

Call Forward-Department Group - Unconditional	***1179-Department Group Number (Access Code)-1-Destination Number***
Call Forward-Department Group - Busy	***1179-Department Group Number (Access Code)-2-Destination Number***
Call Forward-Department Group - No Reply	***1179-Department Group Number (Access Code)-3-Destination Number***
Call Forward-Department Group - Busy/No Reply	***1179-Department Group Number (Access Code)-4-Destination Number***
Call Forward-Department Group - Cancel	***1179-Department Group Number (Access Code)-0***

Call Hold

Put the caller on Hold	***Flash***
Retrieve the caller	***Flash***

Call Park

To Park a Call	***115-Orbit Number***
To Retrieve the Parked Call	***116-Orbit Number***

Call Pick Up

Call Pick Up-General	***4***
Call Pick Up-Selective	***12-Station***

Call Toggle

Call Toggle (Toggle)	***Transfer-1 (for Extended IP phone)***

Call Transfer

Call Transfer to Station	***Speech with Station-Flash-Station (Transfer Target)-OnHook***
Call Transfer to Station (External Number)	***Speech with External Number-Flash-Station (Transfer Target)-OnHook***

Call Transfer to Trunk (External)	***Speech with External Number -Flash-TAC-External Number (Transfer Target)-OnHook***
Call Transfer to Trunk	***Speech with Station -Flash-TAC-External Number-Go OnHook***
Blind Transfer to Mail Box (VMS)	***Flash-1078-Station***

Calling Line Identification Restriction

Enable/disable CLIR	***103***

Cancel All Station Features

Cancel all station features	***1051***

Conference 3-Party

Conference 3-Party	***Flash-*3***

Conference Dial-In

Schedule a Conference	****19-1-Conference Number-Conference Password***
Initiate a Conference	****19-2-Conference Number-Conference Password***
Cancel a Conference	****19-0***

Conference Multiparty

To Temporarily Leave / Rejoin a Conference	***191***
Terminate Conference	***190***

Conversation Recording

Conversation Recording	***Flash-1095***

Department Call

Department Call	***Department Number (3901-3916)***

Direct Inward System Access

Enter DISA Mode	***1079-Station Number-User Password***

Do Not Disturb

Set Do Not Disturb with Text Message	***18-DND Message Number***
Cancel Do Not Disturb	***18-0***

DND Override	***4***

Dynamic Lock

Set Dynamic Timer	***142-User Password-Minutes***
Change Toll Control Level	***141-User Password-Level***

Emergency Conference

Make Emergency Conference	***1177-Department Group Number***
Cancel Emergency Conference	***While in speech with any one party, Go Off-Hook and dial 190***

Emergency Dialing

Dial Emergency Number	***Go Off-Hook and dial Emergency Number or Dial TAC-Emergency Number***

Floor Service

To access Floor Service	***38***

Follow Me

Set Call Follow Me	***135-Station-User Password***
Cancel Call Follow Me	***130***

Forced Answer

Forced Answer	***5***

Forced Call Disconnection

Forced Call Disconnection	***#****

Hotline

Set Hotline	***151-Station***
Set Hotline Timer	***154-Seconds***
Hot Outward Dialing	***152-Trunk Access Code***
HOD with Number	***153-Trunk Access Code-Number-#****
Cancel Hotline/HOD	***150***

Interrupt Request

Interrupt Request	***3***

Last Caller Recall

Last Caller Recall	***1092***

Last Number Redial

Last Number Redial	***7***

Live Call Supervision

Set Live Call Supervision	***1098-Destination Station***

Maid-In

Maid Status from a Room	***1054-Code***

Meet Me Paging

Meet Me Paging-Caller	***1093-Page Zone Number***
Meet Me Paging-Called Party	***1093-Station Number of the Caller***

Message Wait

Message Wait Set/Cancel	***1076-Station-Code***
Message Wait Retrieval	***1077***

Mini Bar

To check the utilities of the room	***1056-Item Number-Quantity***

Mute

Mute	***1052***

OG Trunk Bundle Group

To grab OG Trunk Bundle Group	***0/5/6-OGTBG Index***

Operator

Call to Operator	***9***

Paging

Paging	***1074-Page Zone Number***

Presence

Publishing Presence by extension user	***104-Password-Index No.***
To view Presence Status	***1097-Extension Number***

Raid

Raid	***5***

RCOC

RCOC in DISA Mode	***** on dial tone***

Reminder

Set Reminder	***162-DD-MM-YYYY-HH-MM***
Cancel Reminder	***162-#***
Set/Cancel Voice Guided Reminder	***164-Follow VMS Prompts***

Room Monitor

Room Monitor	***1073-Station***

Selective Port Access

Selective Port Access	***69-Port Type-Port Offset***

User Absent/Present

User Absent	***104-User Password-0***
User Present	***104-User Password-1***

User Password

Change User Password	***114-Old User Password-New User Password***

Walk-In Class Of Service

To Walk-In	***111-1-Your Extension Number-Your User Password***
To Walk-Out	***111-0***

SA Commands

Feature Name	Feature Number	Feature Code
Check-In	001	1072-901
Check-Out	002	1072-902
Guest Name	003	1072-903
Guest Group	004	1072-904
Guest-In/Out	005	1072-905
Guest Title	006	1072-906
Change Check-In Profile of Room	007	1072-907
Change Occupancy Status of Room / Extension	008	1072-908
Change Clean status of Room/Extension	009	1072-909
Room Shift	010	1072-910
Reprint Check Out Report	011	1072-911
Print Room Status Report	012	1072-912
Print Alarm Status Report	013	1072-913
Set DND-Remote	014	1072-001
Set Dynamic Lock settings – Remote	015	1072-002
Set Alarm –Remote	016	1072-003
Assign Call Budget to a station	017	1072-004
Assign/De-assign Mailbox to a Station/Department Group - Remote	018	1072-005
Set Call Forward – Remote	019	1072-006
Set Call forward for all stations-Remote	020	1072-007
Assign Station User Greeting Message	021	1072-008
Display & Acknowledge System Activity	022	1072-009
Display & Acknowledge System Fault	023	1072-010
Station Budget Display	024	1072-011
Change User password of a Station	025	1072-012
Lock/Unlock Keypad	026	1072-013
User Absent / Present	027	1072-014
Change SA password	028	1072-015
Change SA mode timer	029	1072-016
Set Day/Night mode	030	1072-018
Clear System Activity Log	031	1072-022
Start/Abort SAL in Offline mode	032	1072-023
Start/Abort SAL in Online mode	033	1072-024

Feature Name	Feature Number	Feature Code
Clear Mailbox for a range of extensions	034	1072-315
Cancel Dial in Conference	035	1072-026
Start/Abort SFL in Offline mode	036	1072-027
Start/Abort SFL in Online mode	037	1072-028
Start/Abort Online OG Report	038	1072-101
OG Print Filter: To Print calls terminated from SIP	039	1072-108
OG Print Filter: To Print calls Department Bill Group wise	040	1072-109
OG Print Filter: To print calls made on dates	041	1072-110
OG Print Filter: Print calls made between time	042	1072-111
OG Print Filter: To Print calls made to numbers matching with the numbers programmed in the Number List	043	1072-112
OG Print Filter: To Print calls of Duration more than this time	044	1072-113
OG Print Filter: To Print calls of Units more than the units programmed	045	1072-114
OG Print Filter: To Print calls made to account code	046	1072-115
Assign default OG Print filters	047	1072-120
Start/Abort offline report	048	1072-121
Enable/ Disable OG Schedule Reports	049	1072-122
Program Time for Daily OG Scheduled Reports	050	1072-123
Program Day and Time for OG Weekly Scheduled Reports	051	1072-124
Program Date and Time for OG Monthly Scheduled Reports	052	1072-125
Delete calls made on/from date	053	1072-132
To clear SMDR OG buffer	054	1072-133
Start/Abort Internal calls Report	055	1072-136
Set filter to print internal calls with duration greater than that given here	056	1072-138
Start/Abort Offline Internal Call Report	057	1072-141
Enable/ Disable Internal Scheduled Reports	058	1072-142
Program Time for Internal Daily Scheduled Reports	059	1072-143
Program Day and Time for Internal Weekly Scheduled Reports	060	1072-144
Program Date and Time for Internal Monthly Scheduled Reports	061	1072-145
To Clear SMDR Internal Buffer	062	1072-150
Start/Abort Online – IC Report	063	1072-151
Set filter to print all Normal calls	064	1072-152
Set filter to print all Auto Attendant calls	065	1072-153
Set filter to print all Unanswered calls	066	1072-154

Feature Name	Feature Number	Feature Code
Set filter to print all Auto Attendant unanswered calls	067	1072-155
Set filter to print all DISA calls	068	1072-156
Set filter to print all calls with speech duration more than timer	069	1072-157
Set filter to print all calls unanswered for duration more than timer	070	1072-158
Set filter to print all calls kept on hold for duration more than timer	071	1072-159
Set filter to print calls received from SIP	072	1072-166
Set filter to print all IC calls recd. On/from date	073	1072-167
Set filter to print all IC calls recd. At/from-to Time	074	1072-168
Set filter to print all IC calls recd. From nos. matching the External Number List	075	1072-169
Default IC Print filters	076	1072-170
Abort/Start IC Offline Report	077	1072-171
Enable/ Disable IC Scheduled Report	078	1072-172
Program Time for IC Daily Scheduled Reports	079	1072-173
Program Day and Time for IC Weekly Scheduled Reports	080	1072-174
Program Date and Time for IC Monthly Scheduled Reports	081	1072-175
Clear SMDR-IC buffer	082	1072-180
Enable/Disable Call Cost Display for a Station	083	1072-181
Start/Abort Hotel/Motel Activity log in Offline mode	084	1072-176
Start/Abort Hotel/Motel Activity log in Online mode	085	1072-177
Display and Acknowledge Hotel/Motel Activity	086	1072-178
Change Guest VIP Status of Station	087	1072-915
Change Phone Ringing Pattern of Room	088	1072-916
Print Reminder Status Report	089	1072-917
Remote Reminder	090	1072-033
Remote Voice Guided Alarm	091	1072-034
Remote Voice Guided Reminder	092	1072-035
Redirect Messages of a Station	093	1072-314
Enable/Disable Scheduled Alarm Report	094	1072-036
Program Time for Scheduled Alarm Report	095	1072-037
Enable/Disable Scheduled Reminder Report	096	1072-038
Program Time for Scheduled Reminder Report	097	1072-039
Request Database Synchronization to PMS	098	1072-040
Enable/Disable Scheduled Room Status Report	099	1072-041
Program Time for Schedule Room Status Report	100	1072-042
Enable/Disable Scheduled Change of Room Clean Status	101	1072-043

Feature Name	Feature Number	Feature Code
Program Time for Schedule Change of Room Clean Status	102	1072-044
Enable/Disable Internal Call Block For Guest Phones	103	1072-045
Software Version Revision Display of Master Card	104	1072-191
User Definable Fields	105	1072-920
OG Print Filter: To print calls originated from SIP	106	1072-188
Set Budget Type for SIP	107	1072-201
Program Budget Amount for SIP	108	1072-202
Program Free Minutes on SIP	109	1072-203
Call Budget Reset Mode for SIP	110	1072-216
Scheduled Date to Reset Call Budget Statistics on SIP	111	1072-217
Reset consumed Budget Amount /minutes on SIP manually	112	1072-218
Set/Cancel Scheduled Call Forward	113	1072-223
To Broadcast Message	114	1072-301
To record, playback and delete System Voice Prompts	115	1072-302
To record, playback and delete Mailbox Greetings for a Extension	116	1072-303
To record/playback Name of a station/department group	117	1072-304

Acronyms

ACB	Auto Call Back
ANT	Automatic Number Translation
APNS	Apple Push Notification Service
BH	Break Hour
BLF	Busy Lamp Field
CCC	Call Cost Calculation
CCWT	External Call Waiting Tone
CDC	Call Duration Control
CDR	Call Detail Record
CESID	Customer Emergency Services Identification Dialing
CLIP	Calling Line Identification and Presentation
CLIR	Calling Line Identification Restriction
COS	Class of Service
CPTG	Call Progress Tone (Generation)
CUG	Closed User Group
DDI	Direct Dialing-In
DHCP	Dynamic Host Configuration Protocol
DISA	Direct Inward System Access
DND	Do Not Disturb
DNS	Domain Name System
DST	Daylight Saving Time
DTMF	Dual Tone Multi Frequency
FCM	Firebase Cloud Messaging
FIFO	First In First Out
GND	Ground
IC	Incoming call
ICWT	Internal Call Waiting Tone
IMEI	International Mobile Equipment Identity
IP	Internet Protocol
IR	Interrupt
ISP	Internet Service Provider
ITU	International Telecommunication Union
LAN	Local Area Network

LCD	Liquid Crystal Display
LCR	Least Cost Routing
LED	Light Emitting Diode
LIFO	Last in First Out
MAC	Media Access Control Address
MOH	Music on Hold
NH	Non Working Hour
OG	Outgoing
PBX	Private Branch Exchange
PC	Personal Computer
PIN	Personal Identification Number
RBT	Ring Back Tone
RTC	Real Time Clock
SA	System Administrator
SAL	System Activity Log
SE	System Engineer
SFL	System Fault Log
SIP	Session Initiated Protocol
SMDR	Station Message Detail Recording
SP	Service Provider
TAC	Trunk Access Code
TAG	Trunk Access Group
TCP/IP	Transmission Control Protocol/Internet Protocol
TLG	Trunk Landing Group
VMS	Voice Mail System
VMAA	Voice Mail Auto Attendant
WAN	Wide Area Network
WH	Working Hour

Regulatory Information

Customer Information-ACTA

Using FAX Capability:

The Telephone Consumer Protection Act of 1991 makes it unlawful for any person to use a computer or other electronic device, including FAX machines, to send any message unless such message clearly contains in a margin at the top or bottom of each transmitted page or on the first page of the transmission, the date and time it is sent and an identification of the business or other entity, or other individual sending the message and the telephone number of the sending machine or such business, other entity, or individual. (The telephone number provided may not be a 900 number or any other number for which charges exceed local or long-distance transmission charges.) In order to program this information into your FAX machine, Refer user's Guide for the software of the Fax operation, such as 'Win Fax'.

Installation and Repairs

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. Repairs to certified equipment should be coordinated by a representative designated by the dealer/ supplier. Contact the support at Tech.Support@matrixcomsec.com

Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Regulatory Information for Terminals

FCC Part 15B ID : 2ADHNVP310 for SPARSH VP310

FCC Part 15B ID : 2ADHNVP330 for SPARSH VP330

FCC Part 15B ID : 2ADHNVP510 for SPARSH VP510

FCC Class B Information [131]

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications.

However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. this device may not cause harmful interference, and
2. this device must accept any interference received, including interference that may cause undesired operation.

131. Common for SPARSH VP310, SPARSH VP330, SPARSH VP510, SPARSH VP248, VP210.

SPARSH VP248

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 & 19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	SPARSH VP
Model Type:	SPARSH VP248
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above products

Conforms to the following product specification.

EMI/EMC Standard**:**

CISPR 22	: 2005-04 Edition 5.0
IEC 61000-3-2	: 2004-11 Edition 2.2
IEC 61000-3-3	: 2002-03 Edition 1.1
CISPR 24	: 1997 AMD2:2002
IEC 61000-4-2	: 2001-04 Edition 1.2
IEC 61000-4-3	: 2002-09 Edition 2.1
IEC 61000-4-4	: 2004-07 Edition 2.0
IEC 61000-4-5	: 2001-04 Edition 1.1
IEC 61000-4-6	: 2004-11 Edition 2.1
IEC 61000-4-8	: 2001-03 Edition 1.1
IEC 61000-4-11	: 2004-03 Edition 2.0

SAFETY
IEC 60950-1: 2001 Edition 1.0

Supplementary information:
The Product herewith complies with the following directives ;

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 26.05.2016

Comsec Pvt. Ltd. MATRIX Vadodara

CE

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Factory: 39-GIDC, Waghodia-391 760, Dist. Vadodara, India. Ph: +91 2668 262056/57, Fax: +91 2668 262631

SPARSH VP310

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	SPARSH VP
Model Type:	SPARSH VP310
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMCStandard:

EN 55022	: 2010 (Consolidated with Corrigendum (AC) :2011)
IEC 61000-3-2	: 2014
IEC 61000-3-3	: 2013
EN 55024	: 2010
IEC 61000-4-2	: 2008 (Edition 2:2008-12)
IEC 61000-4-3	: 2006 (Edition 3:2006 Consolidated with Am1:2007 & A2:2010)
IEC 61000-4-4	: 2012
IEC 61000-4-5	: 2014
IEC 61000-4-6	: 2013
IEC 61000-4-8	: 2009 (Edition 2: 2009-09)
IEC 61000-4-11	: 2004 (Edition 2:2004)

SAFETY Standard:

IEC 60950-1 : 2005(Edition 2) + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 28.09.2017

Matrix Comsec Pvt. Ltd. MATRIX Vadodara

CE

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Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

SPARSH VP330

TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	SPARSH VP(INTUITIVE TOUCH-SCREEN IP PHONE)
Model Type:	SPARSH VP330
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standard:

EN 55022 : 2010 (Consolidated with Corrigendum (AC) :2011)
IEC 61000-3-2 : 2014
IEC 61000-3-3 : 2013
EN 55024 : 2010
IEC 61000-4-2 : 2008 (Edition 2:2008-12)
IEC 61000-4-3 : 2006 (Edition 3:2006 Consolidated with Am1:2007 & A2:2010)
IEC 61000-4-4 : 2012
IEC 61000-4-5 : 2014
IEC 61000-4-6 : 2013
IEC 61000-4-8 : 2009 (Edition 2: 2009-09)
IEC 61000-4-11 : 2004 (Edition 2:2004)

SAFETY Standard:

IEC 60950-1 : 2005 + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 28.09.2017

MATRIX COMSEC PVT. LTD.
Registered/Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • www.MatrixComSec.com
Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • CIN: U72200GJ1998PTC034047

SPARSH VP510

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 &19-GIDC, Waghodia, Dist: Vadodara 391760 Gujarat, India
Declares that the product/s Product:	Premium IP Phone
Model Type:	SPARSH VP510
Trade Name:	MATRIX
Product Options:	This declaration covers all options of the above product, conforms to the following product specifications:

EMI/EMC Standard:

EN 55022 : 2010
IEC 61000-3-2 : 2014
IEC 61000-3-3 : 2013
EN 55024 : 2010
IEC 61000-4-2 : 2008 (Edition 2:2008-12)
IEC 61000-4-3 : 2010
IEC 61000-4-4 : 2012
IEC 61000-4-5 : 2014
IEC 61000-4-6 : 2013
IEC 61000-4-8 : 2009 (Edition 2: 2009-09)
IEC 61000-4-11 : 2004 (Edition 2:2004)

SAFETY Standard:

EN/IEC 60950-1 : 2006(Edition 2.0) + A1:2009 + A2:2013

Supplementary Information's:
The Product herewith complies with the following directives :

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Mr. Ganesh Jivani
Director
Date: 28.09.2017

Matrix Comsec Pvt. Ltd. Vadodara

CE

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Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

SPARSH VP210

MATRIX
TELECOM | SECURITY

EU DECLARATION OF CONFORMITY

EU DECLARTION OF CONFIRMITY

Manufacture : : MATRIX COMSEC PVT LTD

Manufacture Address : 15 & 19- GIDC , Waghodia, Vadodara-391760 (Gujarat, India)

Trade Name : **MATRIX**

Declare that the DoC is issued under our sole responsibility and belongs to the following products;

Product : **SPARSH VP**

Model/ TYPE : SPARSH VP210

Essential Requirements /Directives		Applied Specifications/ Standards
EMC	2014/30/EU	EN 55032: 2015+A11:2020; EN 55035:2017+A11:2020; EN 61000-3-2: 2019; EN 61000-3-3: 2013+A1:2019; EN 61000-4-2: 2009; EN 61000-4-3: 2006+A2:2010; EN 61000-4-4: 2012; EN 61000-4-5: 2014+A1:2017; EN 61000-4-6: 2014; EN 61000-4-8: 2010; EN 61000-4-11: 2020
LVD/SAFETY	2014/35/EU	IEC 62368-1: 2018
RoHS (RoHS2)	2011/65/EU	EN 50581: 2012

I hereby declare that the equipment named above has been designated to comply with the relevant section of the above reference standards and meet all essential requirements of the specified directives.

Matrix Comsec Pvt. Ltd. Vadodara MATRIX

Mr Ganesh Jivani
Managing Director
Date: 04/11/2020

MATRIX COMSEC PVT. LTD.
Registered/Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • www.MatrixComSec.com
Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • CIN: U72200GJ1998PTC034047

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- The firmware of this product also includes some of the Open-Source software released under GNU General Public License (GPL) Version 2. Terms of this license is printed in full below.
- The source of the open source software used in this product is available on CD, upon written request from:

  R&D Team
  Matrix Comsec Pvt Ltd
  394, Makarpura GIDC,
  Vadodara - 390 010
  Gujarat
  India.
  Customer shall bear the shipping and handling charges.

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