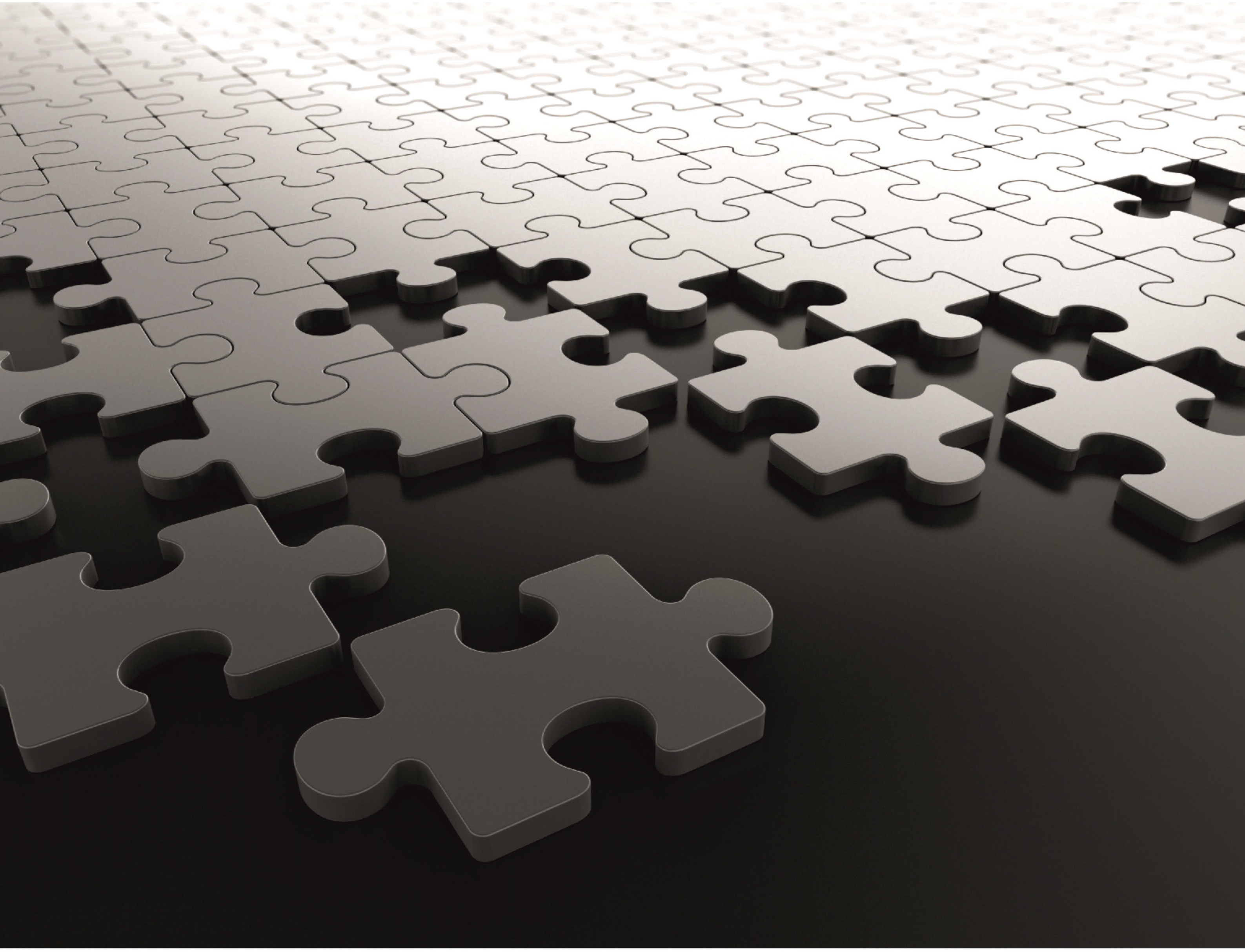
MATRIX SPARSH VP510
User Guide



TELECOM SOLUTIONS

SPARSH VP510

The Premium Standard SIP Phone

User Guide

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

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CHAPTER 1

Introduction

Thank you for choosing the Matrix SPARSH VP510! You have now entered the exciting world of Internet Telephony. We hope you will make optimum use of this intelligent, feature-packed VoIP SIP phone. Please read this document carefully before installing the SPARSH VP510.

Intended Audience

This User Guide is aimed at:

- **Network engineers and network administrators**, who will install, maintain and support the SPARSH VP510. It is assumed that they have some experience in installing phones, are familiar with VoIP technology, how it works, and the various technical terms and functions associated with it.

- **End users**, persons/organizations who will actually use the SPARSH VP510. They include residential consumers, personnel of small and medium businesses, large enterprises, other commercial and public organizations/institutions.

  It is assumed that the End user has some previous experience in operating a key phone. End users are not expected to configure the phone or program its features, but only learn to operate the phone. However, it is anticipated that some of them may have to or want to configure the phone and program the features. Therefore, this document provides instructions on installation and configuration of the phone in as lucid a manner as possible.

Organization of this Document

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

You may use the **Table of Contents** to navigate through this document to the relevant topics or information you want to look up. **Cross-references** are provided in blue fonts with hyperlinks. You can look up the source by clicking the links.

The instructions in this document are written in step-by-step format and contains the following sections:

- **Introduction**: gives an overview of this document, its purpose, intended audience, organization, terms and conventions used to present information and instructions.

- **Know your SPARSH VP510**: describes the phone hardware (LCD, keypad, ports and connectors) and software (Phone User Interface and Web User Interface).

- **Connecting SPARSH VP510:** provides you step-by-step instructions to connect the phone.

- **Phone Home Screen:** provides the information of the various elements on the Idle Screen
- **Configuring SPARSH VP510:** provides instruction to configure the basic settings of the phone.
- **Customizing Your SPARSH VP510:** provides instruction to customize the phone settings according to your country specific requirements and configure parameters such as display, language, volume, time format etc.
- **Making Calls:** explains different ways for making calls.
- **Receiving Calls:** explains receiving calls and the actions you can take.
- **Call Screen Functionality:** explains the call screen elements.
- **Making a Second Call:** explains how you can make a second outgoing call from your phone.
- **Receiving a Waiting Call**: explains how you can handle a second incoming call on your phone.
- **Call Features:** gives step-by-step instructions on using basic features for managing calls.
- **Contacts:** gives step-by-step instructions on accessing and using the contacts.
- **Call Logs:** gives step-by-step instructions on accessing and using the call logs.
- **Advanced Features:** describes in detail the advanced features of the phone; gives step-by-step instructions on how to configure and use these features.
- **System Parameters:** provides the instructions to configure parameters that have system-wide implications.
- **Certificate Management:** allows you to generate the certificates and upload the same.
- **Maintenance:** provides details about maintenance of the phone, including firmware up-gradation and advanced maintenance instructions.
- **Status:** displays the status of the SIP Trunks, Network as well as the System.

Notices

The following symbols have been used to draw your attention to important items.

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 phone.*

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

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

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

Terminology

Throughout this User Guide, the terms “**Phone”** are used synonymously to denote SPARSH VP510. Only for phone specific features the respective phone name is mentioned.

Some specific terms used in this User Guide are defined below:

- **Calling party:** the person making a call.
- **Called party:** the person receiving a call.
- **User:** the person who is in possession of the phone and uses it.
- **Remote user/Remote party/Remote end/Far end:** the person with whom the user interacts.
- **Transferee:** The remote user whose call is to be transferred (first party) to another remote user (second party).
- **Transfer Target:** The remote user to whom the call is to be transferred (second party).
- **Transferor:** The user who transfers the call.

Additional Information

If you encounter any technical problems or have any issues regarding the System, please contact your Dealer/ reseller or the Matrix Customer Care.

The documentation can be found at https://www.matrixcomsec.com/support/telecom-product-manuals/

For product registration and warranty related details, please visit https://www.matrixcomsec.com/warranty/#telecom

CHAPTER 2

Know Your SPARSH VP510 - Standard SIP Phone

SPARSH VP510, the Entry Level IP Phone sets the benchmark for quality performance with elegant design and crystal-clear voice. SPARSH VP510 features a 128 x 64 Graphical LCD Display, SIP Line Keys, High Quality speaker-phone and high definition audio quality.

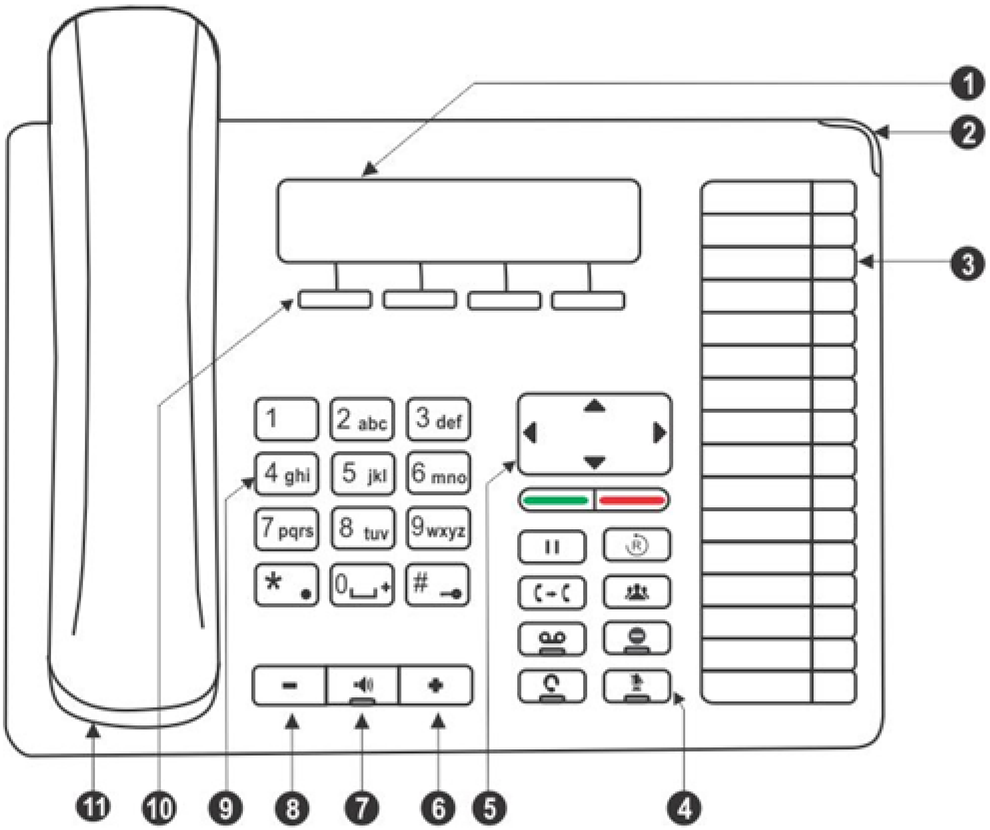
Engineered to deliver full feature access of the System, SPARSH VP510 acts as face of your communication system covering wide array of business environments.

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

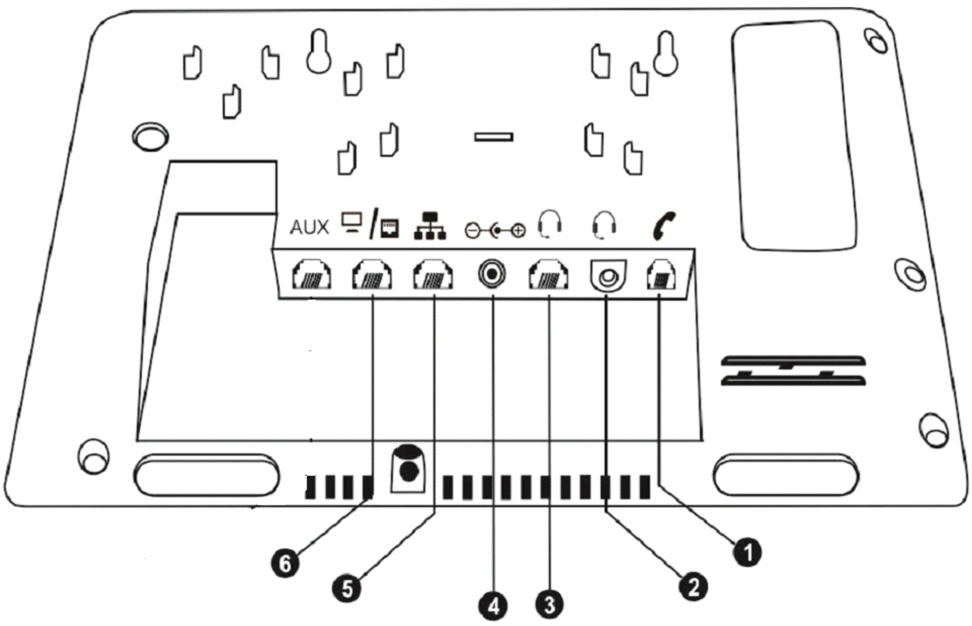
- Minimum 240*64 pixel and above graphical LCD with Back light
- Built-in 16 DSS Keys for Feature, Line, Extension
- HD Voice, HD Handset, HD Speaker
- 4 Context Sensitive Keys
- 5 Fixed Features Keys (With LED) — Voicemail, Headset, Mute, DND, Speaker
- 4 Fixed Function Keys (Without LED) — Hold, Conference, Redial, Transfer
- Integration with ITSP/IP-PBX over SIP Protocol
- HTTP Auto Provisioning
- 16 Direct Station Selection Keys
- 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

Front View



1	LCD Screen
2	Ringer LED
3	DSS (Direct Station Selection) Keys
4	Fixed Feature 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

Bottom View



1	RJ9 Connector to connect Spring Cord of the Handset
2	3.5 mm Audio Jack connector to connect the Headset
3	RJ9 Connector to connect the Headset
4	DJ Jack to connect the Power Adapter (Optional)[a]
5	RJ45 Connector to connect the Router/LAN Switch
6	RJ45 Connector to connect the PC/Computer

a. 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/2A) only.

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. Refer “Customizing Your SPARSH VP510” for details.

Ringer LED

The Ringer LED will glow in Blue (1 sec ON – 500 msec OFF) to indicate incoming 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 (when phone is in the idle state).

- **Down Key:** To scroll downwards while navigating the Menu/sub-menu or to access Network Parameters (when phone is in idle state).

- **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.

- **Cancel Key** **:** To navigate to a previous menu screen.

Fixed Feature Keys

There are 8 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
	Hold	No
	Redial	No
	Transfer	No
	Conference	No
	Voicemail	Single Color - Blue
	Do Not Disturb	Single Color - Blue
	Headset	Single Color - Blue
	Mute	Single Color - Blue

Context Sensitive Keys

SPARSH VP510 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, Call Screen, Transfer Dial Screen, all have different set of features that can be accessed. SPRASH VP510, 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.

To know more, refer to "Context Sensitive Keys (CSK) Programming".

DSS (Direct Station Selection) Keys

There are 16 DSS Keys which that can be assigned to Stations (Extensions) and Trunks and important or frequently accessed features. For example, to dial an extension number you just need to press the DSS Key assigned to that extension and the call will be placed automatically.

To know more, refer to "DSS Keys Programming".

SIP Trunk Status

By default, DSS Keys are assigned to the SIP Trunks. SIP Trunk1 is assigned DSS1 and SIP Trunk2 is assigned DSS2. For details refer to "DSS Keys Programming".

When the DSS Keys are assigned to the SIP Trunks, the LEDs of these Keys displays the Status of the SIP Trunks as follows:

Status of SIP Trunk	Indication
Disabled	OFF
Active/Registered	Continuous ON (Blue)
Not Active	Fast Blink (Red)

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 parties.

You can also configure these for quick access to Contacts. For more information refer to "Speed Dial".

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[1] 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. Make sure you have enabled the **Use Headset** option, refer "Accessories". 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.

Phone Menu

You can access the following features from the Menu of the phone:

Menu option	Description
Contacts	To add, edit, delete names and numbers of contacts.
Call Logs	To view call history of Missed, Answered, Dialed and Rejected calls.
Call Forward	To set/ cancel Call Forward - Always , Busy, No-reply.
Voicemail	To access the Mailbox.
Do Not Disturb	To set/cancel Do Not Disturb on the phone, that is, block incoming calls.
LDAP	To access the contacts of the directory server.
Keypad Lock	To lock the keypad of the phone.
Intercom	To configure and access Intercom.
Auto Answer	To set/cancel Auto Answer.
Hotline	To set/cancel Hotline and Delayed Hotline.
Anonymous call rejection (ACR)	To set/cancel ACR.
CLIR	To set/cancel CLIR.
Settings	To change the following settings: • Network Settings: To change the Network Settings. • SIP Trunk: To configure the SIP Trunk parameters. • Phone Settings: To customize settings of the phone — Volume, Ringer, Display, Time Format, Accessories, Language • Password: To change User/Configuration Password. • Feature: To configure settings of features — Voicemail, Call Waiting, Intercom, Auto Lock. • Speed Dial: To configure the Speed Dial settings.
Phone Info	Displays the phone information.
System Usage	Displays information related to the usage of the phone — Uptime, CPU, RAM, Flash.

1. *Make sure you have connected a compatible Headset to the phone.*

When the phone is in idle state,

- Press the Down Key V , to access the Network Settings.
- Press the Up Key Λ , to access the Phone Settings.

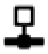
Navigating the Phone Menu

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.
- Press Cancel Key, to navigate to a previous menu screen.

Icon Instruction

Icons appearing on the LCD screen are described in the following table:

Icons	Description
	Network is unavailable
2aB	Alphanumeric
123	Numeric input mode
	Call Forwarded/Forwarded Calls
	Call Hold
	Keypad Lock
	Incoming Calls
	Placed Call/Ongoing Call
	Missed Calls

CHAPTER 3

Getting Started with SPARSH VP510 - Standard SIP Phone

Package Contents

- SPARSH VP510 Phone
- Handset with Handset Cord
- Ethernet Cable
- Foot Stand
- Wall Mounting Template
- Screws and Screw Grips

When you unpack the SPARSH VP510 box, please verify whether the above items are present in the package.

Check the contents for damage. In case any of the above listed items is missing, damaged, or faulty, contact the dealer/reseller. Do not discard any of the package contents or packing materials. For product registration and warranty related details, please visit https://www.matrixcomsec.com/warranty/#telecom

Get your Internet Connection ready

To install SPARSH VP510 you must have:

- Broadband Internet Connection to make/receive calls through Public Internet. If you want to make calls within your network, you do not need an Internet connection.

- A SIP Account with an Internet Telephony Service Provider (ITSP)/IP-PBX. If you want to make Peer-to-Peer calls (i.e. calls made without the intervention of a SIP/Proxy server), you do not need the services of an ITSP/IP-PBX.

Get your Network Information ready

Ask your LAN Administrator/ISP for:

- IP Address
- Subnet Mask
- Gateway Address
- DNS Address

Ask your ITSP for:

- SIP ID/User ID
- Authentication User ID (in most cases same as SIP ID)
- Authentication Password

- Registrar Server Address
- Registrar Server Port

Protecting the Phone and Yourself

Power Supply

- Before you connect the phone to its power source, please read the installation instructions, mentioned in the Quick Start as well as “Connecting SPARSH VP510”.

- The phone can be powered from an AC supply or from the LAN network (PoE).

- If you power the phone from an AC supply, purchase the power adapter from Matrix. The use of any third-party power adapter may cause damage to the phone. Damages to the phone caused by using other power adapters are not covered by Matrix warranty.

- Check the voltage of the AC supply. It must be between 100-240 VAC, 47-63Hz.

- The electric plug and socket must be easily accessible to you at all times so that you can disconnect power from the device, quickly. Remember, the phone does not have a power switch. The only way to disconnect it is to plug out the power supply.

- The power supply must be placed indoors.

- If you power the phone from the LAN network (Power over Ethernet), ensure that the Ethernet switch to which the phone is connected supplies power complying with IEEE 802.3af.

- 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.

Using External Devices

When using external devices like the headset, cables, connectors with the phone, always ensure that they are of good quality, so that phone's performance is not affected.

Matrix does not guarantee the performance of external devices with the phone, as it has no control over the quality of external devices, cables and connectors.

Cleaning the Product

Use a lightly moistened tissue paper or cloth towel to clean the phone surface.

Do not spray or pour cleaning solution directly on the phone as this may cause damage to the phone.

Preparing for Disruptions in Power Supply and Internet Connectivity

You will not be able to make calls during a power outage. All current calls will be disconnected, and any changes you make in the configuration of any phone/feature/network settings will not be saved, if you have not already saved the settings before the power outage.

Use an un-interruptible power supply (UPS) with your VoIP installation to be able to use the phone during power outage.

Dialing Emergency Services

You will not be able to dial through the phone, whenever there is a disruption in power supply and Internet connectivity. Ensure that you have another traditional phone line accessible to you always so that you have immediate access to Emergency Services.

Disposing the Product

This product must be disposed according to the national laws and regulations prevailing in the country where it is installed.

Avoiding Discomfort

To avoid strain or discomfort to your body:

- Place the phone where it is most convenient for you to reach it, without straining any part of your body.
- Do not cradle the handset between your ear and shoulder; use the headset instead.
- Do not expose yourself continuously to loud sounds; keep the volume of the handset receiver and headset at a moderate level.

Protecting Against Security Threats/Risks

As VoIP is a form of communication over the internet, the security threats and risks associated with VoIP are very similar to those inherent to any internet application. Like spam and phishing are common forms of email abuse, Spam over Internet Telephony (unsolicited calls and voice mail), and Spoofing (attacker masquerading as a known or trusted source to trick the receiver into disclosing important and confidential personal information) are common threats in VoIP. Confidentiality of the conversation is another concern. VoIP data sometimes travels unencrypted and it is possible that someone may collect the VoIP data and reconstruct a conversation. Though at present such activity may be a rare occurrence, it may increase as the deployment of VoIP spreads wider. Educate yourself further on the security risks involved in using VoIP and how to protect yourself.

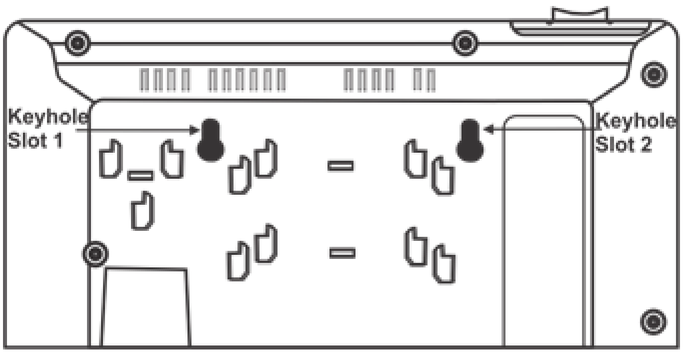
CHAPTER 4 **Connecting SPARSH VP510**

Connecting the Phone

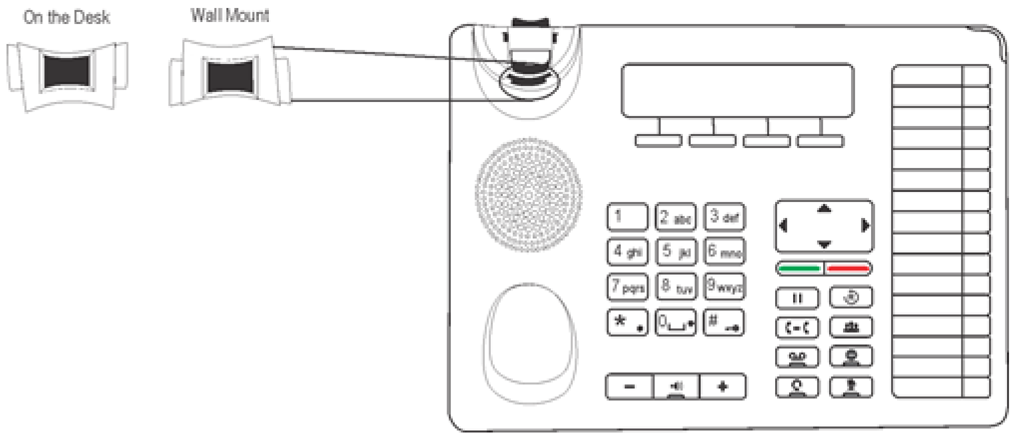
- Unpack the box and verify the package contents. See "Package Contents".
- You can mount the phone on the wall or desk at a convenient location.

Mount SPARSH VP510 on the 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.



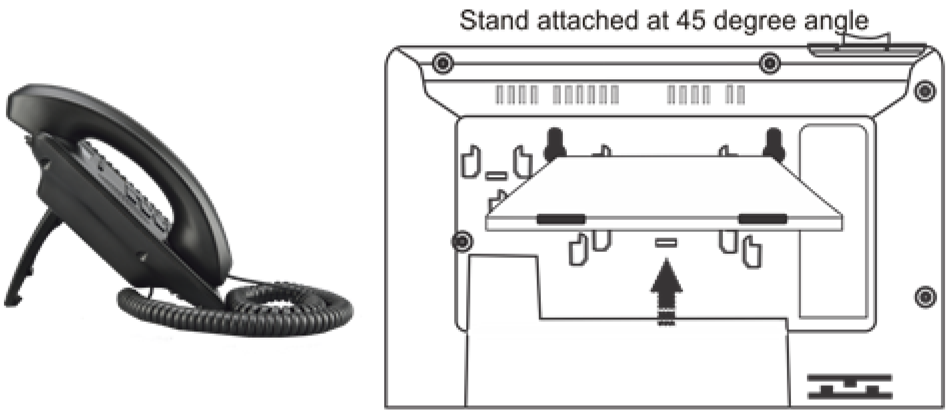
- Now, mount the phone with the screws into the Keyhole Slots.
- Reverse the handset wall mount tab to make sure that the handset remains intact when you mount the phone. Push the handset wall mount tab upwards to remove it from the slot. Rotate it 180 degrees clockwise and push it downwards into the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

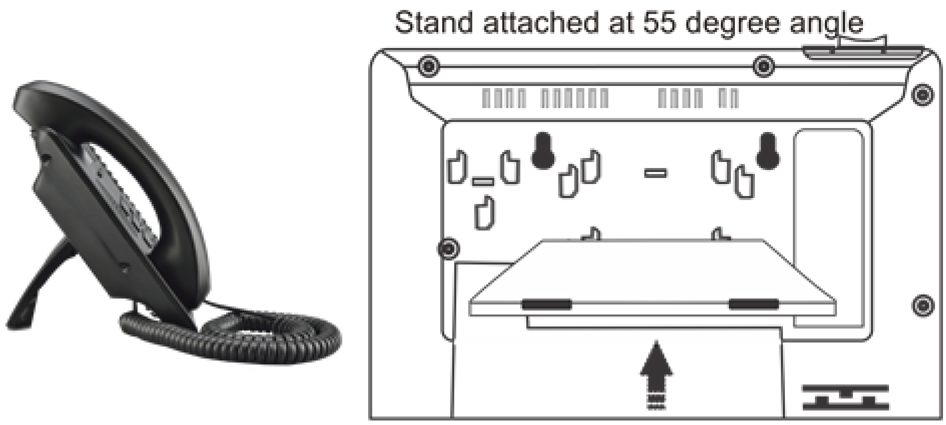


If you are unable to remove the wall mount tab, you may use a tool like a minus screw-driver to remove it.

Mount SPARSH VP510 on the Desk,

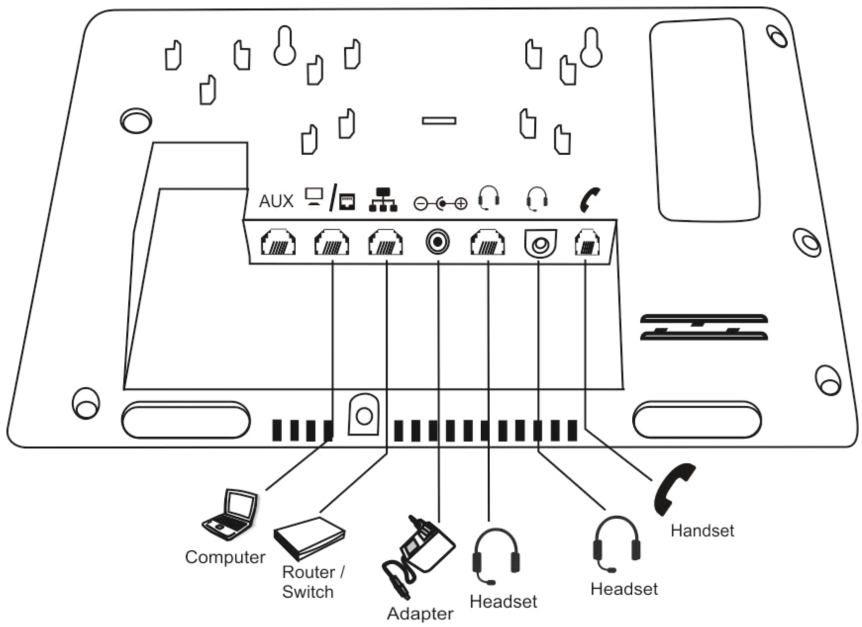
- You can attach the Foot Stand in the following ways — at an angle of 45 degrees or 55 degrees





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

Refer to the diagram below for connectivity.



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.

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 .

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.

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.

Connect the Power Supply

- 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, 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/2A) only. The use of any third-party Power Adapter may cause damage to the phone.

- Switch ON Power Supply.

The SPARSH VP510 Standard SIP Phone can be converted into SPARSH VP510 Extended SIP Phone, if required. To know more, refer “Converting SPARSH VP510 Standard SIP Phone to SPARSH VP510 Extended SIP Phone”.

Powering On

After your phone is powered on:

- By default, the phone will boot in the Extended mode. To convert the mode to Standard, refer to “Converting SPARSH VP510 Extended SIP Phone to SPARSH VP510 Standard SIP Phone”.

SPARSH VP510 with Serial Number: 100160001 and onwards only can be converted to Standard SIP Phones.

- DHCP is enabled on the IP phone by default with which the phone functions as a plug and play device. The phone attempts to contact a DHCP server to obtain valid network settings (e.g., IP address, subnet mask, default gateway address, DNS address and Server Address).
- If you need to change the network parameters of the IP phone manually, refer to “Configuring SPARSH VP510” and “Network Parameters” for instructions.
- For calling you must configure the SIP Trunks. For instructions refer to “Configuring SPARSH VP510” and “SIP Trunk”.
- After the phone starts successfully, the Home Screen appears.

- You may adjust the LCD for brightness and backlight. For instructions, see “Customizing Your SPARSH VP510”.

Converting SPARSH VP510 Extended SIP Phone to SPARSH VP510 Standard SIP Phone

To convert the SPARSH VP510 Extended SIP Phone to SPARSH VP510 Standard SIP Phone, follow the steps given below:

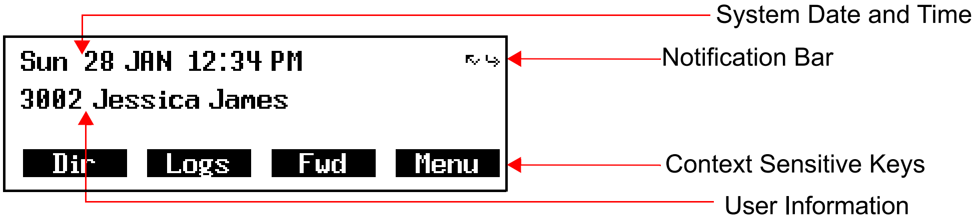
- When the Phone is powered on and the Loading screen appears press #2.

- The message **"Do you want to start in Standard SIP Mode?"** appears.
- Press **Yes** Key.
- The phone will reboot and start as a Standard SIP Phone.The Factory Default values for this mode will be assigned to all the parameters.

CHAPTER 5

Phone Home Screen

The Home screen displays the Name and /or Number, System Date and Time, Notification Bar and the Context Keys.



Name	Description
User Information	Displays your Name and/or Number. icon appears beside the Name (extreme right) when the LAN link is down.
System Date and Time	Displays the System Day, Date and Time.
Notification Bar	The respective icons appear in the Notification Bar when you set the below mentioned features. When Call Forward is set. Refer "Call Forward". When you have missed calls. Refer "Call Logs".

Name	Description
Context Keys / Feature Keys	By default Dir, Logs, Fwd and Menu features are assigned to these keys. You can change the priorities of the features/functions assigned to these keys. To know more, refer to "Context Sensitive Keys (CSK) Programming". Dir Press to make a call by dialing a Name. Refer "Making Calls". Logs Press to view the list of Call Logs. Refer "Call Logs". Fwd Press to set Call Forward. Refer "Call Forward". Menu Press to access the Menu of the Phone.

CHAPTER 6

Configuring SPARSH VP510

Configuring Methods

The IP phones can be configured manually via the Phone User Interface and/or the Web User Interface.

Phone User Interface

An administrator or a user can configure and use the IP phones via Phone User Interface. You can customize your phone using the Menu to access the Phone User Interface.

The default User Password is “1234“. All features are not accessible from the Phone User Interface.

For more information on customizing your phone with the available options from the Phone User Interface, refer “Customizing Your SPARSH VP510”.

Web User Interface

In addition to the Phone User Interface, you can also customize your phone via Web User Interface. In order to access the Web User Interface, you need to know the IP address of your phone. Make sure the Phone and PC from which you need to access the Web User Interface are in the same network.

The Phone has an in-built web server for configuring the phone and offers a Graphical User Interface (GUI), Jeeves for the same.

Default Static IP Address is 192.168.001.021 and the default Configuration Password is 1234.

We have tested the Web User Interface in the following Browsers and Versions:

- *Google Chrome 88.0.4324*
- *Microsoft Edge 88.0.705.50*
- *Mozilla Firefox 85.0*
- *Safari 15 (15.6.1)*

Configuring Basic Settings

To use Jeeves, make sure you have connected the phone with a PC. If you do not have a PC on your desk to connect the phone with, you can grab any PC in the same LAN network as your phone.

Make sure your PC is in the same Subnet as SPARSH VP510. Change the Subnet Mask of the PC, if necessary.

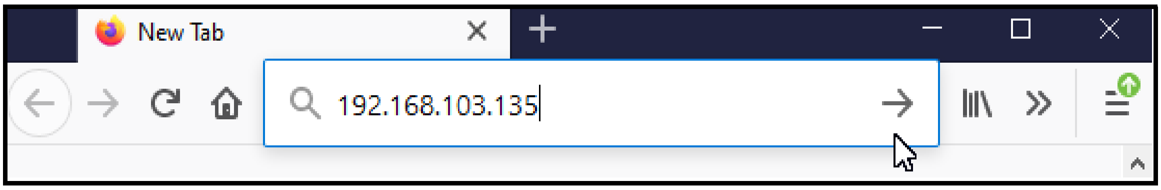
Default Static IP Address: 192.168.001.021
Default Static Subnet: 255.255.255.0
Default Static Gateway: Blank

If you want to access Jeeves *from a LAN PC make sure the IP Address of SPARSH VP510 does not conflict with any other device on the LAN and is in the same subnet as the LAN PC. For assistance consult your LAN Administrator.*

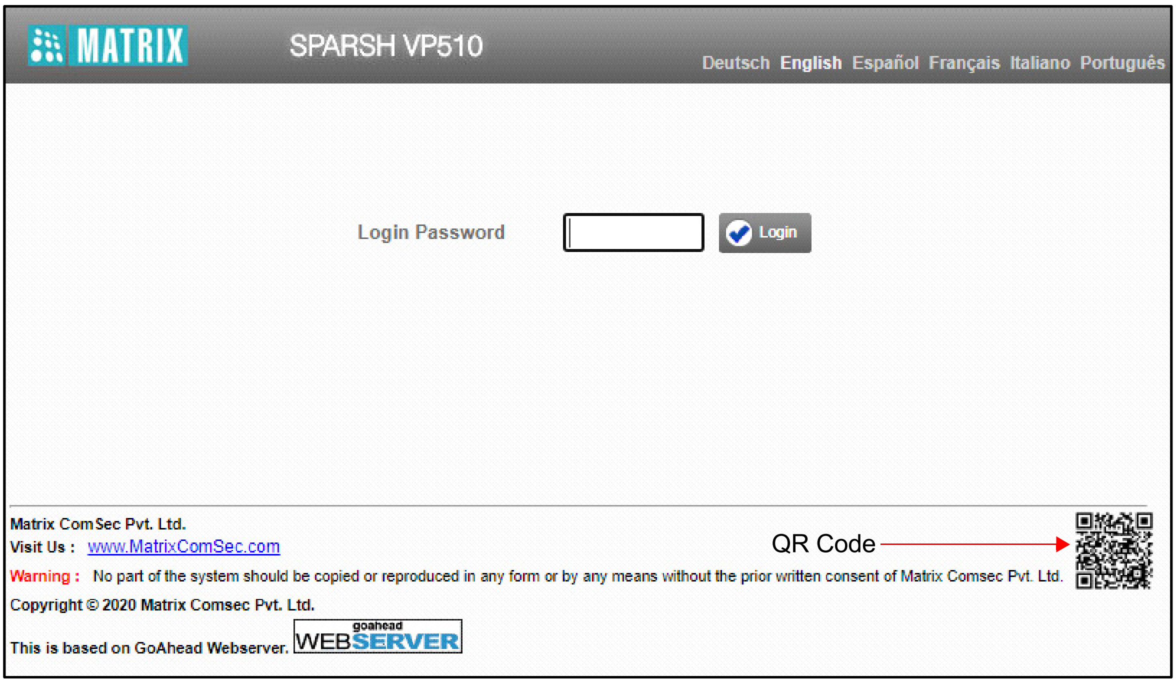
To access and use the Jeeves,

- Open the Web browser [Google Chrome 88.0.4324, Microsoft Edge 88.0.705.50, Mozilla Firefox 85.0, Safari 15 (15.6.1)].

- Enter the current IP Address of the phone in the address bar of the browser.

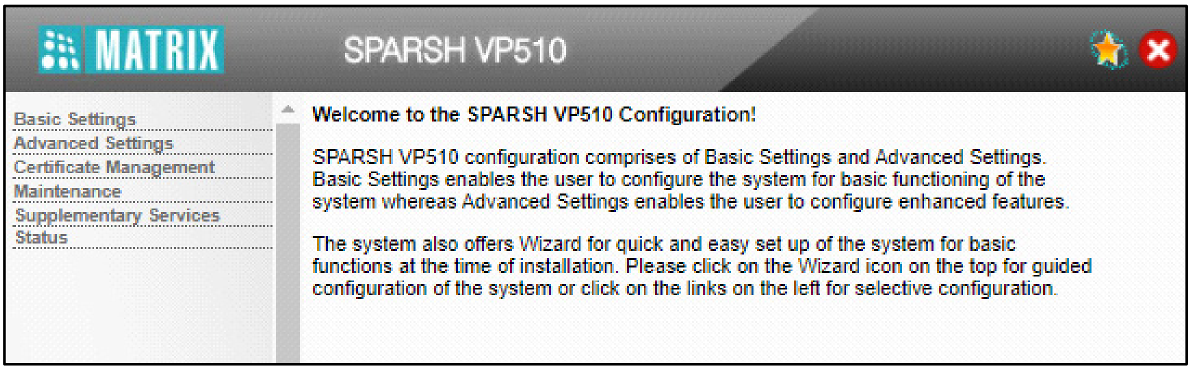


- The login page appears.



- Log into the Jeeves using the **Configuring Password**. Default: 1234. The Home page appears.

You can scan the QR Code to download the documents.



For the ease of installation, as well as to simplify and speed up the process of setting up the SPARSH VP510, the Jeeves offers **Basic Settings.**

Using Basic Settings, you can complete as much as 80 percent of the phone configuration, covering all the basic parameters necessary for the functioning of the phone. To configure advanced features and facilities, you may use **Advanced Settings**.

Detailed information on Basic Settings and instructions for using it for phone configuration are given in the manual later.

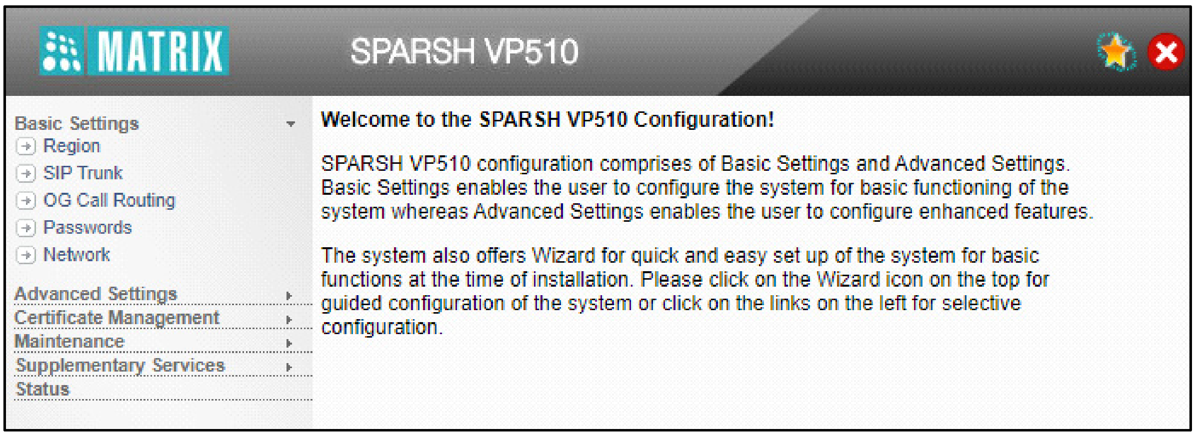
While the Basic Settings provide a fast-track way for configuring the phone, **Advanced Settings**, as the title itself suggests, enable you to configure all the (configurable) parameters of the phone, excluding those under Basic Settings.

The links **Basic Settings**, **Advanced Settings**, **Certificate Management, Maintenance, Supplementary Services** and **Status**, appear on the left pane.

To configure the Basic Settings,

- Click **Basic Settings**.

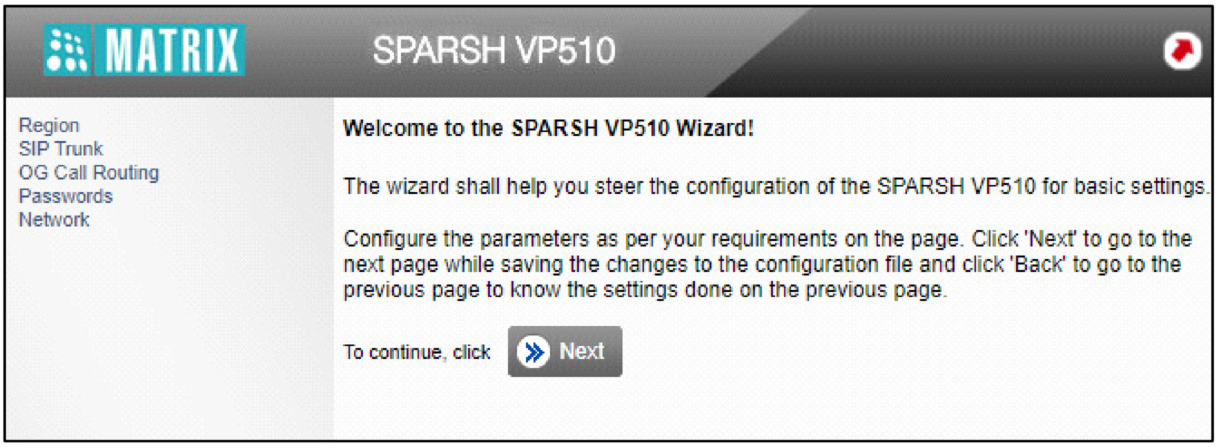
The links to the different basic parameters appear on the left pane.



You may either use the *Wizard* to guide you through the configuration or selectively configure the *Basic Settings* pages.

When you use the *Wizard*,

- Click the Wizard icon on the top right of your screen.



- The **Next** button takes you to the next page, saving the changes you made on the current page.
- The **Back** button returns you to the previous page.
- The **More** button expands parameters on the page.
- The **Less** button collapses parameters on the page.
- The **Default** button assigns factory set values to all the parameters on the page.
- The **Quit** button enables you to exit the Wizard at any stage, saving changes you made before exiting.

When you use selective configuration,

- Click **Basic Settings**.
- Click each parameter link, **Region**, **SIP Trunk**, **Outgoing Call Routing**, **Passwords**, **Network**.
- The selected parameter page opens.
- Set the desired values on the page.
- Click **Submit** to save your settings on the page.

- *The configuring password remains the same for configuring the phone via the Menu and through Jeeves. You can change the configuring password. Please refer the topic "Passwords", for instructions on how to change the configuring password).*

- *Your login session will expire, if* Jeeves *remains idle (no configuration task is performed) for 60 minutes. You must log into* Jeeves *again.*

The instructions provided, describe *selective* configuration of the Basic Settings pages.

- Click **Basic Settings**.

- The following links appear under Basic Settings:
  - Region
  - SIP Trunk
  - Outgoing Call Routing
  - Passwords
  - Network

Region

You can configure the Region using the Web User Interface only.

Configuring Region via Web User Interface

- Log into Jeeves.
- Under **Basic Settings**, click **Region.**

- From the Region drop-down list, select the country where the phone is installed. Default: India.
- Click **Submit**, to save the settings.
- Click **Default**, to set the default Region.

SIP Trunk

SPARSH VP510 supports two SIP Trunks, allowing you to subscribe to two SIP Accounts from the same or from different Internet Telephony Service Providers (ITSP).

You can configure the SIP Trunks using the Web User Interface and certain parameters can also be configured using the Phone User Interface. The Status of the SIP Trunks will be displayed on the Home screen. You can make calls using these trunks. Refer to "SIP Trunk Status".

By default, DSS Keys are assigned to the SIP Trunks. SIP Trunk1 is assigned DSS1 and SIP Trunk2 is assigned DSS2. For details refer to "DSS Keys Programming".

When the DSS Keys are assigned to the SIP Trunks, the LEDs of these Keys displays the Status of the SIP Trunks as follows:

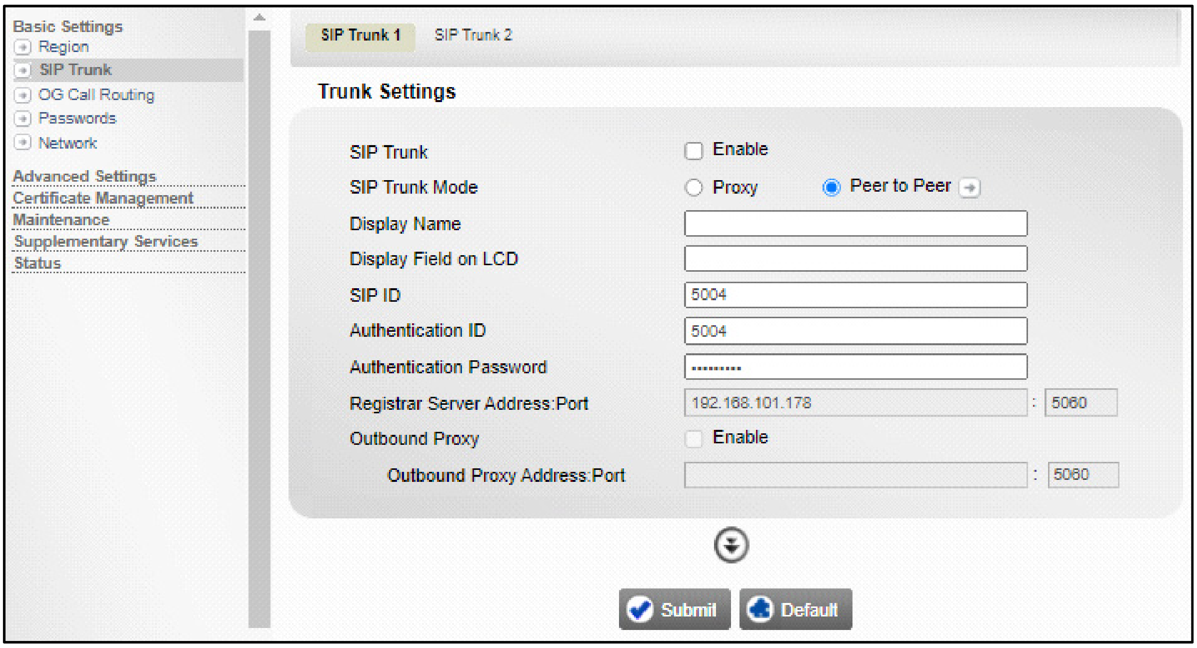
Status of SIP Trunk	Indication
Disabled	OFF
Active/Registered	Continuous ON (Blue)
Not Active	Fast Blink (Red)

To make calls, press the key assigned to the desired SIP Trunk and then you can dial the number manually or press Dir Key or Logs Key. Refer to "Making Calls".

Configuring SIP Trunks via Web User Interface

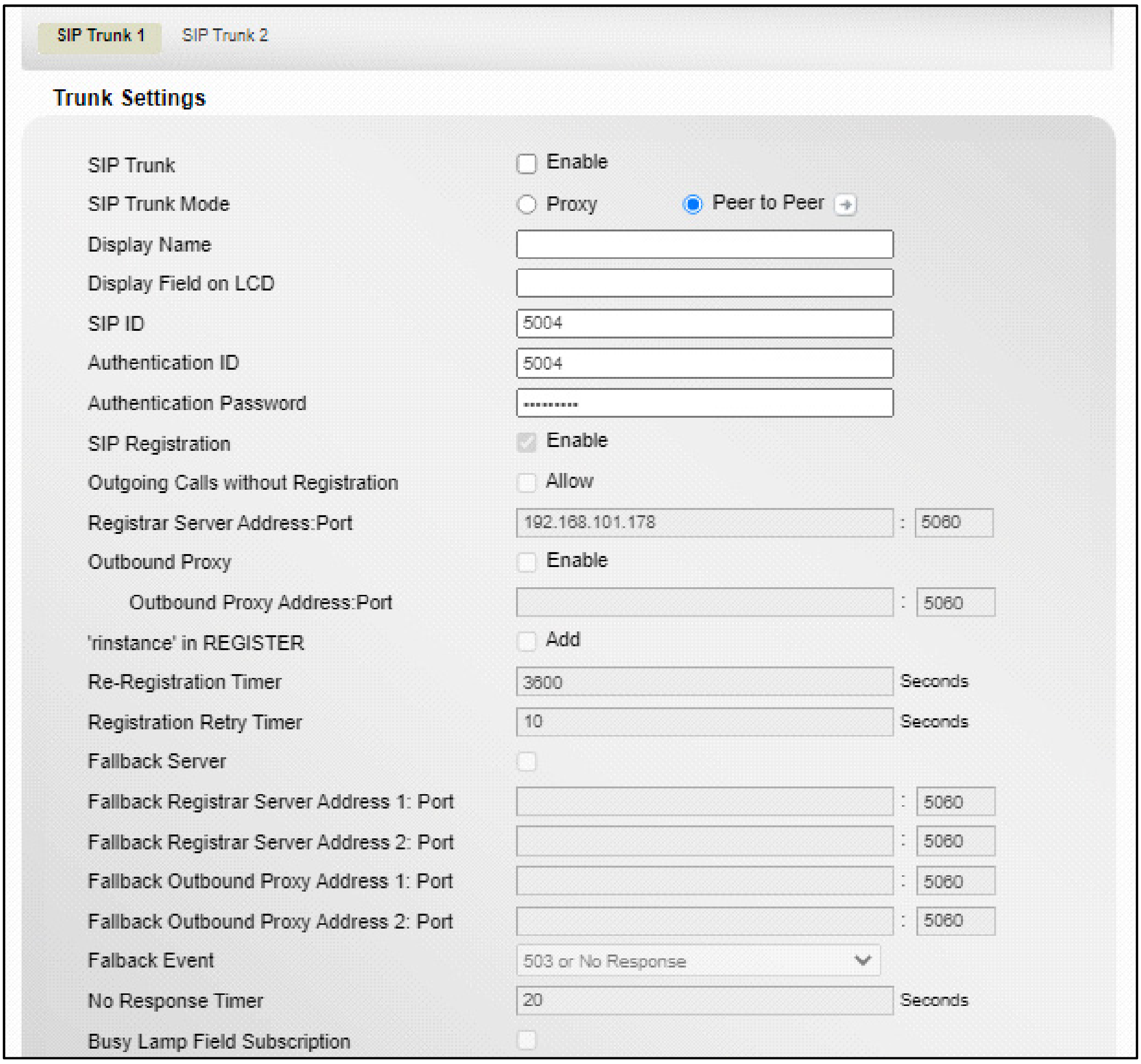
- Log into Jeeves.
- Under **Basic Settings**, click **SIP Trunk.**

- Click the **SIP Trunk 1** tab.



- Click **More** to view all parameters of SIP Trunk 1.

Trunk Settings



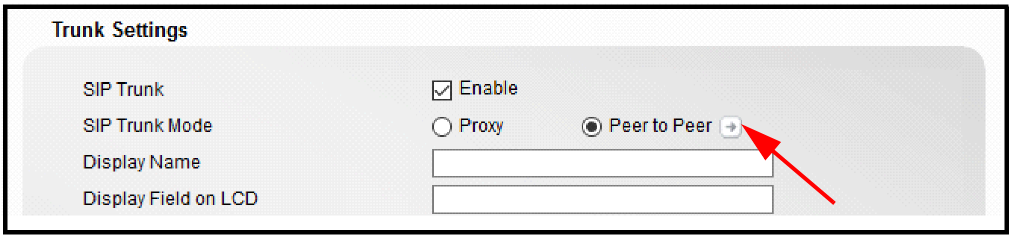
- **SIP Trunk:** Select the check box to enable. The SIP Trunk must be enabled for SPARSH VP510 to be able to route the call from it. You may disable the SIP Trunk, if you do not to use the SIP trunk to route calls. Default: Disabled.

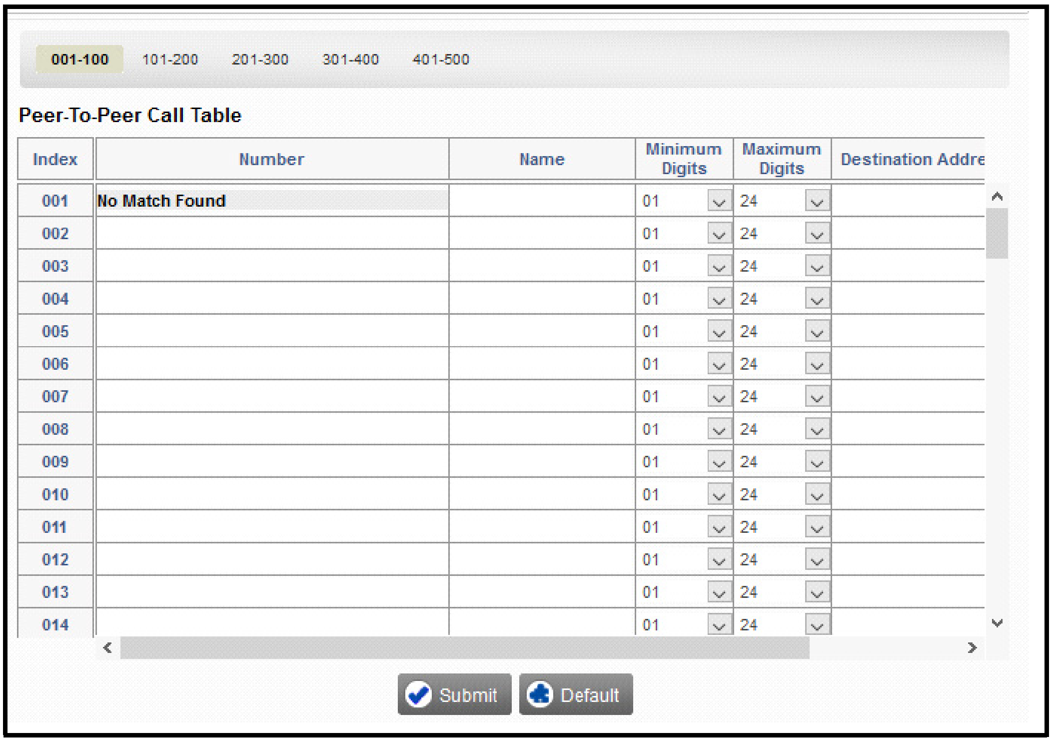
- **SIP Trunk Mode:** SIP Trunks may be Proxy or Peer-to-Peer. Default: Peer-to-Peer. Select the option as per your installation scenario.

  - For **Proxy** SIP Trunks, you must configure the following parameters.
    - SIP ID
    - Registrar Server Address
    - Registrar Server Port
    - Authentication ID
    - Authentication Password as provided by your ITSP.
    - Outbound Proxy Server Address and Port, if your ITSP uses an outbound server.
    - Fallback Registrar Server Address and Port and Fallback Outbound Proxy Address and Port, if required.
    - Busy Lamp Field Subscription, if required.

Ask your Internet Telephony Service Provider (ITSP) from whom you have subscribed the SIP Trunk for this information.

- For **Peer to Peer** SIP Trunks, you must
  - Enable the SIP Trunk.
  - Configure the Peer-to-Peer Table.
  - Click **Settings** , to configure the **Peer-to-Peer Table**. A new window opens.





- You can configure as many as 500 number strings, which are stored against an Index number.

- In **Number**, enter the peer-to-peer number string—prefix or entire number—that will be dialed. The number string must not exceed 24 characters. Default: Blank.

  If the number to be dialed out is <dialednumber@destination address>, for example, 123@abc.com, you must enter 1234 in this field.

- Enter the **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.

- As **Minimum Digits**, define the minimum length of the number string that must be dialed for the system to consider it as a valid number. Default: 01.

If the peer-to-peer number string you dial is shorter than the Minimum Digits you have configured, the system will not dial out the number.

- As **Maximum Digits**, define the maximum length of the number string that must be dialed out for the system to consider it the complete number string. Default: 24.

  If the peer-to-peer number string you dial is longer than the Maximum Digits you have configured, the system will strip off the additional digits and dial out the number.

- In **Destination Address,** enter the domain name or IP Address to where the dialed peer-to-peer number string is to be sent. The Destination Address may consists of up to 40 characters. Default: Blank.

  For example, if the peer-to-peer number to be dialed out is 123@abc.com, enter abc.com as Destination Address. If the number is 1234@192.168.1.197, enter 192.168.1.197 as the Destination Address. The Destination Address can also be in the form of Address: Port number.

- In **SIP Transport**, select the desired protocol for communication — UDP, TCP, TLS.

- Click **Submit** to save entries.

- Close the window.

- To know more about Peer-to-Peer Installations and Configurations refer “Peer-to-Peer Numbers”. Also see “Peer-to-Peer Calls”

- **Display Name:** Enter the name which should appear on the remote user's phone when you make calls. A maximum of 24 characters are allowed. Default: Blank.

- **Display Field on LCD:** Enter the SIP Trunk related information which you wish should be displayed on the LCD of the SPARSH VP510

  The SIP Trunk details that you enter will appear on the LCD display of the phone, when the phone is in idle state. The **Display Field on LCD** may be the name of the person/department who is using the SPARSH VP510, or the name of the ITSP, or any other identifying information you wish to use. Default: Blank. This name will also appear automatically as the **User Name** in Advanced Settings > System > General. For details, refer to “General” in “System Parameters”.

- **SIP ID:** Enter the SIP ID provided by the ITSP. This can be a number or text. For instance, if SIP URI provided by ITSP is 12345@abc.com, enter 12345 in this field. SIP ID may consist of a maximum of 40 characters. Default: Blank.

- **Authentication ID:** Enter User ID provided by the ITSP for registering the SIP trunk with the SIP server. This field is relevant when SIP user ID and Authentication user ID are not the same. User ID can be of a maximum of 40 characters. Default: Blank.

- **Authentication Password:** Enter the authentication password provided by the ITSP. Password can be of a maximum of 24 characters. Default: Blank.

- **SIP Registration:** With this parameter you can select whether or not the phone should send REGISTER message from the SIP Trunk. Default: Enabled (REGISTER message is allowed to be sent from the SIP Trunk). Clear the check box is you do not want to send the REGISTER message.

- **Outgoing Calls without Registration:** Select the check box to enable. SPARSH VP510 will allow outgoing calls to be made from the SIP Trunk, even when the SIP Trunk is not registered. If the check box is cleared, it the will not allow outgoing calls to be made if the status of the SIP Trunk is 'not registered'. Default: Disabled.

- **Registrar Server Address: Port:** Enter SIP Registrar Server Address provided by ITSP. It can be an IP address or domain. SIP registrar server address must not exceed maximum 40 characters. Default: Blank.

Leave the Registrar Server Address blank, if you want to use this SIP Trunk for the Peer-to-Peer application.

Also Enter registrar server listening port provided by ITSP. Valid range: 1024-65534. Default: 5060.

- **Outbound Proxy:** This parameter is relevant only if the ITSP has a SIP outbound server to handle voice calls. Select the check box to enable Outbound Proxy, if your ITSP has a SIP outbound server to handle voice calls. Default: Disabled.

- **Outbound Proxy Address:Port:** Enter the outbound proxy server address provided by your ITSP, if you enable outbound proxy. It can be IP address or domain. Outbound proxy server address can be of a maximum of 40 characters. Default: Blank.

  Enter the Outbound Proxy Server's listening port provided by your ITSP. Valid range: 1024-65534. Default: 5060.

- **‘rinstance' in REGISTER:** 'rinstance' is any random value which can be used by the phone to fetch its own contact binding, i.e. to know the Registration Expiry Timer assigned by the server. Default: Enabled. Clear the check box to disable.

- **Re-Registration Timer:** 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. The SPARSH VP510 sends a registration request before this timer expires to remain registered on the server. Enter the value of timer after which the phone should send registration request to get registered again. Valid range: 00001-65535 sec. Default: 3600 sec.

- **Registration Retry Timer:** Registration Retry timer indicates the period between retries for registration. If the registration attempt fails, SPARSH VP510 sends the registration request on expiry of this timer again. The phone keeps sending the registration request till it gets registered. Valid range: 00001-65535 sec. Default: 10 sec.

- **Fallback Server:** Select the Fallback Server check box, if your Service Provider supports multiple servers in their network.Default: Disabled.

  If you have enabled Fallback Server and Outbound Proxy is disabled, then configure the following:

- **Fallback Registrar Server Address 1: Port / Fallback Registrar Server Address 2: Port:** Enter the IP Addresses of the alternate Registrar Servers and their respective listening ports. The Fallback Registrar Server Address can be of maximum 64 characters. Valid range is 1025 to 65535. Default: 5060.

  If you have enabled Fallback Server and Outbound Proxy is enabled, then configure the following

- **Fallback Outbound Proxy Server Address 1: Port / Fallback Outbound Proxy Server Address 2: Port:** Enter the IP Addresses of the alternate Outbound Proxy Servers and their respective listening ports. The Fallback Outbound Proxy Server Address can be of maximum 64 characters. Valid range is 1025 to 65535. Default: 5060.

- **Fallback Event:** Select the event on occurrence of which the phone should fallback to an alternate Registrar/Outbound Proxy Server, if available.
  - No Response
  - 503 or No Response
  - 5xx or No Response

  Default: 503 or No Response

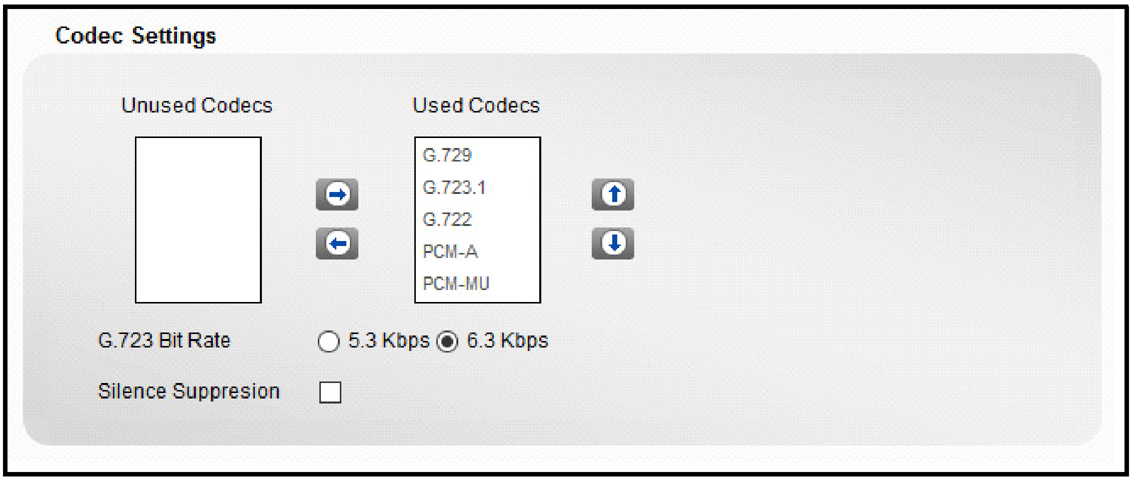
- **No Response Timer:** This timer defines the time period for which the phone will wait for the response from the Server (Registrar/Outbound) for any request. If no valid response is received before the expiry of this timer, the phone will fallback to alternate Registrar/Outbound Proxy Server or Routing Group/Fallback Routing Group for further processing of the call. Valid range is 01 to 99 seconds. Default: 20 seconds.

- **Busy Lamp Field Subscription:** The purpose of Busy Lamp Field is to provide the real-time status information about the availability or status of a particular line (SIP Trunk) /user. Busy Lamp Field Subscription allows one user to monitor the status of another user as well as monitor the status of the line (SIP Trunk). If you wish to monitor users/lines for change in status, then select this check box to enable. Default: Disabled. To know more about the functionality of this feature, refer to "Busy Lamp Field (BLF)".

Voice Mail Settings

To configure and access Voicemail Settings, refer "Voicemail".

Codec Settings



- Select the Codecs in the order of preference from the multiple selection box.

  Codecs are the various voice codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality. You can set 5 Codec options in the order of preference.

  The Codecs supported by the IP Phone in the order of preference, are listed in the **Used Codecs** box:
  - G.729
  - G.723.1
  - G.722
  - PCM-A
  - PCM- µ

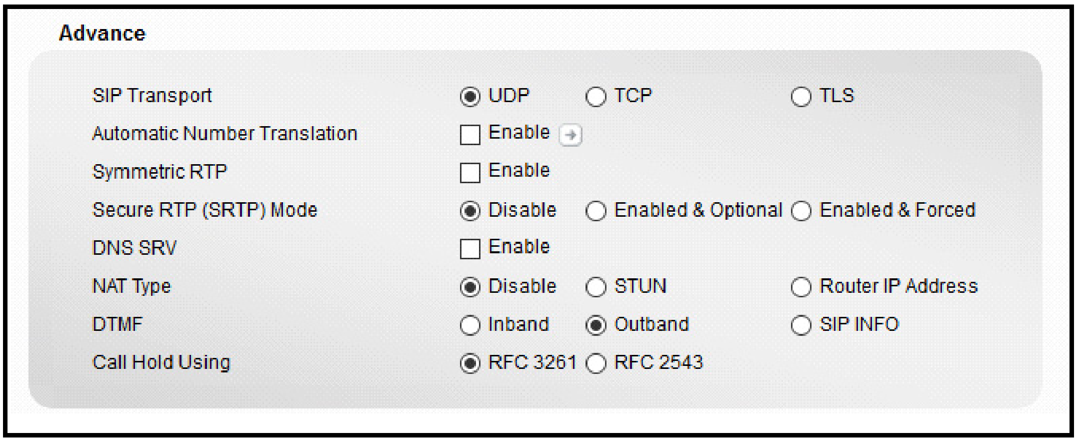
- To remove a Codec from this list, select the Codec and the back arrow.
- The Codec will be moved to the **Unused Codecs** box.
- Select the required Codec from the right list box with your cursor.
- Use the **Up** and **Down** arrows near the **Used Codecs** box to change the order of Codec preference.

The above preference shall be used for both incoming and outgoing calls.

- **G.723 Bit rate:** 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. Default: 6.3 kbps.

- **Silence Suppression:** This is applicable to G.729 codec only. Select this check box to enable only if it is required by your Service Provider. Default: Disabled

Advance



- **SIP Transport:** The SPARSH VP510 supports three options for transporting outgoing SIP messages.

  - UDP: Outgoing messages are transported using UDP.
  - TCP: Outgoing messages are transported using TCP.
  - TLS: Outgoing messages are transported using TLS.

  You may select any one, as per your requirement. Default: UDP.

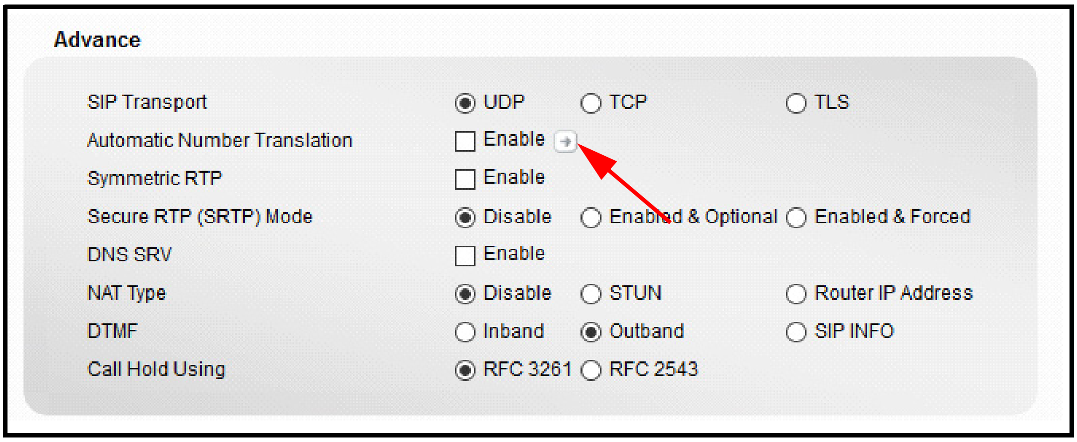
If you have selected TCP or TLS make sure that you enable SIP over TCP or SIP over TLS in the System parameters.

- **Automatic Number Translation:** This parameter is to be configured only if you want to apply the Automatic Number Translation Feature on the SIP Trunk. This feature translates the number dialed by the user into a number that is understood by the ITSP (VoIP) network.

  To know more, read the feature description for “Automatic Number Translation”. To apply this feature on the SIP Trunk, you must configure the related parameters.

  - Select the check box to enable. Default: Disabled

- Click **Settings** , the Automatic Number Translation page opens. Refer to "Automatic Number Translation" to know more.



- **Symmetric RTP:** The use of Symmetric RTP makes it possible for a SIP device to send the RTP packets on the same connection on which it is listening for RTP. This is done only on Peer-to-Peer SIP Trunks.

  Clear the check box, if the IP phone is located on a public IP and you do want outgoing calls to the SIP Client located behind the NAT Router OR if you do not want to receive incoming calls from the SIP Client located behind the NAT router. Default: Enabled.

- **Secure RTP (SRTP) Mode:** Secure Real-Time Transport Protocol (SRTP) encrypts the RTP streams during VoIP phone calls to avoid interception and eavesdropping. The parties participating in the call must enable SRTP feature simultaneously. When this feature is enabled on both phones, the type of encryption to utilize for the session is negotiated between the IP phones. This negotiation process is compliant with RFC 4568. When a user places a call on the enabled SRTP phone, the IP phone sends an INVITE message with the RTP encryption algorithm to the destination phone.

  The callee receives the INVITE message with the RTP encryption algorithm, and then answers the call by responding with a 200 OK message which carries the negotiated RTP encryption algorithm.

  Select the desired option:
  - Disable
  - Enabled & Optional
  - Enabled & Forced

  Default: Disabled.

- **Media Type:** If you select **Secure RTP (SRTP) Mode** as **Enabled & Optional**, then you must configure the Media Type. Select the desired option:
  - AVP
  - SAVP

  Default: AVP

- **DNS SRV:** Select the check box to enable. SPARSH VP510 will send DNS SRV query to the configured domain server. Clear the check box, to send DNS A query to the configured domain server. Default: Disabled.

- **NAT Type:** You can select either of the following options:

  - **Disable:** Select this option if the IP Phone is directly connected to the Public Network.

    - **Router IP address:** Select this option if IP Phone is located behind the NAT router (any type). This option will work only if Outbound is disabled on SIP trunk. If you have selected this option, configure the Router's Public IP Address in the “System Parameters”.

    - **STUN:** STUN is the most widely used protocol by SIP clients when located behind the NAT router. STUN is used to map the public IP address and port of the NAT router behind which the SIP client is located. Select this option if IP Phone is located behind the NAT router other than Symmetric. This option will work only if Outbound is disabled on the SIP trunk. If you have selected this option, configure the STUN server address and port in “System Parameters”.

      Default: Disabled.

- **DTMF:** This parameter defines how the DTMF digits are to be sent over the IP network, when a DTMF digit is pressed. SPARSH VP510 supports the following:
    - Inband
    - Outband
    - SIP INFO

  Default: Out-band.

  Inband means DTMF is combined in audio signal. Outband means digits are to be sent via RTP (RFC 2833). Info means digits are to be sent in SIP Info message. Select In-band/Out-band/SIP INFO.

- **Call Hold using:** SPARSH VP510 supports the following Call Hold options:
    - RFC 3261
    - RFC 2543

  Default: RFC 3261

- Click **Submit** to save your settings.

- Similarly, you can configure SIP Trunk 2.

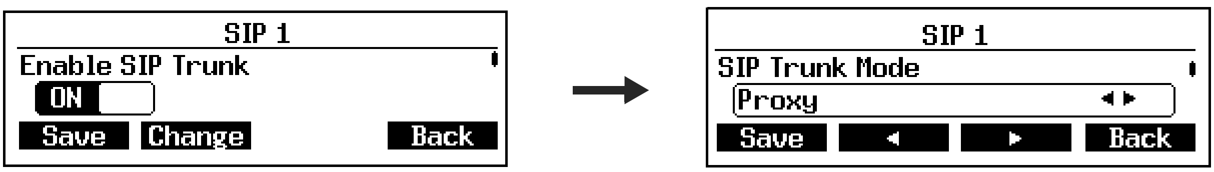
Configuring SIP Trunks via Phone User Interface

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **Settings** and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **SIP Trunk** and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **SIP Trunk 1** or **SIP Trunk 2** and press **Select** Key.

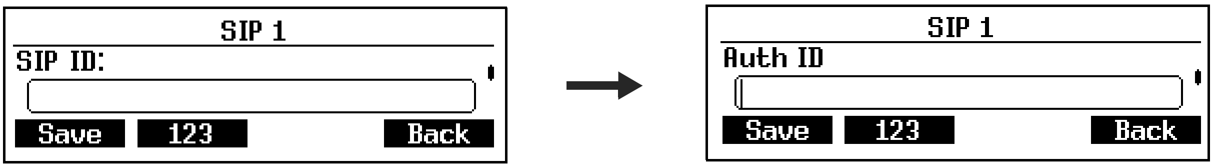


- Scroll using the **Up/Down Navigation** Key to select **Enable SIP Trunk**.

- Press the **Change** Key to turn it **On/Off**.

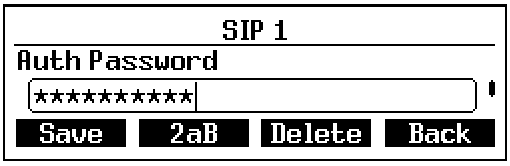
- Scroll using the **Up/Down Navigation** Key to select **SIP Trunk Mode**.

- Scroll using **Right Navigation >** Key or **Left Navigation <** Key, to select the desired Mode — P2P or Proxy.



- Scroll using the **Up/Down Navigation** Key to select **SIP ID** and configure the same.

- Scroll using the **Up/Down Navigation** Key to select **Auth ID** and configure the same.

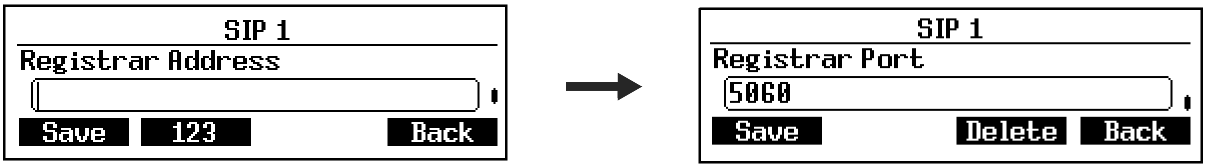


- Scroll using the **Up/Down Navigation** Key to select **Auth Password** and configure the same.

- Click **Save**.

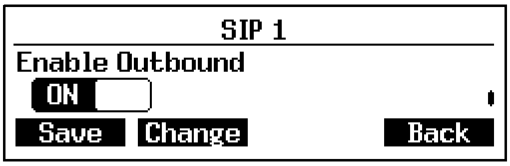
If you have selected SIP Trunk Mode as Proxy, configure the following:

- Scroll using the **Up/Down Navigation** Key to select **Registrar Address** and configure the same.

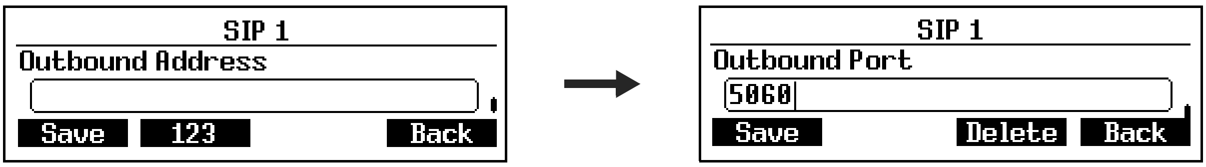


- Scroll using the **Up/Down Navigation** Key to select **Registrar Port** and configure the same.

- Scroll using the **Up/Down Navigation** Key to select **Enable Outbound**.



- Press the **Change** Key to turn it **On/Off**.



- Scroll using the **Up/Down Navigation** Key to select **Outbound Address** and configure the same.

- Scroll using the **Up/Down Navigation** Key to select **Outbound Port** and configure the same.

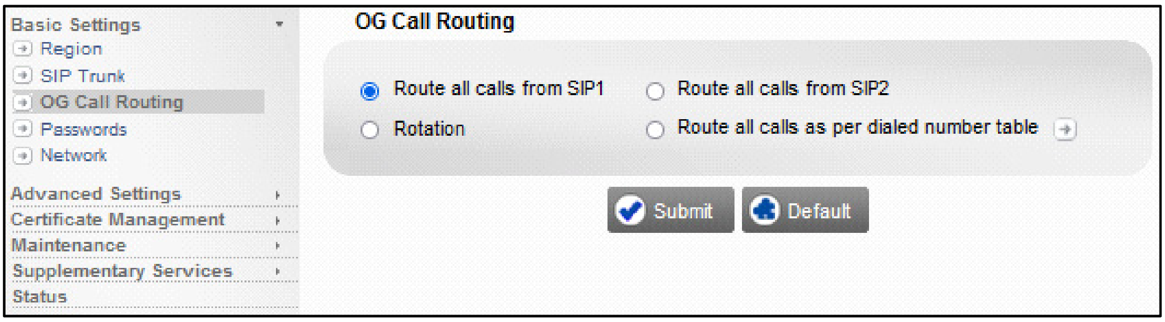
- Press **Save** Key.

Outgoing Call Routing

You can route outgoing calls from the phone through a particular SIP trunk or Rotation or using the Dialed Number Table. You can select the outgoing call routing method via Web User Interface only.

Configuring Outgoing Call Routing via Web User Interface

- Log into Jeeves.
- Under Basic Settings, click **Outgoing Call Routing**.



- Select the desired routing option from the following:
  - Select **Route all calls from SIP1 or SIP2,** to route calls through a particular SIP Trunk**.**

    OR

  - Select **Rotation** if you want the first call to be routed through SIP1 and the next call through SIP2. The phone will select the next SIP trunk only if the SIP Trunk is enabled and active.

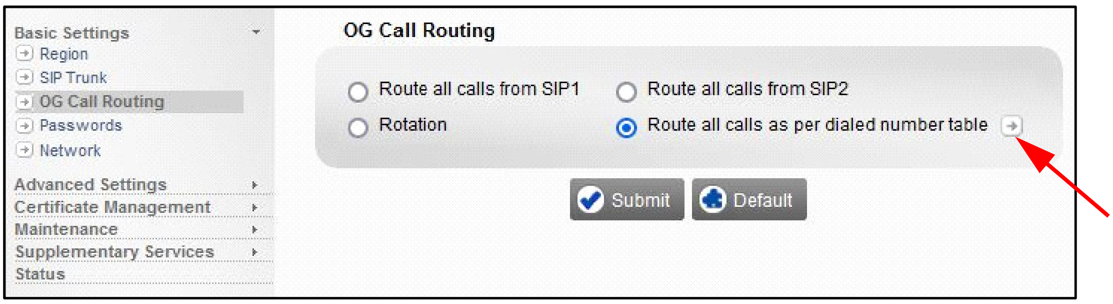
    OR

  - Select **Route all calls as per dialed dumber table,** if you want the phone to route certain numbers through a specific SIP trunk. For this you need to configure the **Dialed Number Table**.

    Default: Route all calls from SIP1.

- If you selected **Dialed Number Table** you must also configure this table.

1. Click **Settings** , the Dialed Number Table page appears.



Dialed Number Table

Index	Number	Minimum Digits	Maximum Digits	Destination Trunk
001	No Match Found	01	24	SIP Trunk 1
002		01	24	SIP Trunk 1
003		01	24	SIP Trunk 1
004		01	24	SIP Trunk 1
005		01	24	SIP Trunk 1
006		01	24	SIP Trunk 1
007		01	24	SIP Trunk 1
008		01	24	SIP Trunk 1
009		01	24	SIP Trunk 1
010		01	24	SIP Trunk 1
011		01	24	SIP Trunk 1
012		01	24	SIP Trunk 1
013		01	24	SIP Trunk 1
014		01	24	SIP Trunk 1

Submit Default

- Each entry in the dialed number table is stored at an Index. For each entry in the table, you must configure the following parameters:
  - **Number:** Enter the number which will be dialed out from the phone (which the phone should match with this Table before dialing out). The number should not be more than 24 characters. Default: Blank.
  - **Minimum Digits:** For the number you entered in the previous field, select the minimum number of digits which the phone should wait to receive before considering it as a valid number. Default: 01.
  - **Maximum Digits:** For the number you entered, select the minimum number of digits which the phone should wait to receive before considering it as End of Dialing[2] and dial out the number. Default: 24.
  - **Destination Trunk:** For the number you entered, define the SIP Trunk through which the dialed number (which matches the minimum and maximum digits) should be routed. Default: SIP Trunk 1.
- Click **Submit** and close the window. See "Dialed Number Table" for more details.

- *If you have set SIP1 as the outgoing call route for all numbers, you can still route a call through SIP2, using the SIP Trunk 2 Key.*
- *However, if you want the phone to use Dialed Number Table to select route, do not select a SIP Trunk before dialing the number. If you select a SIP Trunk before dialing the number, the phone will not match the dialed number with the entries in the Dialed Number Table and the call will be routed through the selected SIP Trunk instead.*

2. *When 'Dialed Number Table' is selected as the option for OG Call Route Selection, the phone will not apply 'Fixed Number of Digits for "End of Dialing".*

Passwords

The SPARSH VP510 allows you to secure the phone by way of protecting the keypad and the phone configuration with a user and a configuration password respectively.

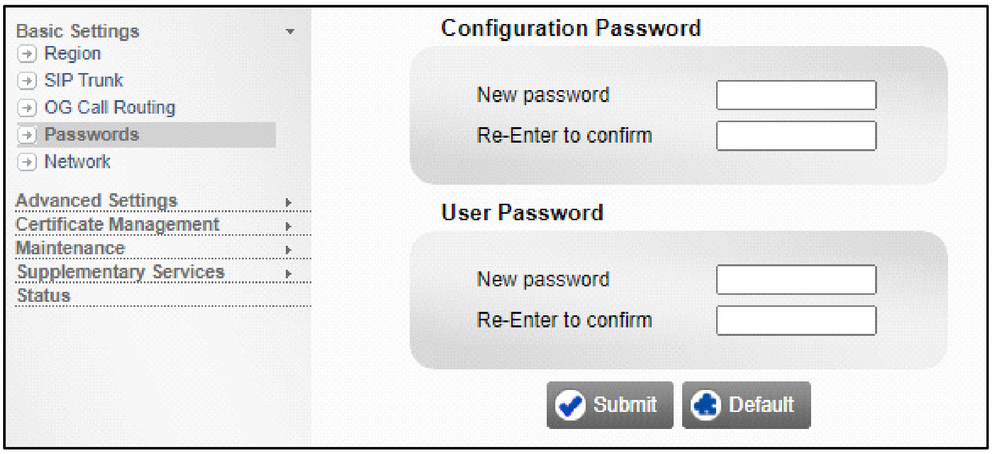
- The default User Password is 1234 for the Phone User Interface.
- The default Configuration Password is 1234 to log into the Web User Interface.

  You can change the Password using the Web User Interface as well as the Phone User Interface. The new User and Configuration Password must consist of minimum 4 characters to a maximum of 8 characters.

When you change passwords for the first time from the Phone User Interface or Web User Interface, you need to use the default User and Configuration Password.

Changing Password via Web User Interface

- Log into Jeeves.
- Under **Basic Settings**, click **Passwords**.



Configuration Password

- **New password:** Enter New Password.
- **Re-Enter to confirm:** Retype the New Password to confirm.

User Password

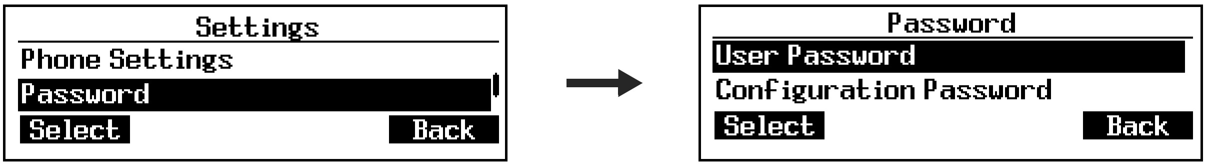
- **New password:** Enter New Password.
- **Re-Enter to confirm:** Retype the New Password to confirm.

Click **Submit**.

Changing Password via Phone User Interface

User Password

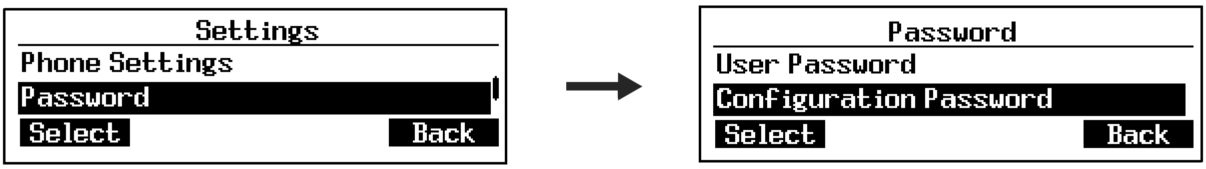
- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **User Password** and press **Select** Key.



- Enter **Old Password**.
- Scroll using the **Up/Down Navigation** and enter **New Password**.
- Scroll using the **Up/Down Navigation** and enter **Re-enter Password**.
- Press **Save** Key.

Configuration Password

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Configuration Password** and press **Select** Key.



- Enter **Old Password**.
- Scroll using the **Up/Down Navigation** and enter **New Password**.
- Scroll using the **Up/Down Navigation** and enter **Re-enter Password**.
- Press **Save** Key.

Once you have changed the Configuration Password, you can log into Jeeves only with the new configuration password.

Network Parameters

For your phone to communicate with the IP network, you must first configure the basic network parameters:

1. IP Address
2. Subnet Mask
3. Gateway Address
4. DNS Address

If you do not have information on any/all of these, ask your LAN Administrator or your ISP.

The Gateway Address, DNS Address, must be configured in order to make IP calls on WAN network.

There are two ways in which to set these network parameters, depending on the IP addressing of the network your phone is connected to.

If your network uses DHCP server on your LAN, the above network parameters will be automatically assigned to your phone by the DHCP server.

If your network uses Static IP addressing, you must set these network parameters manually.

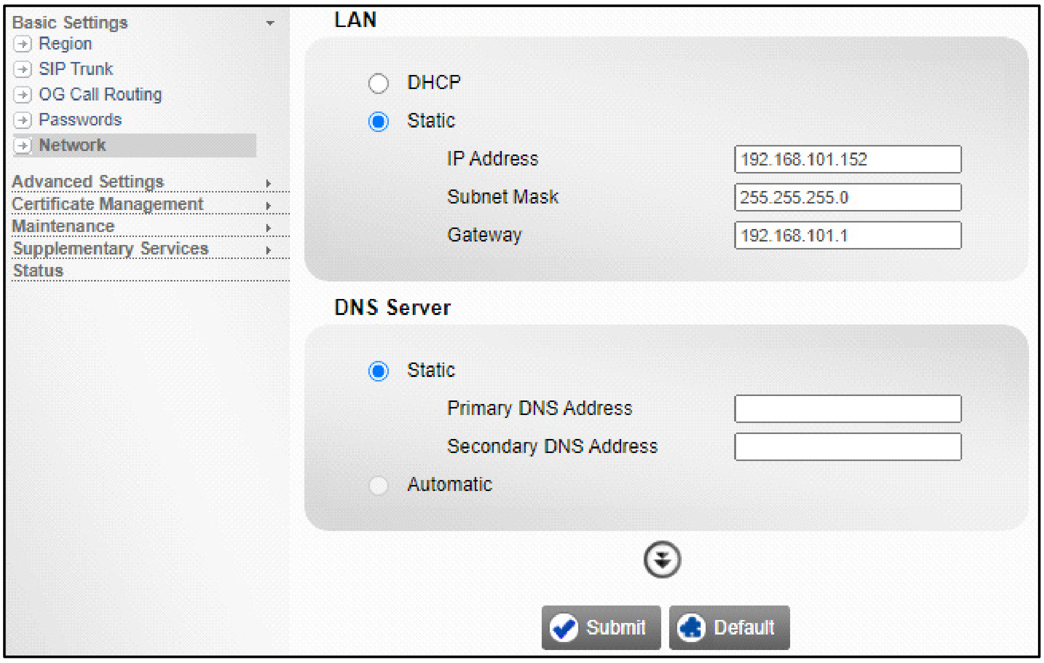
The basic Network Parameters can be set via Phone User Interface, while all the Network Parameters can be configured via Web User Interface.

SPARSH VP510 supports Automatic Provisioning (Auto Configuration). If you have been provided the phone by your ITSP, and it is being configured by the ITSP, the LCD will display the upgrade information after power on. Wait for the phone to be configured automatically by the ITSP.

Configuring Network Parameters via Web User Interface

- Log into Jeeves.

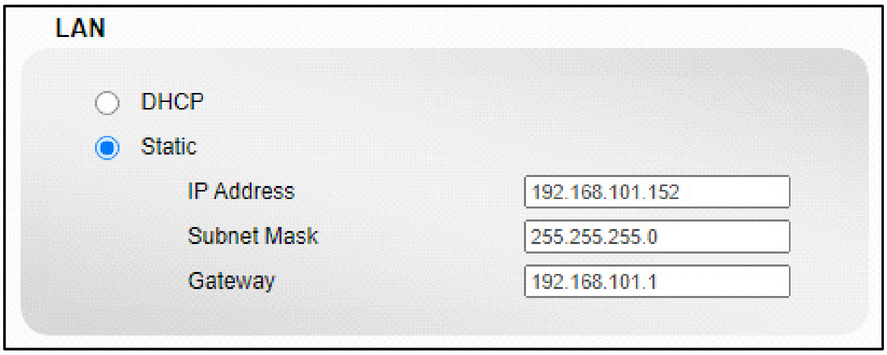
- Under **Basic Settings**, click **Network.**



- Click **More** to view all parameters on this page.

- Configure the following network parameters:

LAN

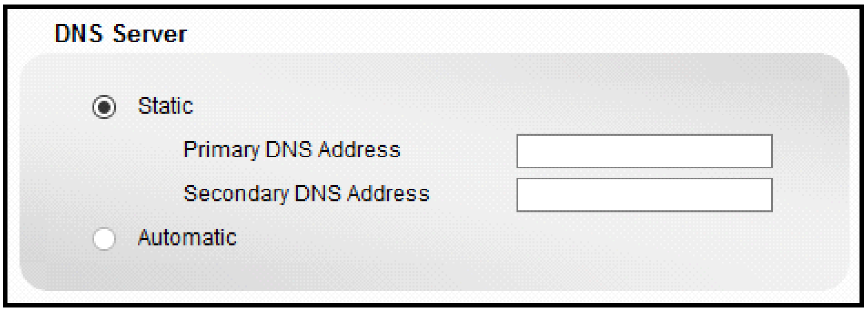


- **Connection Type:** Select a connection type depending on the IP addressing scheme of the network that your phone is connected to: DHCP or Static IP. Default: Static IP.

  DHCP: Select **DHCP** if you want the IP address, Subnet Mask and Gateway address to be assigned by the DHCP server automatically.

  Static: Select **Static**, if you want to assign IP Address, Subnet Mask and Gateway Address manually. Configure all the fields manually, if you have selected this option. Default: Enabled.

DNS Server



DNS stands for Domain Name Server which is used to resolve domain name into IP address. You can select either **Static** or **Automatic**. Default: Static.

DNS stands for Domain Name Server which is used to resolve domain name into IP address. You can select either Static DNS or Automatic DNS. Ask your network administrator if your network provides Automatic DNS.

Automatic DNS: Select Automatic DNS if you want DNS Address and Domain name to be assigned by the DHCP server automatically. This option will be applicable only if you have enabled DHCP as connection type.

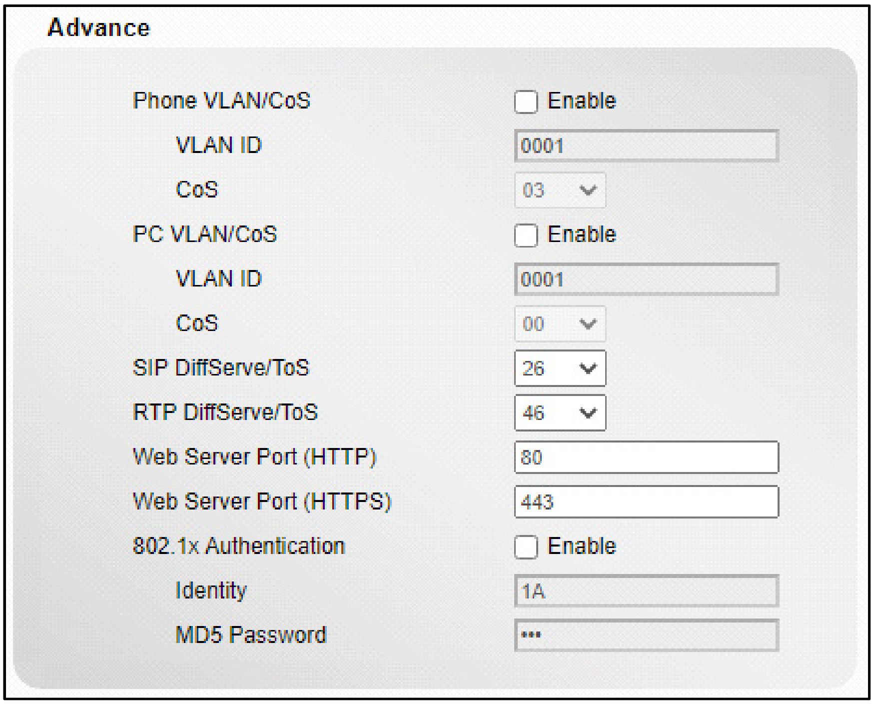
Static DNS: Select Static DNS if you want to configure DNS manually.

Primary DNS Address: Enter Primary DNS Address.

Secondary DNS Address: Enter Secondary DNS Address. Secondary DNS Address is considered when the request to Primary DNS server fails.

- *You should configure DNS Address, only when 'DNS Server' is selected as 'Static'. Default: Blank.*
- *You can configure either Primary DNS Address or Secondary DNS Address or both Primary and Secondary DNS Address.*

Advance



- **VLAN/CoS:** This parameter is to be configured if the SPARSH VP510 is to be connected in VLAN network. To enable the switch correctly route packets generated by the phone and the PCs to each other, they must be tagged with a VLAN header.

  This parameter enables the SPARSH VP510 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 (i.e. packets generated by the PC connected to its PC port).

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

  Default: Disabled.

3. *The IEEE 802.1P standard allows Layer2 switches to prioritize the traffic, thus providing Quality of Service (QoS), i.e. 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.*

Phone VLAN/CoS: Select the check box to enable that is if you want the VLAN ID to be tagged on all packets generated by the phone (SIP, RTP, DNS, ARP, etc). Default: Disabled.

VLAN ID: Enter the VLAN ID that you have assigned to the VLAN in which the IP Phone/s are connected. Valid range: 0-4094. Default: 1.

CoS: Define the CoS (priority) bits in all SIP packets. Valid range: 0-7. Default: 3

PC VLAN/CoS: Select the check box to enable that is if you want VLAN header to be tagged on all packets entering the PC Port and leaving the LAN port of the phone. Default: Disabled.

VLAN ID: This is the same ID as you have assigned to the VLAN in which the PCs are connected. Valid range: 0-4094. Default: 1.

CoS: Define the Layer 2 CoS (priority) bits. Valid range: 0-7. Default: 0.

- **SIP DiffServer/ToS:** SPARSH VP510 will send all SIP messages using SIP QoS setting. Valid range: 00-63, Default: 26.

  RTP DiffServer/ToS: SPARSH VP510 will send all the RTP packets with RTP QoS setting. Valid range: 00-63, Default: 46.

- **Web Server Port (HTTP):** SPARSH VP510 has an embedded web server, known as Jeeves, for system configuration. You may change it as per your requirement. Valid range is 80, 1024-65535. Default:80

- **Web Server Port (HTTPS):** SPARSH VP510 has an embedded web server, known as Jeeves, for secure system configuration. You may change it as per your requirement. Valid range is 443, 1024-65535. Default:443.

- **802.1x Authentication:**802.1x is a standard for network access control that provides an authentication framework for wired as well as wireless networks. It is a part of the IEEE 802 family of standards and is used to enhance network security by ensuring only authorized and authenticated devices can access the network. If you wish to use 802.1x Authentication, configure the following:
  - **Enable:** Select the check box to enable.
  - **Identity:**Enter the user name for authentication.
  - **MD5 Password:** Enter the password for authentication.

  Default: Disabled

Click **Submit** to save the Network parameter changes.

You can either continue further with the configuration of Advanced features and facilities, else you can log out of Web User Interface by clicking on 'Logout' on the top right of the page.

Configuring Network Parameters via Phone User Interface

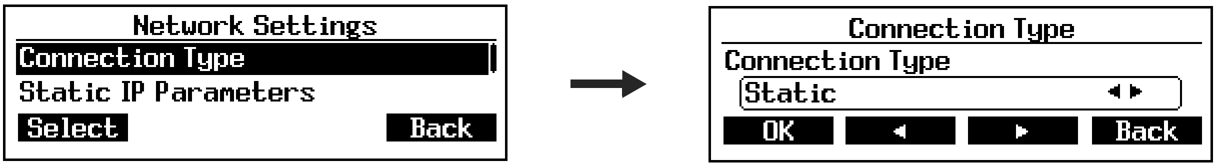
It is recommended that you do not configure the Network Settings on your own as it may result in malfunctioning of your phone. Ask your System Administrator to configure it for you.

To configure the Network parameters from the Menu,

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Network Settings** and press **Select** Key.

Connection Type

- Scroll using the **Up/Down Navigation** Key to select **Connection Type** and press **Select** Key.



- Scroll using **Right Navigation >** Key or **Left Navigation <** Key, to select the desired Connection Type — **DHCP**, **Static**.
- Press **OK** Key.

If your connection type is DHCP,

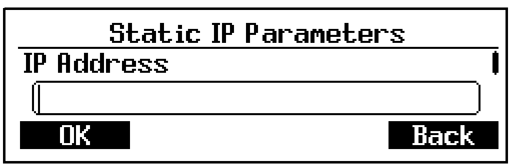
- The phone will be assigned **IP Address**, **Subnet Mask**, **Gateway Address**, **DNS Address** and **Server Address** automatically by the DHCP Server.

If your DHCP Server does not provide DNS Settings and/or Server Settings automatically, you must configure them manually. Refer the steps given in Static.

Static IP Parameters

If you select Static, configure the following.

- Scroll using the **Up/Down Navigation** Key to select **Static IP Parameters** and press **Select** Key.

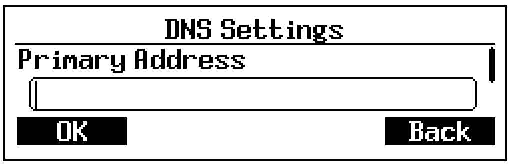


- Scroll using the **Up/Down Navigation** Key to enter the **IP Address**, **Subnet Mask** and **Gateway Address** and configure each of them.

- Press **OK** Key.

DNS Settings

- Scroll using the **Up/Down Navigation** Key to select **DNS Settings** and press **Select** Key.



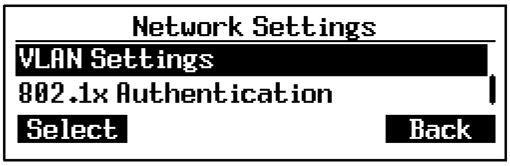
  - Scroll using the **Up/Down Navigation** Key to enter the **Primary DNS Address** and **Secondary DNS Address** and configure each of them.

  - Press **OK** Key.

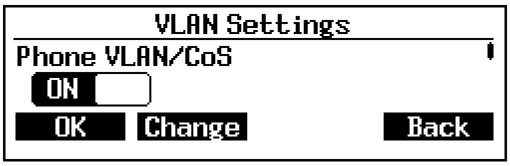
VLAN Settings

If your phone is connected to a Virtual LAN, configure the VLAN Settings. To route packets of the LAN and the PC ports of the phone through a VLAN switch, they must be tagged with a VLAN header. This header consists of a VLAN ID and a Class of Service (CoS).

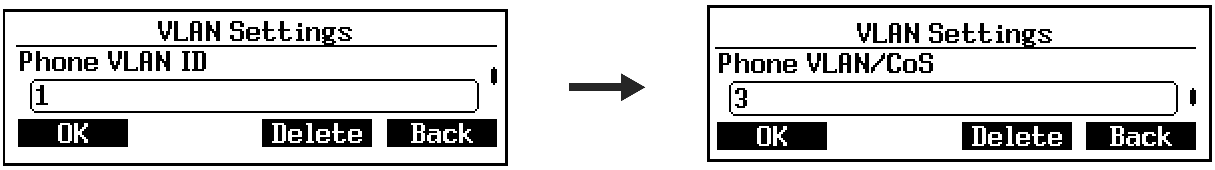
- Scroll using the **Up/Down Navigation** Key to select **VLAN Settings** and press **Select** Key.



Phone VLAN/CoS



- Scroll using the **Up/Down Navigation** Key to select **Phone VLAN/CoS** and press **Change** Key to turn it **On/OFF.**

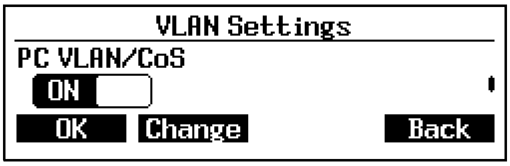


- Scroll using the **Up/Down Navigation** Key to select **Phone VLAN ID** and configure the same.

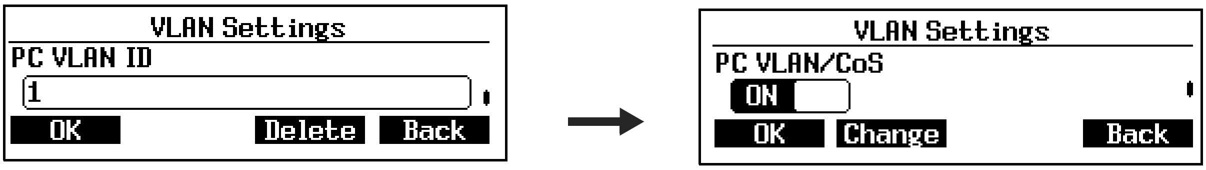
- Scroll using the **Up/Down Navigation** Key to select **Phone VLAN/CoS** and configure the same.

- Press **OK** Key.

PC VLAN/CoS



- Scroll using the Up/Down Navigation Key to select **PC VLAN/CoS** and press **Change** Key to turn it **On**/ **OFF**.

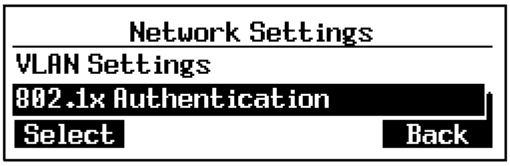


- Scroll using the **Up/Down Navigation** Key to select **PC VLAN ID** and configure the same.
- Scroll using the **Up/Down Navigation** Key to select **PC VLAN/CoS** and configure the same.
- Press **OK** Key.

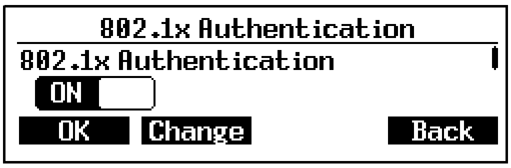
802.1x Authentication

802.1x is a standard for network access control that provides an authentication framework for wired as well as wireless networks. It is a part of the IEEE 802 family of standards and is used to enhance network security by ensuring only authorized and authenticated devices can access the network. To use 802.1x Authentication, configure its parameters.

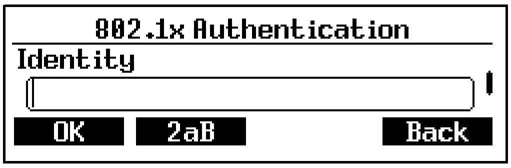
- Scroll using the **Up/Down Navigation** Key to select **802.1x Authentication** and press **Select** Key.



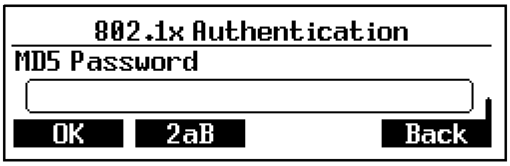
- Scroll using the **Up/Down Navigation** Key to select 802.1x Authentication and press **Change** Key to turn it **On/OFF.**



- Scroll using the **Up/Down Navigation** Key to select **Identity** and configure the User Name for authentication



- Scroll using the **Up/Down Navigation** Key to select **MD5 Password** and configure the Password for authentication.



- Press **OK** Key.
- Press **Save** Key.

CHAPTER 7

Customizing Your SPARSH VP510

You can customize your IP phone personally by configuring certain settings, for example, contrast, language and time & date. You can add contacts to the phone's local directory manually or from call history. You can also personalize different ring tones for different callers.

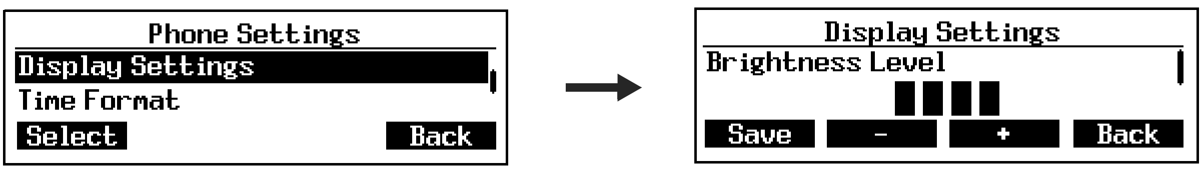
For customizing your phone, refer to the following:

- "Display Settings"
- "Language"
- "Time Format"
- "Call Waiting"
- "Ringer Settings"
- "Volume Settings"
- "Accessories"

Display Settings

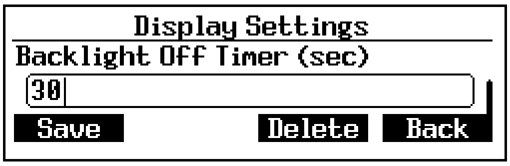
Changing Display Settings via Phone User Interface

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Phone Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Display Settings** and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select the desired option — Brightness Level, Backlight Off Timer.



- You can set the desired Brightness and Contrast levels using the Plus **+** **or** Minus **-** Context Keys.
- Press **Save** Key.

- You can also change the timer to turn off the LCD Backlight.



- Scroll using the **Up/Down Navigation** Key to select **Backlight Off Timer** and enter the maximum time in second after which you want the Backlight to turn Off.

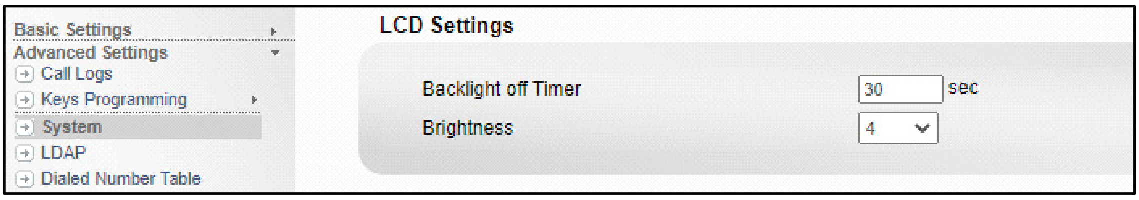
- Click **Save** Key.

*If you set this timer as **000**, the Backlight will always remain on.*

Changing Display Settings via Web User Interface

- Log into Jeeves.

- Under **Advanced Settings**, click **System**.



- Scroll to **LCD Settings**.

- Set the **Backlight off Timer** (Default: 30 sec) and **Brightness** (Default:4) as per your requirement.

- Click **Submit** to save.

Language

Changing Language via Phone User Interface

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.

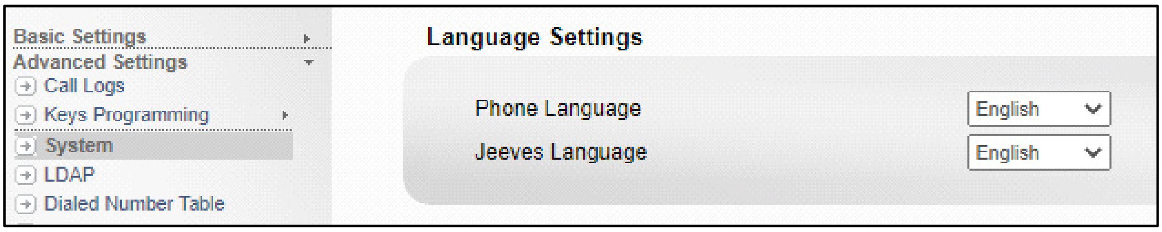
- Scroll using the **Up/Down Navigation** Key to select **Phone Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **Language** option and press **Select** Key.

- Scroll using **Right Navigation >** Key or **Left Navigation <** Key, to select the desired language.
- Press **Save** Key.

Changing Language via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **System**.



- Scroll to **Language Settings**.
- Select **Phone Language** and **Jeeves Language.** Default: English.

  SPARSH VP510 can display the pages of the Jeeves in English, Italian, Spanish, French, German, and Portuguese. Default: English.

  When you login in again later, all the pages of the Jeeves will appear in the language you have selected.

  You can also select a Language of your choice on the Login page of the Jeeves; however, the language you select will be applied for the current session only.

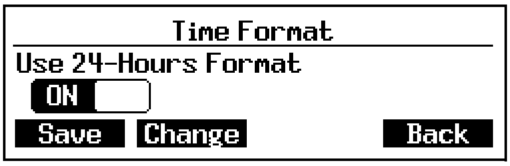
- Click **Submit** to save.

Time Format

Setting Time Format via Phone User Interface

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Phone Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **Time Format** and press **Select** Key.

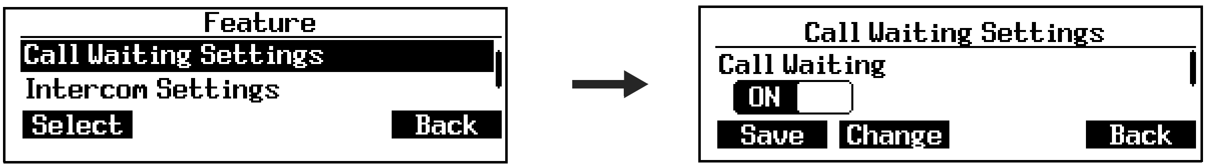


- **Use 24-hour Format** option can be turned **On/Off**. Press the **Change** Key to do so.
- Press **Save** Key.
- Also refer "Date-Time Settings"

Call Waiting

Changing the Call Waiting Settings via Phone User Interface

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Feature** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Call Waiting Settings**.



- Scroll using the **Up/Down Navigation** Key to select Call Waiting or Call Waiting Tone.
- To turn it **On/Off** press **Change** Key.
- Press **Save** Key.

Changing the Call Waiting Settings via Web User Interface

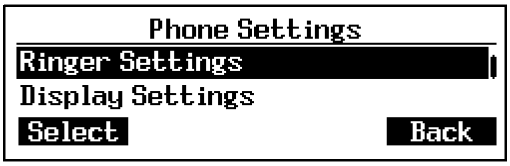
- Log into Jeeves.
- Under **Advanced Settings**, click **System**.

- Scroll to **Call Waiting**.
- By default **Call Waiting** and/or **Call Waiting Tone** check box to are selected (enabled). Clear the check boxes to disable.
- Click **Submit** to save.

Ringer Settings

Changing Ringer Setting via Phone User Interface

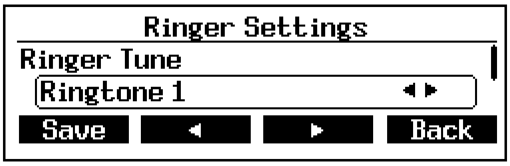
- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Phone Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Ringer Settings**.



- Press **Select** Key.

Ringer Tune

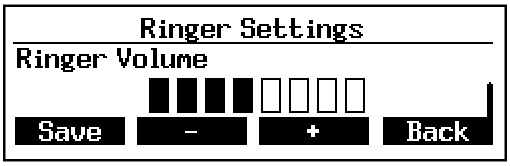
- The **Ringer Tune** setting option appears.



- Scroll using **Right Navigation >** Key or **Left Navigation <** Key, to select the desired Ringtone.
- Press **Save** Key.

Ringer Volume

- Scroll using the **Up/Down Navigation** Key to select the **Ringer Volume** option.



- You can set the desired Volume Level using the Plus **+ or** Minus **-** Context Keys.

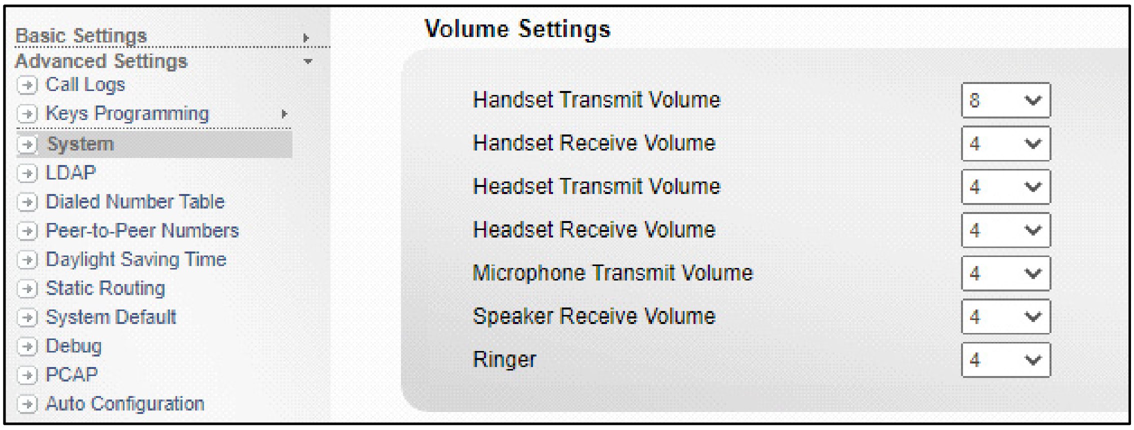
- Press **Save** Key.

Changing Ringer Setting via Web User Interface

You can set the Ringer Volume using Web User Interface.

- Log into Jeeves.

- Under **Advanced Settings**, click **System.**



- Scroll to **Volume Settings**.

- To increase the audibility of the rings for incoming calls, set the **Ringer** volume to the desired volume level.Default:4.

- Click **Submit** to save.

Volume Settings

Changing Volume Settings via Phone User Interface

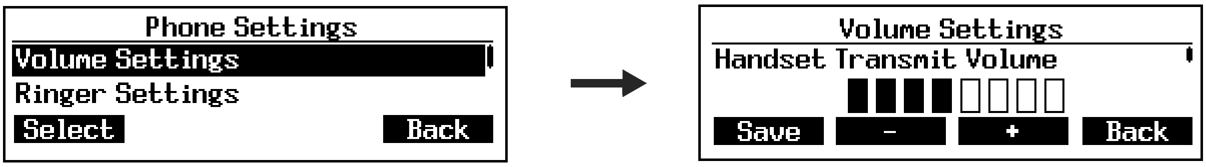
- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **Phone Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **Volume Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select the desired option — Handset Transmit Volume, Handset Receive Volume, Headset Transmit Volume, Headset Receive Volume, Microphone Transmit Volume, Speaker Receive Volume.



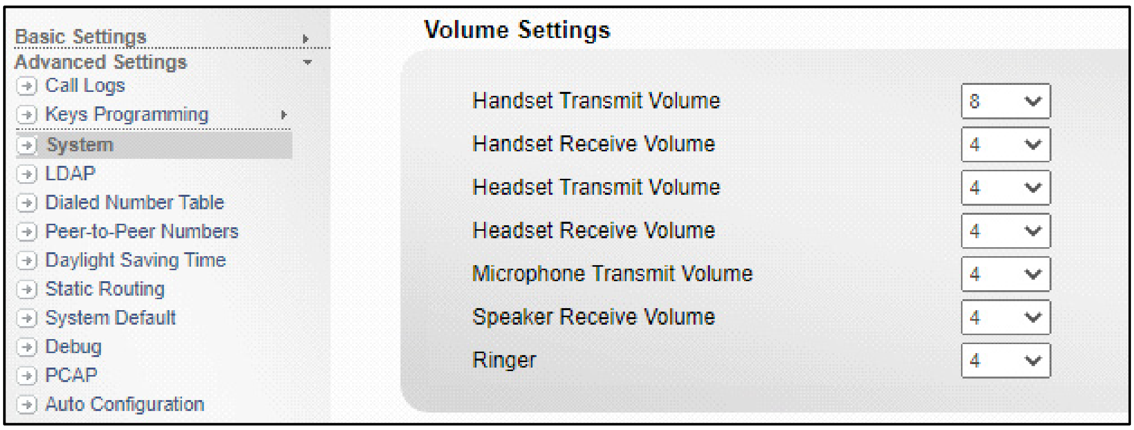
- You can set the desired Volume Level using the Plus **+** **or** Minus **-** Context Keys.

- Press **Save** Key.

Changing Volume Settings via Web User Interface

- Log into Jeeves.

- Under **Advanced Settings**, click **System.**



- Scroll to **Volume Settings**.

- To make your voice audible to the remote user, when using the handset for the call, set the **Handset Transmit Volume** to the desired volume level from 0 to 8. Default: 4.

- To make the remote user's voice audible to you, when using the handset for the call, set the **Handset Receive Volume** to the desired volume level from 0 to 8. Default: 4.

- To make your voice audible to the remote user, when using headset for the call, set the **Headset Transmit Volume** to the desired volume level from 0 to 8. Default: 4.

- To make the remote user's voice audible to you, when using the headset for the call, set the **Headset Receive Volume** to the desired volume level from 0 to 8. Default: 4.

- To make your voice audible to the remote user, when using the speaker for the call, set the **Microphone Transmit Volume** to the desired volume level from 0 to 8. Default: 4.

- To make the remote user's voice audible to you, when using speaker for the call, set the **Speaker Receive Volume** to the desired volume level from 0 to 8. Default: 4.

Accessories

Changing the Accessories Settings via Phone User Interface

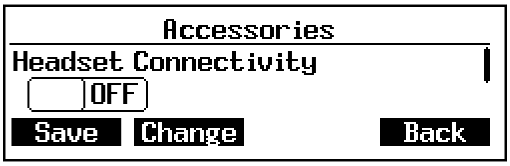
Headset Connectivity

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **Phone Settings** option and press **Select** Key.

- Scroll using the **Up/Down Navigation** Key to select **Accessories** option and press **Select** Key.



- **Headset Connectivity** option can be turned **On/Off**, press **Change** Key to do so.

- Press **Save** Key.

Play Ring on

- Scroll using the **Up/Down Navigation** Key to **Play Ring On** option.

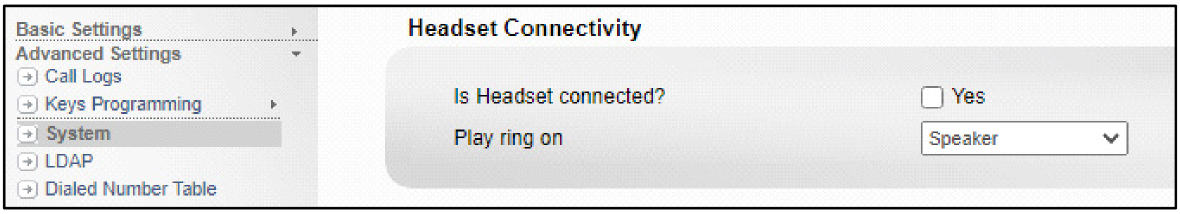
- Scroll using **Right Navigation >** Key or **Left Navigation <** Key, to select the desired option — Speaker, Headset.

- Press **Save** Key.

Headset connectivity via Web User Interface

- Log into Jeeves.

- Under **Advanced Settings**, click **System.**



- Scroll to **Headset Connectivity**.
- Select the **Is Headset connected? Yes** check box to enable. Default: Disabled
- Select the desired option for **Play ring on** — Speaker, Headset. Default: Speaker.
- Click **Submit** to save.

CHAPTER 8 **Making Calls**

There are multiple ways of making calls from the Phone. Among them, most convenient ways include making calls from Keypad or using the Dir Key or Logs Key. You can also make calls to the contacts stored in the directory on the LDAP Server.

Making Calls using Keypad

- Dial the desired number using the Keypad directly or lift the handset to dial the desired number.
- Press **Call** Key.

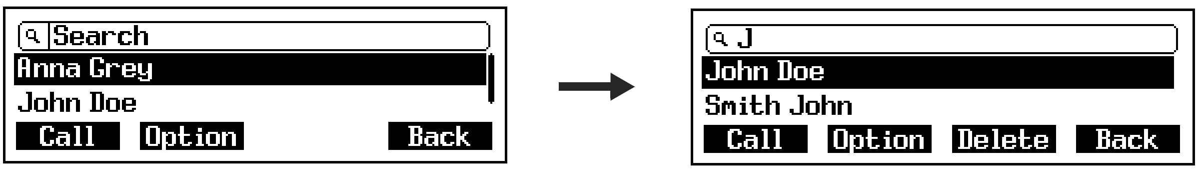
To call using a selective SIP Trunk,

- Press the desired SIP Trunk key. Refer "SIP Trunk" and "Context Sensitive Keys (CSK) Programming" and "DSS Keys Programming".
- Dial the desired number.
- Press **Call** Key.

Making Calls using Dir Key

- Press **Dir** Key.

- Enter the Initial letter(s) of the Contact's name. These are the name configured in the Phone Book.

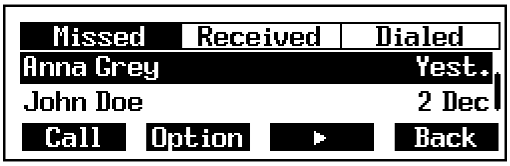


- Scroll using the **Up/Down Navigation** Key to select the Contact from the matching entries.
- Press **Call** Key.

Making Calls using Logs Key

- Press **Logs** Key.

- Press **More >** Key to select the desired tab — Missed Calls, Received Calls, Dialed Calls, Rejected Calls.



- The phone displays the call log details by: Name, Date and Time.
- Scroll using the **Up/Down Navigation** Key to the desired entry and press **Call** Key.

Making Calls Using LDAP

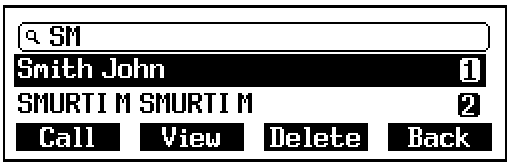
SPARSH VP510 allows you to make calls to the contacts stored in the LDAP Server.

The LDAP Key will be functional only if LDAP is enabled and its parameters are configured. For details, refer to *"LDAP".*

- Press the key assigned to LDAP.

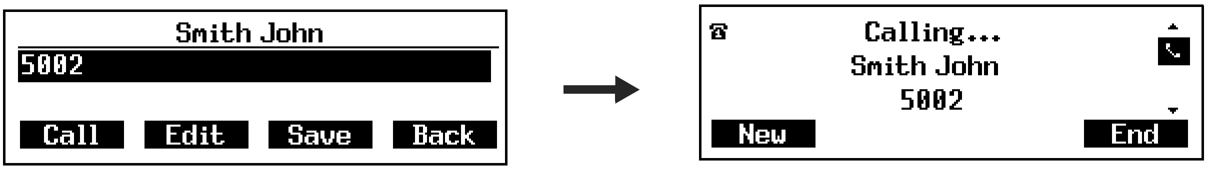
  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **LDAP**.



- Enter the Initial letter(s) of the Contact's name. The contacts name are the names configured in the directory in the LDAP Server.
- Scroll using the **Up/Down Navigation** Key to select the Contact from the matching entries.

- Press **View** Key.



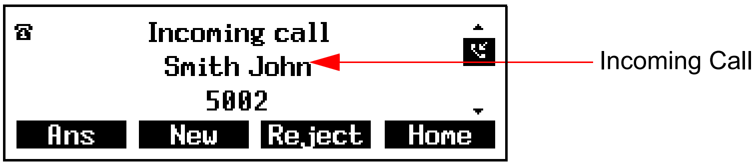
- The details of the selected contact are displayed.

- Press **Call** Key again.

CHAPTER 9

Receiving Calls

You can either answer or reject an incoming call.



- Press **Ans** Key, to answer an incoming call.
- Press **Reject** Key, if you do not wish to take the call.

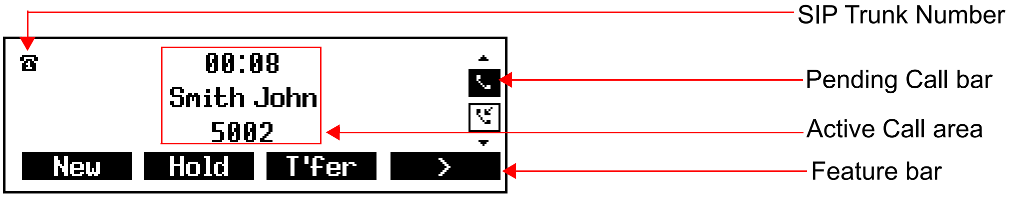
.

- Press **Home** Key, to ignore an incoming call and return to the phone Home screen. If you wish to answer the call, press the respective SIP Trunk DSS Key.

If you have multiple incoming calls, scroll up/down using the **Up/Down Navigation** Key to select the incoming call you wish to answer. Press **Ans** Key. Refer “Handling Multiple Calls”.

CHAPTER 10 Call Screen Functionality

During an active call, you can access the feature and facilities of the System.



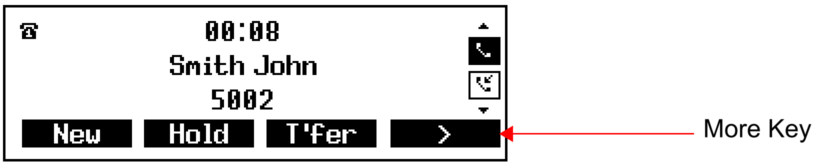
Active Call Area: This displays the User details — Name and Number, as well as the duration of the call.

Pending Call: Refer “Handling Multiple Calls” for details.

SIP Trunk Number: This displays the SIP Trunk number through which the call is routed.

Feature Bar: Displays the call features/facilities that can be accessed. You can change the priority of the features/ functions assigned to these keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

More/Right Navigation > Key: This will appear when you have access to more than 4 features/facilities during the call.



Press **More >** Key, to access other features during an active call.

- Press **New** Key, to make a new call. Refer “Making Calls”.
- Press **Hold** Key, to put the call on Hold. Refer “Call Hold”.
- Press **T’fer** Key to transfer the call to New number or to another held call. Refer “Call Transfer”.
- Press **Conf** Key to create the Conference with a new number or any Held call. Refer “Conference 3-Party”.
- Press **End** Key to end the call.
- Press **Mute** Key, to mute the ongoing call. Refer “Mute”.

Accessing an Active Call from the Home Screen

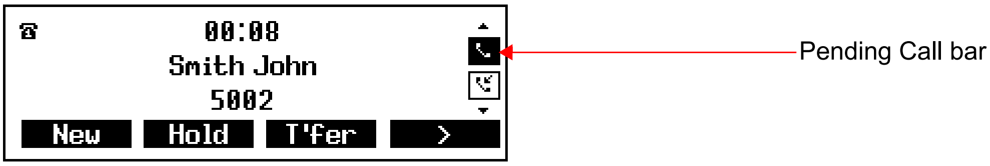
During an incoming or held call, if you wish to access any menu features from the Home Screen, press the **Home** Key.

To resume the held call or answer the incoming call, press the respective SIP Trunk DSS Key,

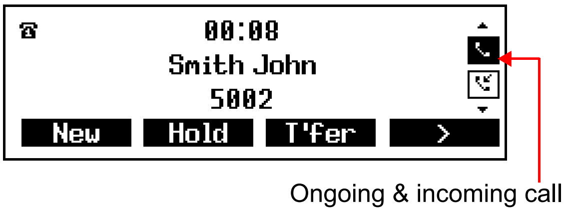
- Press **Unhold** Key, if it is a held call.
- Press **Ans** Key, if it is an incoming call.

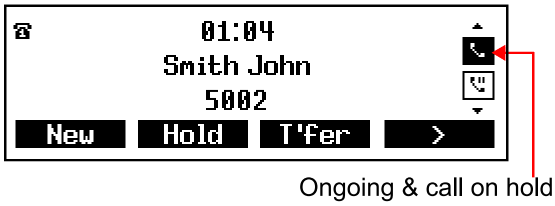
Handling Multiple Calls

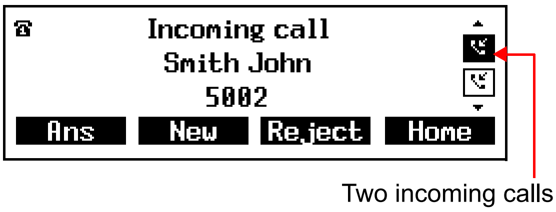
During an ongoing call you can also have held or a waiting incoming call. You can either answer the incoming call or unhold the call put on hold. Similarly, you can also have two incoming calls.These calls appear in the Pending Call bar.



Press the **Up/Down Navigation** Key, to select the respective call. Select the desired Feature Key.







Toggle between Speaker, Handset and Headset

*To use a Headset, make sure you have connected a Headset and have enabled **Use Headset** in Settings. See* “Accessories” for instructions.

You can toggle between a handset, speaker and headset during an active call.

When you are in speech using the **Handset**,

- Press the **Speaker** Key to switch to the Speaker. Replace the Handset.

OR

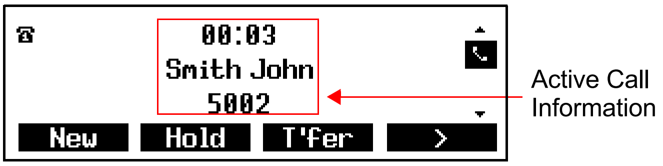
- Press the **Headset** Key to switch to the Headset. Replace the Handset.
- Lift the Handset to switch back to the Handset.

Making a Second Call

You can make a second call using the **Dir** Key, **Logs** Key or **LDAP** Key or using the Keypad when you have an ongoing call or by putting an ongoing call on hold.

To make a Second Call using **Dir** Key, **Logs** Key or **LDAP** Key,

- During an active ongoing call, press **New** Key to view the **Dir** Key, **Logs** Key or **LDAP** Key option.
- Press the desired Key — **Dir** Key, **Logs** Key or **LDAP** Key. The ongoing call is put on hold.



- Enter the Initial letter(s) of the Contact's name in the Search bar.

- Scroll using the **Up/Down Navigation** Key to select the Contact from the matching entries.
- Press **Call** Key.
- After speech if you go On-Hook / Idle, the call of Party 2 will get transferred to the held call.

To make a Second Call by putting a call on hold,

- During an active ongoing call, press **Hold** Key.
- The call will be put on Hold, press **New** Key to make the second call.
- Dial the desired number using the **Dir** Key, **Logs** Key, **LDAP** Key or using the **Keypad**.

  Refer to "Call Hold" to know more.

CHAPTER 12 Receiving a Waiting Call

During an ongoing call, you may receive another call. You can either answer or reject a waiting call.

- During an ongoing call, press **Up/Down Navigation** Key. The incoming call screen appears.
- Press **Ans** Key, to put the first call on Hold and answer the waiting call.
- Press **Reject** Key, to reject the call.
- Press **Home** Key, to return to the Home screen.

CHAPTER 13

Call Features

The Call features include all those features that you can access during a call.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to "Context Sensitive Keys (CSK) Programming".

Call Hold

Call Hold enables you to put an ongoing conversation on hold.

- During an ongoing call, press **Hold** Key or **Hold** ❙❙ Key.

To resume a call put on Hold,

- Press **Unhold** Key or **Hold** ❙❙ Key again.

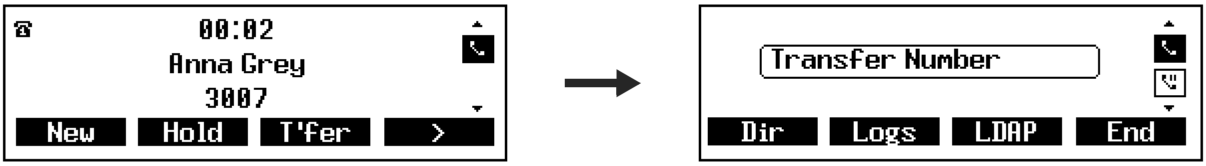
Call Transfer

Call Transfer enables you to relocate an existing call to another number. Calls can be transferred after notifying the destination number about the impending transfer (Attended Transfer) or can be transferred directly without notification (Unattended Transfer).

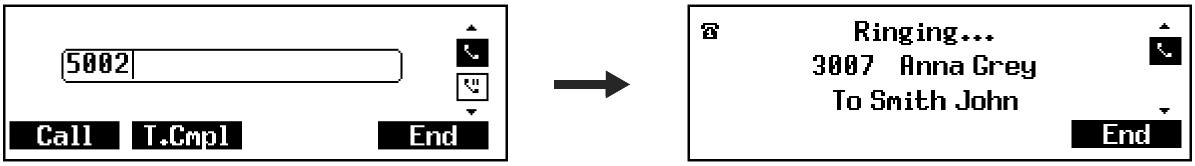
You can change the priority of the features/functions assigned to the Context Keys. To know more, refer “Context Sensitive Keys (CSK) Programming”.

Unattended Transfer

- During an ongoing call, press **T’fer** Key or **Transfer** Key. The ongoing call is put on hold.



- Dial the number of the desired party to whom you want to transfer the call. You can make the call using the Keypad or Dir Key or Logs Key or LDAP Key. To know more, see “Making Calls”.

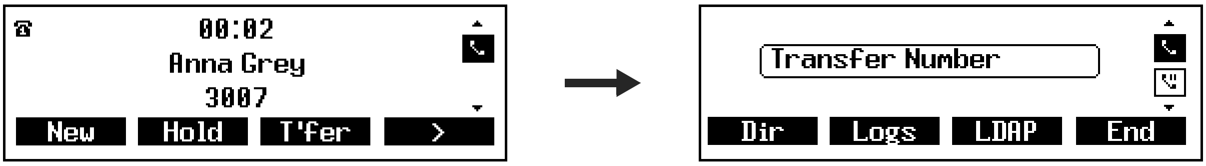


- Press **T.Cmpl** Key. The number starts ringing.
- Press **End** Key. The call is transfered.
- If you do not want to transfer this call to the party on hold and want to unhold the held call, press the **Up/ Down Navigation** Key and then press the desired feature key.

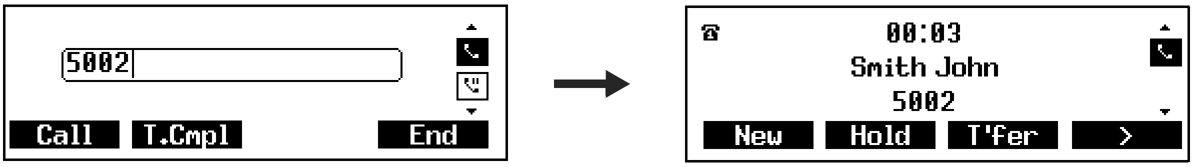
If the party to whom the call is transferred, does not answer, the call will be returned back to you.

Attended Transfer

- During an ongoing call, press **T'fer** Key or **Transfer** Key. The ongoing call is put on hold.



- Dial the number of the desired party to whom you want to transfer the call. You can make the call using the Keypad or Dir Key or Logs Key or LDAP Key. To know more, see "Making Calls".



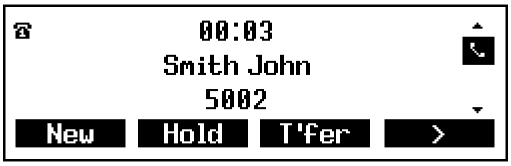
- The dialed party answers the call. Press **T'fer** Key or **Transfer** Key.

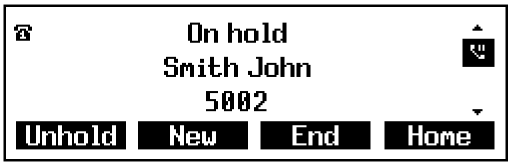
- If you do not want to transfer this call to the party on hold and want to unhold the held call, press the **Up/ Down Navigation** Key and then press the desired feature key.

Call Toggle

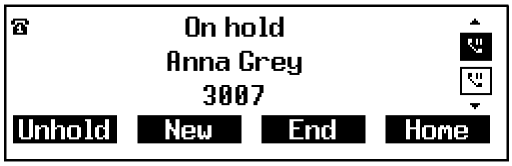
Call Toggle allows you to switch between an active call and a held call.

- During an ongoing call, press **Hold** Key.
- Press **New** Key. Dial the number of the desired party. You can make the call using the **Keypad** or **Dir** Key or **Logs** Key or **LDAP** Key. To know more, see "Making Calls".





- When the dialed party answers the call, press **Up/Down Navigation** Key to select the call on hold. Press **Unhold** Key, speech is established with the party on hold. The active call is put on hold.



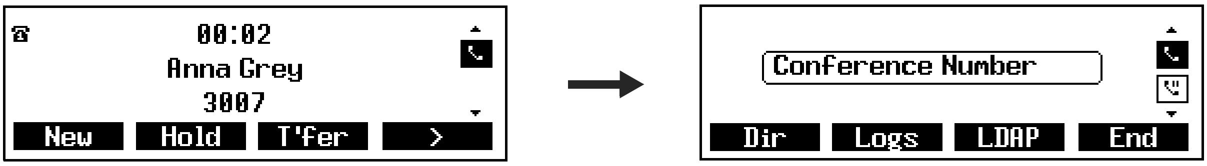
- Repeat the previous step again, to talk to the party on hold.

  In this way, you can talk to both the parties alternately.

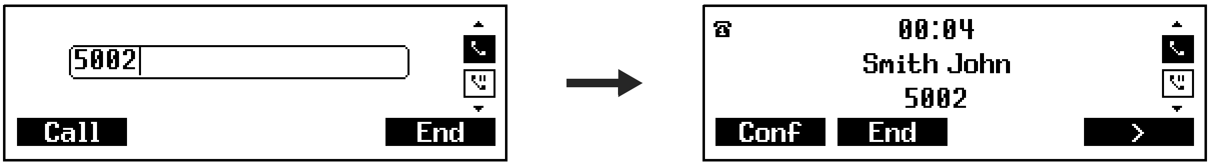
Conference 3-Party

In Conference 3-Party, you can talk to two persons simultaneously. You can merge two separate calls to create a 3-way speech.

- During an ongoing call, press **Conf** Key or **Conference** Key. The active call will be put on hold once the conference key is pressed.

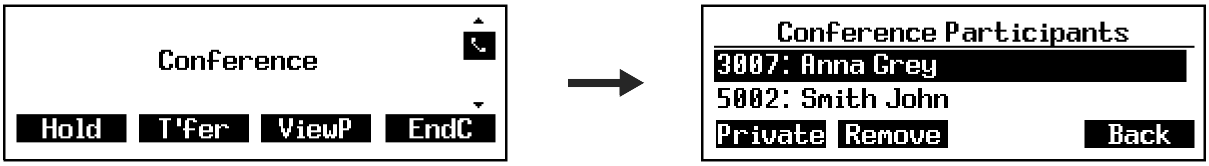


- Dial the number of the desired party with whom you want to establish a conference. You can make the call using the Keypad or Dir Key or Logs Key or LDAP Key. To know more, see “Making Calls”.



- The dialed party answers the call.

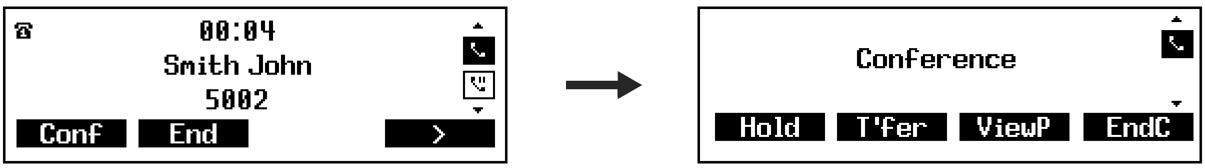
- Using **More >** Key, scroll to and press **Conf** Key or **Conference** Key.



- A 3-party Conference is established. To see the details of the parties in the conference, press **ViewP** Key.

You can also establish the conference with the held call.

- You have one held call and one ongoing call.

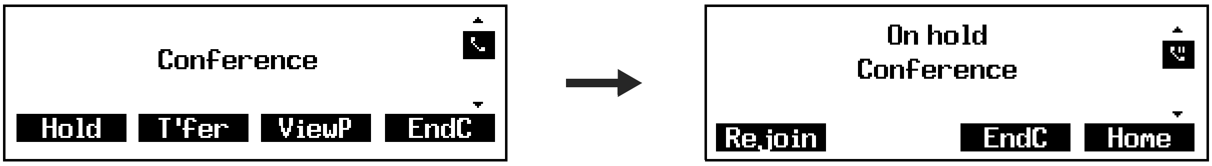


- Using **More >** Key, scroll to and press **Conf** Key or **Conference** Key. A 3-party Conference is established.

Temporary Leave/Rejoin Conference/Permanently Leave Conference

You can leave the conference temporarily and the rejoin again or permanently leave the conference.

- To temporarily leave the conference, during an ongoing conference, press **Hold** Key.



- To rejoin, press **Rejoin** Key.
- To permanently leave the conference, press **T’fer** Key. Speech will be established between the other two parties.

Terminating the Conference

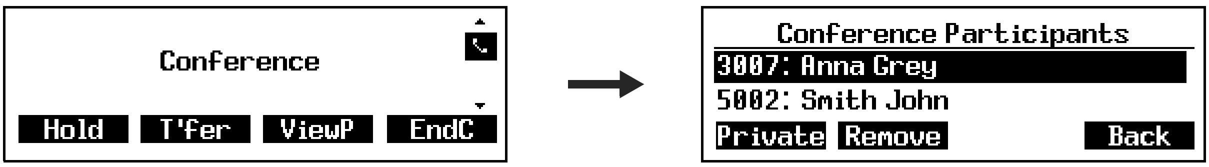
You can terminate the Conference at any point of time.

- Press **EndC** Key during a Conference.

Splitting the Conference to make a Private Talk

You can split the 3-Party Conference into two separate calls and talk to each party separately to make a private talk.

- A 3-party Conference is established. To see the details of the parties in the conference, press **ViewP** Key.

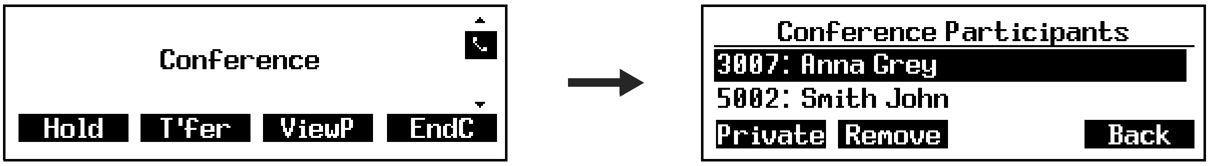


- To have a private talk with any party in the conference, scroll using the **Up/Down Navigation** Key to select the desired party, press **Private** Key, the other will be put on hold.

Remove a Participant

You can remove a participant from the 3-Party Conference.

- A 3-party Conference is established. To see the details of the parties in the conference, press **ViewP** Key.



- To remove any participant from the conference, scroll using the **Up/Down Navigation** Key to select the desired party, press **Remove** Key. The selected party will be disconnected. Speech will be established between other two parties.

Redial

Using Redial you can redial the last number string dialed from your extension.

To make a call to the number,

- Press **Redial** Key.
- The list of last 16 numbers dialed from your extension appears on the LCD.
- Scroll using the **Up/Down Navigation** Key to select the desired number.
- Press **Select** Key.

Headset

Using the Headset feature you can switch the speech path to the Headset directly. To use the Headset, you must enable the **Use Headset** option. For instructions, see "Accessories" in "Customizing Your SPARSH VP510".

You can also get the ring on the headset, if required. For instructions, see "Ringer Volume" in "Customizing Your SPARSH VP510".

To use this feature make sure you have connected a compatible Headset to the phone.

To enable the Headset mode,

- Press Headset Key. The LED of the key glows continuously in Blue.

To disable the Headset mode,

- Press Headset Key.The LED of the key is turned off.

Mute

This feature helps you to disconnect the speech transmission path in the middle of a conversation. You 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.

To mute a call during speech,

- Press Mute Key. The LED of the key glows continuously in Blue.

To unmute a call during speech,

- Press Mute Key again. The LED of the key is turned off.

Intercom

Outgoing Intercom Calls

Intercom is a useful feature in office environment to quickly connect with an operator or secretary. Users can press an intercom key to automatically initiate an outgoing intercom call with a remote extension.

Incoming Intercom Calls

The IP phone supports automatically to answer an incoming intercom call. The phone will play a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it answers an incoming intercom call. You can also enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone, the active call is placed on hold.

The IP phone can process incoming calls differently depending on settings. There are four configuration options for incoming intercom calls:

Intercom Feature	Description
Allow Intercom Calls	Enable or disable the IP phone to answer an incoming intercom call.
Enable Mute	Enable or disable the microphone on the IP phone for intercom calls.
Play Tone	Enable or disable the IP phone to play a tone when it receives an incoming intercom call.
Allow Barge-in	Enable or disable the IP phone to automatically answer an incoming intercom call while there is already an active call on the phone.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to "Context Sensitive Keys (CSK) Programming".

You can also assign a DSS Key to Intercom. To know more, refer to "DSS Keys Programming".

Outgoing Intercom Call via Phone User Interface

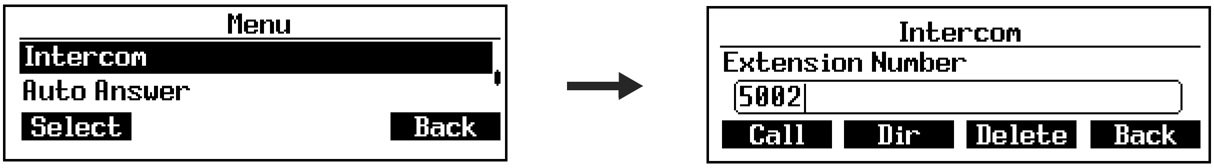
Make sure you have already assigned a DSS/Context Key to Intercom.

- Press the Context/DSS Key assigned to Intercom.

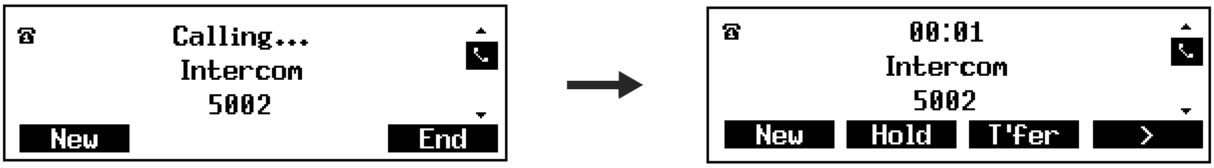
  OR

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **Intercom** option and press **Select** Key.

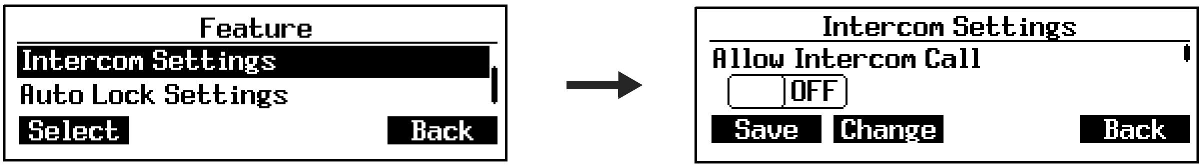


- In **Extension Number**, enter the desired number or press **Dir.** Key to select the desired number.
- Press **Call**.

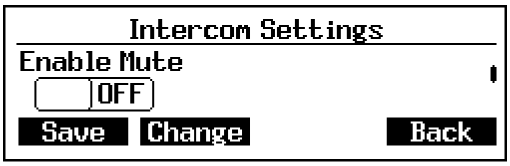


Configuring Incoming Intercom Settings via Phone User Interface

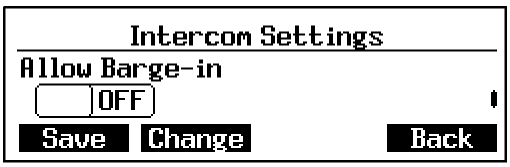
- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Feature** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Intercom Settings** and press **Select** Key.



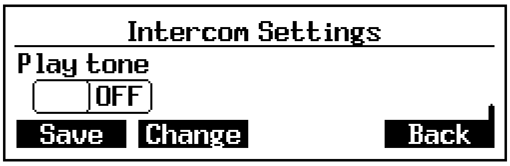
- Scroll using the **Up/Down Navigation** Key to select **Allow Intercom Calls**.
- Press **Change** Key to turn **On/Off,** that is to allow/deny Intercom Calls.



- Scroll using the **Up/Down Navigation** Key to select **Enable Mute**.
- Press **Change** Key to turn **On/Off,** that is to enable /disable Mute.



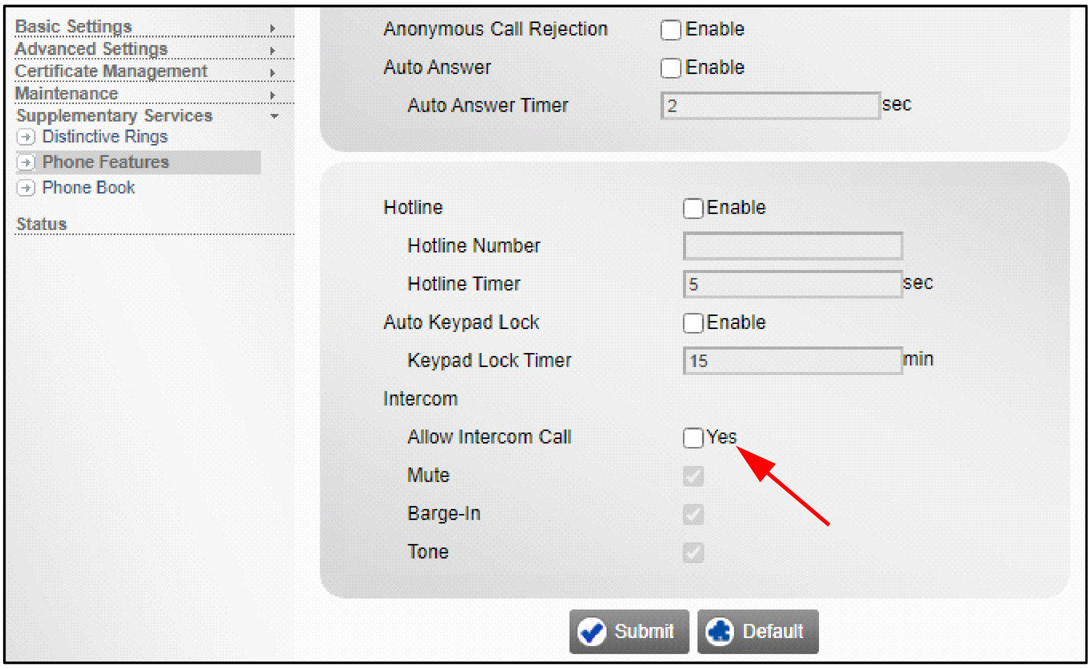
- Scroll using the **Up/Down Navigation** Key to select **Allow Barge-in**.
- Press **Change** Key to turn **On/Off,** that is to allow/deny Barge-in.



- Scroll using the **Up/Down Navigation** Key to select **Play tone**.
- Press **Change** Key to turn Play tone **On/Off**.
- Press **Save** Key.

Configuring Incoming Intercom Settings via Web User Interface

- Log into Jeeves.
- Under **Supplementary Services**, click **Phone Features**.



- Select the **Allow Intercom Call** check box to enable. Default: Disabled.

  You can enable or disable the phone to automatically answer an incoming intercom call. If Allow Intercom is enabled, the phone automatically answers an incoming intercom call. If Allow Intercom is disabled, the phone rejects incoming intercom calls and sends a busy message to the caller.

- Select the respective check box **Mute**, **Barge-In**, **Tone** to enable.Default: Disabled.

- **Mute:** You can mute or un-mute the microphone on the phone for intercom calls automatically. If Intercom Mute is enabled, the microphone is muted for intercom calls. If Intercom Mute is disabled, the microphone works for intercom calls.

- **Barge-In:** You can enable or disable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. If Intercom Barge-In is enabled, the phone automatically answers the intercom call and places the active call on hold. If Intercom Barge-In is disabled, the phone handles an incoming intercom call like a waiting call.

- **Tone:** You can enable or disable the phone to play a warning tone when receiving an intercom call. If Intercom Tone is enabled, the phone plays a warning tone before answering the intercom call. If Intercom Tone is disabled, the phone automatically answers the intercom call without warning.

- Click **Submit**.

Contacts

The Contacts list displays the contacts from the Phone Book. For details refer to “Phone Book”.

You can also save the LDAP contacts in the Phone Book. For details refer to “LDAP”.

You can assign DSS Keys to the desired Contacts. To know more, refer to “DSS Keys Programming”.

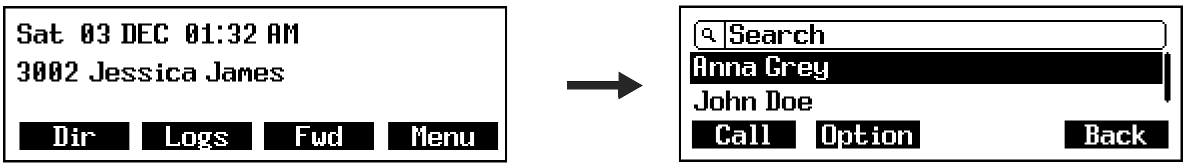
You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

Viewing Contacts via Phone User Interface

- Press **Dir** Key on the Home Screen.

  OR

- Press the **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Contacts**.
- Press **Select** Key.



- Enter the Initial letter(s) of the Contact’s name in the Search bar. These are the name configured in the Phone Book.

- Scroll using the **Up/Down Navigation** Key to the desired Contact from the matching entries.
- You can make a Call, Edit, Delete or view the details of the desired contact. To know more, refer to “Adding Contacts” and “Editing and Deleting Contacts”.

Making a Call to a Contact using DSS Key

Make sure you have already assigned a DSS Key to the desired contact.

To make an outgoing call to the desired contact,

- Press the key assigned to the contact.

Viewing Contacts via Web User Interface

For details refer to "Phone Book".

Adding Contacts

You can add new contacts to the existing Contacts list using the **Add New Contact** option via Phone User Interface. You can view the details of the Contacts you add. This contact will also be updated in the Phone Book in the Jeeves.

Similarly, when you add the contacts in the Phone Book (for details refer to “Phone Book”) using the Web User Interface, these will appear in the list of Contacts displayed in the Phone User Interface (when you press **Dir.** Key or access **Contacts** from the Menu).

Adding Contacts/View Contact Details via Phone User Interface

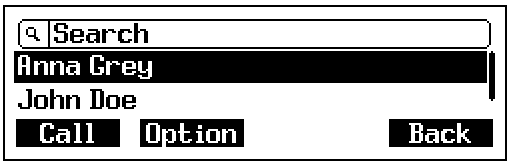
To add a new Contact,

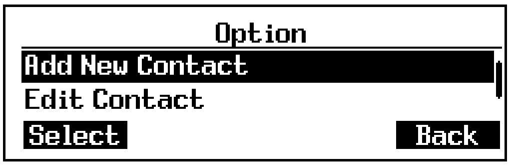
- Press **Dir** Key.

  OR

- Press the **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Contacts**.
- Press **Select** Key.

- Press **Option** Key and scroll using the **Up/Down Navigation** Key to select **Add New Contact.**
- Press **Select** Key.





- Enter the **Name** and scroll using **Up/Down Navigation** Key to enter the **Number**.

- Press **Save** Key. The contact is saved and is automatically updated in the Phone Book also.

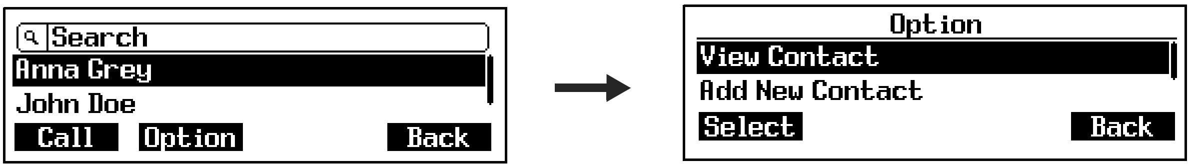
To view the details of the contact,

- Press **Dir** Key.

  OR

- Press the **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Contacts**.
- Press **Select** Key.

- Press **Option** Key and scroll using the **Up/Down Navigation** Key to select **View Contact.**



- Press **Select** Key.

  The details of the contact are displayed.

Adding Contacts/View Contact Details via Web User Interface

For details refer to “Phone Book”.

Editing and Deleting Contacts

You can edit or delete contacts from the Phone User Interface (when you press **Dir.** Key or access **Contacts** from the Menu) as well as the Web User Interface (though the Phone Book). Refer "Phone Book".

When the contact is editted or deleted, the changes will be reflected in both the interfaces.

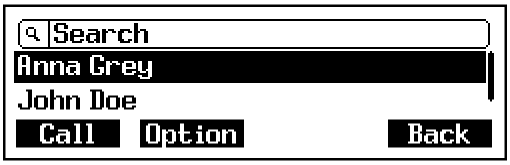
Editing/Deleting Contacts via Phone User Interface

Editing Contacts

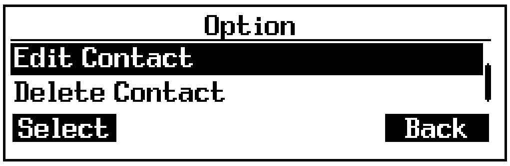
- Press **Dir** Key

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Contacts**.
- Press **Select** Key.
- Enter the Initial letter(s) of the Contact's name in the Search bar.
- Scroll using the **Up/Down Navigation** Key to the desired Contact.



- Press **Option** Key.



- Scroll using the **Up/Down Navigation** Key to select **Edit Contact**. Press **Select** Key.

- Edit the **Name**, if required.

- Scroll using the **Up/Down Navigation** Key to edit the **Number,** if required
- Press **Save** Key.

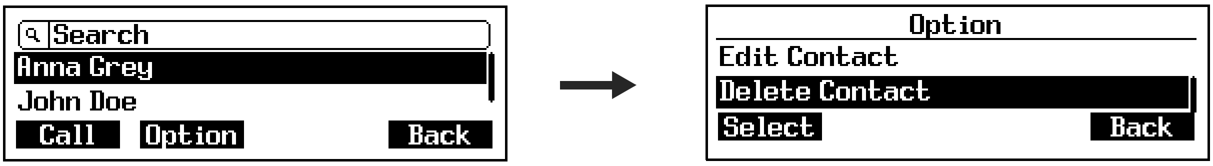
Deleting Contacts

- Press **Dir** Key

  OR

- Press the **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Contacts**.
- Press **Select** Key.

- Enter the Initial letter(s) of the Contact's name in the Search bar.
- Scroll using the **Up/Down Navigation** Key to the desired Contact.
- Press **Option** Key.



- Scroll using the **Up/Down Navigation** Key to **Delete Contact**. Press **Select** Key.
- A confirmation message appears. Press **Yes**. The selected contact is deleted.

Editing/Deleting Contacts via Web User Interface

For details refer to "Phone Book".

CHAPTER 15

Call Logs

Call Logs displays the history of all Missed, Received, Dialed, Rejected as well as Forwarded Calls.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

You can also assign DSS Keys to the desired Call Logs — Missed, Dialed, Received, Rejected. To know more, refer to “DSS Keys Programming”.

You can view the Call Logs from the Phone User Interface as well as the Web User Interface.

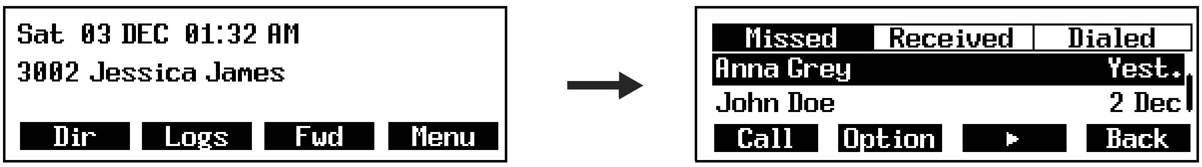
Forwarded list is displayed only in the Web User Interface.

Viewing Call Logs via Phone User Interface

Viewing Call Logs using Logs Key

To view the Call Logs,

- Press **Logs** Key on the Home Screen.



- Scroll using the **Right Navigation >** Key to select the desired Call Log tab — Missed, Received, Dialed, Rejected (calls that were rejected by you as well as incoming calls that were rejected as DND is set on your phone).
- The phone displays the list of last 100 calls. The details displayed are: Name, Date/Time.
- Scroll using the **Up/Down Navigation** Key to the desired entry.
- Press **Call** Key, to make a call.
- Press **Option** Key, you have the following options — Details, Edit before call, Save, Delete, Delete All.
- Press **Back** Key, to return to the Menu Screen.

Viewing Call Logs using DSS Key

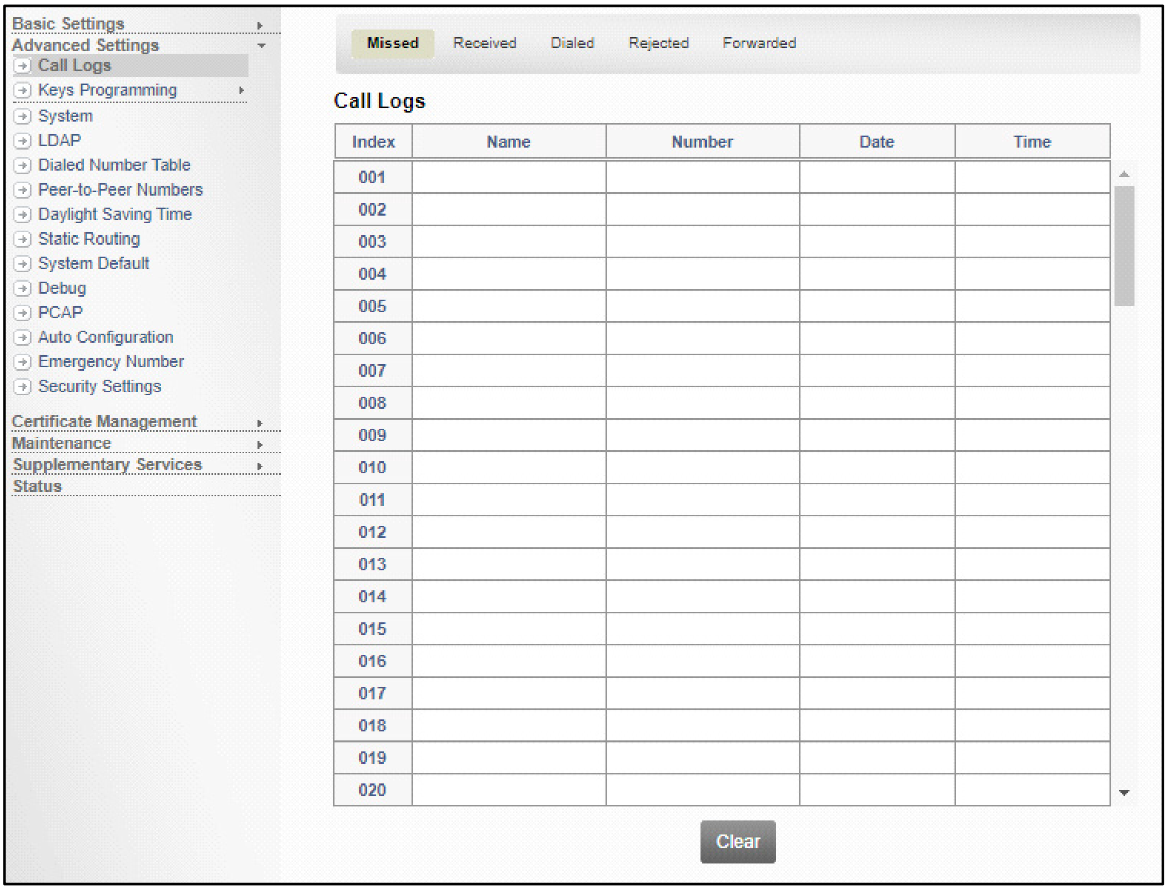
Make sure you have already assigned a DSS Key to the desired Call Log type — Missed, Dialed, Received, Rejected.

To view the call logs,

- Press the key assigned to the Call Log.
- The selected call logs list appears.

Viewing Call Logs via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Call Logs**.



- Click the desired tab — Missed, Received, Dialed, Rejected (calls that were rejected by you as well as incoming calls that were rejected as DND is set on your phone), Forwarded (If you have set Call Forward, then the incoming calls that were forwarded).

  Each tab displays a list of 100 records.

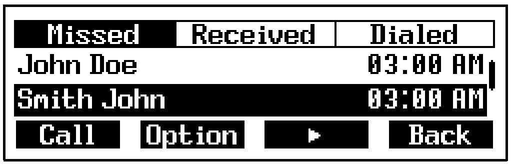
Viewing Details and Adding it to Contacts

You can view the details the of the calls present in the Missed, Received, Dialed or Rejected Call Logs list.The calls present in any log can also be saved.

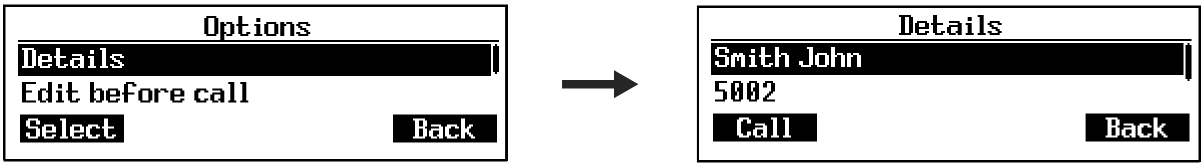
You can view the details of any entry in the Call Log and add it to contacts from the Phone User Interface only.

To view the details of any entry in the Call Log,

- Press **Logs** Key on the Home Screen.
- Scroll using the **Right Navigation >** Key to select the desired Call Log tab — Missed, Received, Dialed, Rejected.
- Scroll using the **Up/Down Navigation** Key to the desired entry and press **Option** Key.



- Scroll using the **Up/Down Navigation** Key to select **Details**.
- Press **Select** Key.

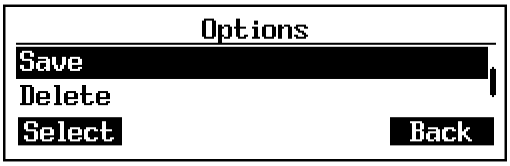


- It displays the Name, Number, SIP Trunk, Date and Time of call.

  You can either make a call directly or select the SIP Trunk through which you wish to place the call. To make a call directly, press **Call** Key. To select the SIP Trunk, select the desired option **SIP 1** or **SIP 2**.

To add any entry in the Call Log to Contacts,

- Press **Logs** Key on the Home Screen.
- Scroll using the **Right Navigation >** Key to select the desired Call Log tab — Missed, Received, Dialed, Rejected.
- Scroll using the **Up/Down Navigation** Key to the desired entry and press **Option** Key.
- Press **Select** Key.



- Scroll using the **Up/Down Navigation** Key to select **Save**.

  If required you can modify the name and number.

- Press **Save** Key.

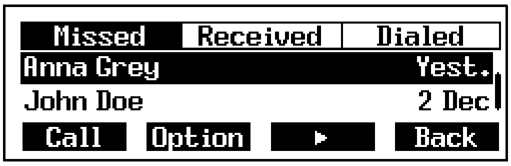
Missed Call Notification

If you have missed any calls,

- a Missed Call Notification appears on the Home Screen.
- the Ringer LED will turn 1 second on and 5 seconds off.

To view the missed calls,

- Press **Logs** Key on the Home Screen.
- Scroll using the **Right Navigation >** Key to select the **Missed** Calls Log.



- The phone displays the list of missed calls.
- Press **Call** Key, to call.
- Press **Option** Key, to view **Details**, **Edit before call, Save, Delete, Delete All.**
- Press **Back** Key, to return to the Menu Screen.

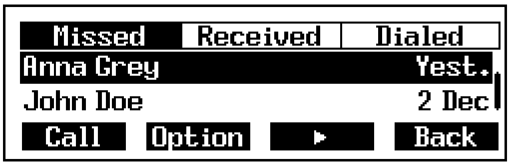
As soon as you access the Missed Calls log, the notification will disappear.

Editing an Entry before Placing a Call

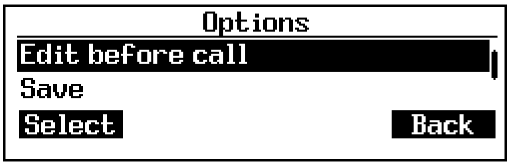
You can edit an number present in the Missed, Received, Dialed or Rejected Call Logs list.

To edit an entry in the Call Log,

- Press **Logs** Key on the Home Screen.
- Scroll using the **Right Navigation >** Key to select the desired Call Log tab — Missed, Received, Dialed, Rejected.



- Scroll using the **Up/Down Navigation** Key to the desired entry.
- Press **Option** Key.



- Scroll using the **Up/Down Navigation** Key to select **Edit before call**.
- Press **Select** Key.
- Edit the **Number** as per you requirement.
- Scroll using the **Up/Down Navigation** Key to select **SIP Trunk**. To edit, scroll using **Right Navigation >** Key or **Left Navigation <** Key, to select the desired trunk.
- Press **Call** Key.

Deleting Call Logs

You can delete a single entry at a time or delete all entries at once from a specific Call Log.

The Call Logs can be deleted from the Phone User Interface as well as the Web User Interface.

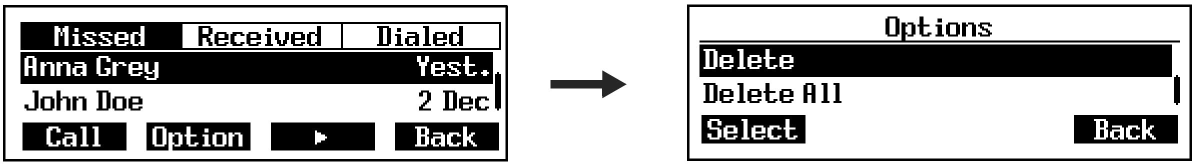
Single entries can be deleted using the Phone User Interface only.

Deleting Call Logs via Phone User Interface

Deleting a Single Entry

To delete an entry from the selected Call Logs list,

- Press **Logs** Key on the Home Screen.
- Scroll using the **Right Navigation >** Key to select the desired Call Log tab — Missed, Received, Dialed, Rejected.
- Scroll using the **Up/Down Navigation** Key to the desired entry.
- Press **Option** Key.

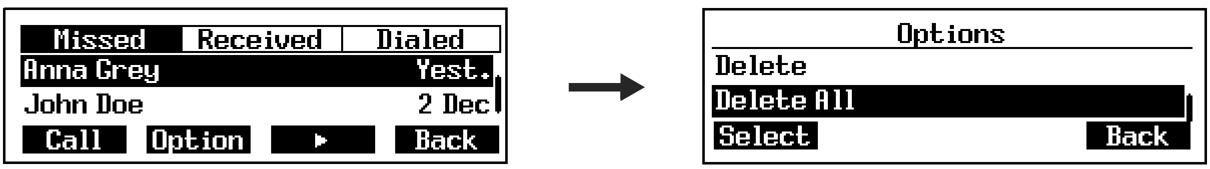


- Scroll using the **Up/Down Navigation** Key to select **Delete**.
- Press **Select** Key.

Deleting all Entries at Once

To delete all the entries from a specific Call Log,

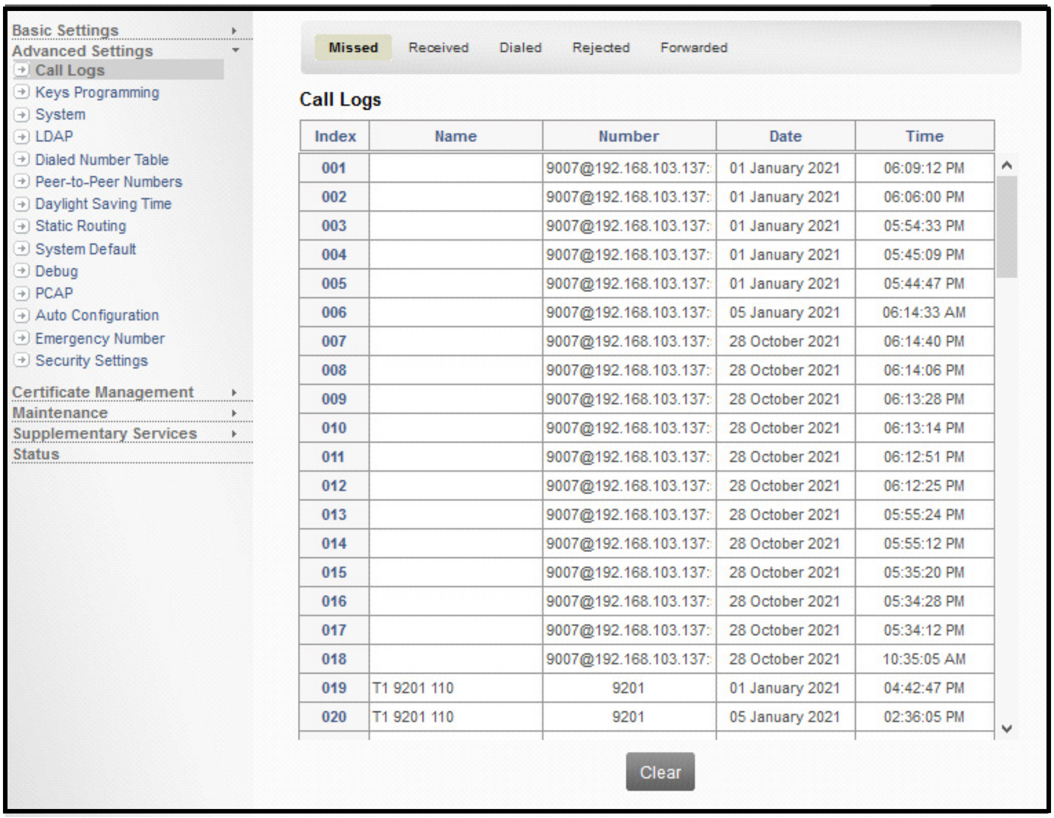
- Press **Logs** Key on the Home Screen.
- Scroll using the **Right Navigation >** Key to select the desired Call Log tab — Missed, Received, Dialed, Rejected.
- Press **Option** Key.



- Scroll using the **Up/Down Navigation** Key to select **Delete All**.
- A confirmation message appears. Press **Yes** Key. All entires in the selected call log tab are deleted.

Deleting Call Logs via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Call Logs.**



- Click the desired tab — Missed, Received, Dialed, Rejected (calls that were rejected by you as well as incoming calls that were rejected as DND is set on your phone), Forwarded (If you have set Call Forward, then the incoming calls that were forwarded).

  Click **Clear**. All the logs in the selected tab will be cleared.

CHAPTER 16 Advanced Features

Auto Answer

Auto Answer allows IP phones to automatically answer an incoming call. IP phones will not automatically answer the incoming call during a call even if auto answer is enabled. Auto Answer delay defines a period of delay time before the IP phone automatically answers incoming calls.

Auto Answer can be set/canceled for all the SIP Trunks or for each SIP Trunk individually depending on the “Feature Access Method” set in “System Parameters”.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

You can also assign a DSS Key to Auto Answer. To know more, refer to “DSS Keys Programming”.

Auto Answer via Phone User Interface

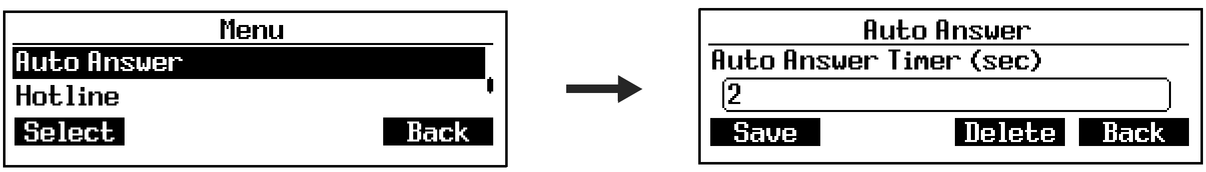
Set/Cancel Auto Answer using CSK/Menu

To enable Auto Answer,

- Press the Context Key assigned to Auto Answer.

  OR

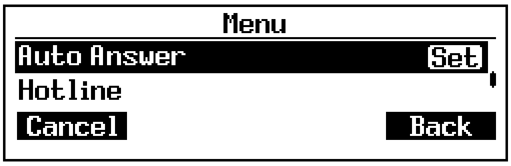
- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Auto Answer**.
- Press **Select** Key.



*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you enable Auto Answer, you need to first select the desired SIP Trunk and then follow the steps as mentioned here.*

- Configure the **Auto Answer Timer** in seconds after which the incoming call should be answered automatically.



- Press **Save** Key. The **Set** icon appears.

To cancel Auto Answer,

- Press the Context Key assigned to Auto Answer.

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Auto Answer**.

*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you disable Auto Answer, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*

- Press **Cancel** Key. The **Set** icon disappears.

Set/Cancel Auto Answer using DSS Key

Make sure you have already assigned a DSS Key to Auto Answer. DSS Key can be assigned either Phone wise or SIP Trunk wise, in this case the DSS Key functionality will be as per the set mode.

To set Auto Answer,

- Press the DSS Key assigned to Auto Answer.

  The LED of the key glows continuously in Red.

To Cancel Auto Answer,

- Press the DSS Key assigned to Auto Answer again.

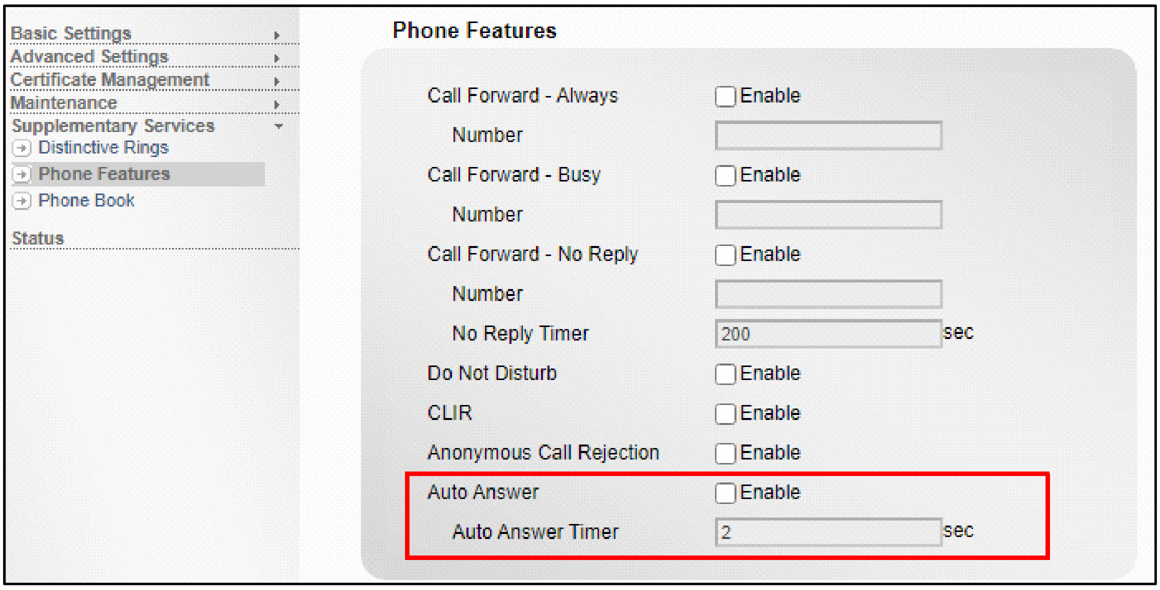
  The LED of the key is turned off.

Enable/Disable Auto Answer via Web User Interface

You can enable/disable Auto Answer depending on the Feature Access Mode set in System parameters.

When Feature Access Mode is set as Phone wise, to enable/disable Auto Answer,

- Log into Jeeves.
- Under **Supplementary Services**, click **Phone Features**.



- Select the **Auto Answer** check box to enable.Default: Disabled.
- Configure the **Auto Answer Timer**. Default:2 sec.
- Click **Submit**.

When Feature Access Mode is set as SIP Trunk wise, to enable/disable Auto Answer,

- Log into Jeeves.
- Under **Supplementary Services**, click **Trunk Features**.
- Select the desired SIP Trunk and follow the same steps to enable/disable Auto Answer as mentioned above.

Anonymous Call Rejection (ACR)

You can use anonymous call rejection to reject incoming calls from anonymous callers. Anonymous Call Rejection automatically rejects incoming calls from callers who deliberately block their identities and numbers from showing up.

The anonymous call rejection on code and anonymous call rejection off code configured on IP phones are used to activate/deactivate the server-side anonymous call rejection feature. They may vary on different servers.

You can configure ACR from the Phone User interface as well as Web User Interface.

ACR can be set/canceled for all the SIP Trunks or for each SIP Trunk individually depending on the “Feature Access Method” set in “System Parameters”.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

You can also assign a DSS Key to ACR. To know more, refer to “DSS Keys Programming”

ACR via Phone User Interface

Set/Cancel ACR using CSK/Menu

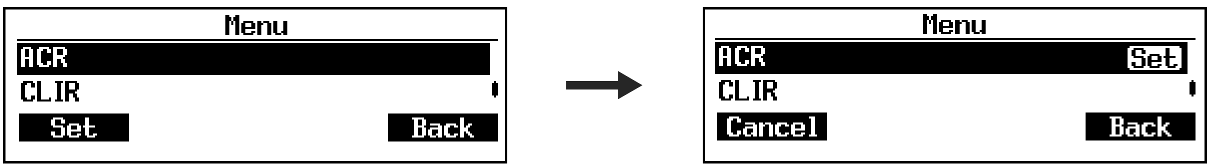
To set ACR,

- Press the key assigned to ACR.

  OR

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **ACR**.



*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you set ACR, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*

- Press **Set** Key. The **Set** icon appears.

To cancel ACR,

- Press the key assigned to ACR.

OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **ACR**.

*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you cancel ACR, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*

- Press **Cancel** Key. The **Set** icon disappears.

Set/Cancel ACR using DSS Key

Make sure you have already assigned a DSS Key to ACR. DSS Key can be assigned either Phone wise or SIP Trunk wise, in this case the DSS Key functionality will be as per the set mode.

To Set ACR,

- Press the key assigned to ACR.

  The LED of the key glows continuously in Red.

To Cancel ACR,

- Press the key again.

  The LED of the key is turned off.

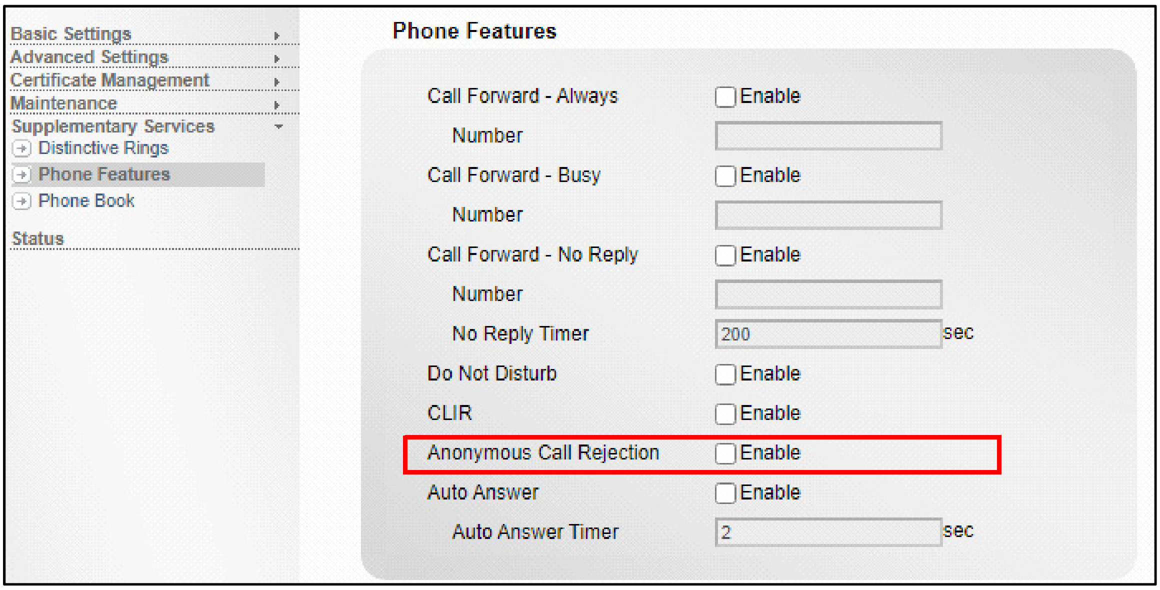
Enable/Disable ACR via Web User Interface

You can enable/disable ACR depending on the Feature Access Mode set in System parameters.

When Feature Access Mode is set as Phone wise, to enable/disable ACR,

- Log into Jeeves.

- Under **Supplementary Services**, click **Phone Features**.



- Select the **Anonymous Call Rejection** check box to enable.Default: Disabled.
- Click **Submit.**

When Feature Access Mode is set as SIP Trunk wise, to enable/disable ACR,

- Log into Jeeves.
- Under **Supplementary Services**, click **Trunk Features**.
- Select the desired SIP Trunk and follow the same steps to enable/disable ACR as mentioned above.

Automatic Number Translation

SPARSH VP510 supports two SIP Trunks, allowing it to register with the SIP Servers of two ITSPs/IP-PBXs.

When outgoing calls are made from the phone, the SIP Trunk is selected on the basis of the **Outgoing Call Routing** configured for that trunk, and the calls are routed from the SIP Server of the ITSP/IP-PBX configured on that SIP trunk.

Generally, persons who use the phone to dial out numbers without knowing the routing mechanism of the phone or the SIP Trunk (ITSP/IP-PBX) that will be selected for making the calls.

Many a time, the number string dialed by the users is not understood by the network through which the call is to be routed. So, SPARSH VP510 with the help of the Automatic Number Translation feature, dials the number in such a way that it is understood by the network and call reaches the correct destination.

Let us understand this feature with the help of an example:

- An organization has registered itself with Pulver.com and Voiptalk.com. SPARSH VP510 is installed for VoIP calls.
- Pulver.com and Voiptalk.com suggest different prefixes for calling different domains.
- Pulver.com suggests its users to dial *777 to make calls to abc.com, whereas Voiptalk.com suggests its users to dial *888 to make calls to abc.com.
- The System Engineer has suggested a set of prefixes to the users of SPARSH VP510 to make calls to other domains to help users remember only one prefix code per domain, and to ease the routing of calls. The System Engineer has suggested users to dial *234 to make calls to abc.com.
- '9874' is the number of a subscriber of abc.com.
- To call this subscriber, the user must dial *2349874 (*234 being the prefix code suggested by the System Engineer to make call to abc.com and 9874 being the number of a subscriber of abc.com).
- The routing logic is so configured that this call will be routed through Pulver.com. However, Pulver.com will not recognize the string *234 as it expects the string *777 and will not route the call correctly.
- Automatic Number Translation resolves this.
- The System Engineer configures the Automatic Number Translation table, with *234 as the Dialed Number, Strip Digits as 4 and add Prefix as *777.
- Now, when a user dials *2349874 from SPARSH VP510, the Automatic Translation logic translates this string to *7779874 and dials it out on Pulver.com. Pulver.com now recognizes this string and routes the call correctly.

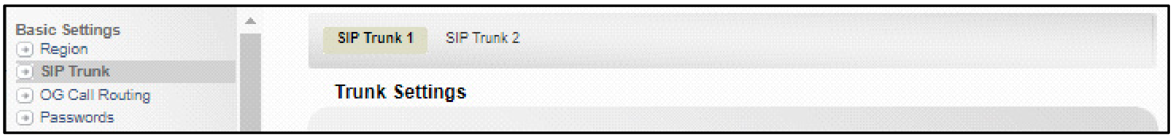
Automatic Number Translation can also be used for Abbreviated Dialing, whereby short digit codes can be dialed in place of long number strings.

Automatic Number Translation can also be used to strip off some digits before dialing. For example, an IP phone user dials an UK number as 0044-xxxx whereas the ITSP expects 44-xxxx. In the Automatic Number Translation

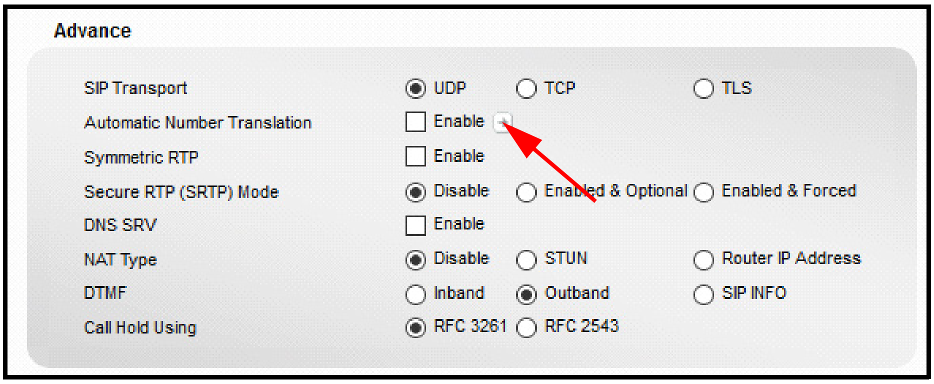
(ANT) table, configure 00 as Dialed Number, Strip Digits as 2 and Add Prefix as Blank. Now, when the user dials 0044-xxxx, the system will strip off 00 and dial out 44-xxxx.

Configuring Automatic Number Translation (ANT) via Web User Interface

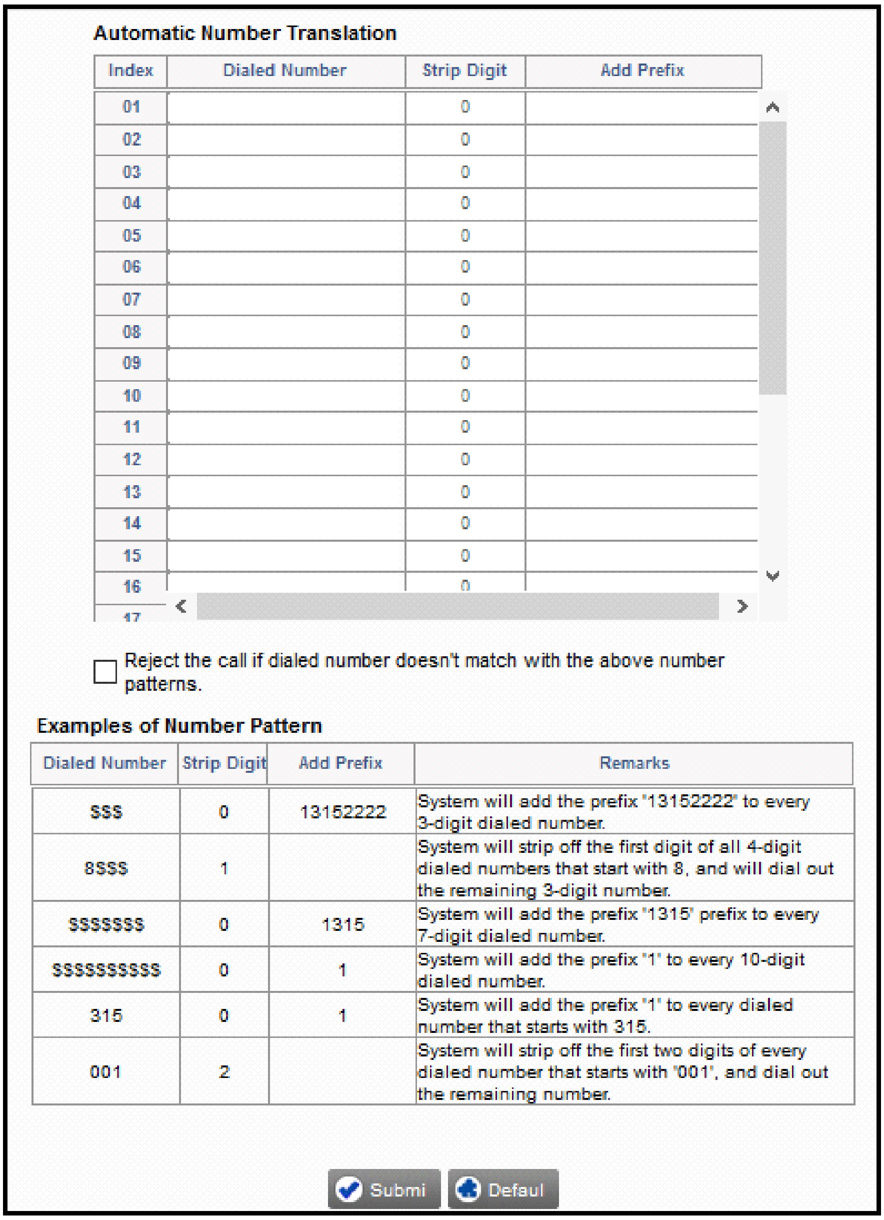
- Make a list of numbers that need to be modified before being dialed out from SIP Trunks, along with the number of Digits need to stripped off as well as the prefix, if required.
- Make must enabled ANT on the desired SIP Trunk. To do so,
- Log into Jeeves.
- Under **Basic Settings**, click **SIP Trunk.**
- Select the desired SIP Trunk, for example, **SIP Trunk 1** tab.



- Scroll to **Advance**, and select the **Automatic Number Translation Enable** check box. Default: Disabled
- Click **Settings** , the Automatic Number Translation page opens.



Configure the following:



Automatic Number Translation

Index	Dialed Number	Strip Digit	Add Prefix
01		0	
02		0	
03		0	
04		0	
05		0	
06		0	
07		0	
08		0	
09		0	
10		0	
11		0	
12		0	
13		0	
14		0	
15		0	
16		0	
17			

☐ Reject the call if dialed number doesn't match with the above number patterns.

Examples of Number Pattern

Dialed Number	Strip Digit	Add Prefix	Remarks
$$$	0	13152222	System will add the prefix '13152222' to every 3-digit dialed number.
8$$$	1		System will strip off the first digit of all 4-digit dialed numbers that start with 8, and will dial out the remaining 3-digit number.
$$$$$$$	0	1315	System will add the prefix '1315' prefix to every 7-digit dialed number.
$$$$$$$$$$	0	1	System will add the prefix '1' to every 10-digit dialed number.
315	0	1	System will add the prefix '1' to every dialed number that starts with 315.
001	2		System will strip off the first two digits of every dialed number that starts with '001', and dial out the remaining number.

Submi Defaul

- **Dialed Number:** Enter the numbers that will be dialed from this SIP Trunk, which need to be modified before being dialed out. The numbers may be complete strings or parts thereof. Maximum 40 characters. Default: Blank.
- **Strip Digit:** Enter the number of digit(s) to be stripped off from the number string entered in the Dialed Number before the number is out-dialed. If you do not want any digits to be stripped, enter '0'. Default:0.
- **Add Prefix:** Enter the digit(s) which are to be added as prefix to the number string entered in the Dialed Number before the number is out-dialed.Maximum 40 characters. Default: Blank.
- Click **Submit** and close the window.

• Follow the same steps to configure ANT on another SIP Trunk.

Busy Lamp Field (BLF)

SPARSH VP510 supports Busy Lamp Field feature. The purpose of Busy Lamp Field is to provide the real-time status information about the availability or status of a particular user using visual indication.

In typical scenario, a BLF is associated with a physical indicator, often a LED light on a phone.The indicator changes color or blinks to reflect the status of the monitored user.

It allows one user to monitor the status of another user. For example, a receptionist might have a BLF set to monitor the status of multiple executives within an organization. In this way, at a glance the receptionist can know whether an executive is busy, available or on a call, facilitating efficient call handling and transfer.

For this feature to work, make sure:

- SIP Trunk Mode is set as Proxy, refer to “SIP Trunk”.
- Busy Lamp Field Subscription is enabled for the desired SIP Trunk, refer to “SIP Trunk”.
- Registrar Server / Fallback Server of the SIP Trunk supports this feature, refer to “SIP Trunk”.
- BLF Keys are configured for the desired users, refer to “Configuring BLF Keys via Web User Interface”.

You can configure BLF Keys using the Web User Interface only.

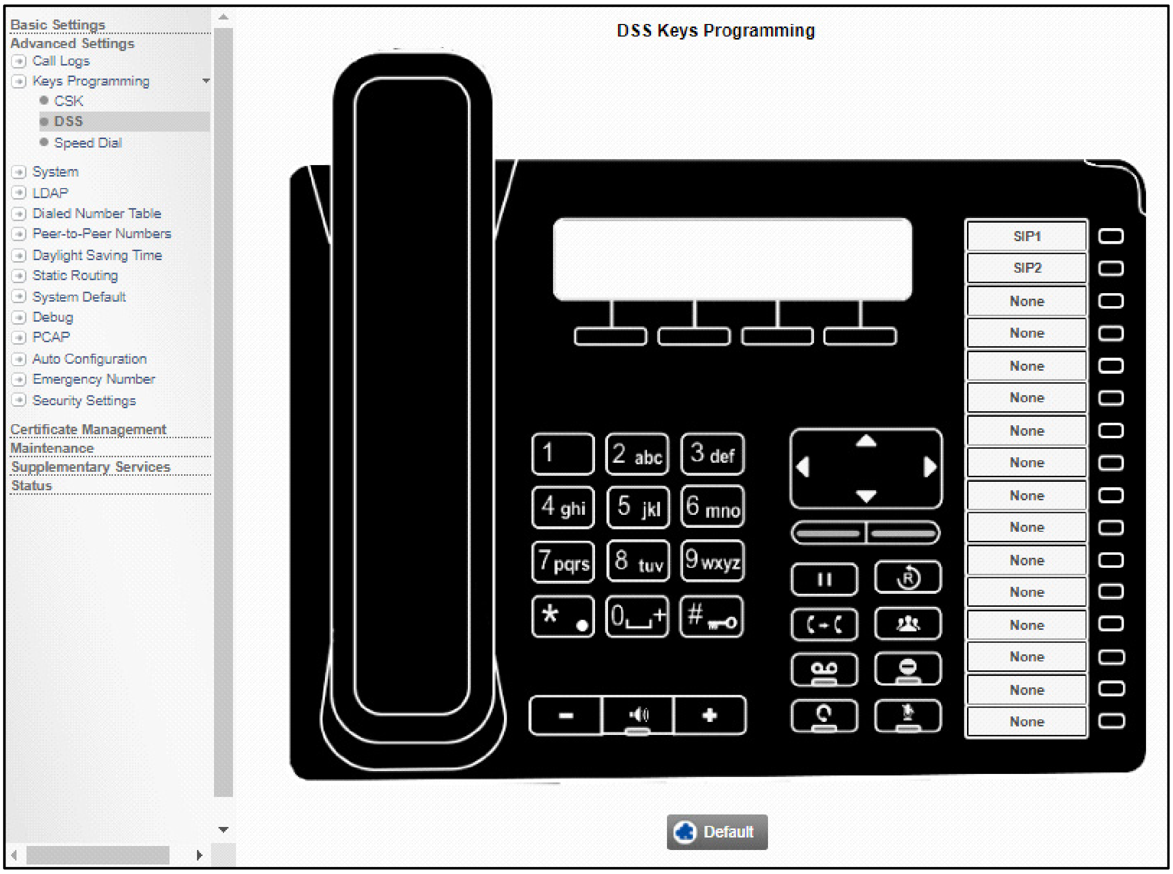
When the BLF Keys are assigned to the desired users whom you wish to monitor, then the status of these users are indicated through the LED of the BLF Keys as follow:

Status of user who is assigned a BLF Key	Description
Idle	OFF
Ringing	Fast Blinking (Red)
Busy	Continuous ON (Red)

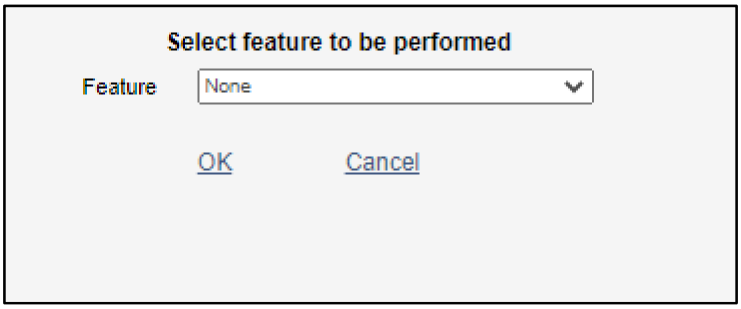
Configuring BLF Keys via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Key Programming**
- Click **DSS.**

The **DSS Keys Programming** page appears.



- The **Select feature to be performed** page appears.

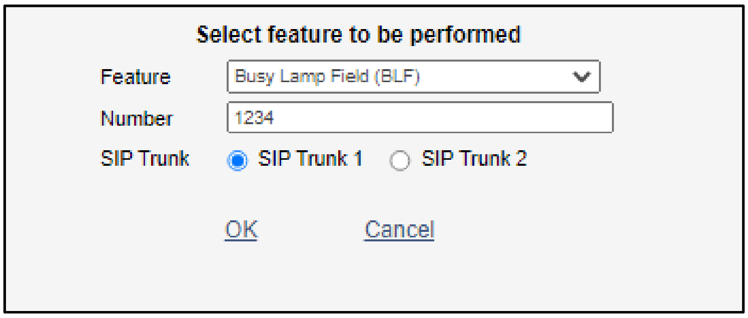


For example if you wish to assign a BLF Key to the extension user 1234.

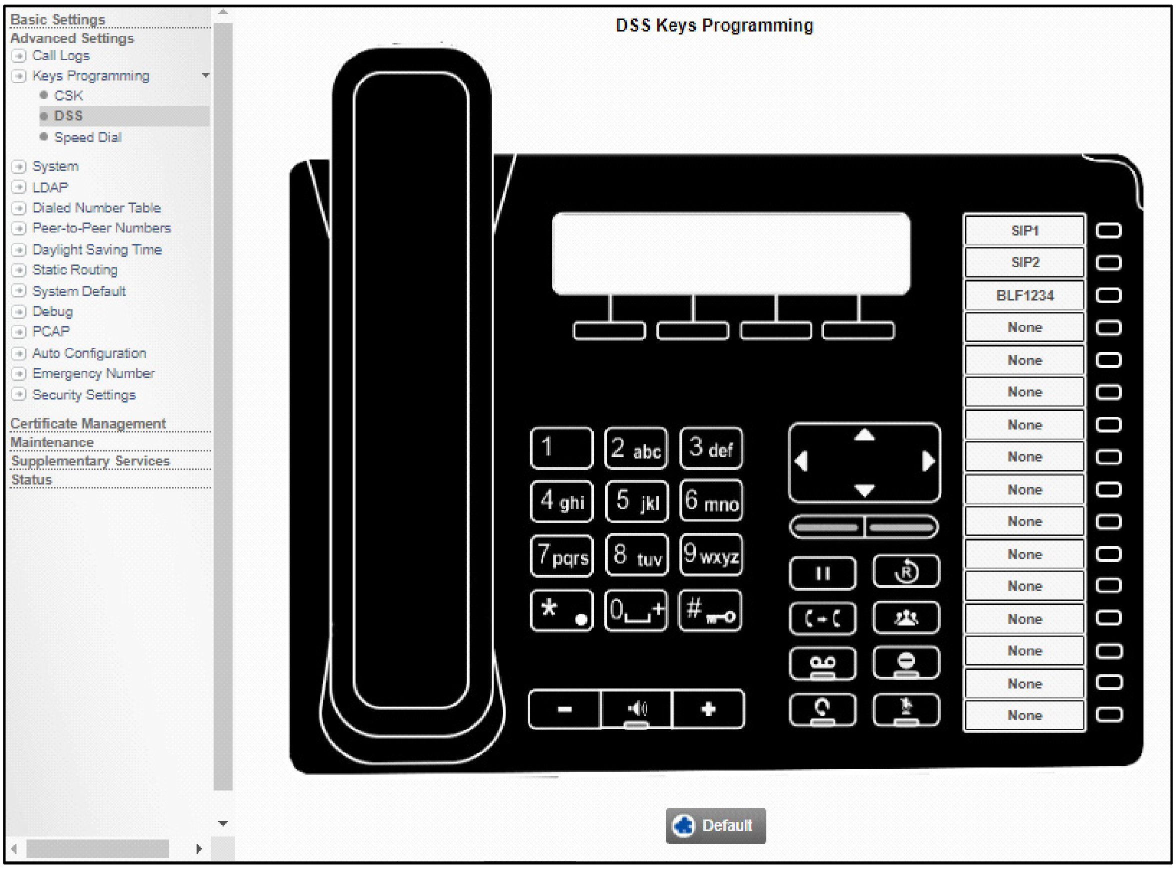
Click the DSS Key3 and configure the following:

- **Feature:** Select **Busy Lamp Field (BLF)** from the drop-down list.
- **Number:** Enter the extension number of the user you wish to monitor.

- **SIP Trunk:** Select the desired SIP Trunk — SIP Trunk 1, SIP Trunk 2.



- Click **OK**.



The BLF feature will appear on the key label as BLF1234, that is BLF feature along with the number of the extension user.

Similarly, you can assign keys to all the desired users.

Accessing BLF Keys

To access the BLF Key,

- Press the BLF Key of the desired extension user.
- Outgoing call will be initiated to the user.

Call Forward

You can use this feature to forward calls to another number, if you are busy or are unable to attend the call.

SPARSH VP510 supports three types of Call Forward:

- **Call Forward-Busy:** Incoming calls are forwarded to another number if your number is busy.
- **Call Forward-No Reply:** Incoming calls are forwarded to another number, if there is no response from your number. For this, you need to set the Call Forward Timer, defining the time (in seconds) that the phone should wait before forwarding an incoming call to the destination number you have configured. All Waiting Calls that are not answered within a particular time set in the Call Forward Timer, will be treated as Call Forward-No Reply.
- **Call Forward-Always:** All incoming calls will be forwarded to the number you have set, without first checking whether your number is busy or there is no reply.

You can either set the same forwarding number, for example: 5678 for Busy, No-reply and Always, or you can set a different forwarding number for each Call Forward type. For example: '5678' for Busy, '7896' for No-reply, and '2525' for Always.

The destination address where the incoming call should be forwarded must not exceed 40 characters, and may be an E.164 number, SIP URI or an IP address.

Peer-to-Peer table will be checked for E.164 numbers.

Call Forward Busy, No-reply and Always can be set via Phone User Interface using the Call Forward Key as well as the Web User Interface.

Call Forward can be set/canceled for all the SIP Trunks or for each SIP Trunk individually depending on the "Feature Access Method" set in "System Parameters".

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to "Context Sensitive Keys (CSK) Programming".

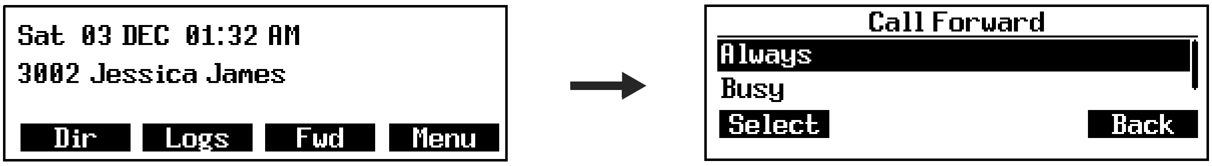
You can also assign a DSS Key to the desired Call Forward Type — Always, Busy, No Reply. To know more, refer to "DSS Keys Programming"

Call Forward via Phone User Interface

Set/Cancel Call Forward using CSK/Menu

To Set Call Forward,

- Press **Fwd** Key on the Home Screen.
- Then scroll using the **Up/Down Navigation** Key to select the type of Call Forward — Always, Busy, No-reply.



- Press **Select** Key.

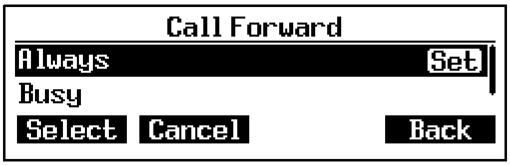
*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you set Call Forward, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*

- To set Call Forward **On Number**, enter the desired number on which you wish to set Call Forward.

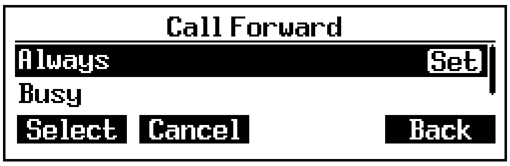
- Press **Save** Key.
- When Call Forward is set, **Set** icon appears.

  The Call Forward set indication also appears on the Home Screen.



To Cancel Call Forward,

- Press **Fwd** Key again on the Home Screen.



- Press **Cancel** Key**.** The Call Forward set indication disappears from the Home Screen.

Similarly, you can set/cancel Call Forward Busy and No-reply.

You can also set/cancel Call Forward from the Phone Menu.

Set/Cancel Call Forward using DSS Key

Make sure you have already assigned a DSS Key to the desired Call Forward type — Always, Busy, No Reply. DSS Key can be assigned either Phone wise or SIP Trunk wise, in this case the DSS Key functionality will be as per the set mode.

To Set Call Forward,

- Press the key assigned to Call Forward.

  The LED of the key glows continuously in Red.

To Cancel Call Forward,

- Press the key again.

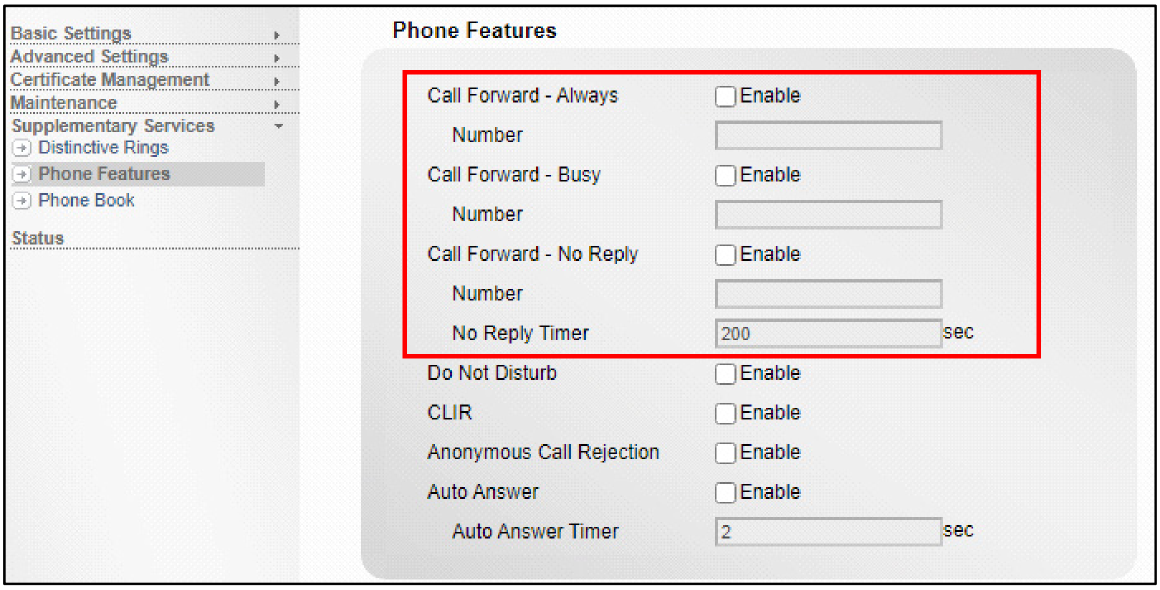
  The LED of the key is turned off.

Enable/Disable Call Forward via Web User Interface

You can enable/disable Call Forward depending on the “Feature Access Method” set in “System Parameters”.

When Feature Access Mode is set as Phone wise, to enable/Disable Call Forward,

- Log into Jeeves.
- Under **Supplementary Services**, click **Phone Features**.



- Select the **Call Forward - Always** check box to enable and configure the desired **Number** (maximum 40 alpha numeric) on which you wish to set Call Forward.
- Select the **Call Forward - Busy** check box to enable and configure the desired **Number** (maximum 40 alpha numeric) on which you wish to set Call Forward.

- Select the **Call Forward - No Reply** check box to enable and configure the desired **Number** (maximum 40 alpha numeric) on which you wish to set Call Forward.

  You may also configure the **No Reply Timer**, if required. This is the time in seconds the phone will wait before forwarding an incoming call to a configured number. Valid Range: 01-99 sec. Default: 45 sec.

  Clear the check box of the desired Call Forward type to disable. Default: Disabled.

When Feature Access Mode is set as SIP Trunk wise, to enable/disable Call Forward,

- Log into Jeeves.
- Under **Supplementary Services**, click **Trunk Features**.
- Select the desired SIP Trunk and follow the same steps to enable/disable Call Forward as mentioned above.

Calling Line Identification Restriction (CLIR)

CLIR is a feature you can use when you do not want to reveal your identity (number) to the person you are calling. In other words, you can make anonymous calls to others using this feature.

This feature can be enabled from the Phone User Interface as well as the Web User Interface.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to "Context Sensitive Keys (CSK) Programming".

You can also assign a DSS Key to CLIR. To know more, refer to "DSS Keys Programming".

CLIR via Phone User Interface

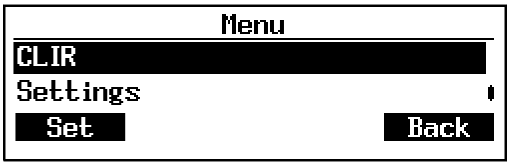
Set/Cancel CLIR via CSK/Menu

To Set CLIR,

- Press the key assigned to CLIR.

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **CLIR**.



*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you set Call Forward, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*

- Press **Set** Key. The **Set** icon appears.

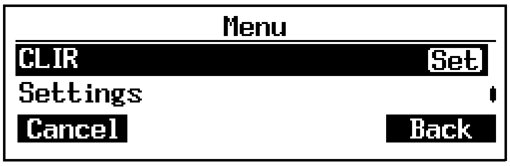
To Cancel CLIR,

- Press the key assigned to CLIR again.

  OR

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **CLIR**.



*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you cancel CLIR, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*

- Press **Cancel** Key. The **Set** icon disappears.

Set/Cancel CLIR using DSS Key

Make sure you have already assigned a DSS Key to CLIR. DSS Key can be assigned either Phone wise or SIP Trunk wise, in this case the DSS Key functionality will be as per the set mode.

To Set CLIR,

- Press the key assigned to *CLIR*.

  The LED of the key glows continuously in Red.

To Cancel CLIR,

- Press the key again.

  The LED of the key is turned off.

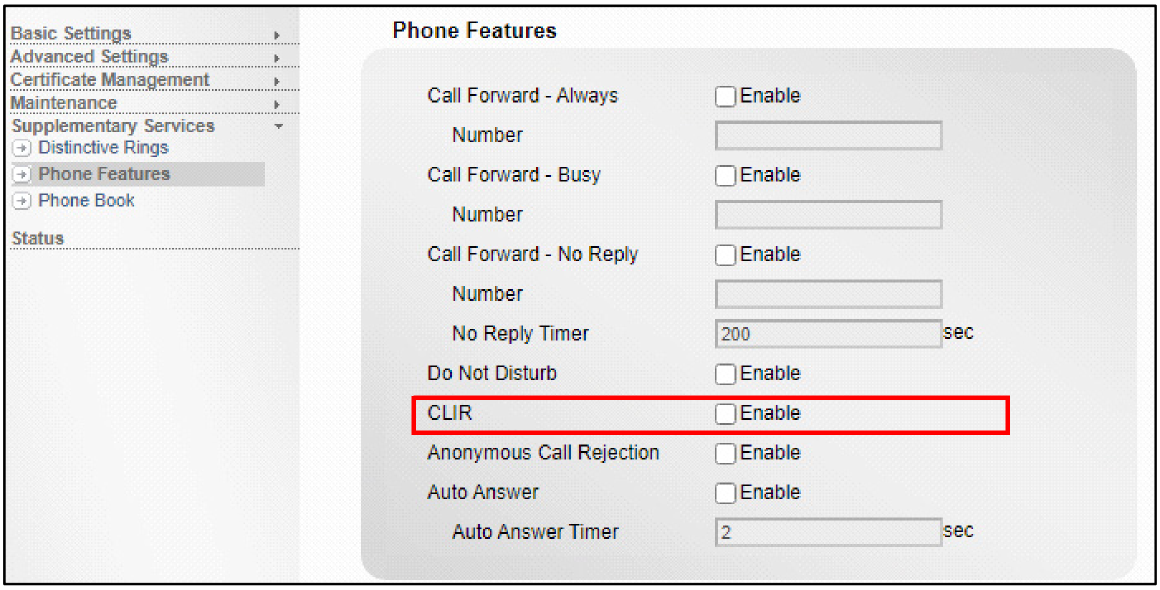
Enable/Disable CLIR via Web User Interface

You can enable/disable CLIR depending on the Feature Access Mode set in System parameters.

When Feature Access Mode is set as Phone wise, to enable/disable CLIR,

- Log into Jeeves.

- Under **Supplementary Services**, click **Phone Features**.



- Select the **CLIR** check box to enable. Default: Disabled.

When Feature Access Mode is set as SIP Trunk wise, to enable/disable CLIR,

- Log into Jeeves.
- Under **Supplementary Services**, click **Trunk Features**.
- Select the desired SIP Trunk and follow the same steps to enable/disable CLIR as mentioned above.

Daylight Saving Time (DST)

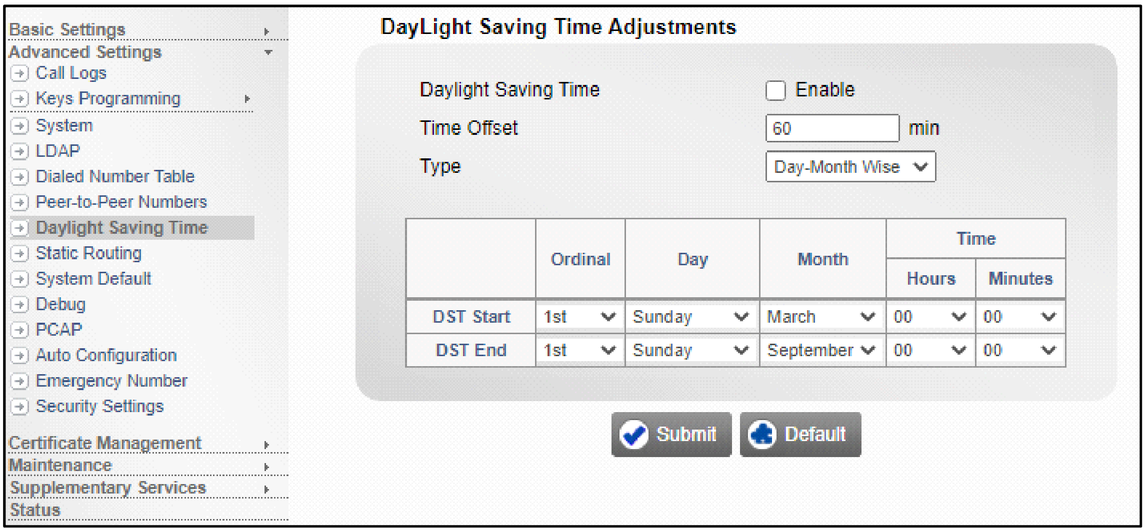
Daylight Saving Time (DST) is the practice of advancing clocks so that afternoons have more daylight and mornings have less. Typically clocks are adjusted forward one hour near the start of spring and are adjusted backward in autumn.

Many countries of the world use it, though the start and end dates of DST vary by location and year. SPARSH VP510 supports Daylight Saving Time adjustment to enable you to set the Date and Time of SPARSH VP510 forward and backward according to the DST convention followed in your country.

You can set DST by: **Day Month Wise** or **Date Month Wise**.

Configuring Daylight Saving Time (DST) via Web User interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Daylight Saving Time**.



- Select the **Daylight Saving Time** check box to enable.Default: Disabled.
- In **Time Offset**, enter the time in minutes which the phone should consider to forward the clock at the start of DST and to set the clock back when DST ends. Default: 60 minutes.
- Select the desired **Type** of DST as:
  - **Day-Month Wise**, if the DST in your country starts and ends on a particular day of the month. For example, if DST starts on the Second Sunday of March and ends on the First Sunday of October.
  - **Date-Month Wise**, if the DST in your country starts and ends on a particular date of the month. For example, if DST starts on October 12 and ends on March 15.

    Default: Day-Month Wise.
  - If you select **Day-Month Wise** option, you need to configure the Start and End time for DST.
    DST Start

- Select the **Ordinal** day of the month when DST begins: 1st, 2nd, 3rd, 4th or 5th.
- Select the **Day** of the month when DST begins: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday.
- Select the **Month** when DST begins: January to December.
- Set the **Time** when you want DST to begin in 24 hours format.

Default: 1st Sunday March, Time 00 hours and 00 minutes.

DST End

- Select the **Ordinal** day of the month when DST ends: 1st, 2nd, 3rd, 4th or 5th.
- Select the **Day** of the month when DST ends: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday, Saturday.
- Select the **Month** when DST ends: January to December.
- Set the **Time** when you want DST to end in 24 hours format.

Default: 1st Sunday September, Time 00 hours and 00 minutes.

When the DST of a particular country starts or ends on the Last Sunday or any other day, for instance, the last Tuesday, last Friday of the month, always set the Ordinal Number as '5th'.

- If you select **Date-Month Wise** option, configure the following parameters:

  DST Start

  - Select the **Month** when DST begins — January to December.
  - Select the **Date** on which DST begins — 1 to 31.
  - Set the **Time** when DST begins in 24 hours format.

  DST End

  - Select the **Month** when DST ends — January to December.
  - Select the **Date** on which DST ends — 1 to 31.
  - Set the **Time** when DST ends in 24 hours format.

- Click **Submit** to save your DST settings.

Dialed Number Table

SPARSH VP510 supports Dialed Number Based Routing, whereby the route for outgoing calls is selected on the basis of the number you have dialed. For example, if you have subscription with different ITSPs, you can instruct the SPARSH VP510 to route outgoing calls to certain numbers through SIP Trunk 1 and certain calls through SIP Trunk 2, using the Dialed Number Table. The phone will select the outgoing call route according to the number you have dialed.

This feature is particularly useful when each ITSP you have subscribed to is offering a better tariff for calls to certain destinations. For example: ITSP A (configured on SIP1) offers low tariffs for calls to the US, Canada and Europe, and ITSP B (configured on SIP2) offers lower tariffs for countries in the Middle-East. You can use Dialed Number based routing to route calls made to numbers in the US, Canada, Europe through SIP1 and those to the Middle-East through SIP2.

This is made possible with the help of a 'Dialed Number Table', in which you can define the outgoing call route for the dialed numbers. You can configure as many as 100 numbers in the Dialed Number Table

For each number that you enter in the table, you must define the length of the number string in terms of Minimum Digits and Maximum Digits, and define a SIP Trunk for routing this number.

With the Dialed Number Table configured, whenever you dial a number, SPARSH VP510 will compare it with the entries in the Table.

If a match is found, it will check the 'Minimum Digits' configured for this entry to consider it a valid number. If the length of the number string matches with the Minimum Digits configured for this entry, and End of Dialing is detected, the phone will consider it as a valid number and dial out the number using the route (SIP Trunk) selected for this entry in the table.

However, if the dialed number string does not match with Minimum Digits, the phone will check the Maximum Digits configured for this entry. If the length of the dialed number string matches with the Maximum Digits configured for this entry, the phone will not wait for further digits to be dialed. It will consider the Maximum Digits defined as End of Dialing, and dial out the number using the route (SIP Trunk) selected for this entry in the table.

This can be illustrated with the following example. You have configured an entry in the Dialed Number Table as follows:

Dialed Number Table

Index	Number	Minimum Digits	Maximum Digits	Destination Trunk
001				
002	#41	5	8	SIP 1

- #41 is being dialed, the phone checks the dialed number table. A match is found with the entry at index 002 on the table. The phone checks the Minimum Digits configured for the entry. The Minimum digits do not match, but End of Dialing is detected. The phone will treat this as an invalid number and will not dial out the number.

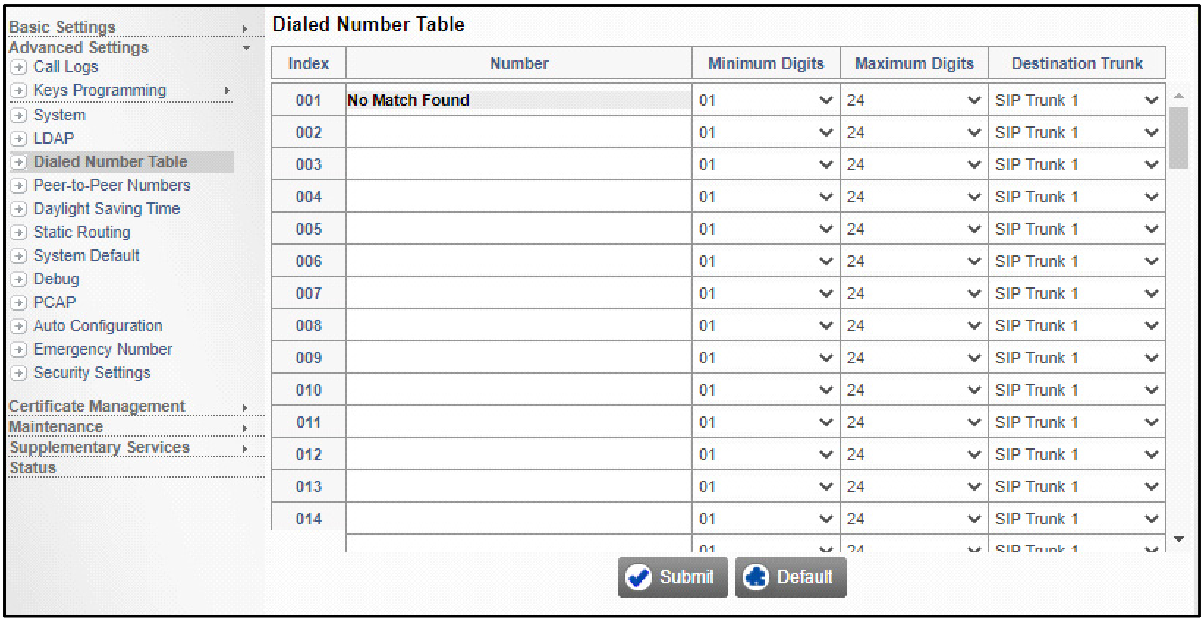
- When #41111 is being dialed, the phone checks the dialed number table. The number matches with the entry at index 002 and the minimum digits configured for this entry. The phone will wait for End of Dialing.

End of Dialing is detected. The phone will treat this as a valid number and dial out the number from SIP1 which is defined as the Destination Trunk for this number.

- When #4111111 is being dialed, the phone checks the dialed number table. The number matches with the Minimum digits as well as the Maximum digits. The phone treats this as a valid number. It does not wait to detect End of Dialing, and dials out the number from SIP1.

Configuring Dialed Number Table via Web User interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Dialed Number Table**.



- Each entry in the dialed number table is stored at an Index. For each entry in the table, you must configure the following parameters:
  - **Number:** Enter the number which will be dialed out from the phone (which the phone should match with this Table before dialing out). The number should not be more than 24 characters. Default: Blank.
  - **Minimum Digits:** For the number you entered, select the minimum number of digits which the phone should wait to receive before considering it as a valid number. Default: 01.
  - **Maximum Digits:** For the number you entered, select the maximum number of digits which the phone should wait to receive before considering it as End of Dialing[4] and dial out the number. Default: 24.
  - **Destination Trunk:** For the number you entered, define the SIP Trunk through which the dialed number (which matches the minimum and maximum digits) should be routed. Default: SIP1.
- Click **Submit** to save.

---

4. *When 'Dialed Number Table' is selected as the option for Outgoing Call Routing, the phone will not apply 'Fixed Number of Digits for "End of Dialing".*

Do Not Disturb

Do Not Disturb (DND) prevents incoming calls from landing on your extension.

DND can be set via Phone User Interface as well as Web User Interface. DND can be set/cancelled for all the SIP Trunks or for each SIP Trunk individually depending on the "Feature Access Method" set in "System Parameters".

You can access Voicemail using — Phone Menu, CSK, DSS Key as well as Fixed Function Key (DND Key).

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to "Context Sensitive Keys (CSK) Programming".

You can also assign a DSS Key to DND. To know more, refer to "DSS Keys Programming".

Do Not Disturb via Phone User Interface

Set/Cancel Do Not Disturb using CSK/Menu

To Set DND,

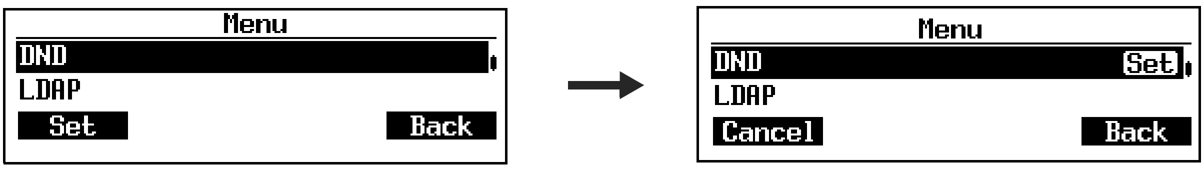
- Press the key assigned to DND.

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **DND**.

*Screens are with reference to **Feature Access Mode** set as **Phone wise**.*

*If it is set as **SIP Trunk wise**, while you set DND, you need to select the desired SIP Trunk and then follow the steps as mentioned here.*



- Press **Set** Key.
- When DND is set, **Set** icon appears.

The Fixed Function Key, that is the DND Key LED glows continuously in Blue.

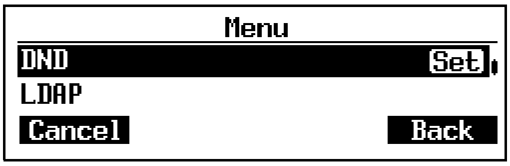
If you have assigned a DSS Key to DND then the LED of the key glows continuously in Red.

To Cancel DND,

- Press the key assigned to DND again.

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **DND**.



- Press **Cancel** Key.
- The message DND canceled is displayed.

The Fixed Function Key, that is the DND Key LED is turned off.

If you have assigned a DSS Key to DND then the LED of the key is turned off.

Set/Cancel Do Not Disturb using DSS Key

Make sure you have already assigned a DSS Key to DND.DSS Key can be assigned either Phone wise or SIP Trunk wise, in this case the DSS Key functionality will be as per the set mode.

To Set DND,

- Press the key assigned to DND.

  The LED of the key glows continuously in Red as well as the LED of the Fixed Function Key, **DND** Key glows continuously in Blue.

To Cancel DND,

- Press the key again.

  The LED of the key is turned off as well as the LED of the Fixed Function Key, **DND** Key is turned off.

Set/Cancel Do Not Disturb using Fixed Function Key

You can Set/Cancel DND using DND Key, only if you have selected Phone wise as the Feature Access Method >Use Features option.

If you have selected SIP Trunk wise s the Feature Access Method >Use Features option and you press the DND Key, it will open DND configuration in the Phone Menu.

To Set DND,

- Press the DND Key.

  The LED of the key glows continuously in Blue.

  If you have assigned a DSS Key to DND then the LED of the key glows continuously in Red.

To Cancel DND,

- Press the DND Key again.

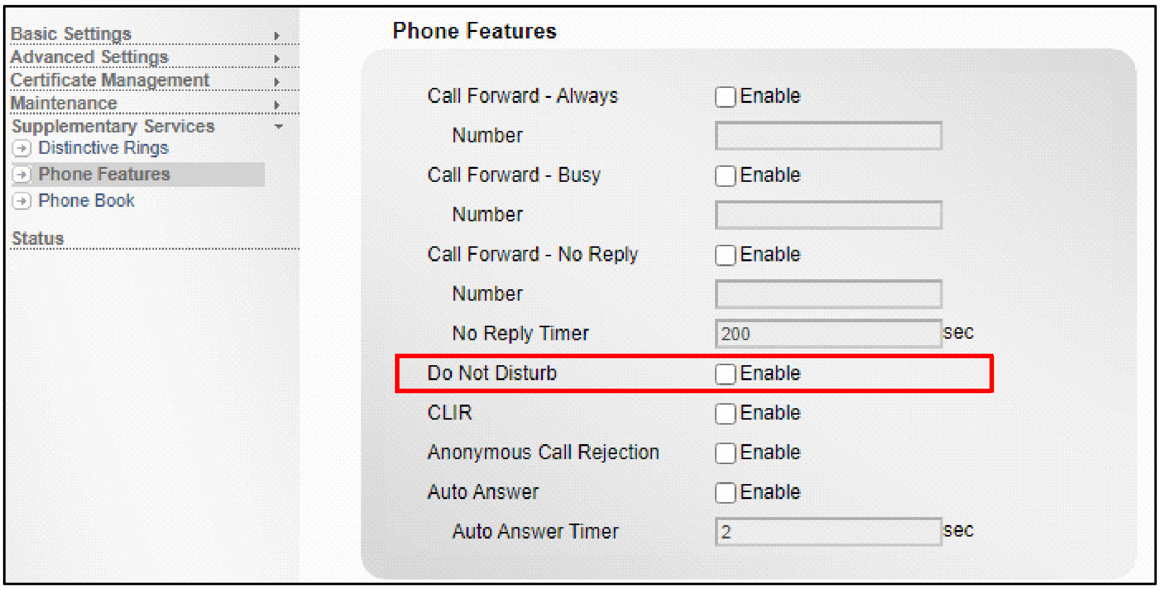
  The LED of the key is turned off as well as the LED of the DSS Key assigned to DND is turned off.

Enable/Disable DND via Web User Interface

You can enable/disable DND depending on the "Feature Access Method" set in "System Parameters".

When Feature Access Mode is set as Phone wise, to enable/disable DND,

- Log into Jeeves.
- Under **Supplementary Services**, click **Phone Features**.



- Select the **Do Not Disturb** check box to enable. Default: Disabled.

Clear the check box to disable.

When Feature Access Mode is set as SIP Trunk wise, to enable/disable DND,

- Log into Jeeves.
- Under **Supplementary Services**, click **Trunk Features**.
- Select the desired SIP Trunk and follow the same steps to enable/disable DND as mentioned above.

Distinctive Rings

Distinctive ring tones allows certain incoming calls to trigger IP phones to play distinctive ring tones. The IP phone inspects the INVITE request for an “Alert-Info" header when receiving an incoming call. If the INVITE request contains an "Alert-Info" header, the IP phone strips out the URL and keyword parameter and maps them to the appropriate ring tone.

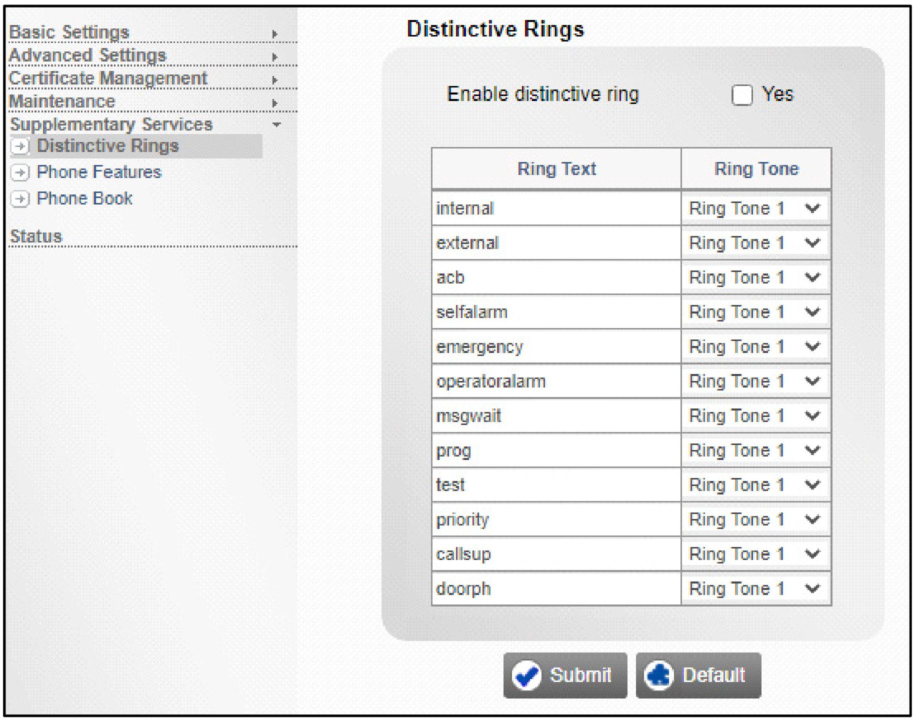
This is used to differentiate different type of calls and it is dependent on Server/Proxy configuration. You can set different ringer tuner for different type of calls. SPARSH VP510 supports distinctive tones for the following types of calls:

- Internal Call
- External Call
- Auto Call Back Call
- Self Ring
- Emergency
- Operator Alarm
- Message Wait Call
- Programming Ring
- Test
- Priority Call
- Door Phone
- Calls Up

Configuring Distinctive Rings via Web User interface

- Log into Jeeves.

- Under **Supplementary Services**, click **Distinctive Rings**.



- Select the **Enable distinctive ring** check box to enable. Default: Disabled.
- You can configure the **Ring Text** and select the desired **Ring Tone** for each call event as per your requirement.
- Click **Submit** to save.

Emergency Number

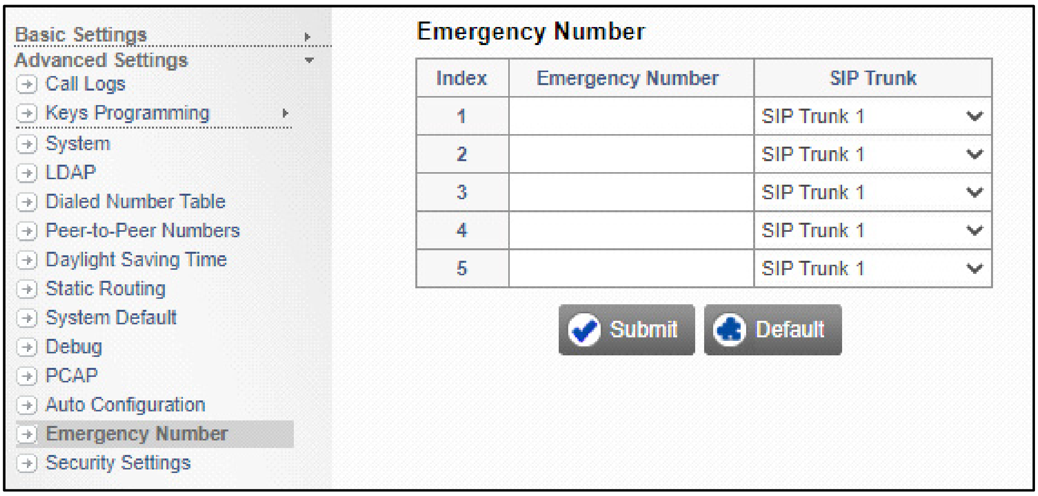
Public telephone networks in countries around the world have a single emergency telephone number (emergency services number), that allows a caller to contact local emergency services for assistance when necessary. The emergency telephone number may differ from country to country. Some countries have different emergency numbers for different emergency services.

You can configure the emergency numbers that can be dialed out from the IP phone in an emergency situation. These numbers will be dialed out even when the Keypad is locked.

To do so, you need to configure the Emergency Number Table, wherein you need to mention the number and the select the SIP Trunk using which the number can be dialed out.

Configuring Emergency Number Table via Web User interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Emergency Number** table.



- **Emergency Number:** Configure the desired Emergency Number against the Index 1 to 5, as per your region.Default: Blank.
- **SIP Trunk:** By default, SIP Trunk 1 is assigned to route the emergency numbers. You can select the desired SIP Trunk as per your requirement.

  Make sure that the trunks configured in the Emergency Number Table, through which the calls are to be routed are not disabled.
- Click **Submit** to save changes.

Dialing the Emergency Number

- Lift **Handset** or press **Speaker** Key.
- Dial the number.

Hotline

If you have a number that you dial very frequently, you can spare yourself the time and effort of dialing that number by configuring it as 'Hotline'. With the Hotline feature, you can configure a particular number such that whenever the phone goes OFF-Hook, the configured number will be dialed out automatically after waiting for a few seconds. You can also configure the number of seconds it should wait before dialing that number using the Hotline Timer.

This feature is particularly useful when you have a long-digit number that you dial very frequently.

You can enable or disable the Hotline feature through the Phone User Interface as well as the Web User Interface. But first you must configure the Hotline Address, i.e. the number you want to be dialed automatically, and the Hotline Timer.

The Hotline Timer is the time in seconds the phone should wait after going OFF-Hook to dial the hotline number that you have set. Default: '0' sec, which means the phone will dial the hotline number set by you immediately after it goes OFF-Hook.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

You can also assign a DSS Key to Hotline. To know more, refer to “DSS Keys Programming”

Hotline via Phone User Interface

Set/Cancel Hotline using CSK/Menu

To set,

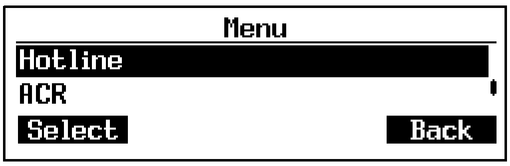
- Press the key assigned to Hotline.

  OR

- Press **Menu** Key.

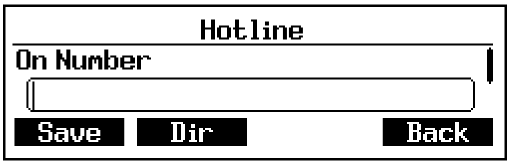
- Scroll using the **Up/Down Navigation** Key to select **Hotline**.

- Press **Select** Key.



- To set Hotline **On Number**, enter the desired number on which you wish to set Hotline.

- To set delayed Hotline, scroll using the **Up/Down Navigation** Key to select **Hotline Timer** and enter the desired value.



- Press **Save** Key. The **Set** icon appears.

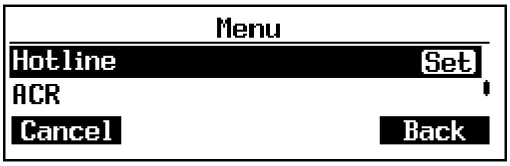
To Cancel,

- Press the key assigned to Hotline again.

  OR

- Press **Menu** Key.

- Scroll using the **Up/Down Navigation** Key to select **Hotline**.



- Press **Cancel** Key.

Set/Cancel Hotline using DSS Key

Make sure you have already assigned a DSS Key to Hotline.

To Set Hotline,

- Press the key assigned to Hotline.

  The LED of the key glows continuously in Red.

To Cancel Hotline,

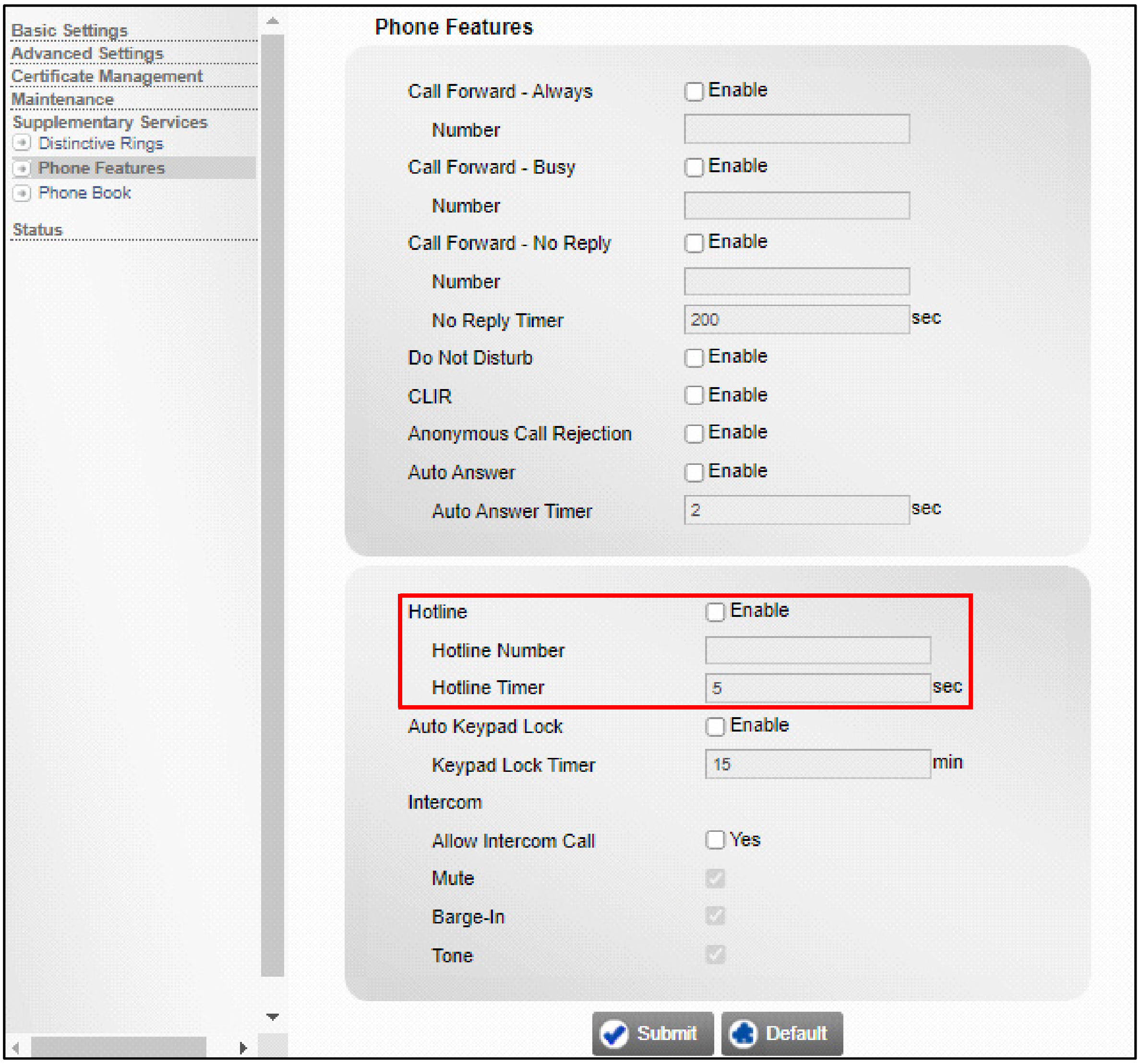
- Press the key again.

  The LED of the key is turned off.

Enable/Disable Hotline via Web User Interface

- Log into Jeeves.

- Under **Supplementary Services**, click **Phone Features**.



- Select the **Hotline** check box to enable.Default: Disabled.
    - Configure the **Hotline Number** (maximum 40 alpha numeric).
    - If you wish to set delayed Hotline, configure the **Hotline Timer.** Valid range: 0-9 sec. Default:0 sec.
- Clear the **Hotline** check box to disable.

Keypad Lock

You can lock your Keypad to avoid misuse of your extension phone, while you are away from your desk.

You can lock the Keypad manually as well as automatically.

You cannot use the default User Password (1234) to Lock the Keypad. Make sure, you have changed it. For detailed instructions, refer to *“Passwords”*

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

You can also assign a DSS Key to Keypad Lock. To know more, refer to “DSS Keys Programming”.

Lock/Unlock via Phone User Interface

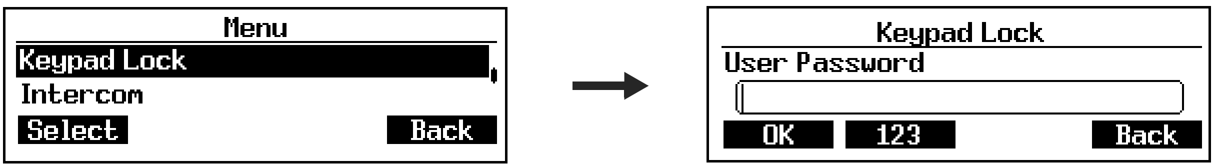
Lock/Unlock using CSK/Menu

To Lock,

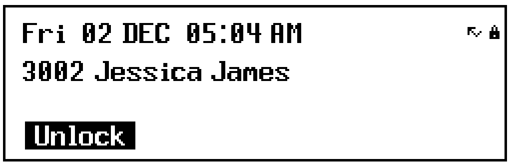
- Press the Context Key assigned to Keypad Lock.

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Keypad Lock** option.



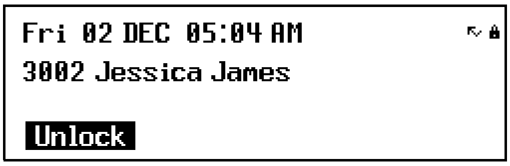
- Press **Select** Key.



- The Keypad is locked and the lock icon appears on the Home Screen.

To Unlock,

- On the Home Screen, press **Unlock** Key.



- Enter the **User Password** and press **OK** Key. The Keypad is unlocked.

Lock/Unlock using DSS Key

Make sure you have already assigned a DSS Key to Keypad Lock

To Unlock,

- Press the key assigned to Keypad Lock.

  Enter the Password to unlock.

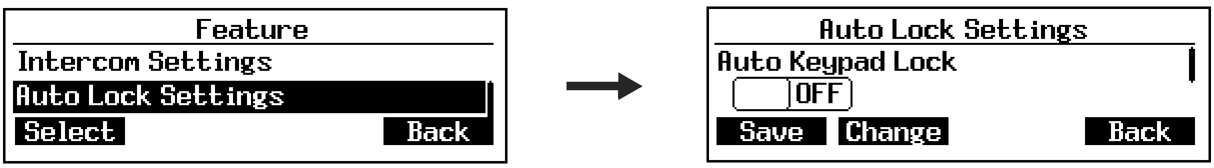
To Lock,

- Press the key again.

  The Keypad is locked.

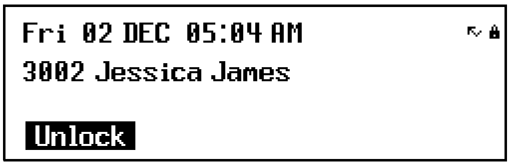
Configuring Auto Keypad Lock via Phone User Interface

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Feature** option and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Auto Lock Settings**.
- Press **Select** Key.



- Press **Change** Key to turn **On/Off** Auto Keypad Lock.

- Scroll using the **Up/Down Navigation** Key to select **Auto Lock Timer (min)** option and press **Select** Key.
- Enter the desired value after which the Keypad should be locked automatically.
- Press **Save** Key.



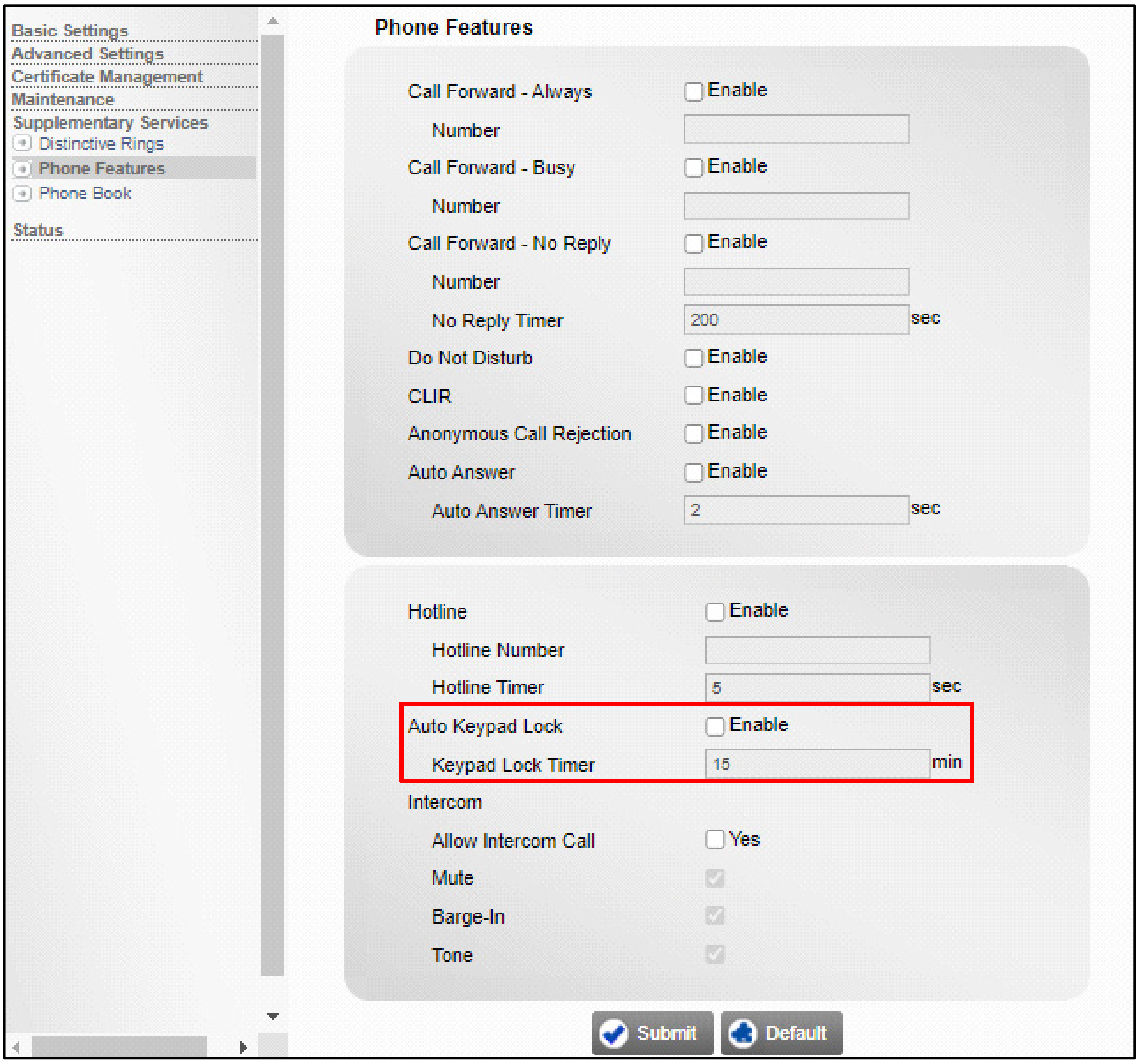
- After the configured timer expires the Keypad is locked and the lock icon appears on the Home Screen.

If your Keypad is locked,
- *you can dial Emergency Numbers.*
- *during a call, you cannot access any Call feature.*

Configuring Auto Keypad Lock via Web User Interface

- Log into Jeeves.

- Under **Supplementary Services**, click **Phone Features**.



- Select the **Auto Keypad Lock** check box to enable. Default: Disabled.

- Configure the **Keypad Lock Timer**. Default: 15 min.

LDAP

Lightweight Directory Access Protocol (LDAP) is a standard and widely implemented protocol for communication between directory client applications and directory servers about the data in directories.

SPARSH VP510 supports LDAP Client (LDAP V2). Organizations that have a corporate directory, i.e. a centralized Phone Book, can use LDAP Client supported by SPARSH VP510 to access and search the Phone Book managed in the centralized LDAP Server.

The Phone Book of SPARSH VP510 allows you to store upto 200 contacts. Additions and updates to the Phone Book are limited to a single phone only. In order for other phones to have the same contact list, these additions and updates must be made individually in all other IP Phones. A centralized Phone Book has the advantage that all IP Phones can access the same directory and all additions and updates are to be made only in a single directory maintained in the LDAP Server.

For accessing a centralized Phone Book managed in the LDAP Server, you must first configure the LDAP Parameters in SPARSH VP510.

In LDAP, data is looked up not in tables, but in trees. Data is not stored in rows and columns but in what are called entries. These entries are much like entries in the phone book. Here is an example of an LDAP entry (text representation):

- dn: uid=sjohn, ou=people, dc=midasbizsolutions, dc=com
- cn: John Simpson
- sn: Simpson
- givenName: John
- mail: s.john@midasbiz.com
- telephoneNumber: +91 265 2660333
- facsimiletelephoneNumber: +91 265 2660338

Each entry is composed of attributes and their values. In the above example, dn: uid=sjohn, ou=people, dc=midasbizolutions, dc=com is the Distinguished Name (DN). A Distinguished Name distinguishes an entry from all others in a directory. Distinguished Name is required by the LDAP server to know where to start the search in the 'tree'. Other attributes in the above entry are: 'cn' for Common Name, 'sn' for Surname, givenName, 'mail', 'telephoneNumber', 'facscimiletelephoneNumber' which take the values John Simpson, Simpson, John, s.john@midasbiz.com, +91 265 2660333, +91 265 2660338, respectively.

When an LDAP search request is sent to the LDAP Server from SPARSH VP510, the attributes defined in the LDAP Parameters of the phone are sent to the server, so that the server searches only those attributes in the entries.

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

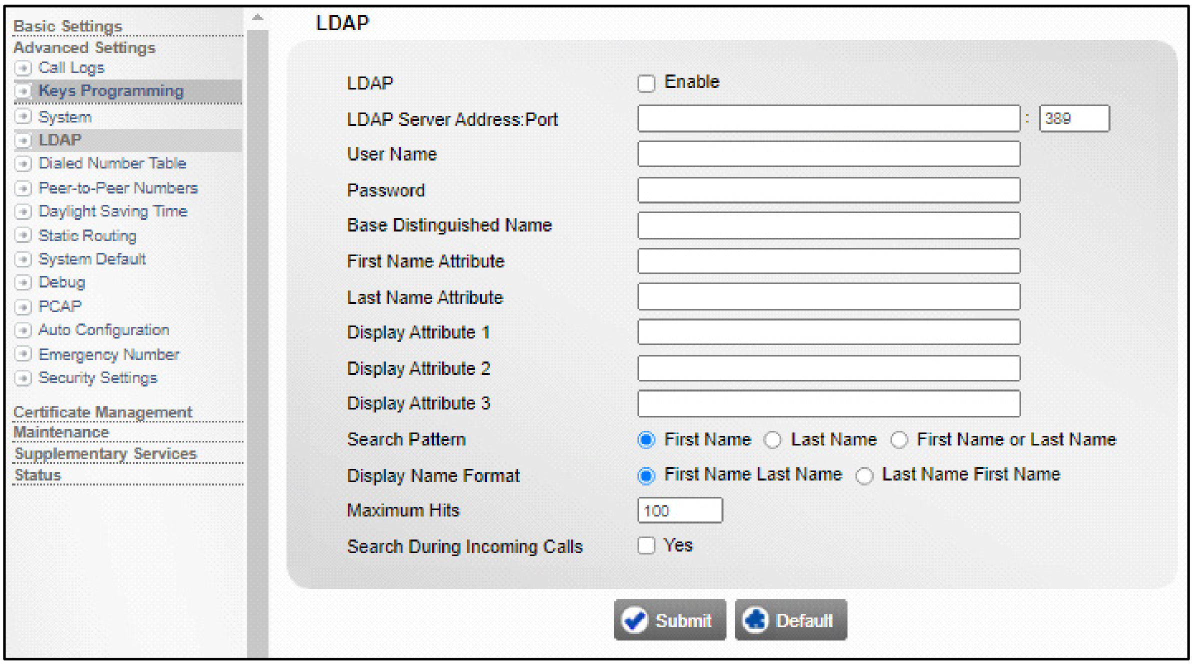
You can also assign a DSS Key to LDAP. To know more, refer to “DSS Keys Programming”.

Configuring LDAP Parameters via Web User Interface

- Log into Jeeves.

- Under **Advanced Settings**, click **LDAP**.

- The LDAP page appears.



- Now configure the following parameters:

  - **LDAP:** Select the check box to enable this feature. Then the phone will contact the LDAP server and retrieve the desired contacts. Default: Disabled.

  - **LDAP Server Address: Port:** Enter the IP Address or the Domain Name of the LDAP Server which is to be accessed by the phone. A maximum of 40 characters. Default: Blank.

    For the Port enter the TCP Listening Port of the LDAP Server. A maximum of 5 digits, from 0 to 9, are allowed. Valid range: 1024 to 65535. Default: 389.

  - **User Name and Password:** If the LDAP Server requires authentication of users (IP Phone) for directory access, ask your Server Administrator for the User Name and Password for your phone and enter the same in these fields.

    Depending on the type of authentication configured in the LDAP Server, configure only the User Name or both User Name and Password.

    For User Name, a maximum of 40 characters are allowed as User Name. Default: Blank.

    As Password, a maximum of 24 characters are allowed. Default: Blank.

  - **Base Distinguished Name:** Specify the Distinguished Name (DN) from where LDAP server should start searching for the requested entry in the database. The use of Distinguished Name reduces the searching time of entries in database.

    A maximum of 40 characters are allowed as Base Distinguished Name. Default: Blank.

  - **First Name Attribute:** An entry in LDAP database may have multiple names: First Name, Last Name, Common Name etc. Each type of name is assigned a specific attribute in the LDAP database. E.g., First Name attribute is defined as 'givenName'. So, you need to define what the LDAP server will understand as the First Name in this field. Enter the attribute for First Name defined in the LDAP Server in this field. If First Name attribute is defined as 'givenName' in the server, enter the same in this field.

This attribute is sent in the Attribute list of LDAP Search Request to get only the respective name in LDAP Response.

A maximum of 40 characters are allowed as First Name Attribute. Default: Blank.

- **Last Name Attribute:** Just as you defined First Name Attribute, enter the attribute for the Last Name defined in the LDAP Server in this field. For example, if the Last Name attribute is defined as 'sn' in the server, enter 'sn' in this field.

  This attribute is also sent in the Attribute list of LDAP Search Request to get only respective name in LDAP Response.

  A maximum of 40 characters are allowed as Last Name Attribute. Default: Blank.

- **Display Attribute 1:** Each entry in the LDAP database can have multiple contact numbers, such as office number, home number, cell number, fixed line number, etc. SPARSH VP510 supports display of upto three numbers of a single contact.

  Now, each type of number is assigned a specific attribute in the LDAP database. For example, Business Phone Number may be defined as 'telephoneNumber' and Home Phone Number may be defined as 'homephone'. If you have decided what kind of contact number (i.e. mobile, fixed line, home, office, etc.) the Server should search for first, then enter the same attribute for that type of number as defined in the LDAP database. For example, if you want the Server to search first by Business Phone number, which is defined as 'telephonenum' in the LDAP Server, enter 'telephonenum' in this field.

  A maximum of 40 characters are allowed as Display Attribute 1. Default: Blank.

- **Display Attribute 2:** If you want a second contact number to be displayed for the same entry being searched, enter the same attribute for that type of number as defined in the LDAP database. For example, if you want the Server to search and display the second contact number by Mobile number, which is defined as 'mobile' in the LDAP Server, enter 'mobile' in this field.

  A maximum of 40 characters are allowed as Display Attribute 2. Default: Blank.

- **Display Attribute 3:** if you want a third contact number to be displayed for the same entry being searched, enter the attribute configured for that type of number in the LDAP database. For example, if you want the server to search the third contact number by residence number, which is defined as 'homeNumber' in the LDAP server, enter 'homeNumber' in this field.

  A maximum of 40 characters are allowed as Display Attribute 3. Default: Blank.

- **Search Pattern:** Entries in the LDAP database may have multiple names, like first name, last name, middle name, etc. You may define how the entries in the LDAP database should be searched, whether by First Name, or by Last Name, or by First Name or Last Name. For example, if you want the entries to be searched by Last Names, select this option as Search Pattern. Default: First Name.

- **Display Name Format:** SPARSH VP510 retrieves only two name attributes from the LDAP Server: First Name and Last Name. You may select the format for displaying the Name as either First Name Last Name or Last Name First Name. Usually, the preferred format for displaying the names is by First

Name and Last Name. However, if desired by the users, you may select Last Name First Name as the display format. Default: First Name Last Name.

- **Maximum Hits:** When an LDAP Search Request is sent, the server returns all the entries in the LDAP response which match the search criteria. You specify the number of entries you want the server to return in the response by configuring this parameter. Valid range: 001 to 100. Default: 100.

- **Search During Incoming Calls:** When there is an incoming call and the caller's number is visible to the called party, but not the name, it is possible to find the name of the caller by sending a search request to the LDAP Server. The phone sends LDAP Search Request to look for the name related to the number, and displays the name as CLI on the called party's phone. To disable this search select the check box. Default: Disabled.

- If you have completed configuring the LDAP Parameters, click **Submit** to save your settings.

Making Calls Using Key assigned to LDAP

SPARSH VP510 allows you to make calls to the contacts stored in the LDAP Server.

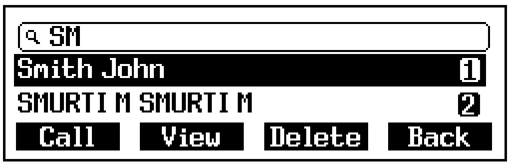
Make sure you have already assigned a DSS/Context Key to the LDAP.

- Press the Context Key or DSS Key assigned to LDAP.

  OR

- Press **Menu** Key.

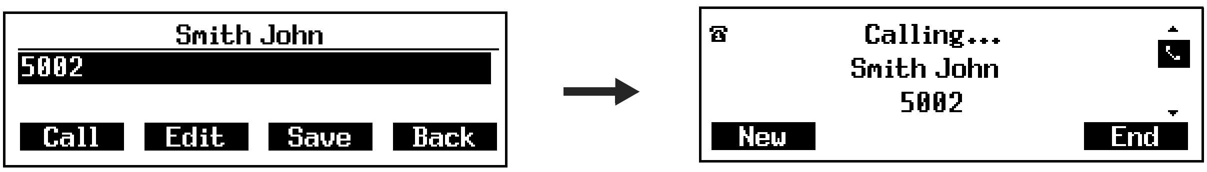
- Scroll using the **Up/Down Navigation** Key to select **LDAP**.



- Enter the Initial letter(s) of the Contact’s name. The contacts name are the names configured in the directory in the LDAP Server.

- Scroll using the **Up/Down Navigation** Key to select the Contact from the matching entries.

- Press **View** Key.



- The details of the selected contact are displayed.

- Press **Call** Key again to dial-out.

Managing LDAP Contacts

You can View details, Edit before call as well as Save the LDAP contacts to the Phone Book.

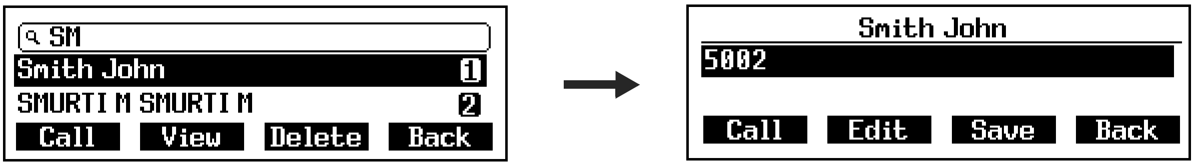
- Press the Context Key or DSS Key assigned to LDAP.

  OR

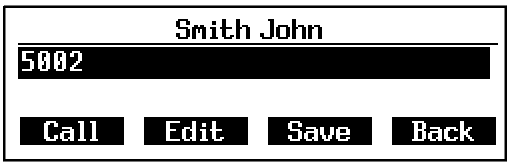
- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **LDAP**.
- Enter the Initial letter(s) of the Contact's name. The contacts name are the names configured in the directory in the LDAP Server.

*While searching if you wish to delete any letter from the Search bar, press the **Delete** Key.*

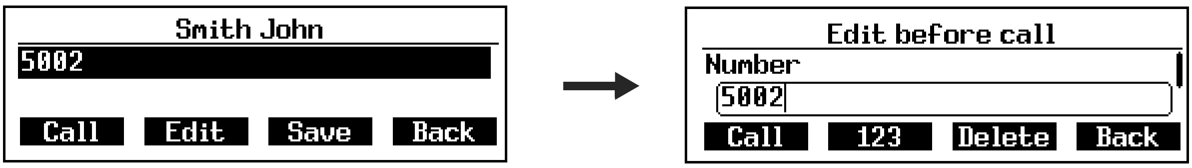
- Scroll using the **Up/Down Navigation** Key to select the Contact from the matching entries.



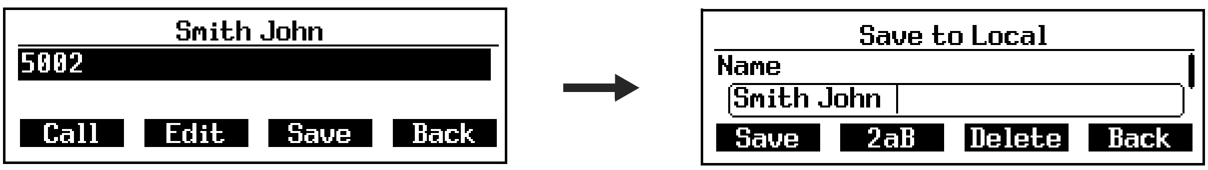
- Press **View** Key. The details of the selected contact are displayed.



- To call, press the **Call** Key.



- To edit the number before making a call, press **Edit** Key.



- To save the contact in the local Phone Book, press **Save** Key.

Voicemail

SPARSH VP510 supports voicemail on each of its SIP Trunks. So, with the SPARSH VP510, you can subscribe to two different Voice Mail Service Providers and retrieve your voice mail messages from all your accounts.

Whenever you receive a message in your voice mail account, the Message Wait Indication appears on the Home Screen and the total number of unread messages (of both SIP Trunks) will be displayed besides Voicemail in the Menu.

Indication of new messages in the mailbox is possible only if the phone receives a notification from the Voice Mail Server from the ITSP.

Voicemail can be set via Phone User Interface as well as Web User Interface.

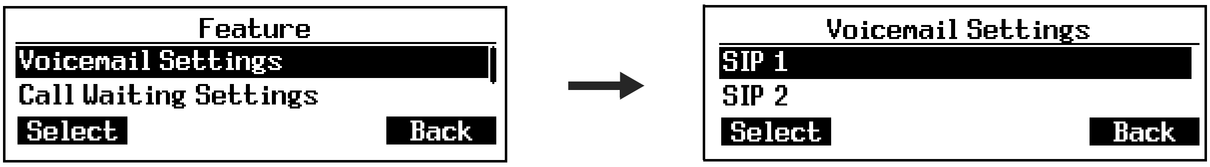
You can access Voicemail using — Phone Menu, CSK, DSS Key as well as Fixed Function Key (**Voicemail** Key).

You can change the priority of the features/functions assigned to the Context Keys. To know more, refer to “Context Sensitive Keys (CSK) Programming”.

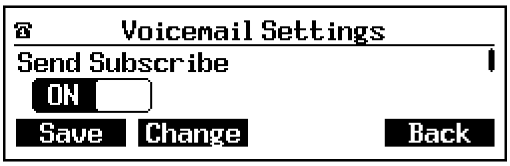
You can also assign a DSS Key to Voicemail. To know more, refer to “DSS Keys Programming”.

Configuring Voicemail Settings via Phone User Interface

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Feature** and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Voicemail Settings** and press **Select** Key.

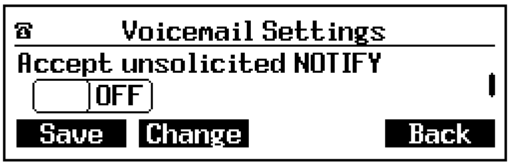


- Scroll using the **Up/Down Navigation** Key to select the desired SIP Trunk and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Send Subscribe.**

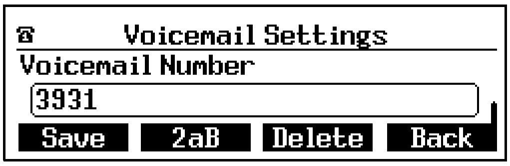


- Press **Change** Key to turn it **On/Off**.

- Scroll using the **Up/Down Navigation** Key to select **Accept unsolicited NOTIFY.**



- Press **Change** Key to turn it **On/Off**.
- Scroll using the **Up/Down Navigation** Key to select **Voicemail Number.**



- Press **Save** Key.

Accessing Voicemails using CSK/Menu

If you have unread Voicemails the LED of the **Voicemail** Key glows continuously in Blue.

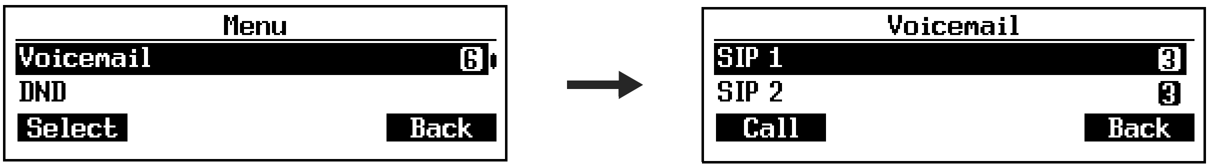
If you have assigned a DSS Key to Voicemail, then the LED of the key glows continuously in Red.

To access the Voicemail,

- Press the CSK assigned to Voicemail.

  OR

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Voicemail** and press **Select** Key



- Scroll using the **Up/Down Navigation** Key to select the desired SIP Trunk and press **Call** Key.
- Follow the voice prompts.
- During the call you can press **End** Key, if you want to release the call.

Accessing Voicemails using DSS Key

Make sure you have already assigned a DSS Key to Voicemail. DSS Key can be assigned either Phone wise or SIP Trunk wise, in this case the DSS Key functionality will be as per the set mode.

If you have unread Voicemails, the LED of the Voicemail key glows continuously in Red as well as the LED of the Fixed Function Key, **Voicemail** Key glows continuously in Blue.

To access the Voicemail,

- Press the key assigned to Voicemail.
- The Voicemail key LED is turned off.

Accessing Voicemails using Fixed Function Key

If you have unread Voicemails, the LED of the **Voicemail** Key glows continuously in Blue.

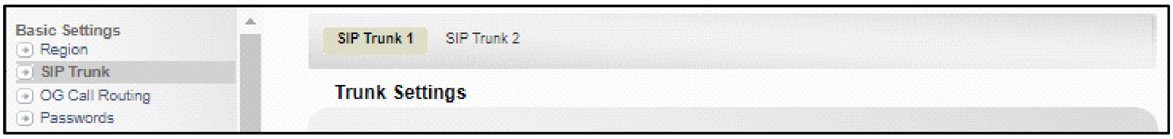
To access the Voicemail,

- Press **Voicemail** Key.
- Follow the voice prompts.

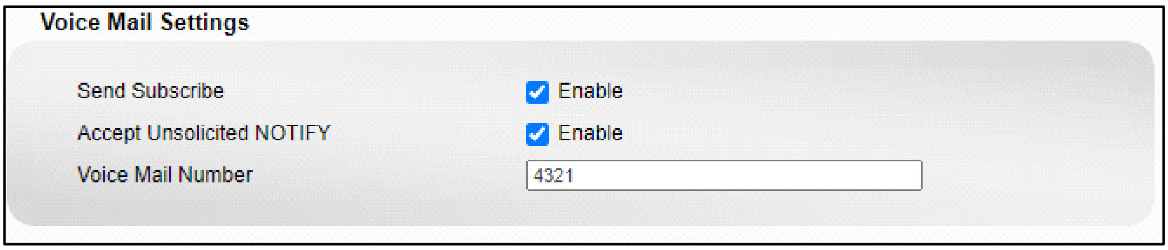
  After you listen to the Voicemails the LED is turned off.

Configuring Voicemail Settings via Web User Interface

- Log into Jeeves.
- Under **Basic Settings**, click **SIP Trunk.**
- Click the **SIP Trunk 1** tab.



Scroll to **Voice Mail Settings** and configure the following:



- **Send Subscribe:** It is a server related feature. The IP phone sends a SUBSCRIBE message to the server for the message-summary updates. The server sends a message-summary NOTIFY within the subscription dialog each time the VM status changes.

  If your Voice Mail server/service provider requires SUBSCRIPTION for Voice Mail, select the Send Subscribe check box. SPARSH VP510 will send SUBSCRIPTION for the voice mail to the mail server. Default: Disabled.

- **Accept Unsolicited NOTIFY:** It is a server related feature.If your server sends Unsolicited Notify then enable this check box. This will enable you to receive MWI notification from the server for which subscription is not sent. Default: Disabled.

- **Voice Mail Number:** Enter the Voice Mail number provided by your Service Provider.

Indication of new messages in the mailbox is possible only if the phone receives a notification from the Voice Mail Server from the ITSP.

Similarly, you can configure the Voicemail settings for SIP Trunk 2.

Peer-to-Peer Calls

Placing and receiving calls over IP network without using a SIP server is called Peer-to-Peer Calling.

As the Peer-to-Peer call application does not require a SIP server, voice communication using this application is 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.

To be able to make Peer-to-Peer Calls, all you must do is configure the network settings of your phone as per the IP addressing scheme of your network (DHCP, Static IP), and enable a SIP Trunk. Your phone will be ready to make Peer-to-Peer calls within the same LAN or VLAN network without a SIP server. You can make calls by dialing the number (i.e. IP address or Domain name) of the other IP phone connected in the same network.

For example, you have an office in city A and another in city B, both offices can be connected to each other using the VLAN services offered by ISP/ITSP. After configuring the network settings at both offices, calls can be made by dialing the IP address of the IP phone connected in the same network. However, if either office uses a Gateway[1] with multiple extensions (with either IP phones or analog phones) connected to it, a particular extension can be reached by dialing the extension number followed by the IP address or Domain Name, which is configured as destination address.

Making Peer-to-Peer Calls

Follow the steps described below to make Peer-to-Peer Calls:

- Install and connect the IP phones to the network at the locations where you want to use the Peer-to-Peer application. Refer the topic “Peer-to-Peer Numbers” and “Network Parameters”.
- Enable the SIP Trunk, you want to use for peer-to-peer calling.
- Set the SIP Trunk Mode of this trunk as Peer-to-Peer.

  Refer “SIP Trunk”.
- Configure the Peer-to-Peer Table. Refer “Peer-to-Peer Numbers”.
- Dial any of the desired numbers configured in the Peer-to-Peer Table. For example: when you dial 456, the phone will dial the corresponding destination address 199.100.100.100:5060 you have configured for this number in the Peer-to-Peer Table.

  As the number gets dialed, the SIP Trunk icon through which the number is dialed out will appear on the LCD.

1. *A device for converting calls from one network type into another (IP calls into analog) and for distributing calls to multiple extensions, e.g. VoIP PBX, VoIP adapter, GSM-VoIP gateway.*

Peer-to-Peer Numbers

A call that is made over the IP network without using a SIP proxy (SIP server) is called a Peer-to-Peer Call. By deploying the IP Phone in a peer-to-peer application, Corporate Offices can use the IP Phone to communicate between their branch offices free of cost.

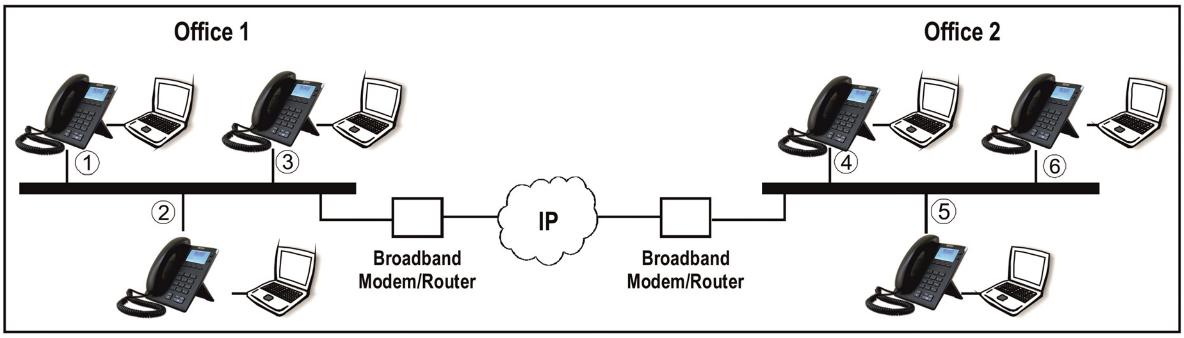
Installing SPARSH VP510 for Peer-to-Peer Application

The following describes how corporate users can connect the IP Phone for peer-to-peer application.

Please read the instructions on configuring Network and SIP Trunks first, before attempting peer-to-peer configuration.

Corporate Users

Installing SPARSH VP510 for a P2P application in corporate offices.



At each office:

- Connect the LAN port of the IP Phone to one of the ports of the LAN switch as shown in the figure.
- Connect the PC to the PC port of IP Phone.
- Connect the power adapter to the IP Phone.
- Power the IP Phone.

Configuring SPARSH VP510 for Peer-to-Peer Application

The Peer-to-Peer application can be configured only using Web User Interface.

Corporate Users

The steps described below are to be followed at both offices:

- To access the Jeeves, using a PC connected to the LAN port of the switch or using a PC connected to PC port of IP Phone, ensure that the LAN port of the IP Phone and the computers in the LAN or the computer connected to PC port of IP Phone are in the same subnet. The default Static IP Address of the IP Phone is 192.168.001.016 and the subnet mask is 255.255.255.0.

- If required, change the IP address and the subnet mask of the IP Phone via Phone User interface as per the IP addressing scheme used by your network.

- Open the web browser of any of the computers connected in the LAN network or computer connected to PC and enter the IP address of the IP Phone in the Address Bar of the browser.

- Log into Jeeves.

- Under **Basic Settings**, click **Network**.

  - Modify the IP address, subnet mask, gateway address etc., as required.

- Under **Advanced Settings**, click **System**.
  - In office1, configure the public IP address of broadband router in **Router Public IP Address**.
  - This should be done for all the IP Phones connected to LAN network of office1.
  - Similarly, in office2, configure the public IP address of the broadband router in **Router IP Address** of all the IP Phones connected to LAN network of office2.

- Decide the numbers of the IP Phones connected to the LAN of office1 and office2.

  - For example: the numbers assigned to the IP Phones connected to the LAN of office1 are as follows:

	IP Address	Number
IP Phone1	192.168.1.11	123
IP Phone2	192.168.1.12	234
IP Phone3	192.168.1.13	345

  - For example: the numbers assigned to the IP Phones connected to the LAN of office2 are as follows:

	IP Address	Number
IP Phone3	192.168.50.11	456
IP Phone4	192.168.50.12	567
IP Phone5	192.168.50.13	678

- Decide the SIP listening port of all the IP Phones connected to the LAN of office1 and office2.

  - For example: the SIP listening port of the IP Phones connected to LAN of office1 are as follows:

	IP Address	SIP UDP Port
IP Phone1	192.168.1.11	5060
IP Phone2	192.168.1.12	5061
IP Phone3	192.168.1.13	5062

  - For example: the SIP listening port of IP Phones connected to LAN of office2 are as follows:

	IP Address	SIP UDP Port
IP Phone3	192.168.50.11	5060
IP Phone4	192.168.50.12	5061
IP Phone5	192.168.50.13	5062

The SIP listening port must be unique to each IP Phone connected in a LAN. However, they can be same for the phones connected in different LANs.

- Decide the RTP listening ports of all the IP Phones connected to the LAN of office1 and office2.
  - For example: RTP listening port of IP Phones connected to LAN of office1 are as follows:

	IP Address	RTP Listening Port
IP Phone1	192.168.1.11	8000
IP Phone2	192.168.1.12	8008
IP Phone3	192.168.1.13	8016

  - For example: RTP listening port of IP Phones connected to LAN of office2 are as follows:

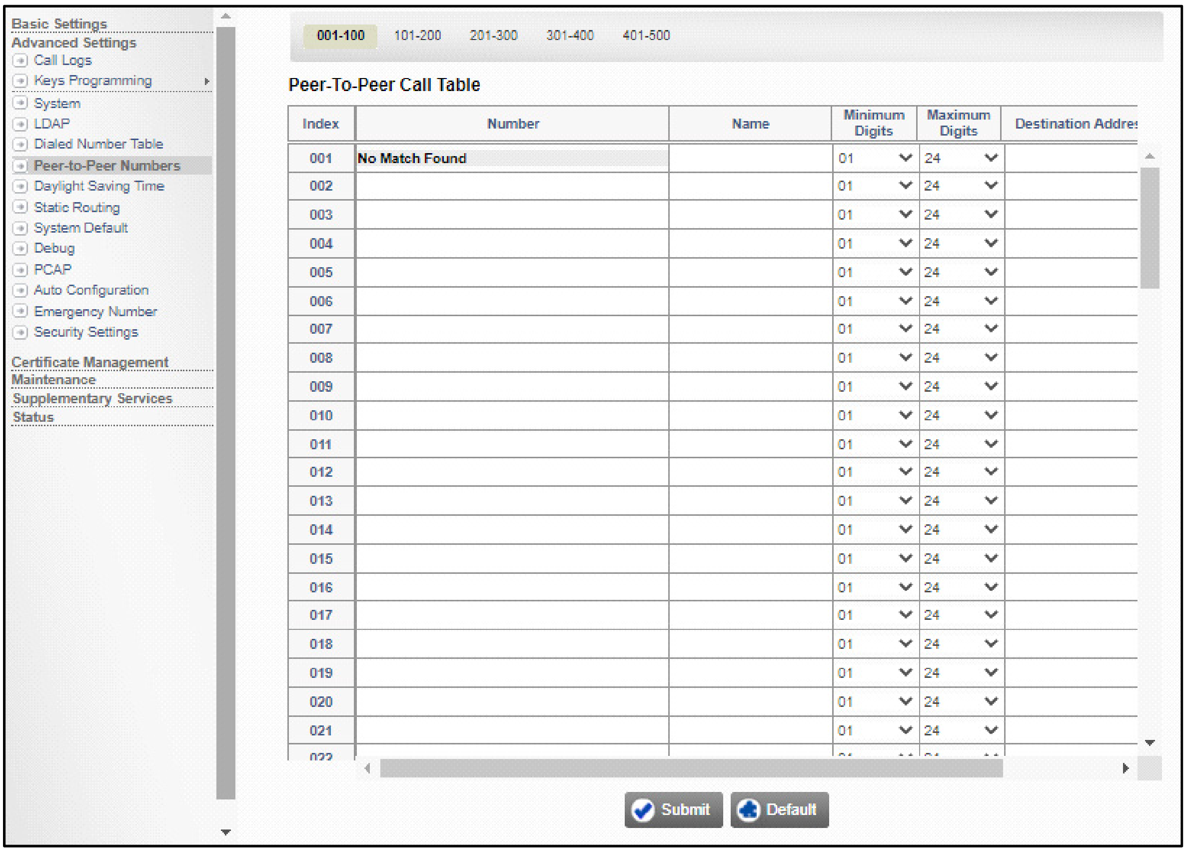
	IP Address	RTP Listening Port
IP Phone3	192.168.50.11	8000
IP Phone4	192.168.50.12	8008
IP Phone5	192.168.50.13	8016

The RTP listening port must be unique to each IP Phone connected in a LAN. However, they can be same for the phones connected in different LANs. Also, the RTP listening ports of phones connected in the same LAN must have an intermediate gap of 8 ports, since the IP Phone supports two simultaneous IP calls and each call needs two UDP ports.

- Under **Advanced Settings**, click **System** and configure the following parameters:
  - Configure the SIP UDP Port to the value decided by you.
  - Configure the RTP Listening Port to the value decided by you.

1. Under **Basic Settings**, click **SIP Trunk** and configure the following parameters:
   - Enable the desired SIP Trunk - SIP 1, SIP 2.
   - Select the SIP Trunk Mode as Peer-to-Peer.
   - Assign the number as SIP ID.
   - Set the **NAT Type** as **Router Public IP Address**.
   - Enable **Symmetric RTP**.
2. Enable the Port Forwarding in your Broadband Router of office1 as follows:
   - Forward port 5060 and 8000~8007 to the IP Address of IP Phone1 viz. 192.168.1.11.
   - Forward port 5061 and 8008~8015 to the IP Address of IP Phone2 viz. 192.168.1.12.
   - Forward port 5062 and 8016~8023 to the IP Address of IP Phone3 viz. 192.168.1.13.
3. Enable the Port Forwarding in your Broadband Router of office2 as follows:
   - Forward port 5060 and 8000~8007 to the IP Address of IP Phone4 viz. 192.168.50.11.
   - Forward port 5061 and 8008~8015 to the IP Address of IP Phone5 viz. 192.168.50.12.
   - Forward port 5062 and 8016~8023 to the IP Address of IP Phone6 viz. 192.168.50.13.

- Under **Advanced Settings**, click **Peer-to-Peer Numbers** and configure the peer to peer table as follows (Office1):



Index	Number	Name	Minimum Digit	Maximum Digit	Destination Address	SIP Transport
001						
002	456		3	3	199.100.100.100:5060	UDP
003	567		3	3	199.100.100.100:5061	UDP
004	678		3	3	199.100.100.100:5062	UDP
:						
500						

Where, 199.100.100.100 is considered as the public IP address of broadband router of office2.

For each phone configure the following information:

- **Number:** This is the number of the called party to be dialed.
- **Name:** This is the name of the called party (serves as identification of the called party).
- **Minimum Digits:** This is the number of digits that the phone should wait to be dialed to consider it a valid number.

- **Maximum Digits:** This is the number of digits that the phone should consider as "End of Dialing" and dial out the number.

- **Destination Address:** This is the corresponding IP Address for the configured Number.

- **SIP Transport:** This is the SIP Transport — UDP, TCP, TLS, you wish to use.

- Similarly, under **Advanced Settings**, click **Peer-to-Peer Numbers** and configure the peer to peer table as follows (Office2):

Index	Number	Name	Minimum Digit	Maximum Digit	Destination Address	SIP Transport
001						
002	3		3	3	59.162.252.82:5060	UDP
003	5		3	3	59.162.252.82:5061	UDP
004	5		3	3	59.162.252.82:5062	UDP
:						
500						

Where 59.162.252.82 is considered as the public IP address of broadband router of Office2.

For each phone configure the following information:

- **Number:** This is the number of the called party to be dialed.

- **Name:** This is the name of the called party (serves as identification of the called party).

- **Minimum Digits:** This is the number of digits that the phone should wait to be dialed to consider it a valid number, and dial out the number.

- **Maximum Digits:** This is the number of digits that the phone should consider as "End of Dialing" and dial out the number.

- **Destination Address:** This is the corresponding IP Address for the configured Number.

- **SIP Transport:** This is the SIP Transport — UDP, TCP, TLS, you wish to use.

- Under **Basic Settings**, click **Outgoing Call Routing** to configure the outgoing call route.

Now, all of your IP Phones are ready to make and receive calls between the branch offices!

Phone Book

You can store the names and numbers of 200 contacts in the Phone Book of the SPARSH VP510. The Phone Book is useful for storing numbers of the desired contacts and making calls to these contacts using the **Dir.** Key.

You can also save the LDAP contacts in the Phone Book. For details refer to "LDAP".

You ca n also assign a DSS Key to the desired contact from the Phonebook.To know more, refer to "DSS Keys Programming".

Adding Contacts to the Phone Book

You can enter the names and numbers of your contacts, edit or delete their details in the Phone Book using either the Web User Interface as well as the Phone User Interface.

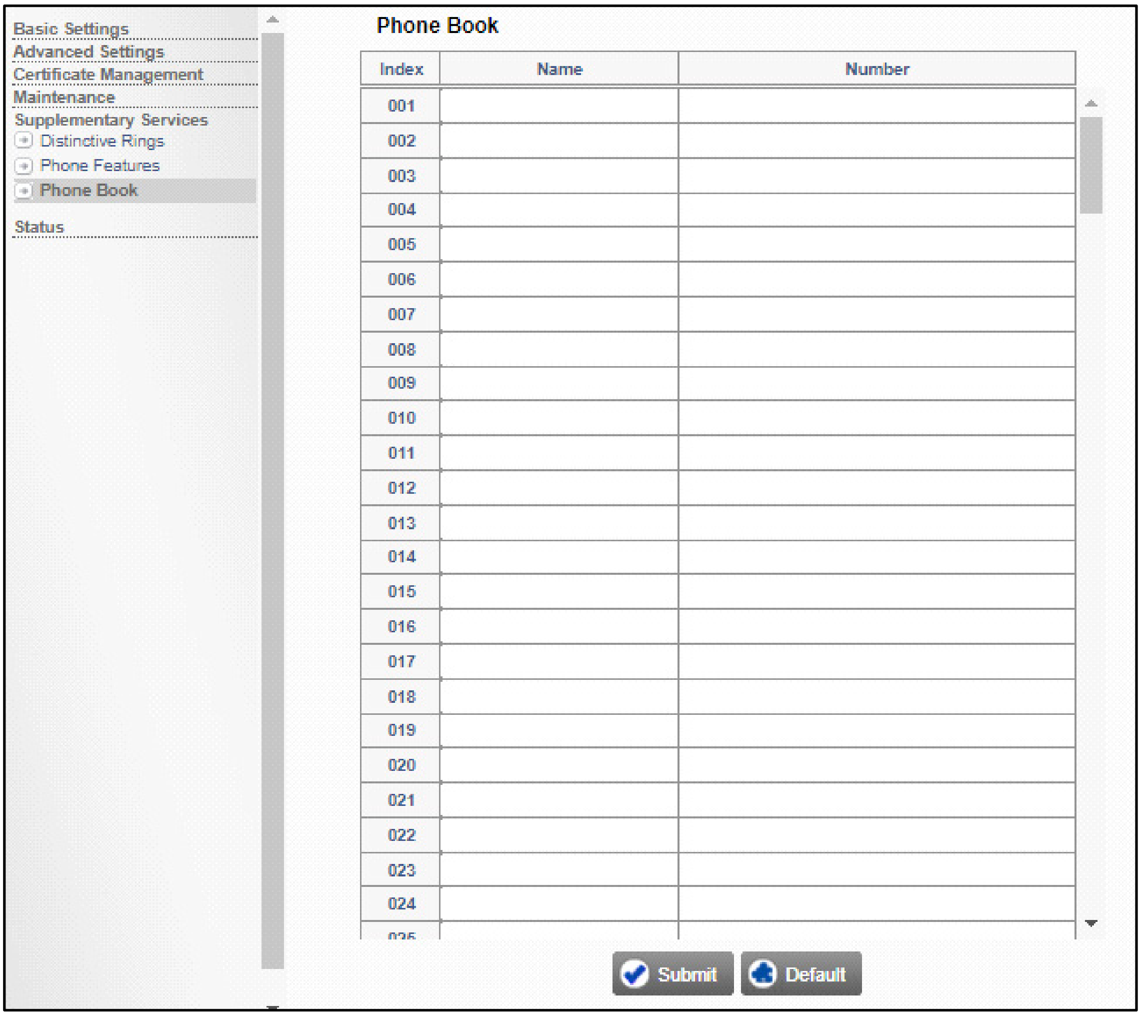
Adding Contacts via Phone User Interface

To Add, Edit, Delete contacts via Phone User Interface, refer "Contacts", "Adding Contacts", "Editing and Deleting Contacts".

Adding Contacts via Web User Interface

- Log into Jeeves.
- Under **Supplementary Services**, click **Phone Book**.

- The Phone Book page appears.



- Enter the names of your contacts under **Name** and their numbers/IP addresses under **Number**. The Name may consist of special characters, numbers or they may be names of persons, businesses, departments, etc. The name must not exceed 24 characters. The Number may be a telephone number, an IP address or SIP URI, not exceeding 40 characters.

- To edit an existing contact name/number, click on it with the cursor and edit (with your keyboard).

- To delete contacts, select the name and its corresponding number and press the delete key on your keyboard.

  In the Phone Book in the Jeeves, the names and numbers of your contacts are stored index wise (each contact, i.e. name and number has an index starting from 01-200 and not in alphabetical order).

If you want to delete all contacts, you must select each name and number manually. Or you can default the page to delete all contacts.

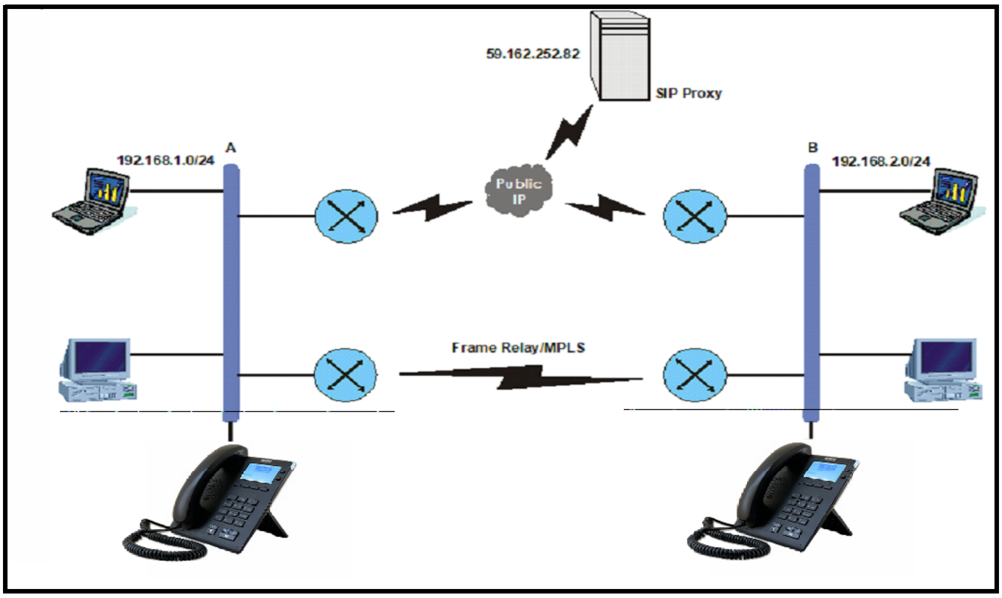
- Click **Submit** to save the changes you have made to the Phone Book (adding, editing, deleting contact details).

For each Entry in the Phone Book, if you fail to enter a corresponding number for a name or a corresponding name for a number, you will be prompted to enter the name/number for the entry. You will not be allowed to submit your entries or modifications in the Phone Book until you have completed the information.

Static Routing

Static Routing Table is required for routing calls in Multiple Gateway Applications. Static Routing Table enables routing of calls between point to point sites connected through MPLS, Frame Relay and to the public Internet at the same time. Consider the example illustrated in the figure below.

SPARSH VP510 is connected in a Frame Relay/ MPLS network.



- In the above figure, two Local Area Networks are connected through Frame Relay/ MPLS network to give access to local resources and also to make Peer to Peer calls.

- These sites are also connected to public IP network for the following reasons:

    - To give internet access to local hosts.
    - To access DID service provided by ITSPs to make PSTN/ GSM calls over IP network.

- IP Phone is connected at both sites.

- Network A and Network B are in different subnets, but SPARSH VP510 has only one default gateway where IP packets can be routed. If the IP Phone is connected behind two Routers, it cannot decide where to route the IP packets.

- So, only Peer to Peer calls between network A and B OR Proxy calls can be made at the same time.

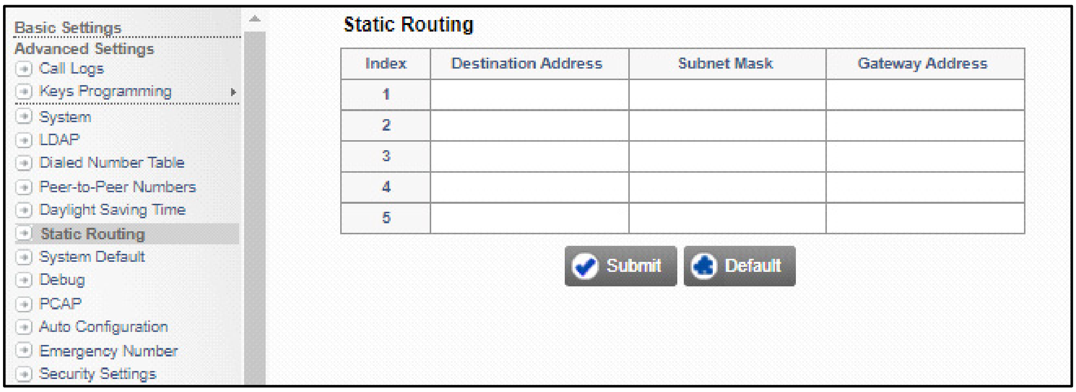
SPARSH VP510 resolves this by way of the Static Routing Table. The Static Routing Table defines the appropriate Gateway Address (OR Router LAN Address) where the IP packets are to be sent. This table makes it possible to route outgoing calls made from the phone to different subnets - Peer-to-Peer or Proxy - to the Gateway as per the called network.

To take the above example further, if the user of SPARSH VP510 (connected in network A) makes an outgoing call to 192.168.2.0/ 24, the call will be routed to Frame Relay/ MPLS link. If the user makes an outgoing call to the SIP proxy server 59.162.252.82, the call will be routed from router which connects IP Phone to public IP network.

To summarize, you need not configure Static Routing when SPARSH VP510 is connected behind a NAT Router. LAN Interface of the NAT Router acts as default Gateway for the IP Phone. Calls initiated from SPARSH VP510 get routed from the LAN port of the NAT Router. But if you have connected multiple offices through MPLS, Frame Relay and want to make and receive Peer-to-Peer Calls between various offices and Proxy calls to the Public Internet at the same time, you need to configure the Static Routing Table in the SPARSH VP510.

Configuring Static Routing via Web User interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Static Routing**.



The Static Routing Table allows you to store up to 5 entries. Each entry is stored against an Index number. For each entry, you must configure the following:

- **Destination Address:** Enter the address of final destination where the call is to be made. This can be an IP address of the end point or the network address where end point resides.
- **Subnet Mask:** Enter the Subnet Mask to be applied on the Destination Address.
- **Gateway Address:** Enter the IP address of the node where the IP packets are to be sent. In most cases, this field specifies an IP address of Router's LAN interface on which the IP Phone is connected. This address must be in the same subnet where IP Phone resides.
- Click **Submit** to save.

The Static Routing Table will be checked each time an outgoing call either Proxy or Peer-to-Peer is made.

If the final destination IP address and IP Phone are not in the same Subnet, the SPARSH VP510 will compare the final destination IP address with the entries configured in the Routing Table.

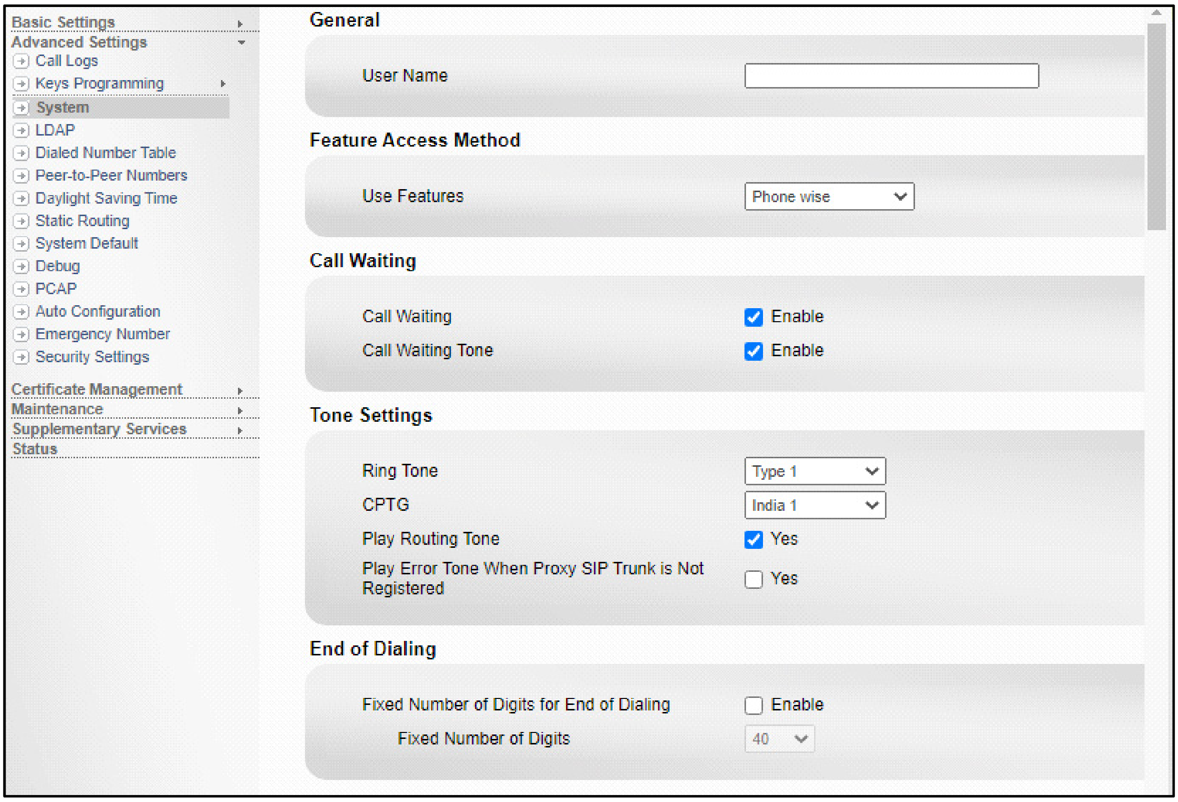
If a perfect match is found, the phone will start sending the IP packets to the corresponding Gateway Address configured.

If no match is found the IP Phone will use the Default Gateway Address configured on Network Parameters to route the call.

CHAPTER 17 System Parameters

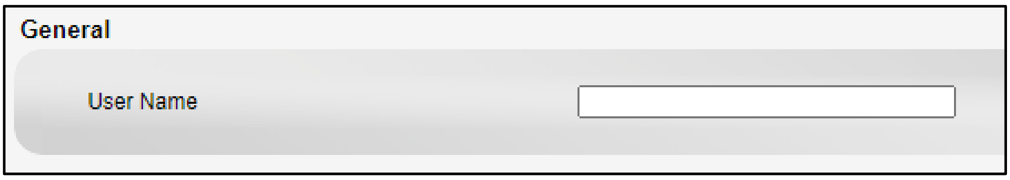
To configure System Parameters,

- Log into Jeeves.
- Under **Advanced Settings**, click **System**. The System Parameters page appears.



- You may configure the following parameters as per your requirements:

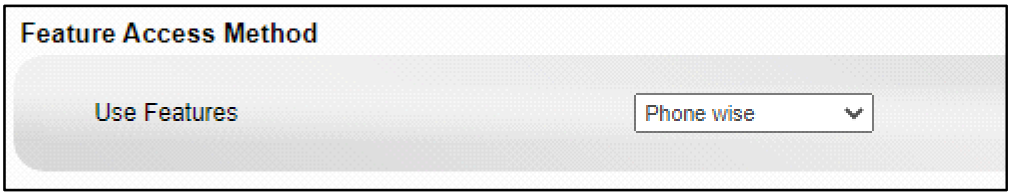
General



- The name you have configured as the **Display Field on LCD** in SIP Trunk appears automatically as the **User Name**. However, if you wish to assign a different name to SPARSH VP510, you can do so by configuring the same as the **User Name**. This name has significance when multiple SPARSH VP510 are connected in the same LAN network. If you have changed the User Name, then this User Name will be displayed on the Phone LCD, however the parameter **Display Field on LCD** in SIP Trunk will not be overwritten with this User Name.For details, refer to "Trunk Settings" in "SIP Trunk".

  The name may have a of maximum 40 characters. Default: Blank.

Feature Access Method



- In **Use Features** set the desired feature access method as **Phone wise** or **SIP Trunk wise**. Default: Phone wise.

  If you select Phone wise, the applicability of the feature will be universal, that is, it will be applicable to all the SIP Trunks.

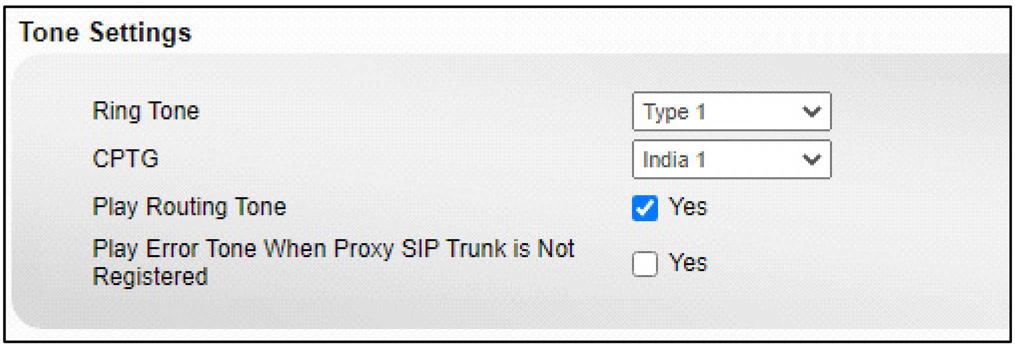
  If you select SIP Trunk wise, the feature will be applicable for the selected SIP Trunk only.

  Example, if you select Phone wise and set DND, then it will be set for all the SIP Trunks, but if you select SIP Trunk wise, then it can be set for each SIP Trunk.

Call Waiting Settings

Refer "Call Waiting" for details.

Tones Settings



- SPARSH VP510 supports different ring types. Select the **Ring Tone** as per your requirement from the drop down list. Default: Type1.

- Select **CPTG** (Call Progress Tone Generation) to match the CPTG of the country where SPARSH VP510 is installed. Default: India.

  The SPARSH VP510 supports country specific Call Progress Tones Generation (CPTG) to simulate the same tones of the local PSTN to which it is connected. The CPTG supported by SPARSH VP510 for different countries are presented in the “Call Progress Tone Generation (CPTG)”.

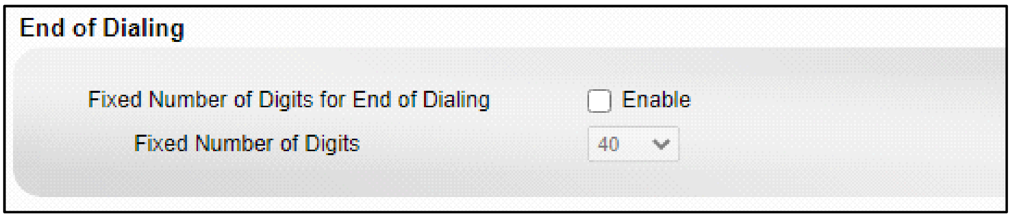
When you set SPARSH VP510 *to factory defaults, CPTG you selected will not be set to default.*

- By default the **Play Routing Tone** check box is selected that is, the phone plays routing tone, clear the check box, if you do not want the phone to play routing tone.

- Select the **Play Error Tone when Proxy SIP Trunk is Not Registered** check box if you want the phone to play error tone when proxy SIP trunk is not registered. Default: Disabled.

End of Dialing

Unlike a normal phone which transmits each digit you press to the telephone exchange for dialing, when you dial a number using this IP phone, you must first enter all the digits/characters and indicate to the phone the 'End of Dialing', i.e. that you have finished dialing.



You can set the following as End of Dialing:

- Select the **Fixed Number of Digits for End of Dialing** check box to enable. Default: Disabled.

- Configure the **Fixed Number of Digits**.

Language Settings

Refer “Language” for details.

Date-Time Settings

- To use a public internet time server for date and time synchronization, select an internet based time server as **NTP Server.** You may select any of the reliable public internet time servers[2] supported by SPARSH VP510:
  - time.windows.com
  - time.nist.gov

2. *You can select from three free, reliable public internet time servers run by the University of Wisconsin-Madison, Microsoft, and the National Institute of Standards and Technology (NIST), to obtain date and time. You can also configure an NTP time server of your preference, other than these. These public internet time servers provide time offset from the Greenwich Mean Time (GMT), and you can select the time according to the time zone of the country you are installing the SPARSH VP510. For instance, if your SPARSH VP510 is installed in India, you can select the time zone for India (GMT+5.30 Calcutta, Chennai, Mumbai, and New Delhi). The time for India is offset from GMT by +5.30 hours. Similarly, if your SPARSH VP510 is installed in Hawaii, select the time zone for Hawaii, which is offset by GMT -10:00 hours, to set the correct time and date.*

Default: time.windows.com

If you want to use an NTP server other than these, select the radio button and enter the IP Address/ domain of the server.

- To synchronize Date and Time of SPARSH VP510 with that of the country where it is installed, select **Time Zone** from the drop down list. Default: (GMT+05:30) Kolkata, Chennai, Mumbai, New Delhi.

  Also refer “Time Format” for details.

- *The Current Date and Time will be displayed on the Status page. See “Status”.*
- *When you set SPARSH* VP510 *to factory defaults, Date and Time settings will not be set to default.*

LCD Settings

Refer “Display Settings” for details.

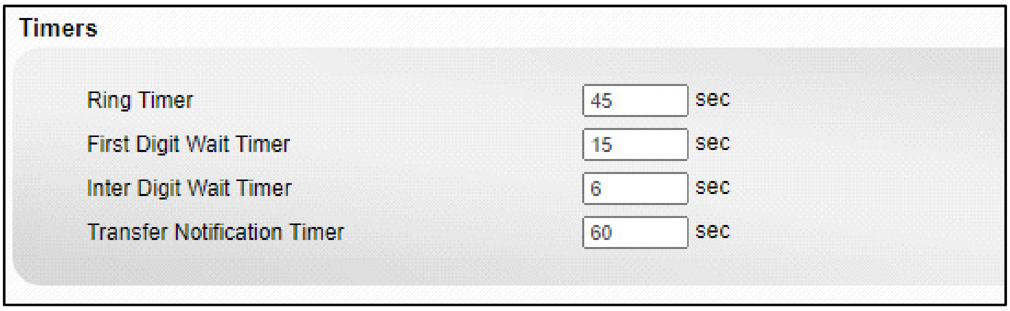
Volume Settings

Refer “Volume Settings” for details.

Headset Connectivity

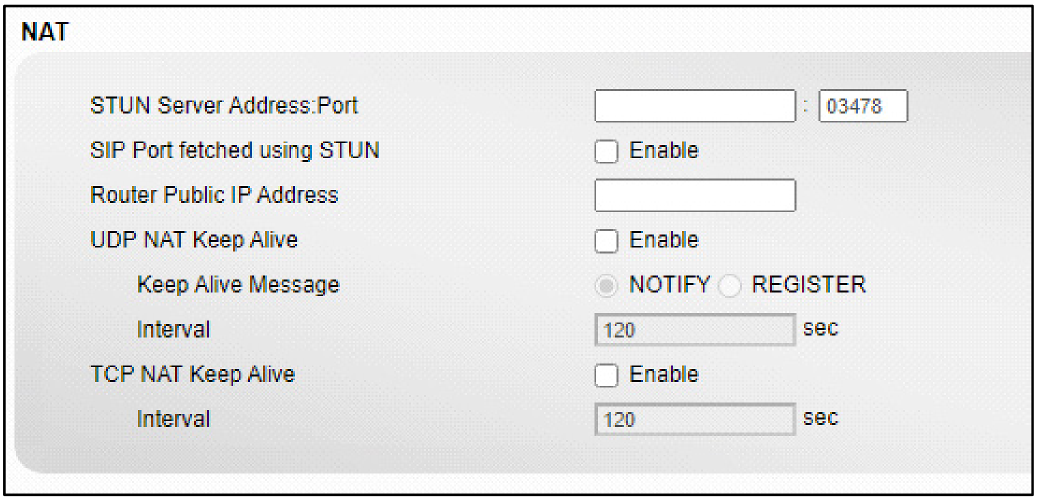
Refer “Accessories” for details.

Timers



- **Ring Timer** is the time in seconds the phone will ring to indicate an incoming call. When the time ends, the phone will stop ringing and go to the idle state. Valid range: 01-99 sec. Default: 45 sec.
- **First Digit Wait Timer** is the time in seconds the phone *will wait for the first digit to be pressed. On expiry of this timer, phone will go in the error state. Valid range: 01-99 sec. Default: 15 sec.*
- **Inter Digit Wait Timer** is the time in seconds the phone will wait between two digits of a number before dialing out that number. On expiry of this timer, the number is considered to be complete and is dialed. Valid range: 01-99 sec. Default: 06 sec.
- **Transfer Notification Timer** is the time in seconds the phone will wait for notification of the status of a transferred call, i.e. whether the transfer target is busy, has answered, has disconnected, etc. Valid range: 01-99 sec. Default: 60 sec.

NAT



- STUN is required if the SPARSH VP510 is located behind the NAT router. STUN server facilitates traversing through most NATs, except symmetric NATs. If your router has symmetric NAT, do not configure STUN.

  This parameter is applicable only if you have selected the option 'STUN' for 'NAT Type' of the SIP Trunk. Refer “SIP Trunk”.

  Enter the **STUN Server Address** (max. 40 characters, Default: Blank) and Port, that is the Listening Port of STUN server. Valid range: 1024-65535. Default: 3478.

- Keep the **SIP Port fetched using STUN** check box disabled, if the IP phone is located behind the NAT router and you have forwarded the SIP listening port of the phone in the router to the phone, you do not need to use the port provided by STUN server as SIP listening port. Default: Disabled.

  Select the check box to enable, if you have not forwarded the SIP port in the router to the IP phone.

- Configure the **Router Public IP Address,** if the IP Phone is located behind the NAT router (any type). This option will work only if Outbound is disabled on the SIP trunk.This parameter is applicable only if you have selected the option 'Routers Public IP address' for 'NAT Type' of the SIP Trunk. Default: Blank. Refer “SIP Trunk”.

- Select the **UDP NAT Keep Alive** check box, when SPARSH VP510 is connected behind a NAT router and SIP messages are transported over UDP, NAT Keep Alive messages must be sent to refresh the binding in the NAT router.Default: Disabled.

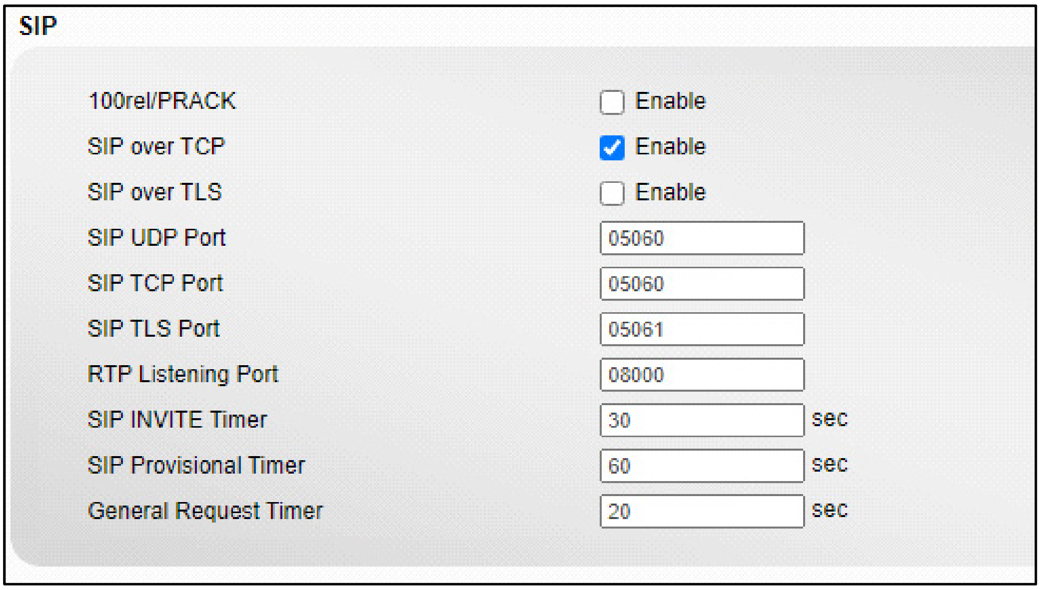
  Select **Keep Alive Message** type if NAT Keep Alive is enabled. Select either REGISTER or NOTIFY. Default: NOTIFY.

  Configure the **Interval** that is, the time period after which the phone should send Keep Alive messages. This time period should be less than the binding timer of the router. Valid range: 001-999 sec. Default: 120 sec.

- Select the **TCP NAT Keep Alive** check box, when SPARSH VP510 is connected behind a NAT router, and SIP messages are transported over TCP, NAT Keep Alive messages must be sent to refresh the binding in the NAT router. Default: Disabled.

Configure the **Interval** that is, the time period after which the phone should send Keep Alive messages. This time period should be less than the binding timer of the router. Valid range: 001-999 sec. Default: 120 sec.

SIP



- 100rel is to be configured if you want to support reliable transmission of (SIP) provisional responses. Select the **100rel** check box, if you want the phone to use 100rel SIP extension for reliable transmission of SIP provisional responses and to use PRACK (Provisional Acknowledgment). Default: Disabled.

  SPARSH VP510 supports transporting of SIP messages over User Datagram Protocol (UDP), Transport Layer Security (TLS) as well as Transfer Control Protocol (TCP) connection.

- By default, the **SIP Over TCP** check box is selected, and you will receive SIP messages over TCP. To be able to send SIP messages over TCP, you must configure the SIP Transport mode as 'TCP' in the SIP Trunk. Clear the check box to disable.

- Select the **SIP Over TLS** check box, if you want to receive SIP messages over TLS. To be able to send SIP messages over TLS, you must configure the SIP Transport mode as 'TLS' in the SIP Trunk. Default: Disabled.

- Configure the **SIP UDP Port.**This port defines the port on which the SPARSH VP510 listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. Valid range: 1024-65534. Default: 05060.

- Configure the **SIP TCP Port**. This port defines the port on which the SPARSH VP510 listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. Valid range: 1024-65534. Default: 05060.

- Configure the **SIP TLS Port**. This port defines the port on which the SPARSH VP510 listens for SIP messages transported over TLS. This port is also used as the source port for sending SIP messages to the remote peer. Valid range: 1024-65534. Default: 05061.

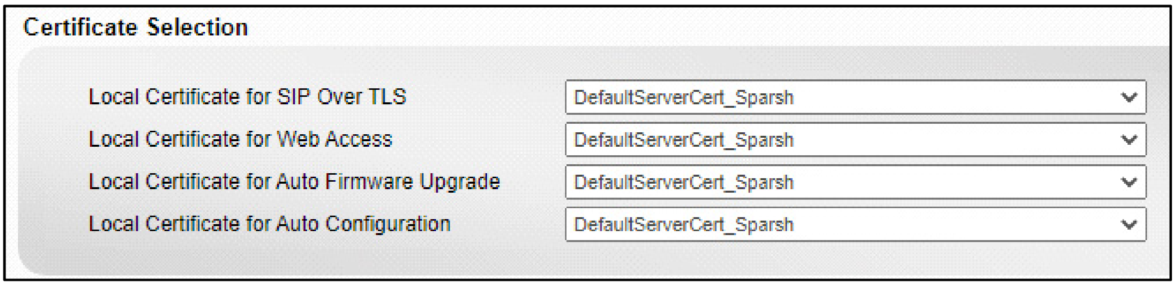
- Configure the **RTP Listening Port**. This port defines the port on which the SPARSH VP510 listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer. Valid range: 1024-65526. Default: 08000.

- **SIP INVITE Timer** is the time in seconds for which 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. Valid range: 010-180 sec. Default: 30 sec.

- **SIP Provisional Timer** is the time in seconds for which 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 SPARSH VP510 terminates the call process and gives error tone to the user. Valid range: 010-180 sec. Default: 60 sec.

- **General Request Timer** 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. Valid range: 10-60 sec. Default: 20 sec.

Certificate Selection



- In **Local Certificate for SIP Over TLS**, select the desired certificate from the installed local certificates that should be sent as the Server Certificate to the Client for secure SIP communication.

- In **Local Certificate for Web Access**, select the desired certificate from the installed local certificates that should be sent as the Server Certificate to the Client for secure HTTP access.

- In **Local Certificate for Auto Firmware Upgrade**, select the desired certificate from the installed local certificates that should be sent as the Server Certificate to the Client for secure Auto Firmware Upgrade.

- In **Local Certificate for Auto Configuration**, select the desired certificate from the installed local certificates that should be sent as the Server Certificate to the Client for secure Auto Configuration.

CHAPTER 18

Keys Programming

Key Programming enables quick access to features/facilities, trunks as well as extensions.There are various types of keys — Context Sensitive Keys, DSS Keys as well as Dial Pad Digit Keys that can be programmed.

For details regarding configuring each type of key, refer to:

- “Context Sensitive Keys (CSK) Programming”
- “DSS Keys Programming”
- “Speed Dial” for configuring Dial Pad Digit Keys.

Context Sensitive Keys (CSK) Programming

SPARSH VP510 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 using the Web User Interface only.

The screens — Idle Screen, Call Screen, Transfer Dial Screen, all have different set of features that can be accessed. SPRASH VP510, 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 Call Screen and Transfer Dial Screens you can set the priorities of the features.

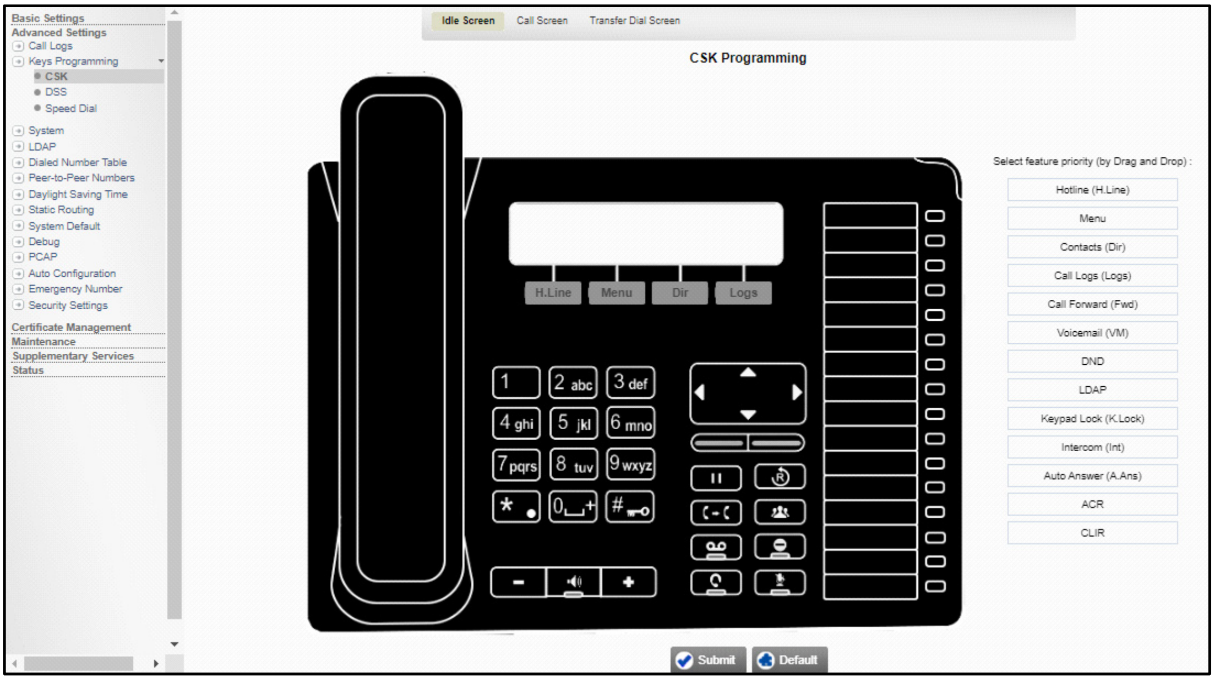
Refer to the details mentioned below for details:

Type of Screen	List of features that can be assigned to the Context Keys as well as features for which priorities can be set
Idle Screen	Contacts
	Call Logs
	Call Forward
	Menu
	Voicemail
	DND
	LDAP
	Keypad Lock
	Intercom
	Auto Answer
	Hotline
	ACR
	CLIR
Call Screen	New Call
	Hold
	Transfer
	Conference
	Record[a]
	End Call
Transfer Dial Screen	New Call
	Transfer Complete
	SIP Trunk 1
	SIP Trunk 2
	End Call

a.The Record functionality will be available in future release.

Customizing the Key Template via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Keys Programming**.
- Click **CSK**. The **CSK** Programming page appears.



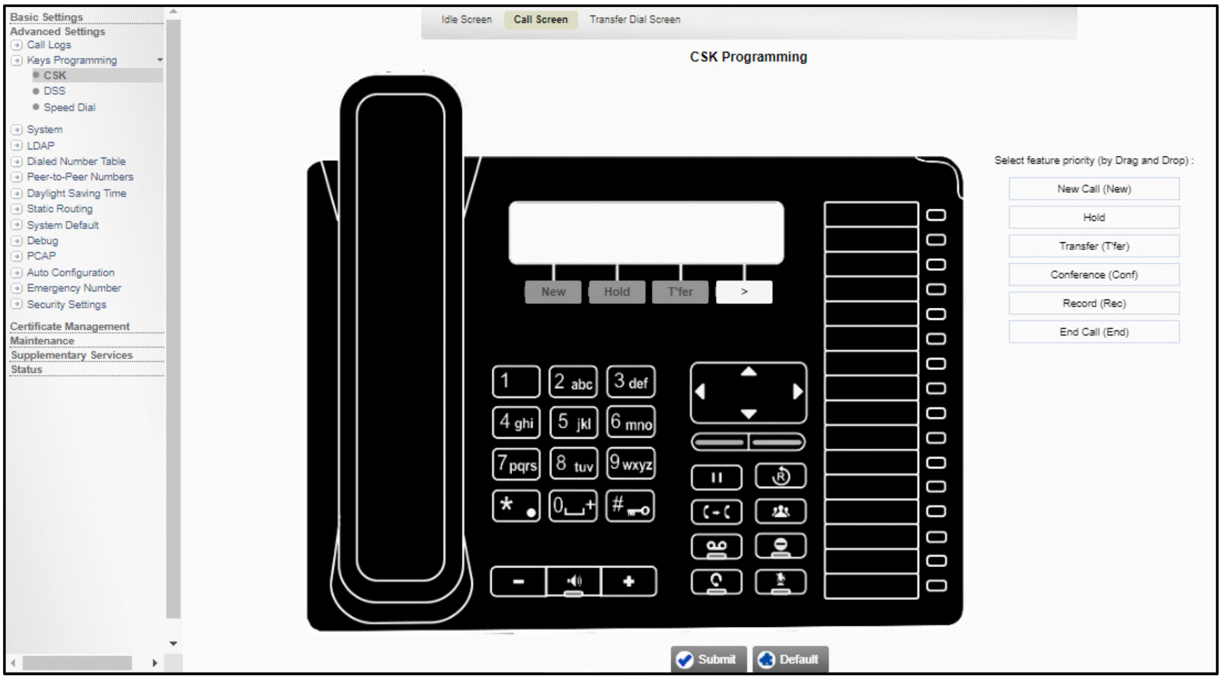
- Click **Idle Screen**.
- Each Context key, 1 to 4 can be assigned features.
- The feature assignment cum priority list appears on the right. You can change the feature assignments/ priorities as per your preference.
- To set the priority, drag and drop the features in the order of your preference. 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 **Call Screen** or **Transfer Dial Screen** and can set the feature priorities as per your preference.

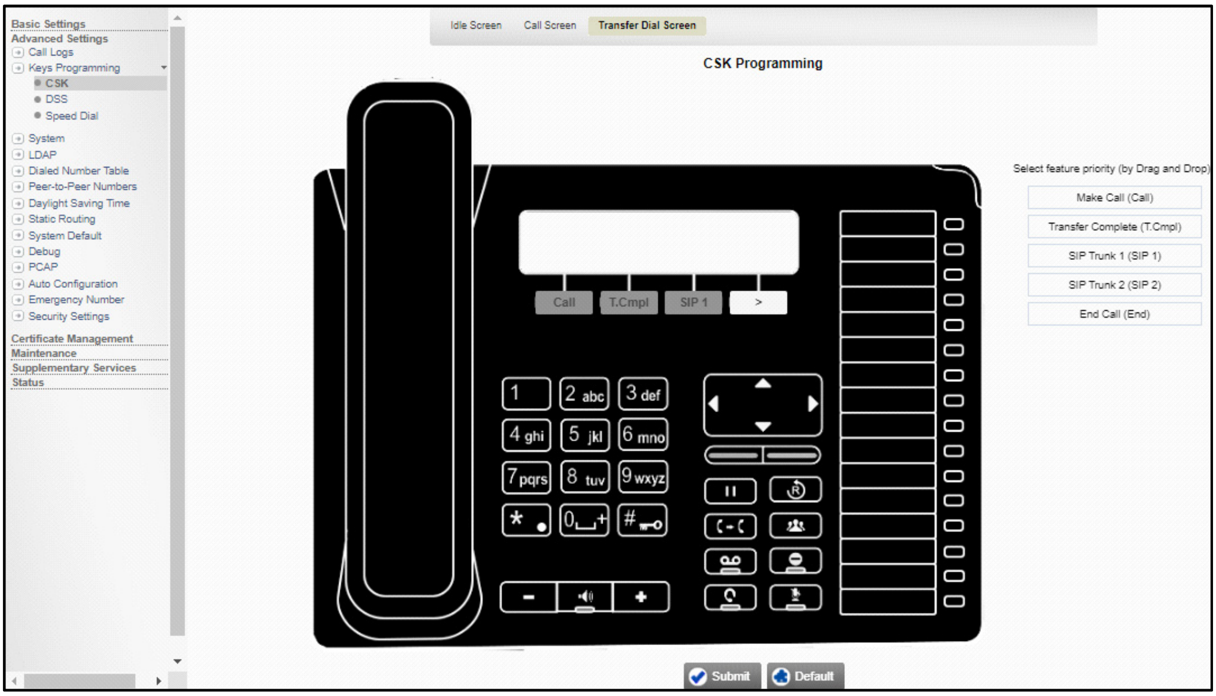
In the Call Screen and Transfer Dial Screens only priorities can be set.

Call Screen



The Record functionality will be available in future release.

Transfer Dial Screen



In **Call Screen** and **Transfer Dial Screen** the 4th Context Key will always be assigned to **More >** feature**.**

DSS Keys Programming

The DSS (Direct Station Selection) Keys can be customized as per your needs to provide you quick access to Contacts, Trunks as well as Features/Functions. You can configure DSS Keys using the Web User Interface only.

Once the DSS Keys are configured the Key label displays the same. Refer to the table below to know the Labels that will be displayed as per the configuration.

Feature	Label Description	Status and LED Indication
SIP Trunk 1	SIP1	Active/Registered: Continuous ON (Blue) Disabled: OFF
SIP Trunk 2	SIP2	Active/Registered: OFF Disabled: Continuous ON (Blue)
Call Waiting	C.Wait	Set: Continuous ON (RED) Cancel: OFF
Contacts	DIRIndex at which the Number of the Contact is stored in the Phonebook.	Idle: OFF The key assigned to the number you are in speech with: Continuous ON (RED) The key assigned to the number you have kept on hold: Slow Blinking (Blue) The key assigned to the number you are calling or from which you are being called: Fast Blinking (Blue)
Call logs - Missed	Logs.M	-
Call logs - Dialed	Logs.D	-
Call logs - Received	Logs.RCV	-
Call logs - Rejected	Logs.RJ	-
Call Forward Always	Fwd.A	Set: Continuous ON (RED) Cancel: OFF
Call Forward Busy	FwdBusy	Set: Continuous ON (RED) Cancel: OFF
Call Forward No Reply	FwdNR	Set: Continuous ON (RED) Cancel: OFF
LDAP	LDAP	-
Keypad Lock	K.Lock	-
Intercom	Intercom	-

Feature	Label Description	Status and LED Indication
Auto Answer	A.Ans	Set: Continuous ON (RED) Cancel: OFF
Hotline	H.line	Set: Continuous ON (RED) Cancel: OFF
Anonymous Call Rejection	ACR	Set: Continuous ON (RED) Cancel: OFF
Caller ID Restriction	CLIR	Set: Continuous ON (RED) Cancel: OFF
Busy Lamp Field (BLF)	BLFExtension Number	
Voicemail	VM	No Voicemail: OFF Unread Voicemail: Continuous ON (RED)
DND	DND	Set: Continuous ON (RED) Cancel: OFF

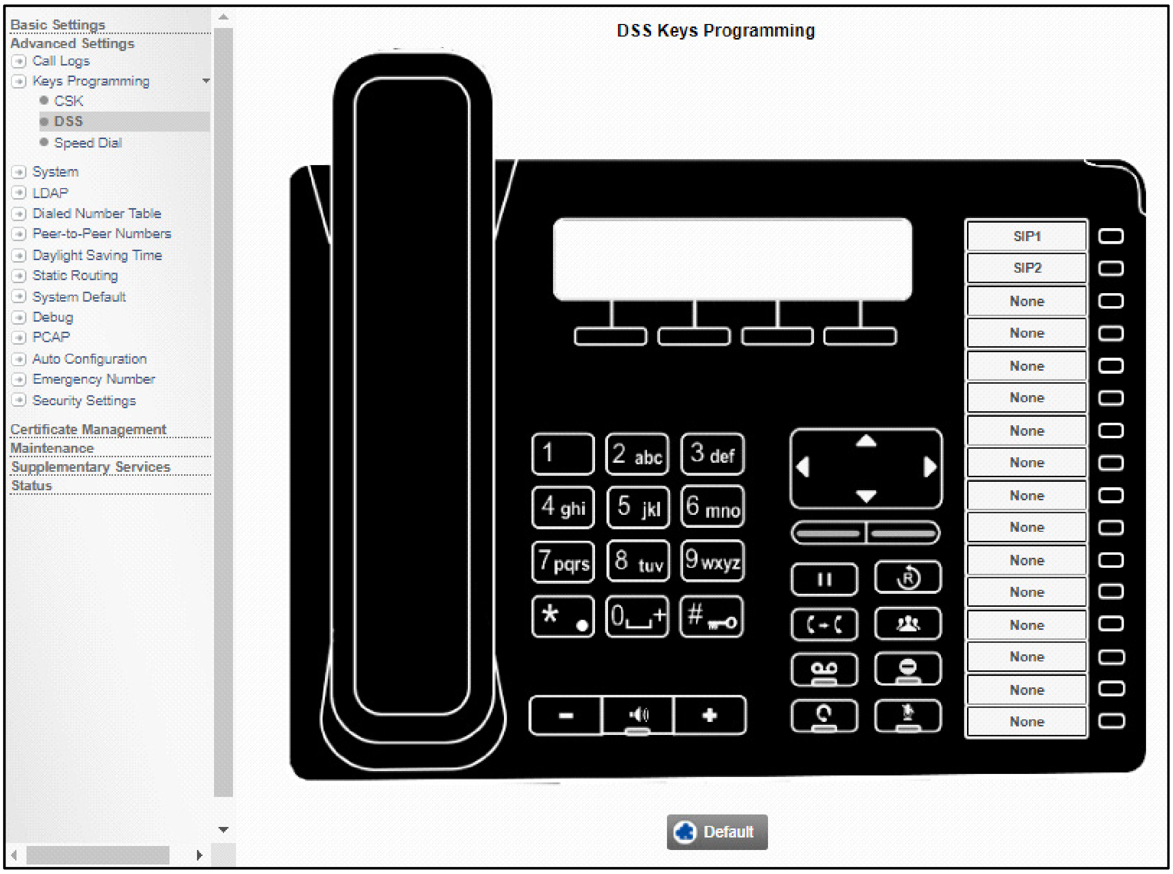
Feature/Facilities assigned to the DSS Keys can be accessed only when the Phone is in the Idle State.

Certain Features can be set Phone wise as well as SIP Trunk wise, such features can be assigned DSS Keys as well. The DSS Key functionality will be as per the set mode. To set the desired Feature Access Mode, refer to "System Parameters".

Assigning DSS Keys via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Key Programming.**

- Click **DSS.** The **DSS Keys Programming** page appears.

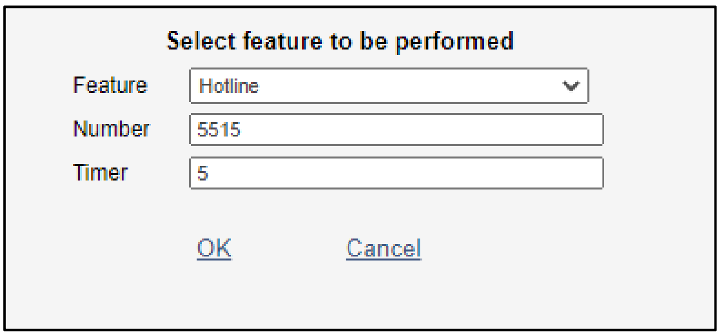


- By default, **SIP Trunk 1** and **SIP Trunk 2** is assigned to DSS1 and 2. However, you can change the same if required. All other DSS Keys are not assigned anything.

  Let us consider the following example as per which you wish to customize:

DSS Key Number	Assigned Feature/Facility/ extension
DSS 3	Hotline
DSS 4	Contacts

- Click DSS 3, the **Select feature to be performed** page appears.

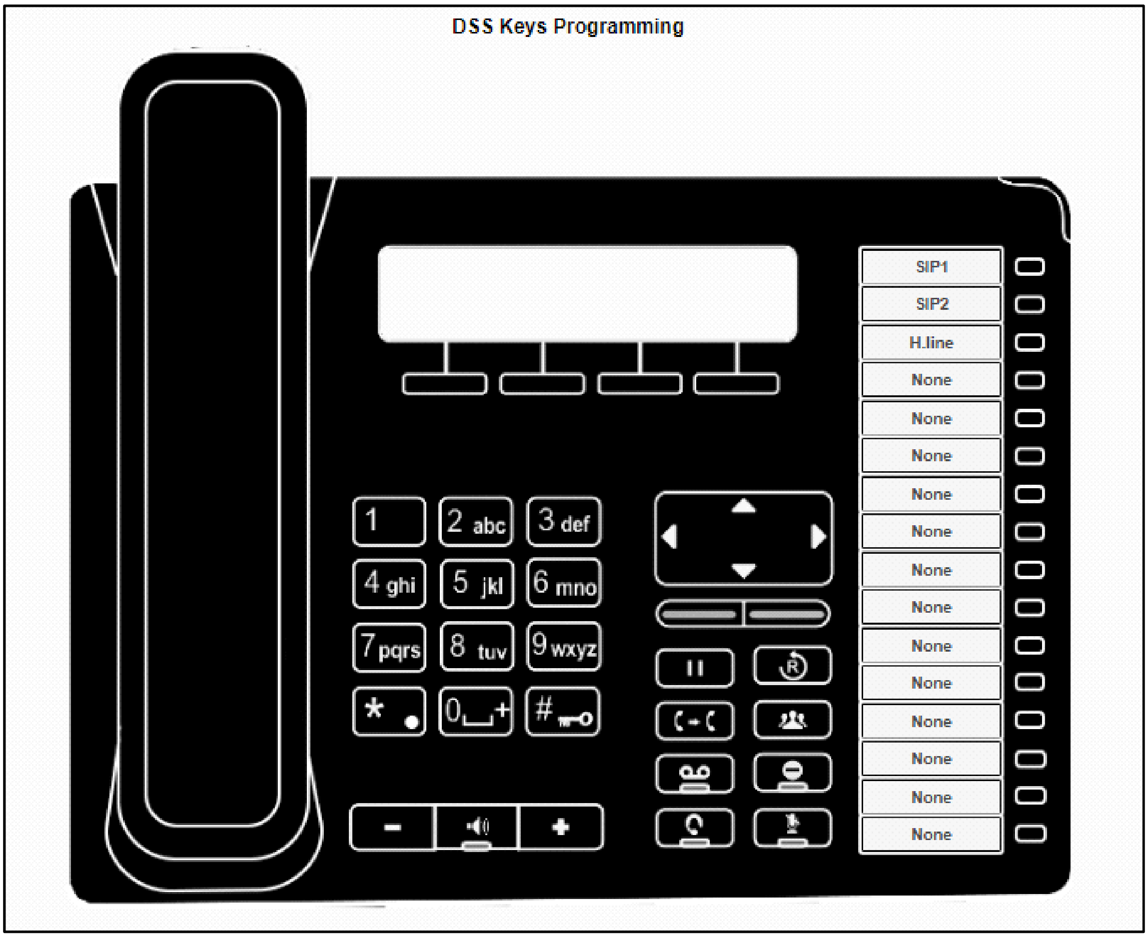


- **Feature:** Select **Hotline** from the drop-down list.

- **Number:** Enter the number for which you wish to set Hotline.

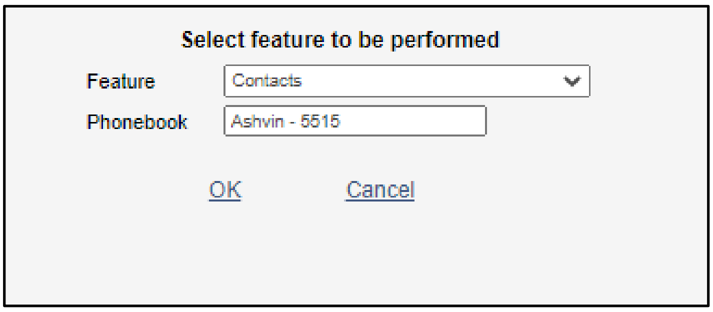
- **Timer:** Enter the time for which the phone should wait (after going OFF- Hook) before automatically dialing out this number. By default the phone will wait for 5 seconds. If you want the phone to immediately dial out the number once OFF-hook, configure the timer value as 0.

- Click **OK**.



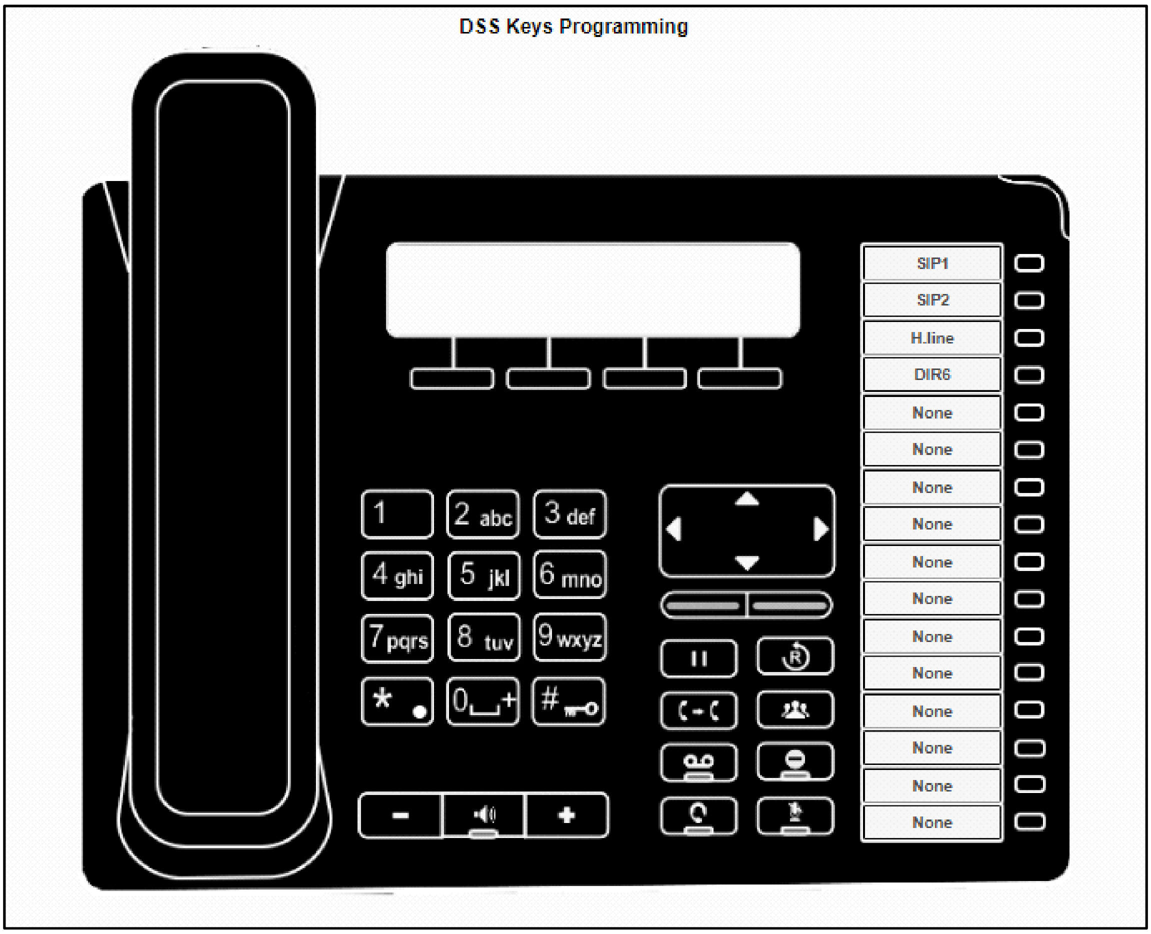
The Hotline feature will appear on the key label as **H.line**.

Now, click DSS 4, the **Select feature to be performed** page appears.



- **Feature:** Select **Contacts** from the drop-down list.

- **Phonebook:** Enter the desired number from the Phonebook for which you wish to assign this key.
- Click **OK**.



The desired Contact is assigned the key and the key label appears as **Dir6**, that is Directory (Phonebook) and Index Number of the Phonebook at which this contact is saved.

Similarly, you can configure other DSS Keys.

If you wish to assign default values to all the DSS Keys,

- Click **Default** on the **DSS Keys Programming** page.

DSS532 is not supported in SPARSH VP510 Standard SIP Mode.

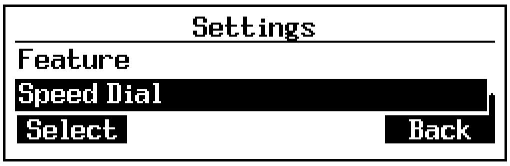
Speed Dial

As the name itself suggests, this feature offers you a quick way to dial a number. You can dial a number on the press of a single key, saving you the effort of pressing several digits or searching the number in your contact list in the Phone.

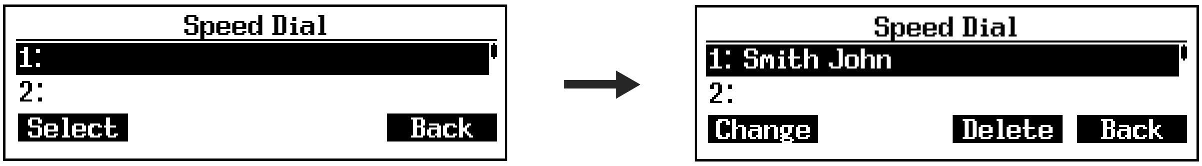
For this feature to work, you must first configure the name and numbers of these contacts in the Phone Book. Then these can be assigned to a key (that is the Dial Pad Key numbers, 1 to 9). Press the desired Dial Pad Key assigned to the contact whenever you wish to quickly dial out the number. As the phone dials, the Name of the contact will appear on the LCD.

Assigning Speed Dial Keys via Phone User Interface

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Settings** and press **Select** Key.
- Scroll using the **Up/Down Navigation** Key to select **Speed Dial** and press **Select** Key.



- Each Number from 1 to 9, represents the Dial Pad Key of the Phone.



- Scroll using the **Up/Down Navigation** Key to select the desired number (1 to 9) and press **Select** Key.
- The list of contacts appears. Scroll using the **Up/Down Navigation** Key to select the desired contact and press the **Select** Key.

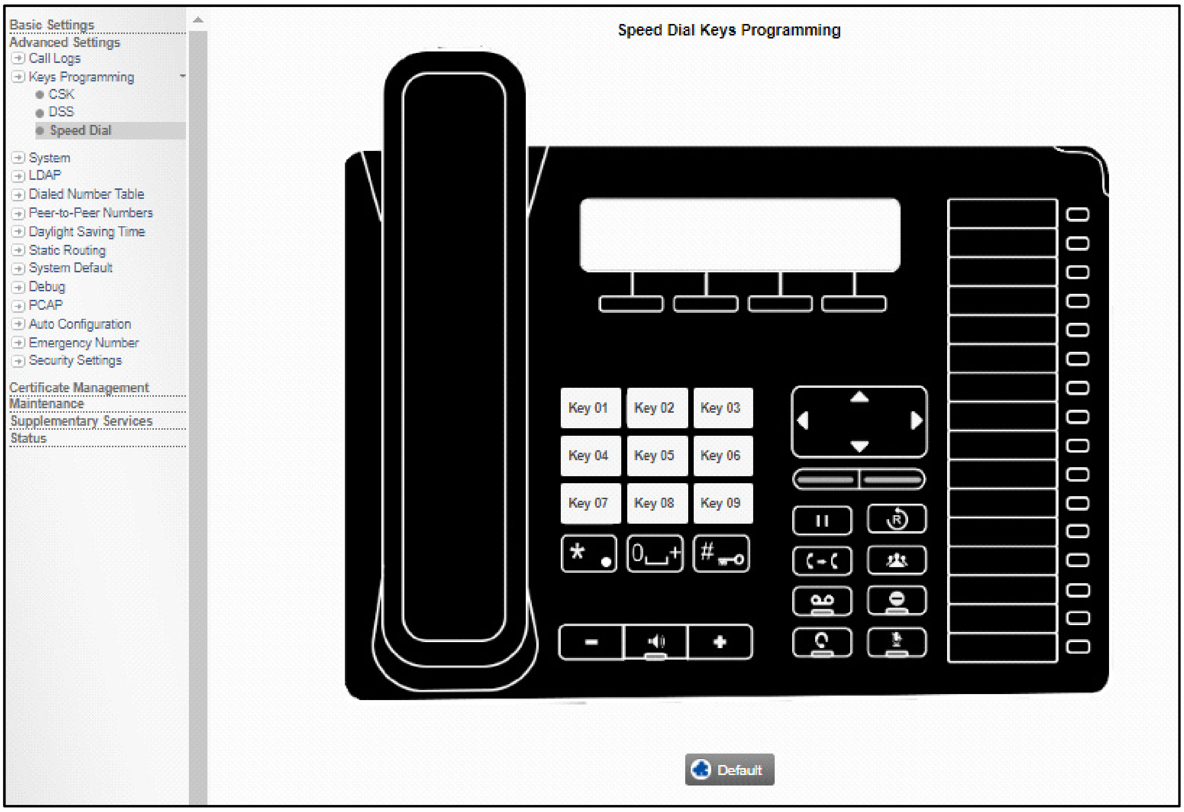
Dialing using Speed Dial Key via Phone User Interface

- Press the Dial Pad digit Key to which the desired contact has been assigned.

- The number of the contact will be out-dialed automatically.

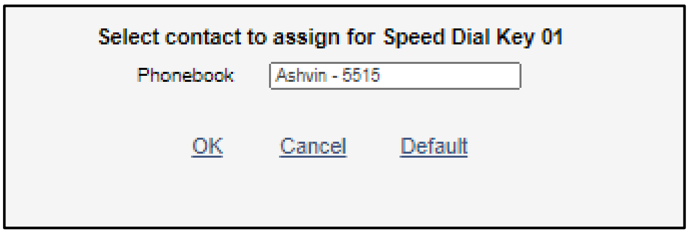
Assigning Speed Dial Keys via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Key Programming.**
- Click **Speed Dial**. The Speed Dial Keys Programming page appears.



- By default, there are no numbers assigned to the Dial Pad Keys. Click the desired Dial Pad Key number (for example Key 01) you wish to assign to a contact.

- The **Select contact to assign for Speed Dial Key 01** page appears.

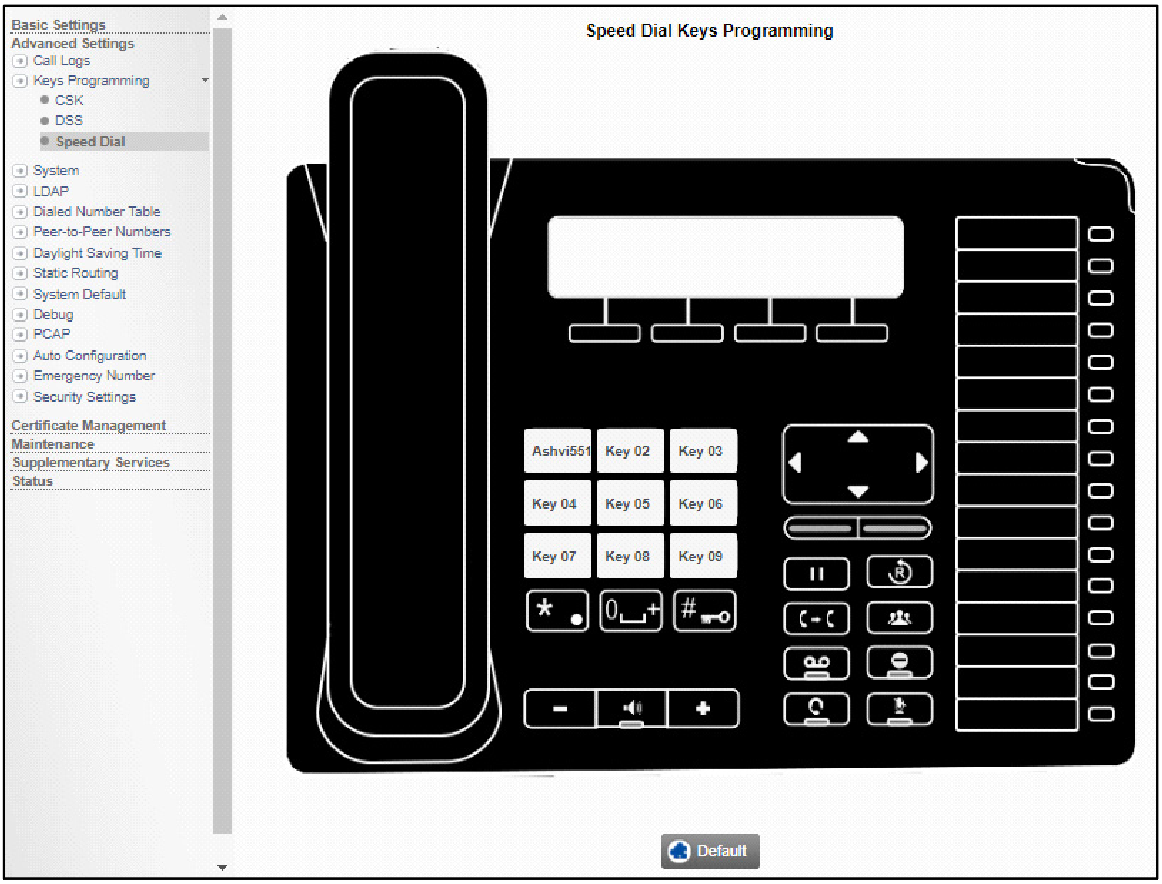


- Click within the **Phonebook** text box. The list of configured contacts appear. Click the desired contact.

OR

- You can also search for the desired contact. To do so,
  - Enter the initial letters of the contact name or number.
  - The list of matching entries appears.
  - Select the desired contact from the list.

- Click **OK** to assign this contact to the Dial Pad Key number.



The label will display the Contacts Name (upto 5 characters) and Number (upto 5 characters). In this case the label of Key 01 is displays Ashvin5515.

- If you wish to assign the default value to the selected Dial Pad Key number,
  - Click the same Dial Pad Key number again.
  - The **Select contact to assign for Speed Dial Digit** page opens.
  - Click **Default**. The selected key is assigned the default value and you return to the **Speed Dial Keys Programming** page.
- If you wish to assign default values to all the Dial Pad Key numbers,
  - Click **Default** on the **Speed Dial Keys Programming** page.

CHAPTER 19

Certificate Management

SPARSH VP510 supports certification for TLS, Web Server, Firmware Upgrade and Configuration Upgrade.

The two types of Certificates supported are: **Self-Signed Certificate** and **CA Signed Certificate**.

Self-Signed Certificate

A self-signed certificate is created by the clients themselves or by the Servers and then given to their clients. It means that you yourself become the Certificate Authority (CA), create a CA Certificate and sign it. The self-signed certificate is faster to create but is not signed by a trusted CA Organization. The self-signed certificate must be installed in the trusted list of clients that connects over TLS with the Server. Because the certificate has been self-signed, the signature is not likely to be in the clients' trust file, hence, they need to add it.

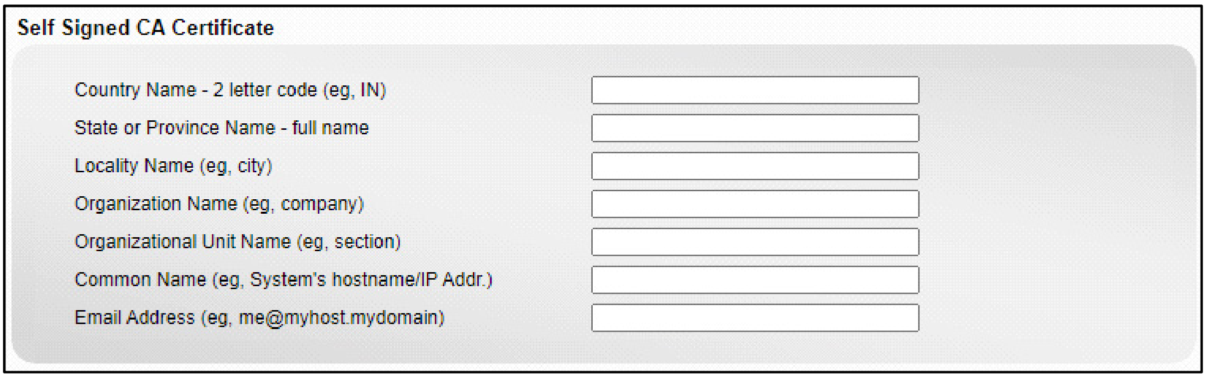
If you select **Self-Signed Certificate**, you need to do the following:

1. Create a Self-Signed CA Certificate.
2. Create a System Certificate (Self-Signed Certificate).

Generating a Self-Signed CA Certificate

- Under **Certificate Management**, click **Generate**.

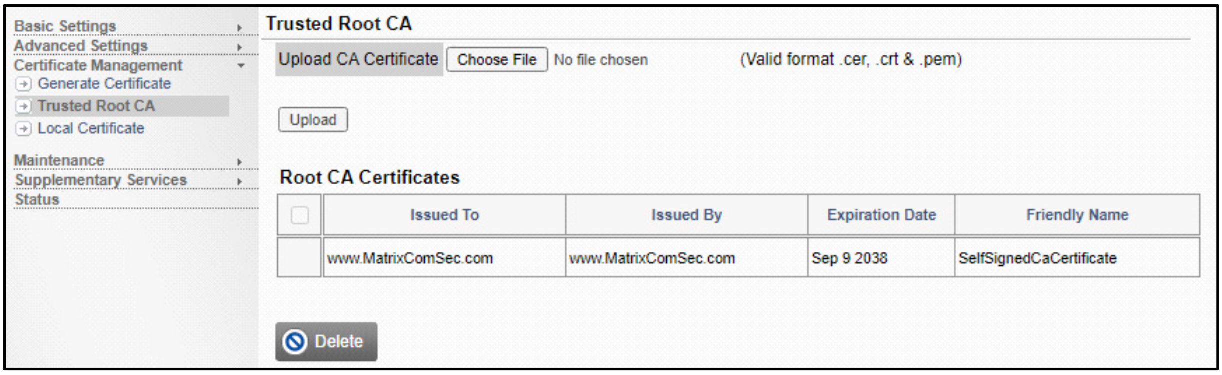
- If you select **Self Signed CA Certificate,** configure the following parameters.



  - In **Country Name - 2 letter code (e.g. IN)**, enter the name of your country.
  - In **State or Province Name - full name**, enter the full name of your state or province.
  - In **Locality Name (e.g. city)**, enter the name of your city.
  - In **Organization Name (e.g. company)**, enter the name of your organization where SPARSH VP510 is installed.
  - In **Organizational Unit Name (e.g. section)**, enter the name of the unit or section or domain of your organization, where SPARSH VP510 is installed.
  - In **Common Name (e.g. System's hostname/IP Addr.)**, enter your SPARSH VP510's host name or IP Address. This Common Name serves as the distinguishing factor.
  - In **Email Address (e.g. me@myhost.mydomain)**, enter your host's e-mail address.
- Click **Generate**, to generate this self-signed CA Certificate.

Once you generate self-signed certificate, you must send it to your clients so that they install it in their trusted list.

- To do this, click **Download.** Save the file at the desired location.
- Under **Certificate Manager**, click the **Trusted Root CA.** The CA Certificate you created appears in the **Root CA Certificate** table.



- If you want to upload other CA Certificates, in **Upload CA Certificate** browse the location at which the certificate is saved and click **Upload.** The CA Certificate you uploaded appears in the **Root CA Certificate** table. Valid format are .cer, .crt and .pem.

- To delete a CA Certificate, select the check box of the respective Root CA Certificate and click **Delete.**

Generating a System Certificate (Self-Signed Certificate)

After creating a Self-Signed CA Certificate, you can either,

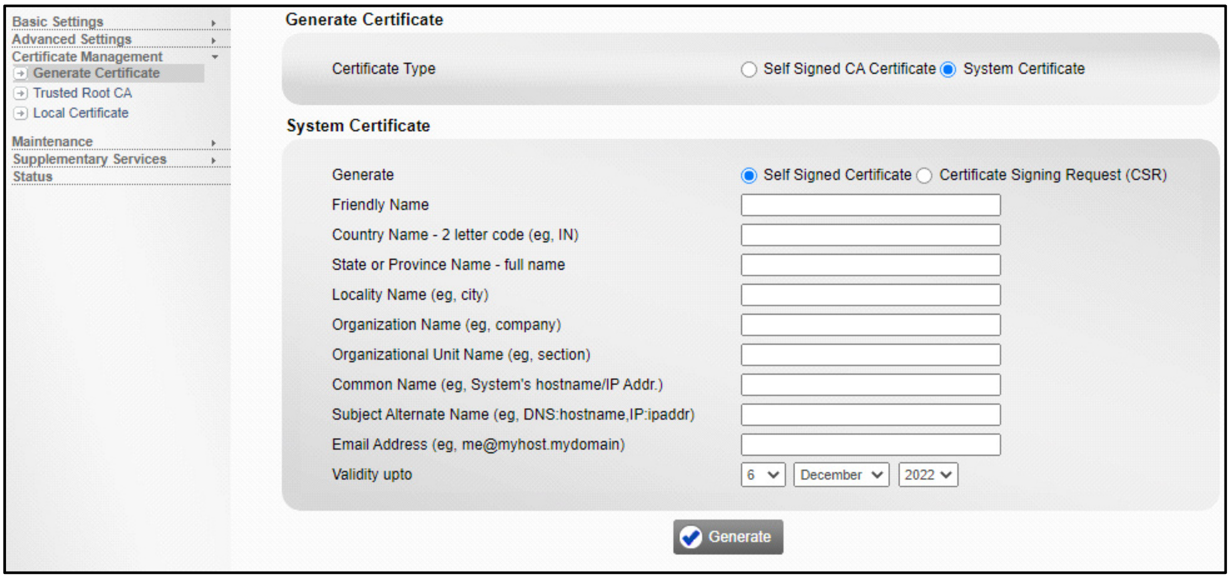
- generate a System Certificate for your clients. These System Certificates can then be given to the respective clients.
  or
- the Clients can prepare their own System Certificates. For this you need to send them the CA Certificate created by you.
  or
  generate a Certificate Signing Request (CSR), if you want the Certificate to be signed by a third party.

If the clients prepare their own certificates, you need to send your CA Certificate to all the clients. The clients must upload the same in their system. Similarly, all the clients must send their CA Certificates to you and you must upload the same in your system. To avoid this, it is recommended that you create the Certificates and then provide it to your clients.

To create the System Certificate,

- Under Certificate Management, click **Generate.**

- If you select **System Certificate**, configure the following parameters.



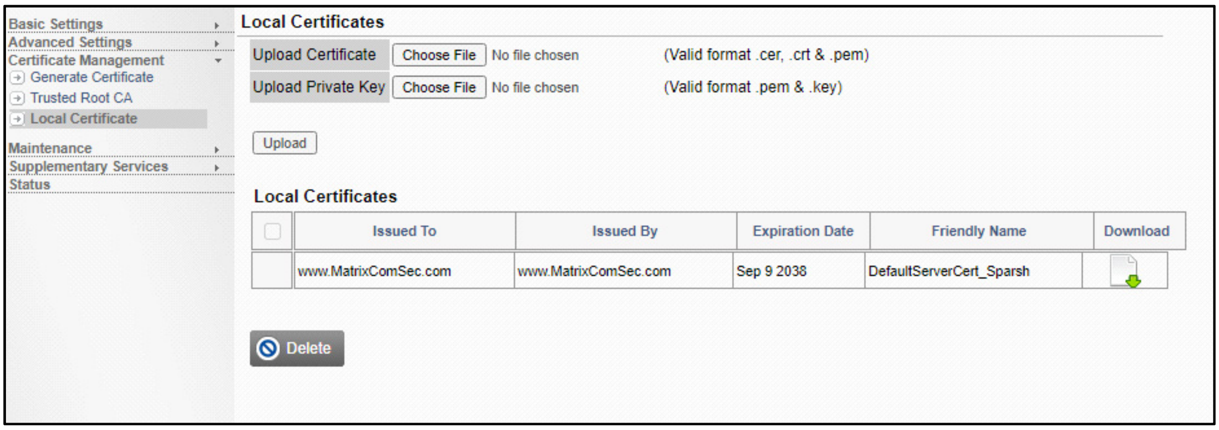
- In **Generate**, select the type of certificate you want to create. You must select **Self-Signed Certificate.**

- In **Friendly Name**, enter the name you want to assign to the certificate.

- In **Country Name - 2 letter code (e.g. IN)**, enter the name (two letter code) of your country.

- In **State or Province Name - full name**, enter the full name of your state or province.
- In **Locality Name (e.g. city)**, enter the name of your city.
- In **Organization Name (e.g. company)**, enter the name of your organization where SPARSH VP510 is installed.
- In **Organizational Unit Name (e.g. section)**, enter the name of the unit or section or domain of your organization, where SPARSH VP510 is installed.
- In **Common Name (e.g. System's hostname/IP Addr.)**, enter your SPARSH VP510’s host name or IP Address. This Common Name serves as the distinguishing factor.
- In **Subject Alternate Name (e.g. DNS:hostname,IP:ipaddr)**, enter the name of the multiple domain separated by comma (if the same certificate is to be issued for multiple domain of the organization).
- In **Email Address**, enter your host’s e-mail address.
- In **Validity Upto**, select the date till which this certificate will be valid.

- Click **Generate**, to generate this System Certificate.
- Under **Certificate Management**, click **Local Certificate**. The generated certificate appears in the **Local Certificates** table.



- If you want to upload other System Certificates, in **Upload Certificate** browse the location at which the certificate is saved. Along with the certificate you also need to upload the Private Key, in **Upload Private Key** browse the location at which the key is saved and click **Upload**.

  The System Certificate you uploaded appears in the **Local Certificates** table.Valid formats for certificate are .cer, .crt and .pem. Valid format for key are .pem and .key (Base64 encoded ASCII file).

- To delete a System Certificate, select the check box of the respective Certificate and click **Delete**.
- To download the System Certificate, click **Download** .

CA Signed Certificate

Certificate Authority (CA) is a trusted organization which creates and sells TLS Certificates to websites. *CA Signed Certificates* are the TLS Certificates which are created by such trusted CAs, signed and sold to any applicant. These certificates contain a public key and the identity of the owner; and it is upto the CA to verify the owner's (applicant's) credentials. 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 certain period of time.

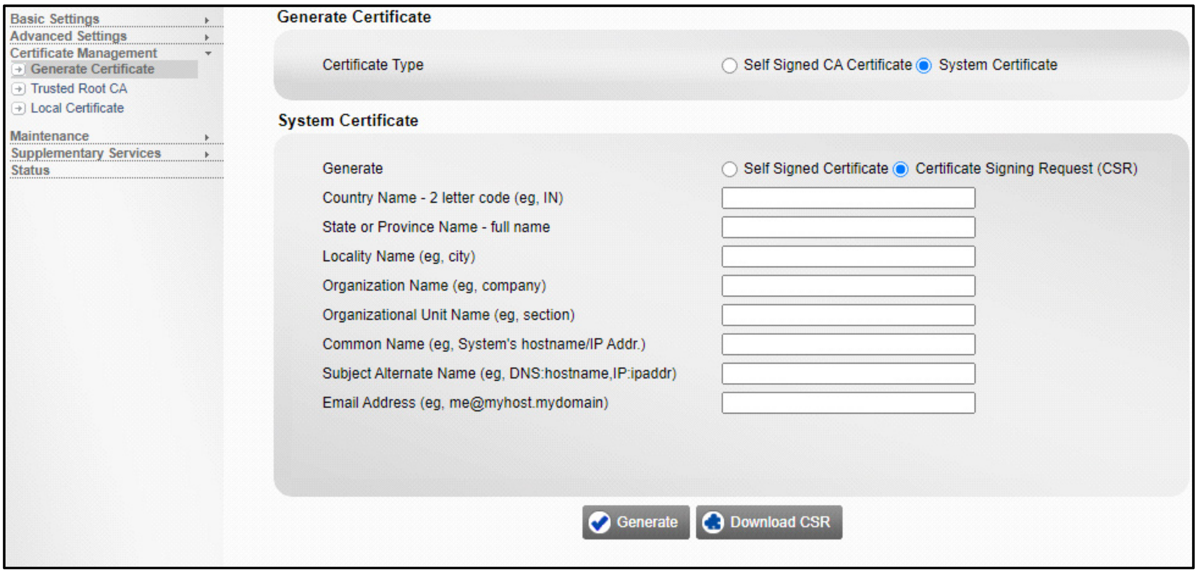
If you want to get a **CA Signed Certificate**, you need to do the following:

1. Generate and enroll the Certificate Signing Request (CSR).
2. Get the Certificate Signing Request (CSR) verified and signed by the Certified Authority (CA).

Generating the Certificate Signing Request

- Under **Certificate Management**, click **Generate.**

- Select **System Certificate** and configure the following parameters.



- In **Generate**, select the type of certificate you want to create. You must select **Certificate Signing Request (CSR)**.

- In **Country Name - 2 letter code (e.g. IN)**, enter the name (two letter code) of your country.

- In **State or Province Name - full name**, enter the full name of your state or province.

- In **Locality Name (e.g. city)**, enter the name of your city.

- In **Organization Name (e.g. company)**, enter the name of your organization where SPARSH VP510 is installed.

- In **Organizational Unit Name (e.g. section)**, enter the name of the unit or section or domain of your organization, where your SPARSH VP510 is installed.

- In **Common Name (e.g. System's hostname/IP Addr.)**, enter your SPARSH VP510's host name or IP Address. This Common Name serves as the distinguishing factor.

- In **Subject Alternate Name (e.g. DNS:hostname,IP:ipaddr)**, enter the name of the multiple domain separated by comma (if the same certificate is to be issued for multiple domain of the organization).

- In **Email Address (e.g. me@myhost.mydomain)**, enter your host's e-mail address.

- Click **Generate**, to generate this System Certificate.

- To send the certificate to the signing authority, click **Download CSR**. The Certificate and the Key downloads.

Enrolling the Certificate Signing Request with CA

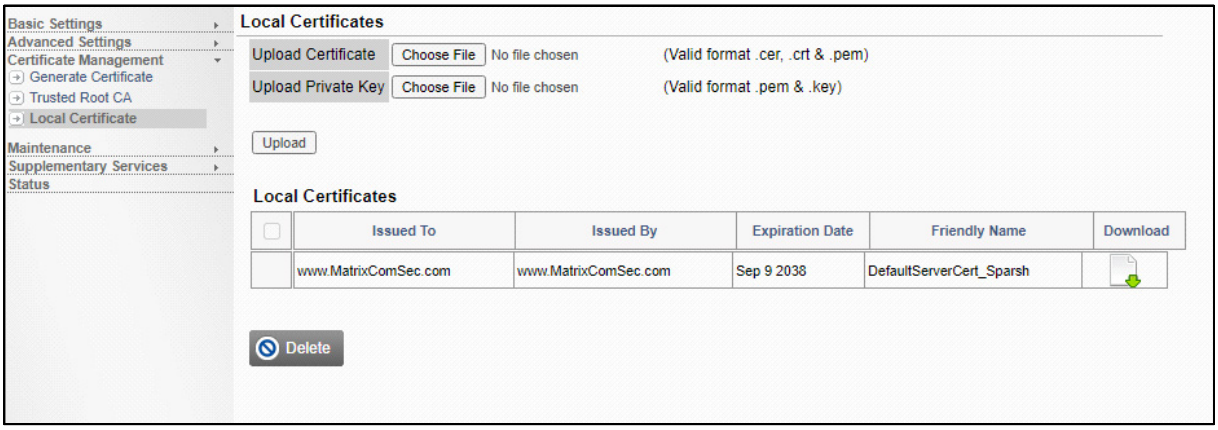
Enrollment is a process of obtaining a certificate from any trusted third party (CA). After you have generated the Certificate Signing Request (CSR), you must contact any authorized third party that issues TLS Certificates to companies or web owners, such as Thawte, VeriSign, etc. and enroll the Certificate 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 year. After the Certificate Signing Request (CSR) has been validated and signed by the CA, it becomes the CA Signed Certificate.

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.

After you receive the signed certificate, you must:

- Under **Certificate Management,** click **Local Certificate**.



- In **Upload Certificate** browse the location at which the certificate is saved. Along with the certificate you also need to upload the Private Key, in **Upload Private Key** browse the location at which the key is saved and click **Upload**

  The System Certificate you uploaded appears in the **Local Certificates** table.Valid formats for certificate are .cer, .crt and .pem. Valid format for key are .pem and .key (Base64 encoded ASCII file).

  To delete a System Certificate, select the check box of the respective Certificate and click **Delete**.

To download the System Certificate, click **Download** .

CHAPTER 20 *Maintenance*

Firmware Upgrade

You can upgrade the firmware of SPARSH VP510

- From a Personal Computer, that is Manual Upgrade
- From the Auto Configuration Server, that is Auto Upgrade

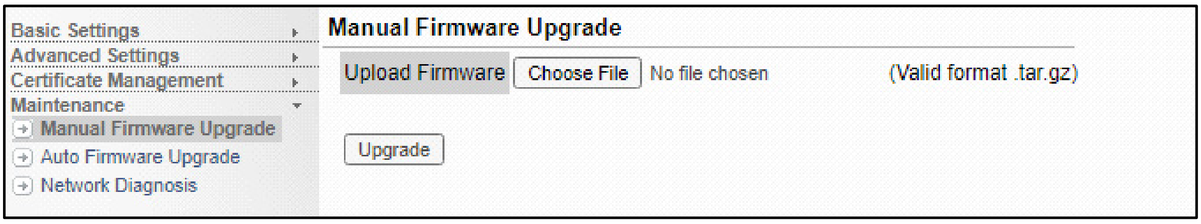
Configuring Firmware Upgrade parameters via Web User interface

Manual Firmware Upgrade

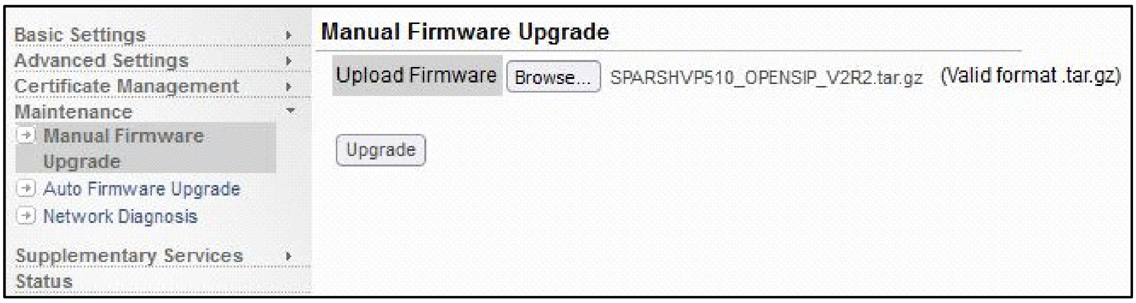
For the Firmware file contact our Technical Support Team.

You can upgrade firmware of SPARSH VP510 with the firmware files stored on your computer. To do so,

- Log into Jeeves.
- Under **Maintenance**, click **Manual Firmware Upgrade.**



- Click the **Browse** to reach the location on the local disk on which the firmware files are stored. Select the required firmware files from the location on the local disk. The file with .tar or .gz extension with maximum file size as 10 MB can be uploaded.



- The path to the file appears besides **Upgrade Firmware**.

- Click **Upgrade**.

Auto Firmware Upgrade

Using Auto Firmware Upgrade, SPARSH VP510 can automatically download the firmware files stored at a central location: Provisional Server.

This feature is useful for ITSPs that have Provisioning Servers to store the firmware files. ITSPs can update the firmware of SPARSH VP510 provided to their customers from a centralized location without physically visiting the customer premises.

For the Firmware as well as HTML (matrix_firmware.html file) file, contact our Technical Support Team. The Technical Support Team will provide the firmware file with the name as: SPARSHVP510_OPENSIP_VxxRyy.tar.gz

To perform Auto-Firmware Upgrade,

- ITSPs must store the following Auto Firmware Upgrade files of SAPRSH VP510 on the Provisioning Server.

  - matrix_firmware.html file
    Open the html file and replace the name as SPARSHVP510_VxxRyy. Mention the actual xx and yy, for example 01 and yy as 03. That is, the name will be SPARSH VP510_V01R03.

  - Rename the firmware file provided by Support as SPARSHVP510_VxxRyy.tar.gz

  - Make sure both the firmware file and html files are placed in the same folder.

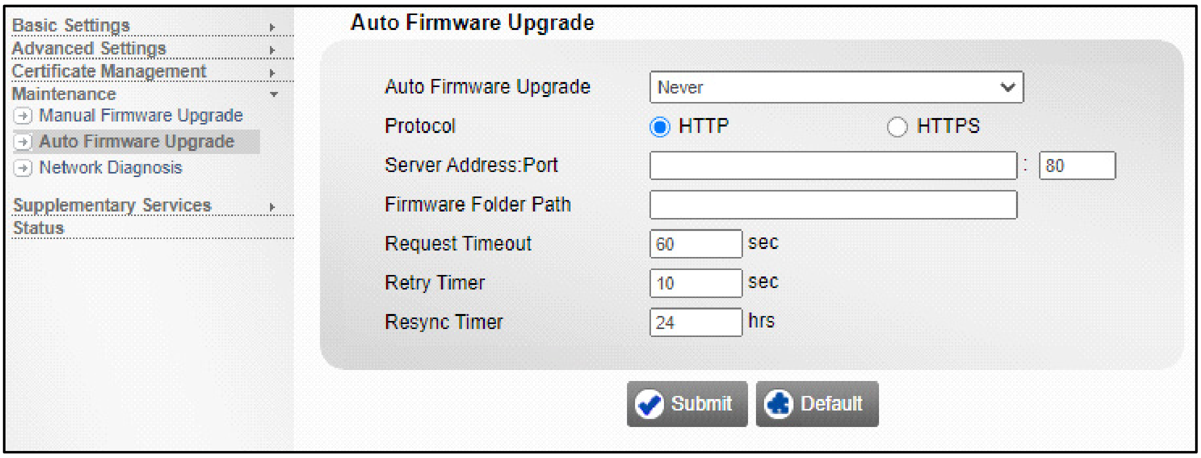
- The following parameters must be configured in the SPARSH VP510.
  - IP Address of the Provisional Server.
  - Path of the Folder (containing the configuration file as well as html file) on the Provisional Server.
  - The protocol to be used: HTTP, HTTPS.

- When SPARSH VP510 installed at a customer site gets connected to the ITSP network, it will automatically compare its current firmware with the firmware files stored on the Provisioning Server. The matrix_firmware.html file helps SPARSH VP510 decide which firmware it should upgrade to.

- After SPARSH VP510 decides the Firmware Version/Revision to upgrade to, it will send the request for the firmware files to the Provisioning Server. Once the respective firmware files are received, SPARSH VP510 will upgrade its current firmware with the new firmware without the intervention or assistance of a technician.

To configure Auto Firmware Upgrade parameters,

- Log into Jeeves.
- Under **Maintenance**, click **Auto Firmware Upgrade**.



- Select the desired **Auto Firmware Upgrade** option — At Each Power On, At Next Power On, Resync Periodically, At Each Power On and Resync Periodically, At Next Power On and Resync Periodically. Default: Never.
  - **At Each Power ON:** Select this option if you want SPARSH VP510 to check for updates in the firmware at each power on.
  - **At Next Power ON:** Select this option if you want SPARSH VP510 to check for updates in the firmware during the next power on. Once Auto Upgrade is done successfully, the option gets changed to **Never**.
  - **Resync Periodically:** Select this option if you want SPARSH VP510 to resynchronize periodically with the Provisional Server to check for the updates. If you select this option, Auto Upgrade will be performed every time the Resync Timer expires.
  - **At Each Power ON & Resync Periodically:** Select this option if you want SPARSH VP510 to perform the Auto Upgrade during each power on and also to resynchronize periodically with the Provisional Server to check for the updates every time the Resync Timer expires.
  - **At Next Power ON & Resync Periodically:** Select this option if you want SPARSH VP510 to update its configuration during the next power on, and also want SPARSH VP510 to resynchronize periodically with the Provisional Server to check for the updates on expiry of the Resync Timer. Once Auto Upgrade is done successfully at next power on, the option gets changed to **Resync Periodically**.
- Select the desired **Protocol** to be used by the Provisional Server to upgrade the firmware. SPARSH VP510 generates the request to the Provisional Server according to the protocol you select. You may select **HTTP** or **HTTPS**. Default: HTTP.

- **Server Address: Port:** Enter the IP Address/Domain and the Port of the Provisional Server on which the firmware files of SPARSH VP510 are stored.

  The default Port differs as per the protocol you select. For HTTP, the Default Port is 80. For HTTPS, the Default Port is 443. You can change the port as per your requirement. Valid Port Range: 1024 to 65535.

- **Firmware Folder Path:** Specify the path of the folder on the Provisional Server where the firmware file is stored. Default: Blank.

- Configure the **Request Timeout.** this is the time for which SPARSH VP510 will try to connect to the Provisional Server to fetch the firmware using HTTP or HTTPS. This timer specifies for how long SPARSH VP510 should wait for successful binding.

  Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

- Configure the **Retry Timer**. If SPARSH VP510 does not get any response from the Provisional Server, it will wait for the duration of the Retry Timer, before sending the firmware request again.The range of Retry Timer is 001-999 seconds. Default: 10 seconds.

- Configure the **Resync Timer**. This is timer after which SPARSH VP510 will generate new firmware request to the Provisional Server.The range of Resync Timer is 001-999 hours. Default: 24 hours.

- Click **Submit** to save.

You may also view the status of Auto Upgrade from the Web User Interface. Refer “Status”.

Auto Configuration

This feature allows you to configure SPARSH VP510 from a centralized location/server, also known as the **Auto Configuration Server (ACS)**, which is usually maintained by the ITSP.

This feature is useful for the ITSPs that have deployed a large number of SPARSH VP510. ITSPs can store the configuration files of each SPARSH VP510 on the Auto Configuration Server (ACS) by a specific name, for example, **matMACAddress.xml** where 'mat' is a fixed string and 'MACAddress' is the MAC Address of the phone. For the **Auto Configuration File** contact the Matrix Support Team. ITSPs can also use this feature to upgrade the software of the phones.

When the SPARSH VP510 connects to the ITSP network, it will automatically download its configuration file stored on the ACS. The customer can thus start using the SPARSH VP510 without the intervention or assistance of a technician, making it a 'plug-and-play' device.

To ensure security, the ITSP can also encrypt the configuration file stored on the ACS. In this case, the password to decrypt the file must be provided to the customer.

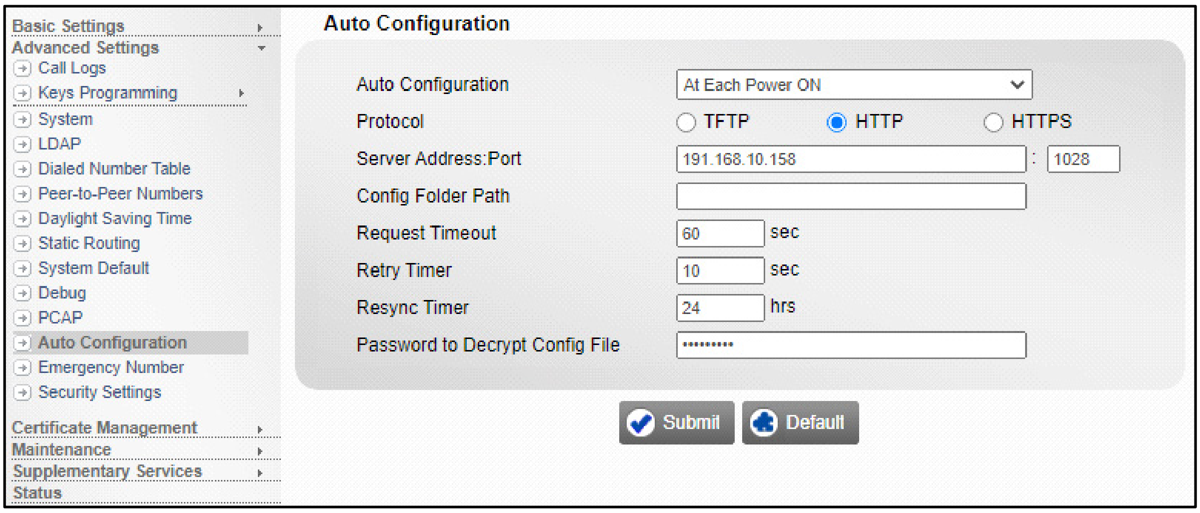
To use this feature, you must obtain following information from the ITSP,

- IP address/Domain Name of the Auto Configuration Server.
- Path of the Folder (containing the configuration file) on the Auto Configuration Server.
- Password to decrypt the configuration file (if encryption is used).
- The protocol to be used: TFTP, HTTP or HTTPS.

Configuring Auto Configuration parameters via Web User interface

- Log into Jeeves.

- Under **Advanced Settings**, click **Auto Configuration**.



- Select the desired **Auto Configuration** method to be used to fetch the configuration file. Default: Never.

  - **Never:** Select this option if you do not want SPARSH VP510 to check for updates in the configuration.

- **At Each Power ON:** Select this option if you want SPARSH VP510 to check for updates in the configuration at each power on.

- **At Next Power ON:** Select this option if you want SPARSH VP510 to check for updates in the configuration during the next power on. Once Auto-Configuration is done successfully, the option gets changed to **Never**.

- **Resync Periodically:** Select this option if you want SPARSH VP510 to resynchronize periodically with the ACS to check for the updates. If you select this option, Auto-Configuration will be performed every time the Resync Timer expires.

- **At Each Power ON & Resync Periodically:** Select this option if you want SPARSH VP510 to perform the Auto-configuration during each power on and also to resynchronize periodically with the ACS to check for the updates every time the Resync Timer expires.

- **At Next Power ON & Resync Periodically:** Select this option if you want SPARSH VP510 to update its configuration during the next power on, and also want SPARSH VP510 to resynchronize periodically with the ACS to check for the updates on expiry of the Resync Timer. Once Auto-Configuration is done successfully at next power on, the option gets changed to **Resync Periodically**.

- Select the **Protocol** used by the Auto Configuration Server to upgrade the configuration. SPARSH VP510 generates file transfer request to the configuration server according to the protocol you select. You may select **TFTP**, **HTTP or HTTPS**. Default: HTTP.

- Specify the Server Address:Port of the Auto Configuration Server. Default: Blank.

  The Auto Configuration Server Address is the IP Address/Domain Name of the ACS where the configuration file of SPARSH VP510 is stored.

  The server address needs to be configured manually, enter Auto Configuration Server's IP address from which SPARSH VP510 should download the configuration file. Default: Blank.

  The default Port differs as per the protocol type you select. For TFTP, the Default Port is 69, for HTTP, the Default Port is 80. and for HTTPS, the Default port is 443. You can change the port as per your requirement. Valid Port Range: 1024 to 65535.

- In **Config Folder Path**, specify the path of the folder on the Auto Configuration Server from where the configuration files are to be downloaded. Default: Blank.

- Configure the **Request Timeout.** Request Timeout is the time for which the SPARSH VP510 tries to connect to the Auto Configuration Server for TCP/TLS binding.

  Request Timeout is applicable only for HTTP/HTTPS as TFTP works on UDP and no connection is established with ACS in case of TFTP.

  Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

- Configure the **Retry Timer**. This timer is used if the connection is not established with Auto Configuration Server before the expiry of the Request Timeout. In such a case, SPARSH VP510 will try to reconnect automatically after this time interval.

Enter the required time interval in seconds. The range of Retry Timer is 001-999 seconds. Default: 10 seconds.

- Configure the **Resync Timer**. Resync Timer is the time after expiry of which SPARSH VP510 will send the resync request to check if there is any change in the configuration files. This timer is applicable only when you select one of these Auto-Configuration options: **Resync Periodically** or **At Each Power ON & Resync Periodically** or **At Next Power ON & Resync Periodically**.

  Enter the required Resync Timer in hours. The range of Resync Timer is 001-999 hours. Default: 24 hours.

- Enter the **Password to Decrypt Config File** as provided by your ITSP. During Auto Configuration, if SPARSH VP510 receives an encrypted configuration file, it will decrypt the file using this password.

  The password may consist of 40 characters (maximum). Default: matrix123

The password is case-sensitive, make sure you enter the password in the same format as given to you by your ITSP.

- Click **Submit** to save.

- If you want to assign factory set values to all the parameters on this page, click **Default**.

You may also view the status of Auto Configuration from the Web User Interface. Refer “System”.

Debug

Debugs are logs of actions and events that take place on any system. These logs are useful for troubleshooting and system security.

SPARSH VP510 supports Syslog[3] Client for debugging. Syslog Client enables the phone to send debug messages in syslog format to the remote 'Syslog Server' on IP network. You can view the debug messages on the remote server.

The phone also supports multiple debug levels, which include:

- SIP
- System
- Call
- User Interface
- Network
- Media
- Vopp
- Communication

You can select any of these debug levels, and 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.

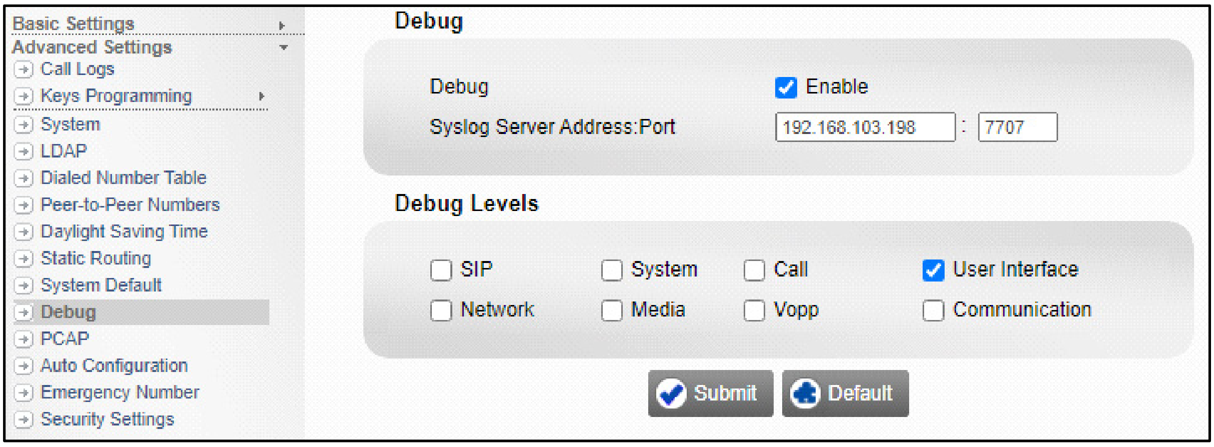
To be able to use this feature, you must enable Syslog, configure the Syslog (Remote) Server Address and define the Server Port on which Syslog will listen for debug messages.

Syslog uses the UDP as transport protocol and listens on the port 514 (the default listening port).

Configuring Debug parameters via Web User interface

- Log into Jeeves.

- Under **Advanced Settings**, click **Debug**.



3. *Syslog is one of the protocols used extensively for sending debug messages, and is defined in RFC 3164.*

- Configure the following parameters:
  - Select the **Debug** check box to enable. 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.

  - In **Syslog Server Address:Port,** enter the IP Address of the Syslog Server. A maximum of 15 digits, including 0-9 and '.' (dot) are allowed.

    Enter the address of the Listening Port of the Syslog Server from 1024-65535; 514. By default the remote server port address is 514.

- Under **Debug Levels**, select the check boxes of the desired debug levels— SIP, System, Call, User Interface, Network, Media, Vopp, Communication — to enable. The phone will send debug messages only for the level you have selected. By default, all debug levels are selected. Clear the check box to disable for debug level you do not want.
- Click **Submit** to save.

PCAP

PCAP or packet capture consists of intercepting and logging the traffic passing over a digital network or a part of 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, analyze network problems, debug client/server communications, debug network protocol implementations.

SPARSH VP510 supports PCAP Trace, which you can use to detect and diagnose network related problems, for example, when the SIP trunk is not getting registered, or any SIP related feature is not functioning.

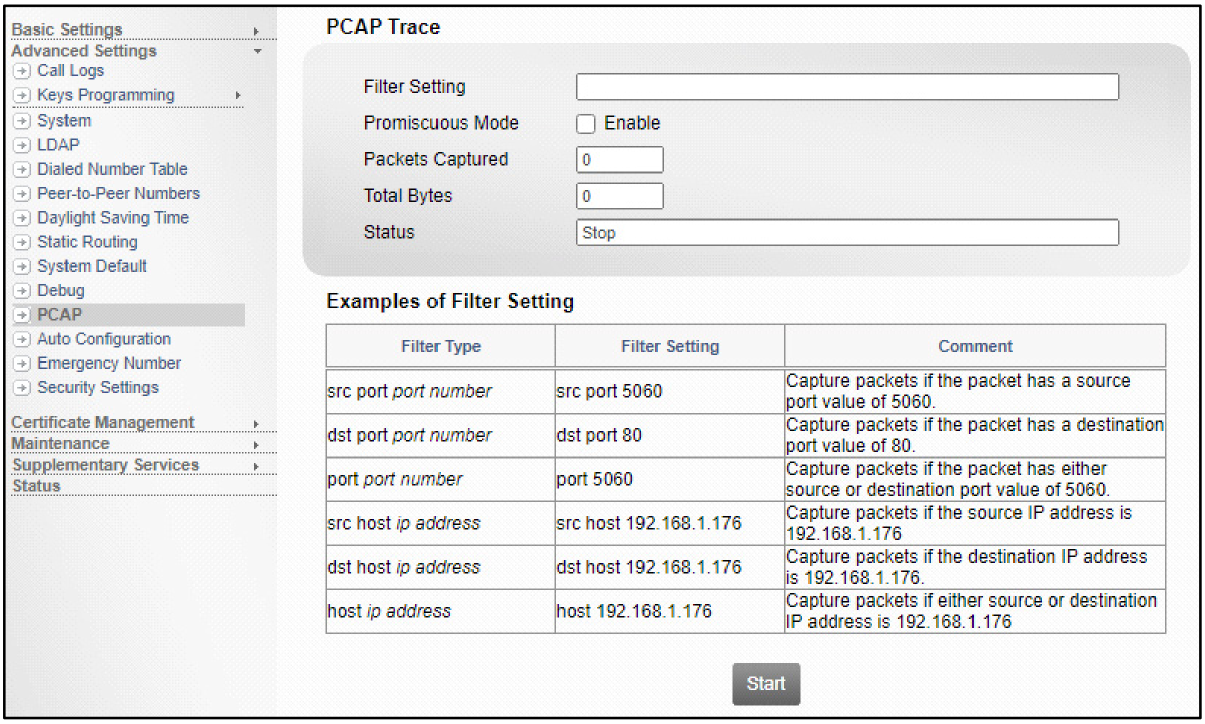
Packets traveling over a network are captured and saved in the IP Phone. You can save these trace files (packets captured by the phone) on a PC and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

A maximum of 1 MB of packets can be captured and stored in the IP Phone.

SPARSH VP510 also supports Filters and 'Promiscuous' mode for capturing packets, which you can use to specify the types of data packets to be captured.

Configuring PCAP via Web User interface

- Log into Jeeves.
- Under **Advanced Settings**, click **PCAP**.



- Decide the type of packets to be captured and set the Filter accordingly. The Filter Settings parameter should be maximum 60 characters in length. Default: Blank. So all packets will be captured.

Refer to the following examples to know how to set the Filters.

Examples of Filter settings:

- To capture only SIP traces:
  - ***Filter Settings = port 5060***
    where, 5060 is the SIP Port number for which the traces are to be captured.

- To capture packets which are transmitted from the phone, i.e. from IP address 192.168.1.181:
  - ***Filter Settings = src 192.168.1.181***

- To capture packets which are received for the phone, i.e. to IP address 192.168.1.181:
  - ***Filter Settings = dst 192.168.1.181***

- To capture packets which are transmitted from the phone and received by the phone i.e. IP address 192.168.1.181:
  - ***Filter Settings = src 192.168.1.181 or dst 192.168.1.181***

- Select the **Promiscuous Mode** check box to enable, if required. Default: Disabled.

  When you enable Promiscuous mode, the SPARSH VP510 will capture all network traffic. However, this will work only in a non-switched environment.

  When Promiscuous Mode is disabled, the phone will capture only traffic that is directly related to it. Only traffic to, from or routed through the SPARSH VP510 will be picked up by the PCAP Trace.

'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

- Click **Start** to begin the capturing of the packets.

- Click **Stop** to stop the packet capturing.

  OR

  Wait for the phone to stop packet capturing. The phone stops packet capturing once the maximum allotted memory of 1 MB (RAM) is utilized.

- The **Status** displays the current activity of packet capturing. It will display the status messages received from the PCAP Library such as, running, off, parsing error etc.

  Number of Packets and bytes captured as per the filter setting will be displayed as **Packets Captured** and **Total Bytes**.

- When the packet capturing is stopped (by you or the phone), click **Save Trace File** to save the files on your PC or another PC.

  A dialog box will open. You can select the path for saving the trace file.

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.

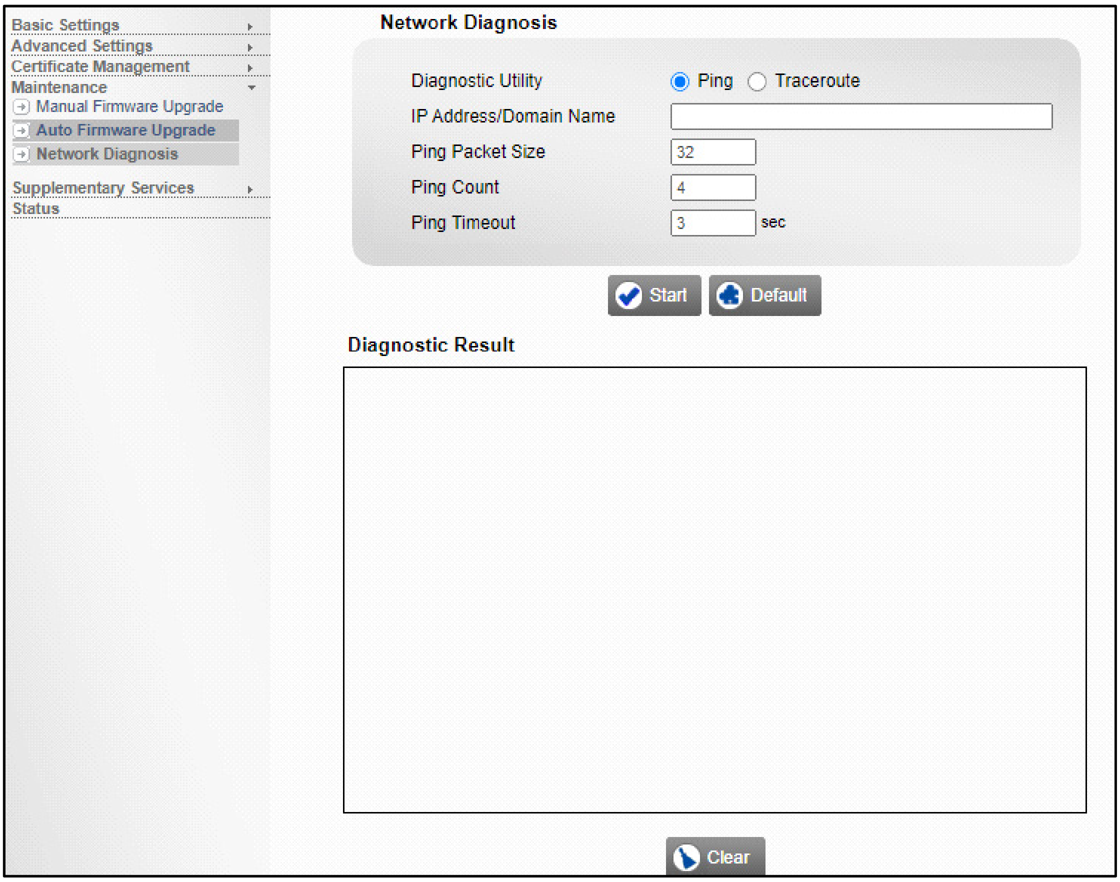
- Now, you can open the trace files using Wireshark/Ethereal or any other similar software which supports opening of trace files.

Network Diagnosis

SPARSH VP510 provides you an option to check the Internet/WAN connectivity using Ping and Traceroute as the diagnostic tools.

Configuring Network Diagnosis parameters via Web User interface

- Log into Jeeves.
- 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**.

  The Diagnostic result will appear on the screen.

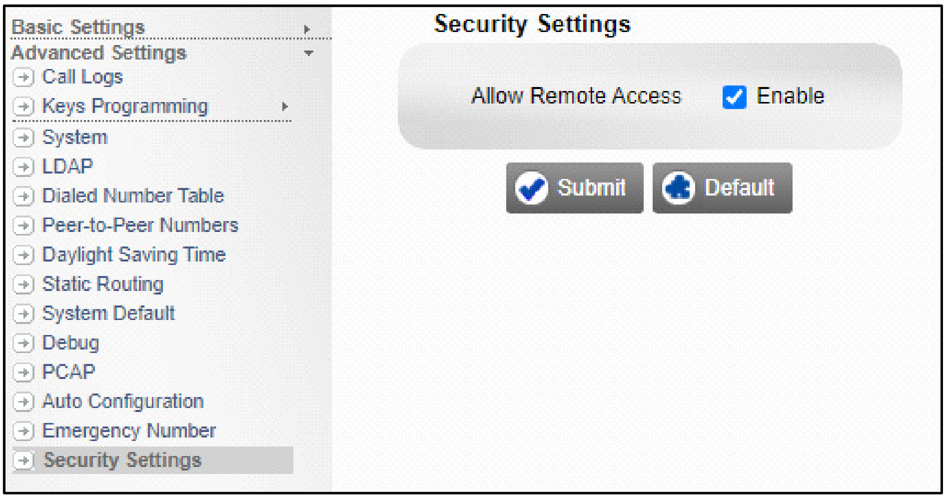
- To clear the Diagnostic result, click **Clear**.

Security Settings

Security Settings allows you to access SPARSH VP510 remotely.

Configuring Security Settings via Web User Interface

- Log into Jeeves.
- Under **Advanced Settings**, click **Security Settings**.



- Select the **Allow Remote Access** check box, if you wish to allow remote access of the Phone.Default: Disabled.
- Click **Submit**.

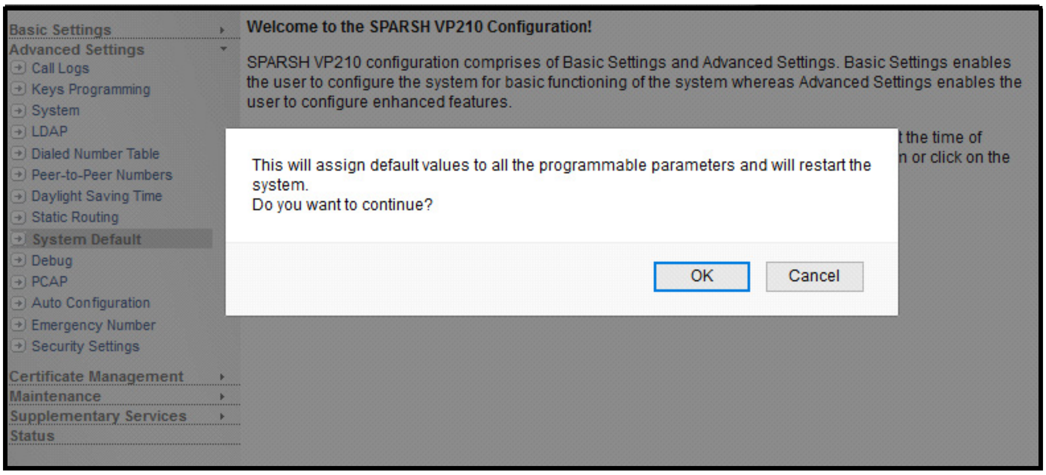
System Default

When you restore default settings using the Web User Interface, all the parameters will be assigned default values except the following:

- Network Parameters
- Password (User Password and Configuration Password)

Setting the System to Default via Web User interface

- Log into Jeeves.
- Under **Advanced Settings**, click **System Default.**
- The following alert message will pop-up:



- Click **OK** to restore the default factory settings.
- Click **Cancel** if you want to continue with the current settings of the phone.

When you default the IP phone, you will lose all the saved information like your Phone Book, Call Logs, Peer-to-Peer table, Do Not Disturb, Voice Mail Addresses, Network and SIP configuration settings.

So, exercise great caution. Use this feature only if you want to reconfigure the IP phone entirely. For instance, when the phone is to be installed at a new location/allocated to another user, or while trouble shooting.

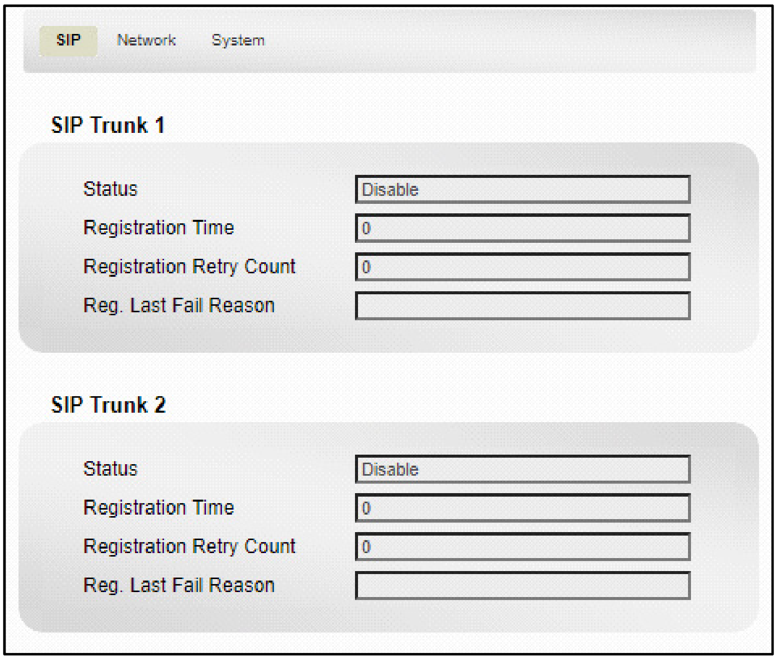
CHAPTER 21 **Status**

You can view the status of the SIP Trunks, Network as well as the System, Network.

Viewing Status via Web User Interface

- Log into Jeeves.
- Click **Status.**
- Click the respective tab — SIP, Network, System.

SIP



The following parameters will be displayed for SIP1 and SIP2.

- **Status:** In this field, different status may be displayed, each of which are described briefly in the table below.

Status	Description
Disable	Shows that SIP Trunk is disabled.
Registering	Shows that SIP Trunk is in registering state.
Active	Shows that SIP Trunk is registered with the SIP server.
Failed	When trunk status is failed.
Invalid	Other reasons.

- **Registration Time:** The SIP Trunk is registered with the Registrar Server for a particular time period, after which it has to be re-registered. The registrar server indicates the time remaining for re-registration of the SIP Trunk. The same is displayed in this field as Registration Time.

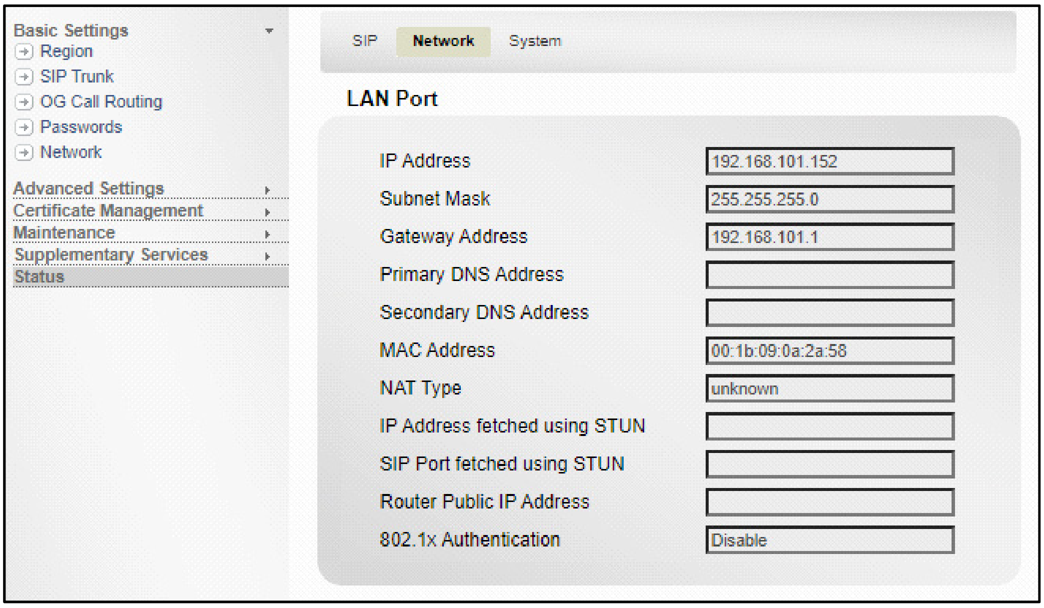
- **Registration Retry Count:** This field displays the total number of register messages which are sent to the registrar server for registering SIP Trunk.

- **Reg. Last Fail Reason:** This field displays the reason for failure of SIP Trunk registration with the registrar server. The different reasons for registration failure that may appear in this field are:

Reason for Failure	Description
Message Send Fail	This reason is displayed when registration request sent to registrar server fails.
Failed to create Register Client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ the number of SIP clients to register is greater than the number programmed in the stack.
Failed to Detach Register Client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ the number of SIP clients to register is greater than the number programmed in the stack.
Failed To send Request	This reason is displayed when DNS server is not programmed.
Local Failure	This reason is displayed when DNS query fails.
Response Timeout	This reason is displayed on the expiry of the General Request Timer.
Error Response	This is error response code.
No Contact Header in 2xx	This reason is displayed when no contact address is received in 2xx response from the SIP server.
Authentication Fail	This reason is displayed when the SIP server does not authenticate the client.
STUN Address is not Programmed	This reason is displayed when STUN is enabled but address is not configured.
STUN Query Fail	This reason is displayed when a query to the STUN server fails.
Outbound address is not programmed	This reason is displayed when Outbound is enabled but Outbound address is not configured.
Router IP Address is not Programmed	This reason is displayed when Router's IP Address is to be used in signaling but the address is not programmed.
TCP Transport Disabled	TCP mode is disabled

Reason for Failure	Description
TLS Transport not initialized	TLS is not initialized
Invalid	Other reasons

Network



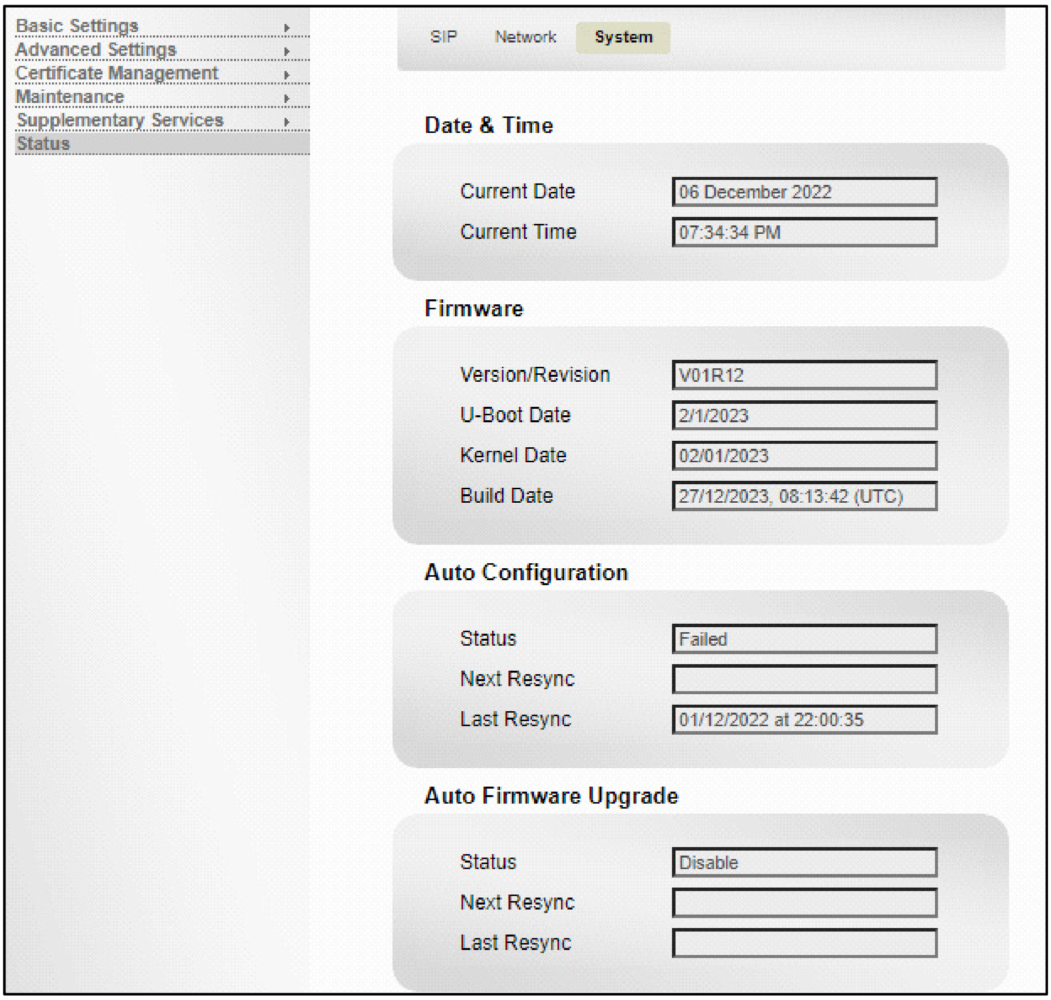
LAN Port

- **IP Address:** This field displays the IP Address of the LAN port.
- **Subnet Mask:** This field displays the Subnet Mask Address of the LAN port.
- **Gateway Address:** This field displays Gateway Address assigned to the LAN Port of SPARSH VP510. This field displays the IP Address of the LAN port of the Router.
- **Primary DNS Address:** This field displays the Primary DNS address of SPARSH VP510.
- **Secondary DNS Address:** This field displays the Secondary DNS address of SPARSH VP510.
- **MAC Address:** This field displays MAC Address assigned to the LAN Port of SPARSH VP510.
- **NAT Type:** This field displays the NAT Type, if STUN is enabled in SPARSH VP510. Commonly used NAT types are:
  - Unknown
  - Open
  - Conenat
  - Restrictednat
  - Portrestrictednat
  - Symmetricnat
  - Symmetricfirewall
  - Blocked

- **IP Address fetched using STUN:** This field displays the IP address fetched using STUN, if STUN server address is configured.

- **SIP Port fetched using STUN:** This field displays the SIP port fetched using STUN if STUN server address is configured.
- **Router Public IP Address:** This field displays the Routers Public IP Address if configured.

System



Date &Time

- **Current Date:** Current Date is displayed in this field.

.

- **Current Time:** Current Time is displayed in this field.

Firmware’

- **Version/Revision:** Current Software Version-Revision of SPARSH VP510 is displayed in this field.

.

- ***U-Boot Date:*** Current U-Boot date is displayed in this field.

- **Kernel Date:** Date of current Kernel of the phone is displayed in this field.

Auto Configuration

- **Status:** Displays the different status of Auto Configuration, each of which are described briefly in the table below.

Status	Description
Disable	This reason is displayed when Auto Configuration type is set as Never.
Failed	This reason is displayed when Auto Configuration is not successful.
Failed to Connect to Host	This reason is displayed when Auto Configuration server is not reacheable or is in different subnet.
Syncing files	This reason is displayed when Auto Configuration process is running and the system is receiving files from the server.
File Not Found	This reason is displayed when the Configuration file is not present on the server or path for the file is invalid.
Parsing File Failed	This reason is displayed when Auto Configuration file is received from the server but parsing fails.
Password Decrypt Failed	This reason is displayed when the encrypted file is received from the server but phone fails to decrypt it.
Server Not Found	This reason is displayed when Server IP Address is not reacheable.
Unknown Error	This reason is displayed when there is an internal application error.
Connecting	Auto Configuration status connecting
Partial	Auto Configuration status partial
Success	Auto Configuration status success
Version Revision Same	Auto Configuration status file version revision same
Downgrade Not Allowed	Auto Configuration status downgrade not allowed
Invalid	Other reasons

- **Next Resync:** Date and time at which the phone will again resync with the server is displayed in this field.
- **Last Resync:** Date and time at which the phone had last resynchronized with the server is displayed in this field.

Auto Firmware Upgrade

- **Status:** Displays the different status of Auto Firmware Upgrade, each of which are described briefly in the table below.

Status	Description
Disable	This reason is displayed when Auto Firmware Upgrade type is set as Never.
Failed	This reason is displayed when Auto Firmware Upgrade is not successful.
Failed to Connect to Host	This reason is displayed when Auto Firmware Upgrade server is not reacheable or is in different subnet.

Status	Description
Syncing files	This reason is displayed when Auto Firmware Upgrade process is running and the system is receiving files from the server.
File Not Found	This reason is displayed when the Firmware file is not present on the server or path for the file is invalid.
Parsing File Failed	This reason is displayed when Auto Firmware Upgrade file is received from the server but parsing fails.
Password Decrypt Failed	This reason is displayed when the encrypted file is received from the server but phone fails to decrypt it.
Server Not Found	This reason is displayed when Server IP Address is not reacheable.
Unknown Error	This reason is displayed when there is an internal application error.
Connecting	Auto Upgrade status connecting
Partial	Auto Upgrade status partial
Success	Auto Upgrade status success
Version Revision Same	Auto Upgrade status file version revision same
Downgrade Not Allowed	Auto Upgrade status downgrade not allowed
Invalid	Other reasons

- **Next Resync:** Date and time at which the phone will again resync with the server is displayed in this field.
- **Last Resync:** Date and time at which the phone had last resynchronized with the server is displayed in this field.

Phone Info

In Phone Info[4], you can view your phone information.

To access Phone Info,

- Press **Menu** Key.
- Scroll using the **Up/Down Navigation** Key to select **Phone Info** and press **Select** Key.

Parameter	Description
MAC Address	It displays the unique MAC Address of the phone.
IP	It displays the IP Address of the phone.
MASK	It displays the Subnet Mask of the phone.
GW	It displays the Gateway of the phone.
P.DNS	It displays the Primary DNS Address.
S.DNS	It displays the Secondary DNS Address.
802.1x	It displays the whether it is enabled or disabled.
FIRM	It displays the Version-Revision of the phone’s firmware along with the date.
KERNEL	It displays the Kernel Date of the phone.
Uboot	It displays the Uboot Date of the phone.

4. *You can view Phone Status, even if the phone is not registered with the ITSP/IP-PBX.*

Appendix

Call Progress Tone Generation (CPTG)

Index Number	Region	Dial tone		Ring Back Tone		Busy Tone		Error Tone		Progress Tone		CCWT	
		Frequency	Cadence (sec)	Frequency	Cadence (sec)	Frequency	Cadence (sec)	Frequency	Cadence (sec)	Frequency	Cadence (sec)	Frequency	Cadence (sec)
1	Region 1	440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	440	0.75on 0.75off	440	0.25on 0.25 off	350+440	0.1on 0.9off	350+440	0.1on 0.1off 0.1on 2.7off
2	Region 2	400	Continuous	400	0.6on 0.2off 0.2on 2.0off	400	0.5on 0.5off	400	0.25on 0.25 off	400	1.5on 0.1off	400	0.2on 4.8off
3	Region 3	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off	440	0.25on 0.25 off	350+440	0.1on 0.9off	440+480	0.1on 0.1off 0.1on 2.7off
4	Argentina	425	Continuous	425	1.0on 4.0 off	425	0.3on 0.2off	425	0.3on 0.4off	425	0.1on 0.9off	425	0.3on 10.0off
5	Australia	425*25	Continuous	400*25	0.4on 0.2off 0.4on 2.0off	425	0.375on 0.375off	425	0.375on 0.375off	425*25	0.1on 0.9off	425	0.2on 0.2off 0.2on 4.4off
6	Brazil	425	Continuous	425	1.0on 4.0 off	425	0.25on 0.25off	425	0.25on 0.25 off	425	0.1on 0.9off	425	0.05on 1.0off
7	Canada	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off	480+620	0.25on 0.25off	350+440	0.1on 0.9off	440	0.3on 10.0off
8	China	450	Continuous	450	1.0on 4.0off	450	0.35 on 0.36off	450	0.7on 0.7off	450	0.1on 0.9off	450	0.4 on 4.0off
9	Egypt	425*50	Continuous	425*50	2.0on 1.0off	425*50	1.0on 4.0off	450	0.5on 0.5off	425*50	0.1on 0.9off	425*50	0.1on 0.1off 0.1on 2.7off
10	France	440	Continuous	440	1.5on 3.5off	440	0.5on 0.5off	440	0.25on 0.25off	440	0.1on 0.9off	440	0.3on 10.0off
11	Germany	425	Continuous	425	1.0on 4.0off	425	0.48on 0.48off	425	0.24on 0.24off	425	0.1on 0.9off	425	0.2on .2off .2on 5.0off
12	Greece	425	0.2on 0.3off 0.7on 0.8off	425	1.0on 4.0off	425	0.3on 0.3off	425	0.15on 0.15off	425	0.1on 0.9off	425	0.3on 10.0off 0.3on 10.0off
13	India 1	400*25	Continuous	400*25	0.4on 0.2off 0.4on 2.0off	400	0.75on 0.75off	400	0.25on 0.25off	400*25	0.1on 0.9off	400	0.2on 0.1off 0.2on 7.5off
14	Indonesia	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off	425	0.25on 0.25off	425	0.1on 0.9off	425	0.15on 0.15off 0.15on 10.0off
15	Iran	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off	425	0.25on 0.25off	425	0.1on 0.9off	425	0.2on 0.2off 0.2on 10.0off
16	Iraq	400	0.4on 0.2off 0.4on 1.5off	400	Continuous	400	1.0on 1.0off	400	0.25on 0.25off	400	0.1on 0.9off	400	0.1on 0.1off 0.1on 2.7off
17	Israel	400	Continuous	400	1.0on 3.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	0.1on 0.9off	400	0.5on 10.0off
18	Italy 1	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off	425	0.2on 0.2off	425	0.1on 0.9off	425	0.4on 0.1off 0.25on 0.1off 0.15on 5.0off
19	Japan	400	Continuous	400*25	1.0on 2.0off	400	.5on .5off	400	0.25on 0.25off	400	0.1on 0.9off	400*25	0.5on 2.0off 0.05on 0.45off 0.05on 3.45off

20	Kenya	425	Continuous	425	0.67on 3.0off 1.5on 5.0off	425	0.2on 0.6off 0.2on 0.6off	425	0.2on 0.6off	425	0.1on 0.9off	425	0.1on 0.1off 0.1on 2.7off
21	Korea	350+440	Continuous	440+480	1.0on 2.0off	480+620	0.5on 0.5off	480+620	0.3on 0.2off	350+440	0.1on 0.9off	350+440	0.25on 0.25off 0.25on 3.25off
22	Malaysia	425	Continuous	425	0.4on 0.2off 0.4on 2.0off	425	0.5on 0.5off	425	0.5on 0.25off	425	0.1on 0.9off	425	0.1on 0.1off 0.1on 2.7off
23	Mexico	425	Continuous	425	1.0on 4.0off	425	0.25on 0.25off	425	0.25on 0.25off	425	0.1on 0.9off	425	0.1on 0.1off 0.1on 2.7off
24	New Zealand	400	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	0.1on 0.9off	400	0.2on 3.0off 0.2on 5.0off

Converting SPARSH VP510 Standard SIP Phone to SPARSH VP510 Extended SIP Phone

To convert the SPARSH VP510 Standard SIP Phone to SPARSH VP510 Extended SIP Phone, follow the steps given below:

SPARSH VP510 with Serial Number: 100160001 and onwards only can be converted to Extended SIP Phones.

- When the Phone is powered on and the Loading screen appears press #2.

- The message **"Do you want to start in Standard SIP Mode?"** appears.
- Press **Yes** Key.
- The phone will reboot and start as an Extended SIP Phone. The Factory Default values for this mode will be assigned to all the parameters.

Refer to the SPARSH VP510 (Extended) User Guide and respective Server Manuals — SARVAM UCS, PRASAR UCS and ANANT UCS to know more. The documentation can be found at https://www.matrixcomsec.com/support/telecom-product-manuals/

Technical Specifications

SPARSH VP510

VoIP	
VoIP Protocols	SIP v2, SDP, RTP, RFC 2833
Network Protocol	IPv4, TCP, DHCP
Voice CODECS	G.722 Wideband, G.711 A/μ-Law, G.723, G.729
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 Adapter (100-240 VAC, 50 - 60 Hz) (Optional)
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 to 45 degree C
Storage Temperature	-20 to 70 degree C
Operating and Storage Humidity	5 to 95%, RH, Non-Condensing

SPARSH VP510

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.	Screws and Screw Grips (pack)	1

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- If the product is subjected to abuse, damage, misuse, neglect, fire, power flow, acts of God, accident.
- If the product is installed or used in combination or in assembly with the products that are not supplied or authorized by Matrix or are of inferior quality or design than Matrix supplied products, which may cause reduction or degradation in functionality.
- If the product is operated outside the product's specifications or used without designated protections.
- If the completely filled warranty cards have not been received by Matrix within 15 days of the installation.

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END OF TERMS AND CONDITIONS

How to Apply These Terms to Your New Programs

If you develop a new program, and you want it to be of the greatest possible use to the public, the best way to achieve this is to make it free software which everyone can redistribute and change under these terms.

To do so, attach the following notices to the program. It is safest to attach them to the start of each source file to most effectively convey the exclusion of warranty; and each file should have at least the "copyright" line and a pointer to where the full notice is found.

```
<one line to give the program's name and a brief idea of what it does.>
Copyright (C) <year>  <name of author>

This program is free software; you can redistribute it and/or modify
it under the terms of the GNU General Public License as published by
the Free Software Foundation; either version 2 of the License, or
(at your option) any later version.

This program is distributed in the hope that it will be useful,
but WITHOUT ANY WARRANTY; without even the implied warranty of
MERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE.  See the
```

GNU General Public License for more details.

You should have received a copy of the GNU General Public License along
with this program; if not, write to the Free Software Foundation, Inc.,
51 Franklin Street, Fifth Floor, Boston, MA 02110-1301 USA.

Also add information on how to contact you by electronic and paper mail.

If the program is interactive, make it output a short notice like this
when it starts in an interactive mode:

```
    Gnomovision version 69, Copyright (C) year name of author
    Gnomovision comes with ABSOLUTELY NO WARRANTY; for details type `show w'.
    This is free software, and you are welcome to redistribute it
    under certain conditions; type `show c' for details.
```

The hypothetical commands `show w' and `show c' should show the appropriate parts
of the General Public License. Of course, the commands you use may be called
something other than `show w' and `show c'; they could even be mouse-clicks or
menu items--whatever suits your program.

You should also get your employer (if you work as a programmer) or your
school, if any, to sign a "copyright disclaimer" for the program, if
necessary. Here is a sample; alter the names:

```
 Yoyodyne, Inc., hereby disclaims all copyright interest in the program
`Gnomovision' (which makes passes at compilers) written by James Hacker.

  <signature of Ty Coon>, 1 April 1989
  Ty Coon, President of Vice
```

This General Public License does not permit incorporating your program into
proprietary programs. If your program is a subroutine library, you may
consider it more useful to permit linking proprietary applications with the
library. If this is what you want to do, use the GNU Lesser General
Public License instead of this License.

Regulatory Information

FCC Class B Information

FCC Part 15B ID : 2ADHNVP510

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.

MATRIX
TELECOM | SECURITY

DECLARATION OF CONFORMITY

DECLARATION OF CONFORMITY

MANUFACTURER : MATRIX COMSEC PVT. LTD.

MANUFACTURER 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 VP510

MODEL/TYPE : SPARSH VP510E

Essential Requirements /Directives		Applied Specifications/ Standards
EMC	2014/30/EU	EN 55032: 2015+AC:2016+A11:2020 EN 55035: 2017+A11:2020 EN 61000-4-2: 2009 EN 61000-4-3: 2006 + A1: 2007 + 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: 2004 + A1: 2017 EN 61000-3-2: 2019, EN 61000-3-3: 2013+A1:2019
LVD/SAFETY	2014/35/EU	IEC 62368-1: 2014 EN 62368-1: 2014 + A11:2017
RoHS (RoHS2)	2011/65/EU	EN IEC 63000:2018

I hereby declare that the equipment/product 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 *

Ganesh Jivani
Managing Director
Date: 30/11/2021

MATRIX COMSEC PVT. LTD.
Head Office: 394-GIDC, Makarpura, Vadodara-390010, India. Ph: +91 265 6137222 • CIN: U72200GJ1998PTC034047
Manufacturing: 19-GIDC, Waghodia, Dist. Vadodara-391760, India. Ph: +91 75678 75244 • Email: Inquiry@MatrixComSec.com
R&D: 190-GIDC, Makarpura, Vadodara-390010, India. • www.MatrixComSec.com

Disposal of Products/Components after End-Of-Life

Main components of Matrix products are given below:

- **Soldered Boards:** At the end-of-life of the product, the soldered boards must be disposed through e-waste recyclers. If there is any legal obligation for disposal, you must check with the local authorities to locate approved e-waste recyclers in your area. It is recommended not to dispose-off soldered boards along with other waste or municipal solid waste.

- **Batteries:** At the end-of-life of the product, batteries must be disposed through battery recyclers. If there is any legal obligation for disposal, you may check with local authorities to locate approved batteries recyclers in your area. It is recommended not to dispose off batteries along with other waste or municipal solid waste.

- **Metal Components:** At the end-of-life of the product, Metal Components like Aluminum or MS enclosures and copper cables may be retained for some other suitable use or it may be given away as scrap to metal industries.

- **Plastic Components:** At the end-of-life of the product, plastic components must be disposed through plastic recyclers. If there is any legal obligation for disposal, you may check with local authorities to locate approved plastic recyclers in your area.

After end-of-life of the Matrix products, if you are unable to dispose-off the products or unable to locate e-waste recyclers, you may return the products to Matrix Return Material Authorization (RMA) department.

Make sure these are returned with:
- proper documentation and RMA number
- proper packing
- pre-payment of the freight and logistic costs.

Such products will be disposed-off by Matrix.

"SAVE ENVIRONMENT SAVE EARTH"

E-Waste Management and Handling Rules

E-waste is a popular, informal name for electronic products nearing the end of their useful life. E-wastes are considered dangerous, as certain components of some electronic products contain materials that are hazardous, depending on their condition and density. The hazardous content of these materials pose a threat to human health and environment. Discarded electronics products such as circuit boards, batteries, wires and other electronic accessories if improperly disposed can leach lead and other substances into soil and groundwater. Many of electronic products can be reused, refurbished or recycled in an environmentally sound manner so that they are less harmful to the ecosystem.

Benefits of E-waste Recycling leach

Electronics Recycling Conserves Natural Resources

There are many materials that can be recovered from old electronic products. These materials can be used to make new products, thus reducing the need for the new raw materials. For instance, various metals can be recovered from circuit boards and other electronics can be recycled.

Electronics Recycling Supports the Community

Donating your old electronics plays an important role in the provision of refurbished products which can be of great help to certain industries, small organizations and non-profitable organizations. It also helps individuals gain access to technology that they could not have otherwise afforded.

Electronics Recycling Creates Employment Locally

Considering that around 90 percent of electronic equipment is recyclable, electronics recycling can play a significant role in creating employment. This is because new firms dealing with electronics recycling will form and existing firms will look to employ more people to recover recyclable materials. This can be triggered by the increase in the demand for electronics recycling.

Electronics Recycling Helps Protect Public Health and the Environment

Many electronics have toxic or hazardous materials such as mercury and lead, which can be harmful to the environment if disposed in trashcans. Reusing and recycling electronics safely helps in keeping the hazardous materials from harming humans or the environment. For example, certain electronic components and batteries are hazardous since they have lead in them. Printed circuit boards contain harmful materials such as cadmium, lead, mercury and chromium.

Instead of keeping old electronics or dumping them in landfills, recycling or reusing them is an appropriate option that should be supported by individuals and organizations. Considering the benefits of electronics recycling, it is very important that people in various parts around the world embrace this concept.
Creates Jobs

E-waste recycling creates new jobs for professional recyclers and creates a second market for the recycled materials.

Do's & Don'ts

Do's:

- Always look for information on the catalogue with your product for end-of-life equipment handling.
- Ensure that only Authorized Recyclers/Dismantler handle your electronic products.
- Always call at our toll-free No's to Dispose products that have reached end-of life.
- Always drop your used electronic products, batteries or any accessories, when they reach the end of their life at your nearest Authorized E-Waste Collection Points.
- Always disconnect the battery from product and ensure any glass surface is protected against breakage.

Don'ts:

- Do not dismantle your electronic Products on your own.
- Do not throw electronics in bins having "Do not Dispose" sign.
- Do not give e-waste to informal and unorganized sectors like Local Scrap Dealer/ Rag Pickers.
- Do not dispose your product in garbage bins along with municipal waste that ultimately reaches landfills.

E-Waste Management Plan

M/s. MATRIX COMSEC PVT LTD has partnered with **E-Waste Recyclers India (EWRI)** to comply with the new India E-Waste management and handling rules in providing drop-of centers and environmentally sound management of end of life electronics.

EWRI has obtained authorizations from the appropriate governmental agency for their processing facilities. EWRI will receive and recycle customer returned equipment, including all the e-waste. Customers can drop their e-waste in the drop-box provided at various collection centers of EWRI.

A list of collection centers along with the address is mentioned below.

The customers can also call on the following toll free number (1800-102-5679) from Monday to Friday between 10:00 AM to 5:30 PM to get details about the collection centers.

Collection Centers:

State/ City	Location	Logistic	Address	Toll-Free Number
Delhi	Rangpuri	Professional Logistics	Rangpuri, Milakpur Kohi Rangpuri, Rangpuri, New Delhi - 110037	1800-102-5679
Gurugram	Gurugram	Professional Logistics	295, LIG Colony, Sector 31, Gurugram, Haryana - 122022	1800-102-5679
Jharkhand	Dhanbad	Professional Logistics	Sardar Patel Nagar, Dhanbad, Jharkhand - 826004	1800-102-5679
Noida	Salarpur Khadar	Professional Logistics	2, Gejha Rd, Goyal Colony, Salarpur Khadar, Sector 102, Noida, Uttar Pradesh - 201304	1800-102-5679
Mumbai	Vashi	Professional Logistics	Plot-92,gala no 01,Sector 19C Vashi Navi, Mumbai - 400705	1800-102-5679

State/ City	Location	Logistic	Address	Toll-Free Number
Pune	Vallabh Nagar	Professional Logistics	No.3/20,Near Ashok Sah Bank, Vallabh Nagar, S.T.Stand Road, Pimpri, Pune - 302021	1800-102-5679
Odisha	Cuttack	Professional Logistics	Cuttack, Odisha	1800-102-5679
Hyderabad	Secunderabad	Professional Logistics	4,Block-3,4th Shatter at 179, MPR Estates Near Old Check Post Old Bowaenpally Secunderabad, Hyderabad - 500011	1800-102-5679
Bangalore	Yeshwanthpur	Professional Logistics	No.44 1st floor 2nd main D.D.U.T.T.L. Yeshwanthpur, Bangalore - 560022	1800-102-5679
Mangalore	Bhathery Road Boloor	Professional Logistics	Opp. Hindustan Lever Ltd, Sulthan, Bhathery Road Boloor, Mangalore (KA) - 575003	1800-102-5679
Jharkhand	Ranchi	Professional Logistics	Ranchi, Jharkhand	1800-102-5679
Chennai	Sennerkuppam	Professional Logistics	27,Sakthi Nagar Phase-II, Sennerkuppam, Near Bisleri Water Plant, Chennai - 600056	1800-102-5679
Rajasthan	Jaipur	Professional Logistics	A-81, 200 ft. By Pass, Heerapura, Jaipur, Rajasthan - 302021	1800-102-5679
Bokaro	Odisha	Professional Logistics	Cuttack, Odisha, India	1800-102-5679
Guwahati	Kundil	Professional Logistics	HN-34, Kundil Nagar Basistha Chariali, Near Parbhat Apartment, Guwahati - 781029	1800-102-5679
Lucknow	Kanpur Road	Professional Logistics	S-175,Ist Floor Transport Nagar Near RTO Kanpur Road Lucknow - 226004	1800-102-5679
Madhya Pradesh	Indore	Professional Logistics	284 AS-3 Scheme No.-78,Vijay Nagar, Indore, Madhya Pradesh	1800-102-5679
Ahmedabad	Pushp Penament	Professional Logistics	Shop No D-18, Pushp Penament, Behind Mony Hotel, Isanpur, Ahmedabad	1800-102-5679
Patna	Malyanil buddha	Professional Logistics	Dr. A.K Pandey (IPS) Malyanil buddha Colony, Patna (Bihar) - 800001	1800-102-5679
Andhra Pradesh	Vishakapatnam	Professional Logistics	Shop No.8, New Gajuwaka, Opp. High School Road, Vishakapatnam, Andhra Pradesh - 530026	1800-102-5679
Chandigarh	Pharbhat Road	Professional Logistics	Shop no:-19, Pharbhat Road, Opp:- Tennis Academy, Zirakpur, Chandigarh, Punjab	1800-102-5679

State/ City	Location	Logistic	Address	Toll-Free Number
Kolkata	B.T. ROAD DUNLOP	Professional Logistics	156A/73, Northern Park, B.T. Road Dunlop, Kolkata -700108	1800-102-5679
Odisha	Bhubaneswar	Professional Logistics	Acharya Vihar - jaydev Vihar Rd, Bhubaneswar, Odisha	1800-102-5679
West Bengal	Asansol	Professional Logistics	Shop No-4 Asansol Station Bus Stand Road, Munshi Bazar, Asansol, West Bengal - 713301	1800-102-5679

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