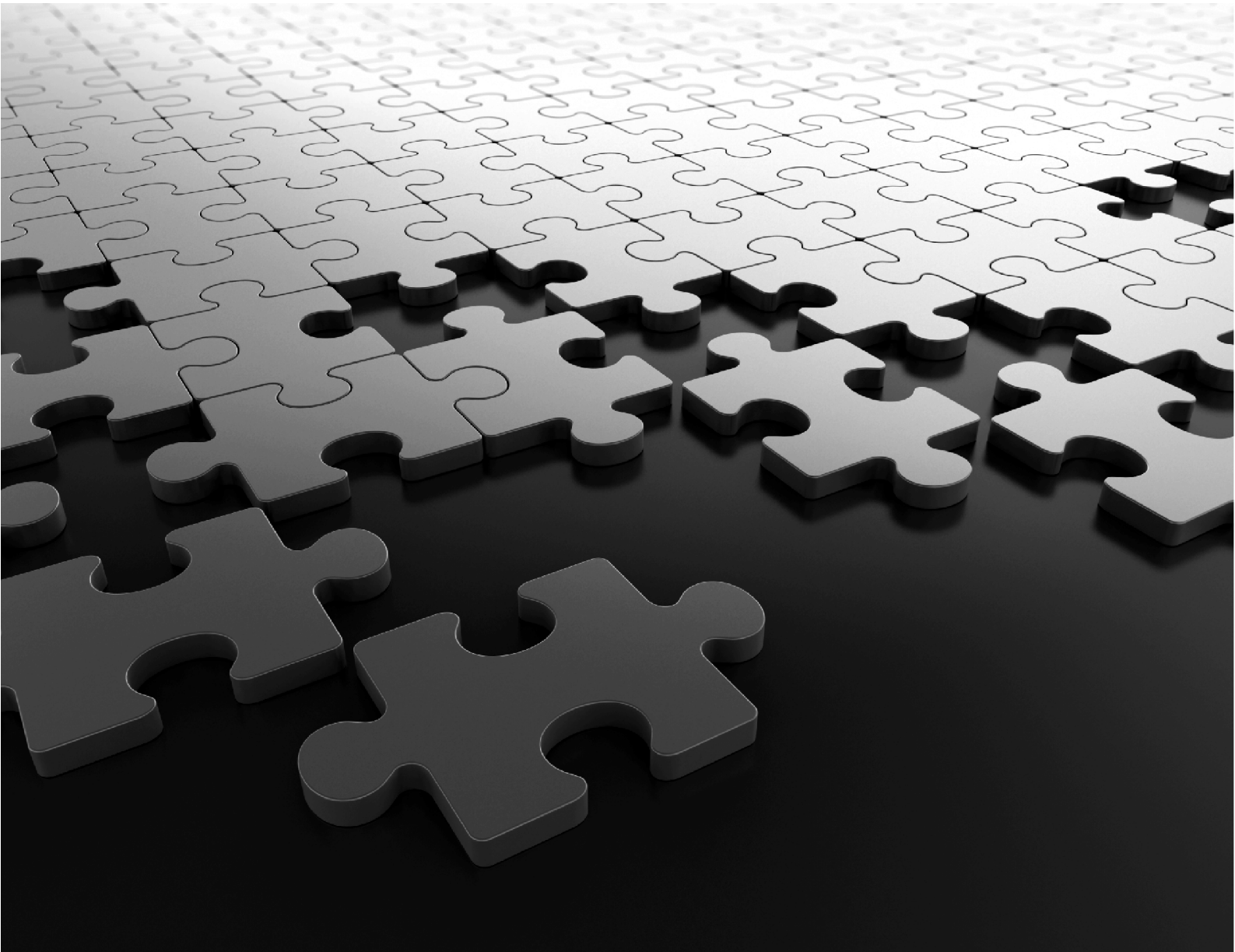


# SETU VGRX System Manual





## **SETU VGRX** Radio-over-IP Gateway with Integrated GSM/CDMA Connectivity

### **System Manual**



# Documentation Disclaimer

Matrix Comsec reserves the right to make changes in the design or components of the product as engineering and manufacturing may warrant. Specifications are subject to change without notice.

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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*Version 1*

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Thank you for choosing SETU VGRX! Please read this document carefully before installing and using your SETU VGRX.

## About this System Manual

This System Manual provides detailed instructions for installing, configuring and using the SETU VGRX.

For instructions on quick installation and operation of this system, you may refer *SETU VGRX Quick Start* shipped with the system.

For instructions on using the features of the SETU VGRX, refer *SETU VGRX User Card*. The documentation can be found at <https://www.matrixtelesol.com/product-manuals.html>

For product registration and warranty related details, please visit <http://www.matrixcomsec.com/product-registration-form.html>

## Intended Audience

This System Manual is aimed primarily at **Network and System Engineers**, who will install, configure and maintain the SETU VGRX.

System Engineers are persons who customize the system configuration to meet the requirements of the organization/users. It is assumed that they have some experience in installing and configuring VoIP-Radio Gateways.

Parts of this document containing description of telephony features are aimed at **End Users**, who are the persons/ organizations who will actually use the SETU VGRX.

## Organization of this Document

This System Manual contains the following chapters:

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

**Know Your SETU VGRX:** Provides an overview of SETU VGRX.

**Installing SETU VGRX:** Contains information on how to install SETU VGRX and configure it using the web-based programming tool, Jeeves.

**Basic Settings:** Provides instructions for configuring the basic parameters of SETU VGRX, which are sufficient to get the system into operation.

**Advanced Settings:** Contains instructions for configuring the more advanced features and facilities of SETU VGRX.

**Features:** Describes the telephony features of SETU VGRX and provides instructions for End Users on using these features.

**Maintenance:** Provides instructions for back-up, generating reports and debugging.

**Status:** Describes the indicators of the System, Network, SIP Trunk, Mobile Ports and FXO Ports status.

## How to Read this System Manual

This System Manual 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 and the Index to navigate through this document to the relevant topic or information you want to look up.

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

## Conventions used in this System Manual

The following symbols have been used for notices to draw your attention to important things:



**Note:** *It indicates something that requires your special attention or it reminds you of something you need to do when you are using the SETU VGRX.*



**Tip:** *It indicates a helpful hint giving you an alternative way to operate the SETU VGRX or carry out a procedure more efficiently.*



**Caution:** *It indicates an action or condition that is likely to result in malfunction or damage to the SETU VGRX or your property.*



**Warning:** *It indicates a hazard or an action that will cause damage to the SETU VGRX and/ or cause bodily harm to the user.*

## Terminology used in this System Manual

In this system manual, following words are used synonymously.

- **SETU VGRX, System** and **Gateway**
- **Login Password, SE Password** and **Jeeves Password**

SETU VGRX supports both GSM (2G/3G) and CDMA modules. Therefore, Mobile Port can be either GSM or CDMA. In the GSM Port, SIM Card is installed and in the CDMA Port, RUIM Card is installed. Consider SIM as RUIM, if CDMA module is installed in your SETU VGRX.

Some of the terms specific to this document are defined below:

<b>Term</b>	<b>Usage in the document</b>
<b>System Engineer (SE)</b>	The person who installs, configures and maintains SETU VGRX.
<b>User</b>	The person who uses SETU VGRX.
<b>Caller / Calling party</b>	The person who make calls using SETU VGRX.
<b>Callee / Called party</b>	The person to whom calls are made using SETU VGRX.
<b>Source / Originating Port</b>	A port from which a call originates.
<b>Destination / Terminating Port</b>	A port on which a call terminates.

Using this System Manual, we hope, you will be able to install, operate and make optimum use of your SETU VGRX. However, if you encounter any technical problems, please contact your dealer/reseller or Matrix Customer Care.



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## Overview of SETU VGRX

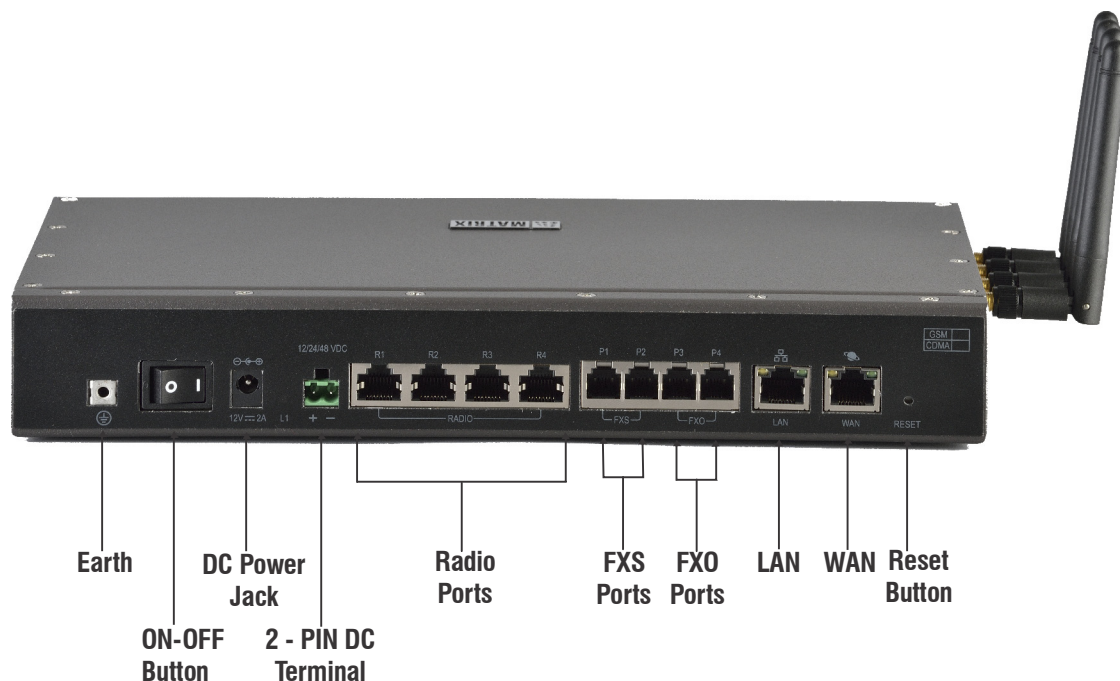
SETU VGRX is a gateway that provides the convergence between IP network, GSM/CDMA network and different Radio networks. It allows you to establish communication path between different Radio networks over Ethernet/IP and GSM network. Efficient communication is very important in emergency situations like war, natural calamities, civil unrest. SETU VGRX plays important role in offering strong and reliable communication in such extreme situations. It is best suited for the organizations that use multiple radio systems operating in HF/VHF/UHF bands.


SETU VGRX is available in two configurations:

- **SETU VGRX (GSM)** with 9 SIP Trunks, 4 Radio Ports, 4 Mobile Ports, 2 FXO Ports and 2 FXS Ports.
- **SETU VGRX (CDMA)** with 9 SIP Trunks, 4 Radio Ports, 4 Mobile Ports, 2 FXO Ports and 2 FXS Ports.

For a complete list of Hardware and Software features, refer [“Product Specifications”](#) in the Appendix.

## Ports and Connectors



Port	Connector	Description
Earth	--	To connect the Telecom Earth.
On-Off Button	--	To On or Off the SETU VGRX.
12VDC-2A	DC Jack	To connect 12VDC, 2A Power Adapter.
12/24/48 VDC	2- PIN DC Terminal	To connect 12/24/48 VDC Battery to feed power.
L1	--	LED labeled as <b>L1</b> will glow red when reverse voltage is detected on the Battery Terminal.
R1 to R4 (Radio)	RJ45	To connect Radio Devices such as Radio Handsets, Radio Base Stations.
P1 & P2 (FXS)	RJ11	To connect standard Telephone Instruments, or a Fax Machine or a PBX.
P3 & P4 (FXO)	RJ11	To connect PSTN lines or a PBX.
LAN Port	RJ45	To connect a Computer or a LAN Switch.
WAN Port	RJ45	To connect to the IP network over a DSL Modem or Router or a LAN Switch.
Reset Button	--	To restart the system or to restore the default LAN IP Address.
	SMA (Female)	To connect the Antenna for the Mobile Ports.
MOB <sup>a</sup>	--	To connect to CDMA/GSM/UMTS network <sup>b</sup> .

*a. The SIM Holder is located on the Main Board.*

*b. When the 3G module is installed in the system, you must disable **Call Waiting** on the SIM before inserting it into the system to prevent current calls from being disconnected.*

## LEDs

SETU VGRX has a Power LED, a Status LED (STS) and ten port LEDs as shown below:

- The LEDs **R1** to **R4** are assigned to the Radio Ports.
- The LEDs **P1** and **P2** are assigned to the FXS Ports.
- The LEDs **P3** and **P4** are assigned to the FXO Ports.
- The LEDs **M1**, **M2**, **M3** and **M4** are assigned to the Mobile Ports.

The LEDs indicate the status of the ports and various events occurring on the ports, including errors.

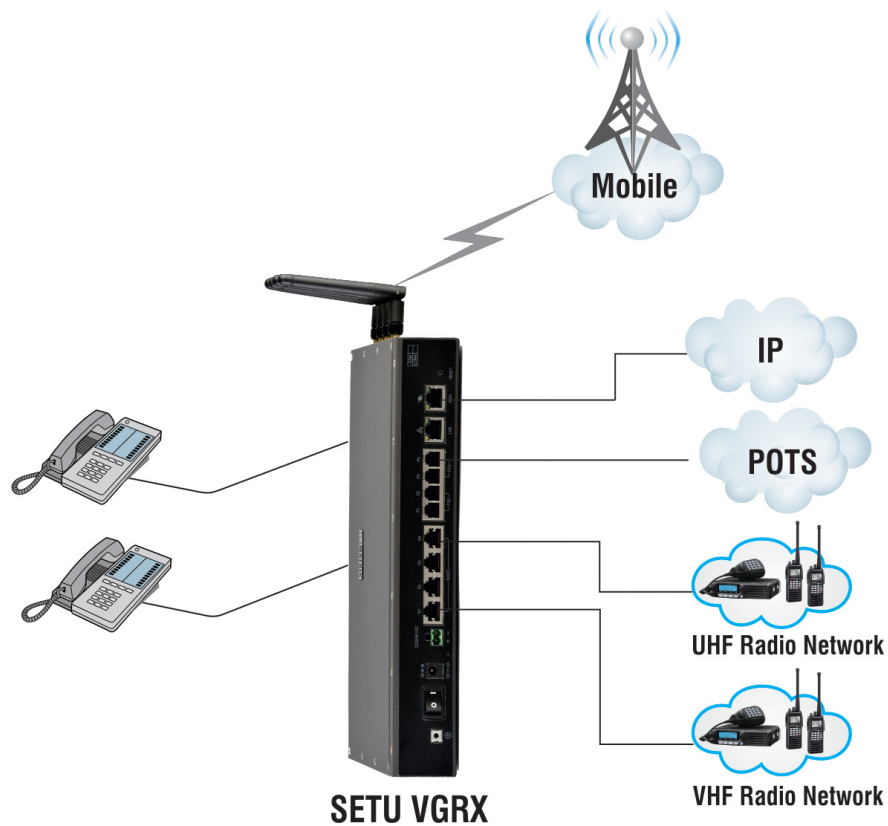


SETU VGRX is easy to install and operate. The built-in web server, *Jeeves*, allows you to configure the system parameters and features on-site as well as from a remote location.

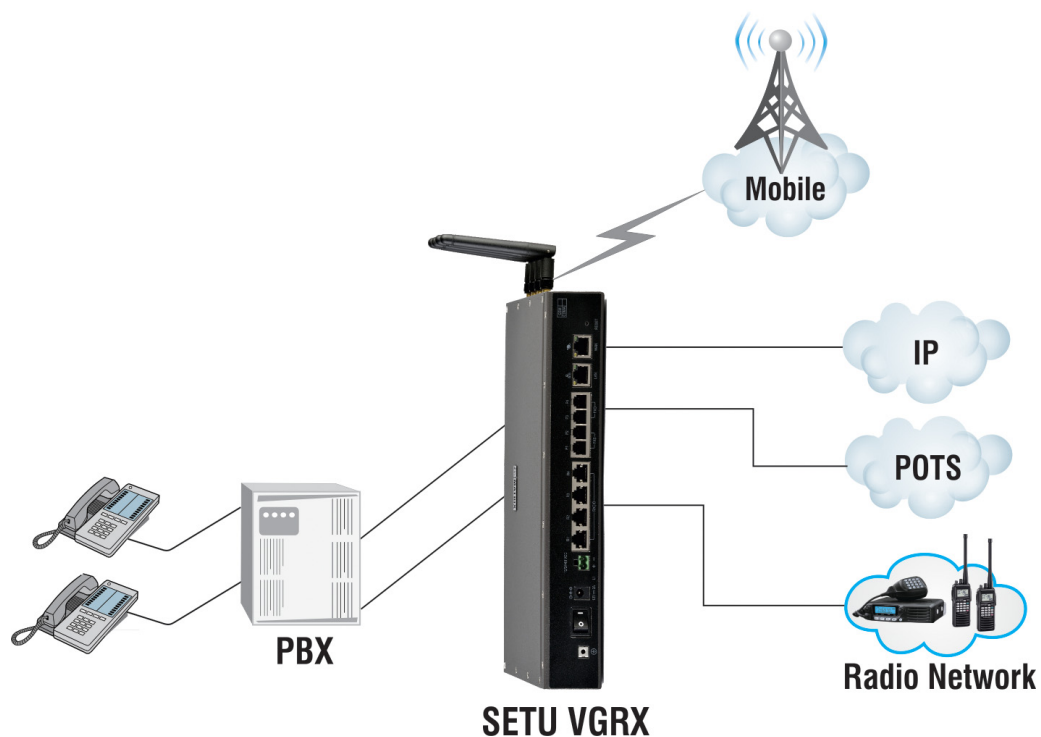
# Applications of SETU VGRX

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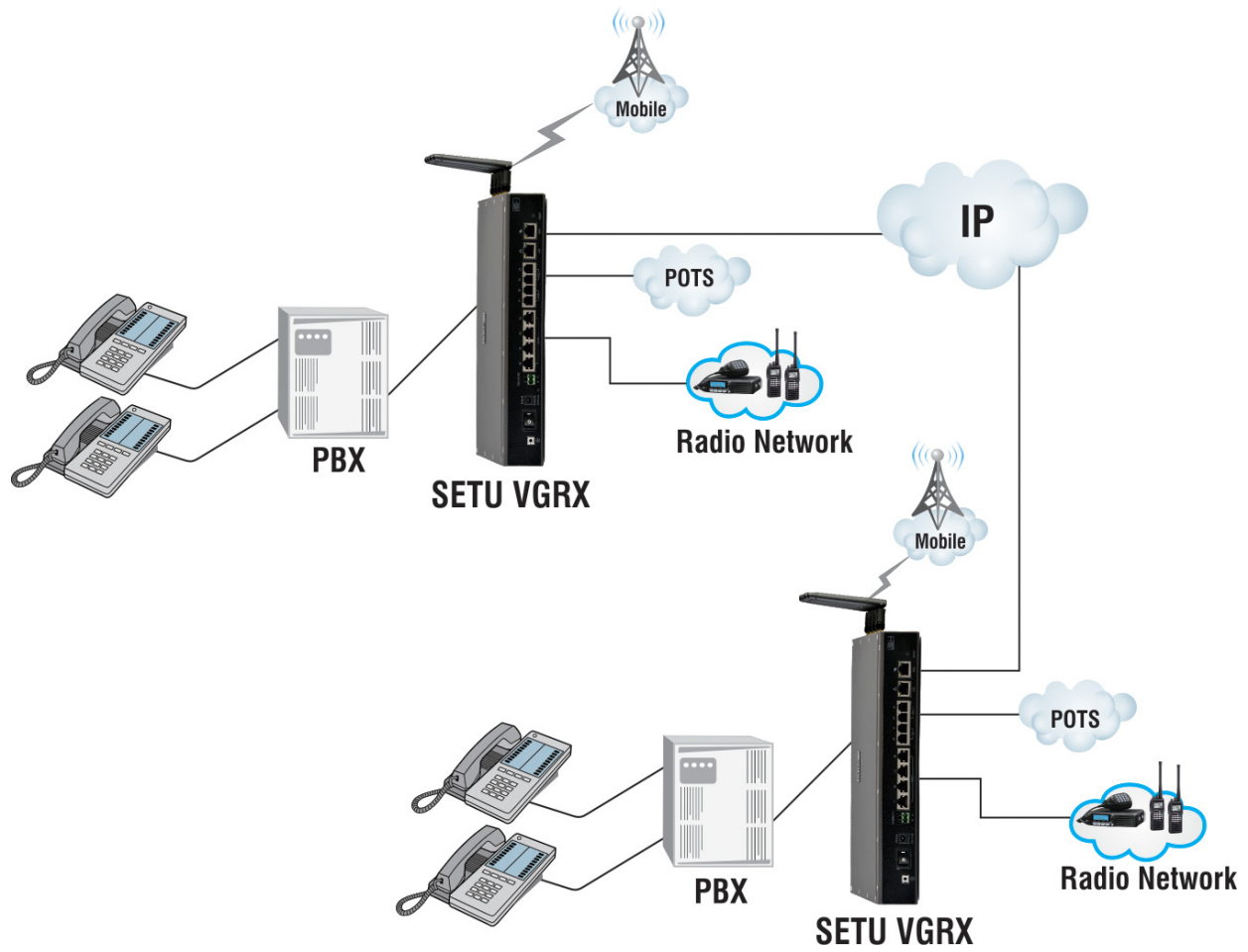
## SETU VGRX: As a Stand-Alone Gateway



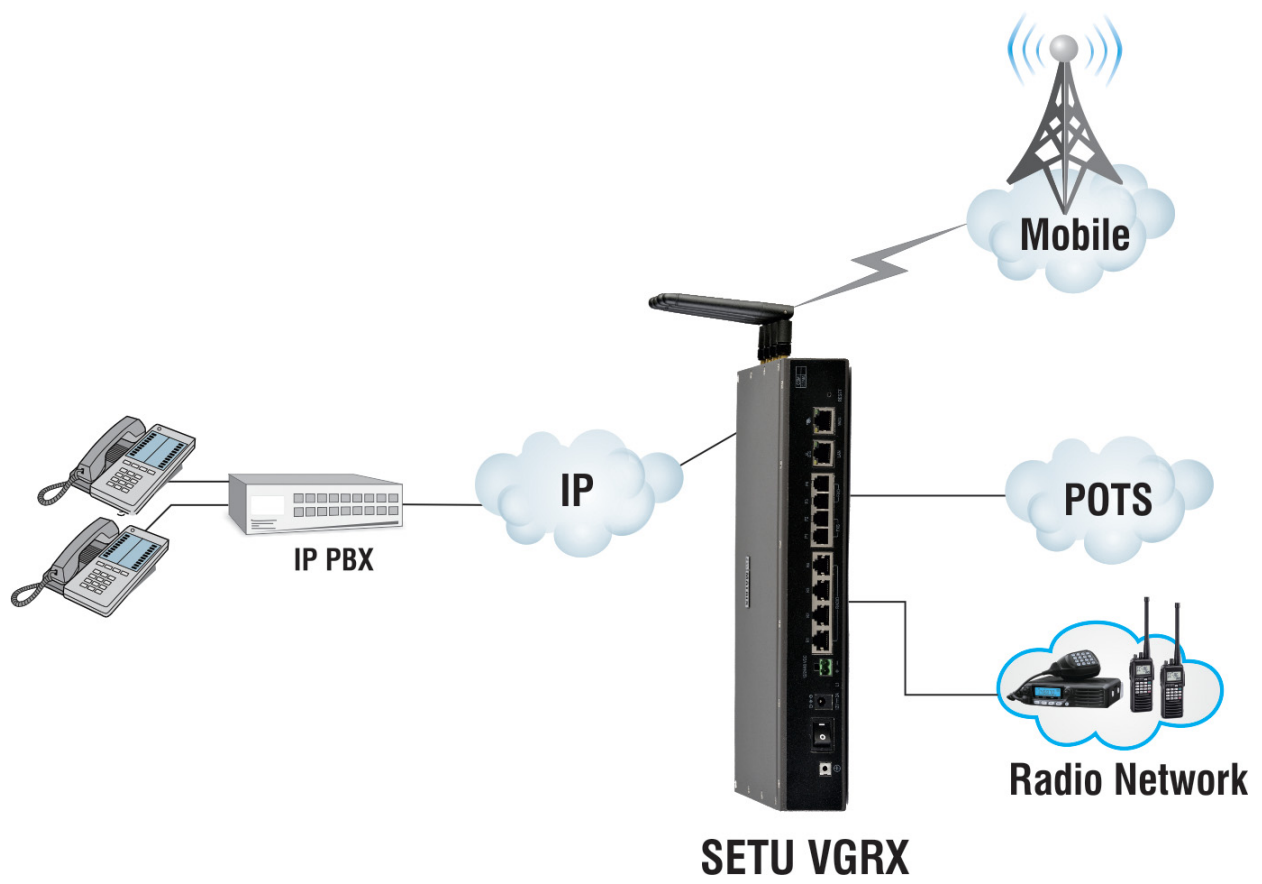
## SETU VGRX: Radio-VoIP-Mobile Gateway for Traditional PBX



## SETU VGRX: The Gateway connecting two Radio networks over IP



## SETU VGRX: Radio-Mobile-FXO Gateway for IP PBX





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## Before You Start

Before you begin to install and set up the hardware of SETU VGRX, make sure you have the following ready:

- A suitable location to install SETU VGRX.
- Power supply.
- A SIP Account from an ITSP to test VoIP connectivity.
- A standalone computer or a computer connected in a LAN to access Jeeves, the web-based configuration tool of SETU VGRX.
- Appropriate cables and connectors to set up and test the WAN interface of SETU VGRX and the LAN connection.
- An Analog Trunk Line from the CO/PSTN to connect to the FXO Port.
- At least one standard telephone instrument to connect to the FXS Port. You can also connect a fax machine or a PBX.
- Necessary telecom wiring for the lines and devices to be connected to the FXS and FXO Ports.
- Standard, good quality, twisted pair telephone cables with 0.5mm conductor diameter and RJ11 plugs for the FXS and FXO Ports.
- A Radio device to check Radio connectivity.
- A SIM Card to test Mobile connectivity.

Well begun is half done; plan your hardware installation well.

## Protect SETU VGRX and Yourself

For safe and efficient operation, observe the guidelines and all necessary safety precautions given in here. While installing as well as using any electronic appliance, take every safety precaution to reduce the risk of fire, electric shock and injury to persons. Read and understand all the instructions given in the manual.

- Do not install the system at any of the below locations:
  - in any area where it is directly exposed to sunlight, excessive cold or humid atmosphere.
  - any area where sulfuric gases are produced and where there are thermal springs.
  - at any place which is sensitive to vibrations or frequent and strong shocks.
  - at dusty places or places where it comes in direct contact with oil or water.
  - near any water source like a wash bowl, kitchen sink, bath tub or near a swimming pool.
  - on movable or unstable surfaces, which may cause the product to fall and get damaged.
- Always wear an electrostatic discharge preventive wrist strap or belt and use a grounding mat when handling the system.
- Unplug the system from the power outlet before cleaning. Do not use liquid cleaners, use only a dry and soft cloth.
- Do not turn on the power supply until the installation is complete.
- Never open SETU VGRX in power ON condition.
- Operate the system within the recommended power supply voltage range.
- Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
- Take the system to a qualified service person for repair work.
- Unplug the system from the power outlet and contact the qualified service personnel under the following conditions:
  - If liquid has been spilled onto it.
  - If it has been exposed to rain or water.
  - If it has been dropped or the cabinet has been damaged.
  - If it does not operate normally.

## **Telecom Earth (Ground)**

To protect the system from extremely high voltage currents resulting from lightning strikes, you must install a lightning protector on an outside (CO) line.

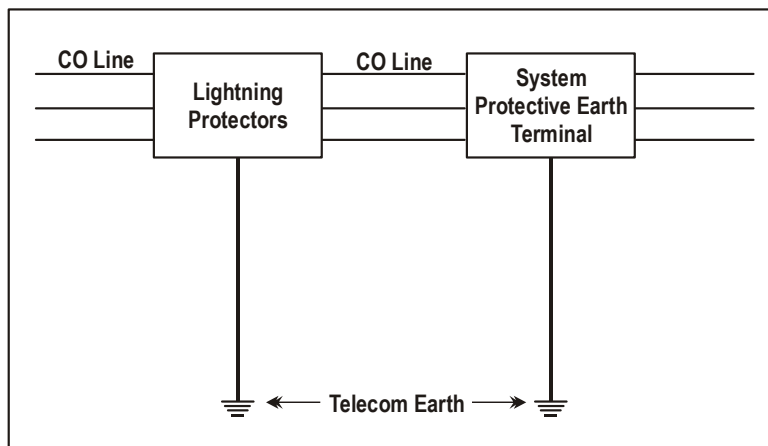
Telecom Earth (Ground) is the most important safety procedure to prevent electrical shocks and fires. It protects the system from lightning strikes, electrical transients, static discharges, electromagnetic interference and surges of telephone lines.

A dedicated terminal for earthing is provided in the port side panel of the system to which the telecom earth should be connected. The advantage of having a dedicated earth is that there is no risk of back voltage. There are chances that if the earthing is not perfect, instead of providing protection to the system, it may damage the system.



*Make sure your electrical earthing is proper and separate from the telecom earth used for the system.*

Protecting the system from high current surges is achieved by installing primary protection device. A lightning protector is a primary protection device which is used to prevent a dangerous surge from entering the building and damaging the system. For equipment installed in a more exposed environment, it is necessary to protect the system with primary protectors such as PPMs. With the development of electronic equipment, problems due to lightning surges have increased. A dangerous surge can occur if a telephone line comes in contact with a power line. A lightning protector should be installed on an outside (CO) line to prevent a dangerous surge from entering the building and damaging the system. The best place for the insertion of the primary protection is the cable entry point of the building, shelter or equipment housing.



This is not always possible but every attempt should be made to place the primary protection as close as possible to the entry point of the cables into the building, shelter or equipment housing. Hence, the system should be installed with lightning protectors and in addition to that, grounding (connection to earth ground) is very important to protect the system.



*Refer [“How to Make the Telecom Earth”](#) for detailed information.*

## Protecting the System from Static charges

While installing the system or servicing the system, care must be taken to provide a path to the static charges. It is strongly recommended that the system engineer should touch a grounded object before touching the system before installation or maintenance tasks.

## Battery

SETU VGRX contains a 3VDC/18mAh (Li-Al) alloy-Manganese Dioxide Coin Battery (ML 1220 - Rechargeable) of diameter 12.5mm and height 2.0mm. The Battery should be replaced only by authorized dealers of Matrix. End Users must not attempt to replace it.



*There is risk of explosion if the Battery is replaced in an incorrect manner. Please dispose-off used Batteries.*

## Disposal

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

## Warning for RF Safety

This product complies with the RF exposure guidelines as per standard FCC 47 CFR part 2. We recommend that you take the following safety measures:

- Keep the RF Antenna at least 20cm away from other electronic and radio transmission devices.
- Keep the RF antenna at a place at least 20cm away from people's vicinity.
- Do not place the magnetic storage media near the system.
- People carrying medical implants like cardiac pacemakers are advised to maintain appropriate distance from the system. They are also advised to avoid being in the vicinity of the product for a long time.

## Getting Started

- Select an appropriate site to install SETU VGRX, considering the safety precautions listed earlier in this chapter.
- Unpack SETU VGRX and verify the package contents.

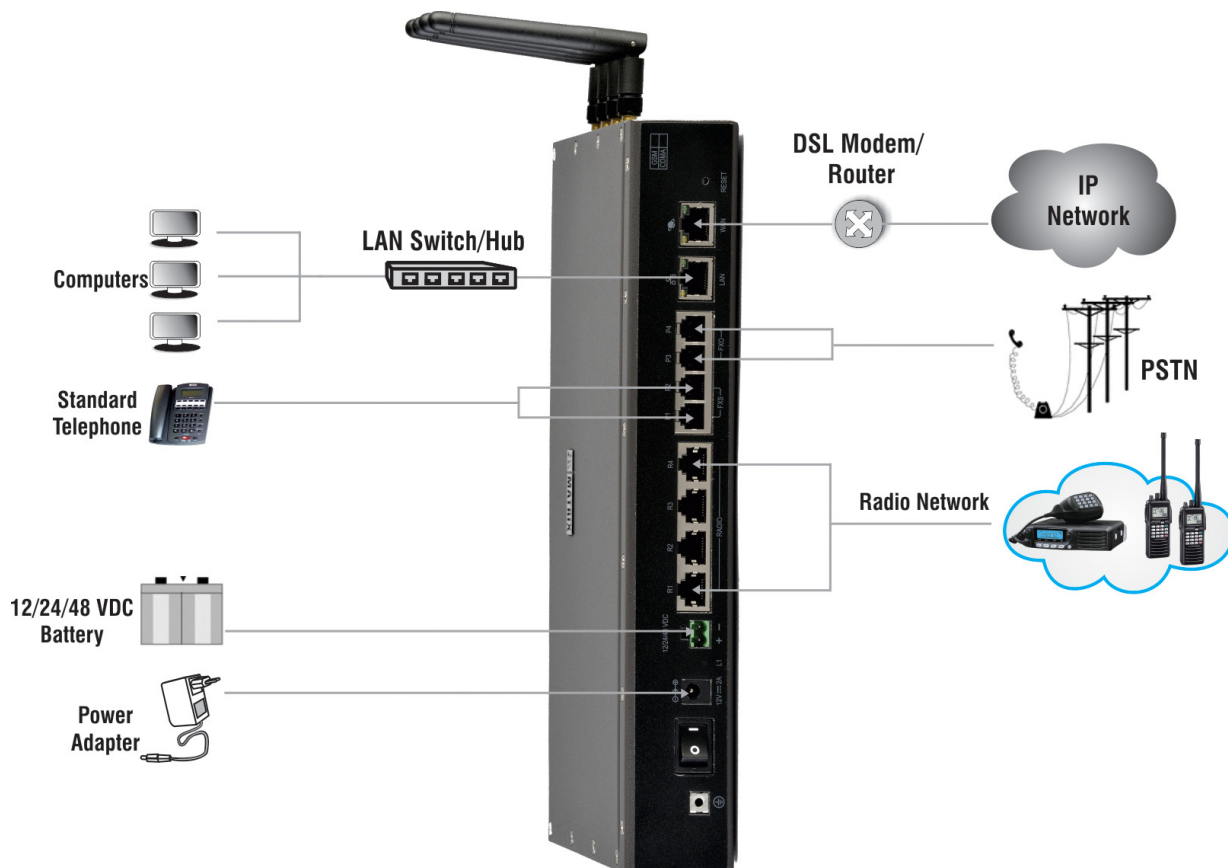
Sr. No.	Item Name	Quantity
1	SETU VGRX Unit	1
2	Power Adapter - 12VDC, 2 A (Country Specific)	1
3	Ethernet Cable (RJ45)	1
4	Cable with RJ45 connectors on both ends	1
5	Cable with RJ45 connector at one end and Amphenol Male connector on the other end	1
6	Cable with RJ45 connector at one end and Amphenol Female connector on the other end	2
7	Cable with RJ45 connector at one end and the free loose wires on the other end	1
8	Line Cord (RJ11)	4
9	GSM Antenna with SMA Connector	4
10	Screw M4/12 for Earthing	1
11	Screw M5/25	2
12	Screw Grip	2
13	SETU VGRX Quick Start (printed copy)	1
14	Wall Mounting Template	1

Make sure the above listed items are present in the package. In case any of these is missing or damaged, contact the dealer/distributor from whom you have purchased it.

- Place the system at the selected location.
- You may mount the system on a wall. Refer the mounting template for dimensions and accordingly drill the holes on the wall.

# Connecting SETU VGRX

SETU VGRX has a WAN Port, a LAN Port, a Reset Button, 4 Radio Ports, 4 Mobile Ports, 2 FXO Ports, 2 FXS Ports, 9 SIP Trunks, a 12/24/48 VDC Battery Terminal, a Power Jack and 15 LEDs.

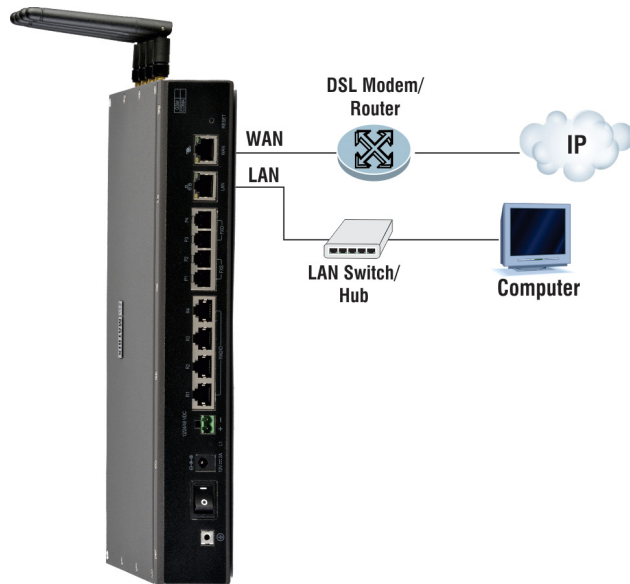


## Connecting to the IP Network

- Connect the **WAN Port** of SETU VGRX to the IP Network—a DSL modem/router or a LAN Switch—using the Ethernet cable supplied for the port.

### *If connecting to the Public IP Network,*

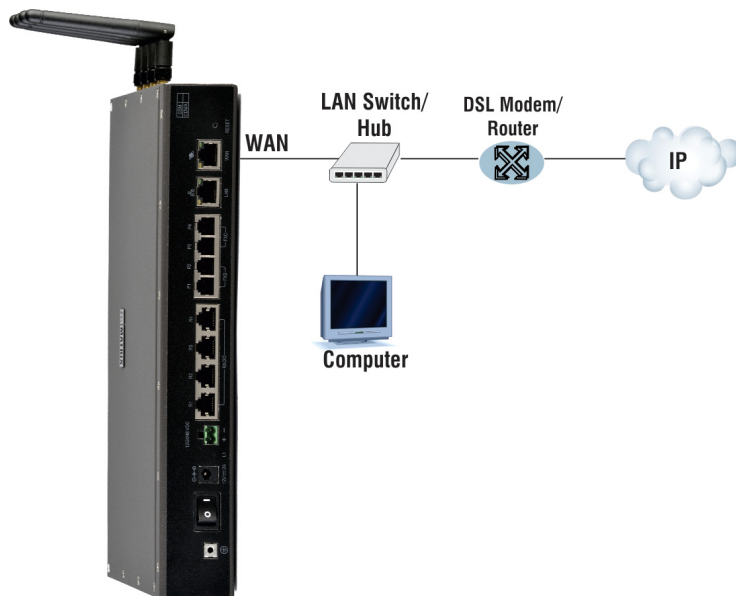
- Plug one end of the Ethernet cable into the WAN Port of SETU VGRX and the other end into the DSL modem/Router.



SETU VGRX

### *If connecting to a Private Network (Behind a NAT Router),*

- Plug one end of the Ethernet cable into the WAN Port of SETU VGRX and the other end into the LAN Switch/Hub.



SETU VGRX

The default IP Address of the WAN Port is: **192.168.1.100**

The default Subnet Mask of the WAN Port is: **255.255.255.0**

## Connecting to the Mobile Network

Make sure the site you have installed the system has sufficient network signal strength.

SETU VGRX supports both GSM (2G/3G) and CDMA modules. If CDMA module is installed in your SETU VGRX, following parameters will not be applicable.

- Band Selection
- CLIR
- Emergency Number
- Network Selection
- Preferred Network Mode
- Route Calls returned Unconnected to Original Caller
- SIM PIN
- SIM Balance Inquiry
- SIM Recharge
- SMSC
- SMS

## Enabling SIM PIN Protection



*If your SETU VGRX has a CDMA module, make sure you have disabled the PIN protection before installing the RUIM in the CDMA Port.*

You can protect the SIM Card from unauthorized use with a Personal Identification Number (PIN) on the SIM (in consultation with the customer/owner of the SIM).

To enable SIM PIN protection,

- Get a mobile handset. Insert the SIM into the mobile handset.
- From the mobile handset enable PIN protection.
- Assign a value as the SIM PIN. SIM PIN can be of minimum 4 and maximum 8 digits.
- Remove the SIM from the mobile handset.

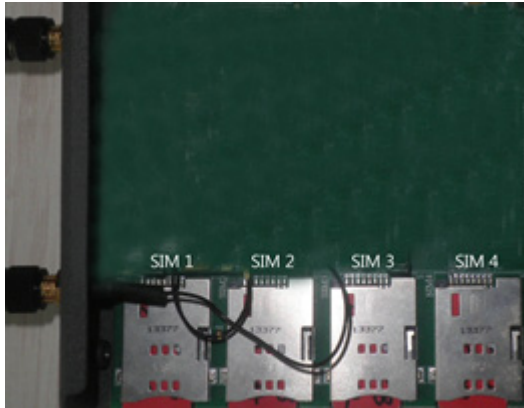


- *If you have enabled SIM PIN protection and you enter wrong SIM PIN thrice consecutively, your SIM Card will be blocked. To unblock, you must contact your Service Provider.*
- *If you do not want to use PIN protection, insert the SIM in the mobile handset and disable PIN protection. Remove the SIM Card from the mobile handset and insert it in the Mobile Port's SIM holder.*
- *If your SETU VGRX has a 3G module, you must disable Call Waiting in the SIM Card before inserting it into the SIM Holder of the Mobile Port. This will prevent current calls from being disconnected whenever there is a waiting call on the Mobile Port.*

## Inserting the SIM Card in the Mobile Port

- Make sure you are wearing an electrostatic discharge preventive wrist strap or belt and power supply is switched off. Unplug the adapter, if you have connected it.
- Unscrew and remove the cover of SETU VGRX. Keep the cover and screws aside.

- The Mobile Ports are located on the Main Board.



- Insert the SIM Card into the SIM holder—SIM1, SIM2, SIM3, SIM4—of the Mobile Port with the contact side of the SIM Card facing down.
- Replace the cover and secure the cover with the screws.
- Fix the Mobile Antenna to the connector.

## Connecting to the CO Network

- To the **FXO Ports**, connect the Analog Trunk lines from your CO Network/PSTN.

You may also connect a PBX to the FXO Port.

## Connecting Telephone instruments

- To the **FXS Ports**, connect standard single line telephones using standard telephone cables with RJ11 plugs.

You may also connect a Fax machine or a PBX to the FXS Port.

## Connecting Radio Devices

- To the **Radio Ports**, connect the Radio devices using the cables supplied with the product for the Radio Ports.
- Plug the RJ45 cable connector into the Radio Port.
- If you are connecting Radio devices using the cable with RJ45 connectors at one end and the loose wires on the other end, refer to the Pin out details given below:

Connector	Color	PIN Number	Signaling
RJ45-1 to RJ45-4	Orange & White	1	PTT
	Orange	2	PTT_RTN
	Green & White	3	Rx-
	Blue	4	Tx+
	Blue & White	5	Tx-
	Green	6	Rx+
	Brown & White	7	Unused
	Brown	8	Unused

## Power ON SETU VGRX

- Check the mains voltage at the power plug from where the power supply is to be fed to the system. It should be as per the specifications mentioned in the ["Product Specifications"](#).
- Make sure system's earthing is proper.
- Connect the **Power Adapter** into the power jack, and plug it into a power outlet.
- Switch ON the power supply and observe the reset cycle.

## LED Indication

At Power ON, Power LED will turn ON (Continuous Green). Other LEDs will follow the sequence summarized in the table below, during initialization.

System Status <sup>a</sup>	STS	R1	R2	R3	R4	P1	P2	P3	P4	M1	M2	M3	M4	Time in msec
Power ON - UBOOT	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
Kernel UP	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
Application and LED Driver Loaded	OFF	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
Few Initialization (i.e SysConfig, Resolver, SysLog etc.)	OFF	ON	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	

System Status <sup>a</sup>	STS	R1	R2	R3	R4	P1	P2	P3	P4	M1	M2	M3	M4	Time in msec
Few Initialization (i.e WebJvs, Call Manager, PortCnfg etc.)	OFF	ON	ON	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
VOPP Program Download Success	OFF	ON	ON	ON	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
All Init Done, System goes Live	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	1000 msec
	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	ON	1000 ms
	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	1000 msec
	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	1000 msec
	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	1000 msec
	(Continuous Green blinking as given in Last 2 steps)													

a. Red color text indicates that the red led will glow and Green color text indicates that the green led will glow.

During initialization, the Mobile Port LEDs (M1 to M4) will indicate the following error/event/status:

LED Status - Cadence (in milliseconds) 1 cadence = 4000msec	Color	Event/State/Status
200ms On- 200ms Off	Red	GSM Initialization
500ms On - 500ms Off	Red	PUK required
500ms On - 500ms Off - 500ms On - 500ms Off - 500ms On - 1500ms Off (3 Blinks)	Red	SIM PIN Required
500ms On - 500ms Off - 500ms On - 2500ms Off (2 Blinks)	Red	SIM PIN Wrong
500ms On - 3500ms Off (1 Blink)	Red	SIM Absent
1sec On - 1sec Off	Red	Mobile network absent

During normal functioning, Mobile Port LEDs will display following error/events/status:

LED Status	Color	Event/State/Status
Continuous OFF	-	Port Idle/Disable
400ms On - 200ms Off - 400ms On - 3000ms Off (2 Blinks)	Red	Incoming Ring Event
2000 ms On - 2000 ms Off	Red	Off-hook Event

LED Status	Color	Event/State/Status
Continuous On	Red	Speech
Continuous On	Red	Sending SMS/ Processing Balance Inquiry / Processing Balance Recharge

During normal functioning, FXS/FXO Port LEDs (P1 to P4) will display following error/events/status:

LED Status	Colour	Event/State/Status
Continuous OFF	-	Port Idle/Disable
400ms On - 200 ms Off - 400ms On - 3000 ms Off (2 Blinks)	Red	Incoming Ring Event
2000 ms On - 2000 ms Off	Red	Off-hook Event
Continuous On	Red	Speech



*During Off-hook state, FXS Port LED glows Red (2000 ms On - 2000 ms Off). However, if the system is in programming mode, the LED of the port from which you entered the programming mode will glow continuously Red as in Speech state. After exiting the programming mode, the LED will again glow Red (2000 ms On - 2000 ms Off) displaying Off-hook event.*

During normal functioning, Radio Port LEDs (R1 to R4) will display following error/events/status:

LED Status	Color	Event/State/Status
Continuous OFF	-	Port Idle/Disable
200ms On - 200ms Off - 200ms On - 3400ms Off (2 Blinks)	Red	Incoming Ring Event
2000 ms On - 2000 ms Off	Red	Off-hook Event <sup>a</sup>
Continuous On	Red	Speech

*a. A call initiated by the Radio Port but not matured i.e. the two way speech is not established.*

During initialization, System LED (STS) will display following error/events/status:

LED Status	Colour	Comment
Continuous On	Red	VoPP program download fail.
2000 ms On (Red) 5000 ms On (Green)	Red and Green	Modules not detected.
2000 ms On - 2000 ms Off	Red	T1 DSP 5502 is not detected.
1sec On - 1sec Off	Green	SETU VGRX started successfully. Network link is Up. SIP stack is Up. CDR buffer is not full.
500 ms On - 500ms Off - 500 ms On - 500ms Off - 500 ms On - 500ms Off - 500 ms On - 500ms Off (4 Blinks)	Green	Network link is down. SIP stack is down. CDR buffer is not full.

LED Status	Colour	Comment
500 ms On - 500ms Off - 500 ms On - 500ms Off - 500 ms On - 1500ms Off (3 Blinks)	Green	Network link is Up. SIP stack is down. CDR buffer is not full.
500 ms On - 500ms Off - 500 ms On - 500ms Off - 500 ms On - 500ms Off - 500 ms On - 500ms Off (4 Blinks)	Red	Network link is down. SIP stack is down. CDR buffer is full.
1sec On - 1sec Off	Red	Network link is up. SIP stack is up. CDR buffer is full.
500 ms On - 500ms Off - 500 ms On - 500ms Off - 500 ms On - 1500ms Off (3 Blinks)	Red	Network link is up. SIP stack is up. CDR buffer is full.

When the Reset Cycle is completed, you may configure the system using the embedded web server, *Jeeves*.

# Configuring SETU VGRX

---

SETU VGRX provides an embedded web server with a Graphic User Interface (GUI), *Jeeves*, for configuration.

To access Jeeves, you will need to connect a computer to SETU VGRX.

## Connecting a Computer

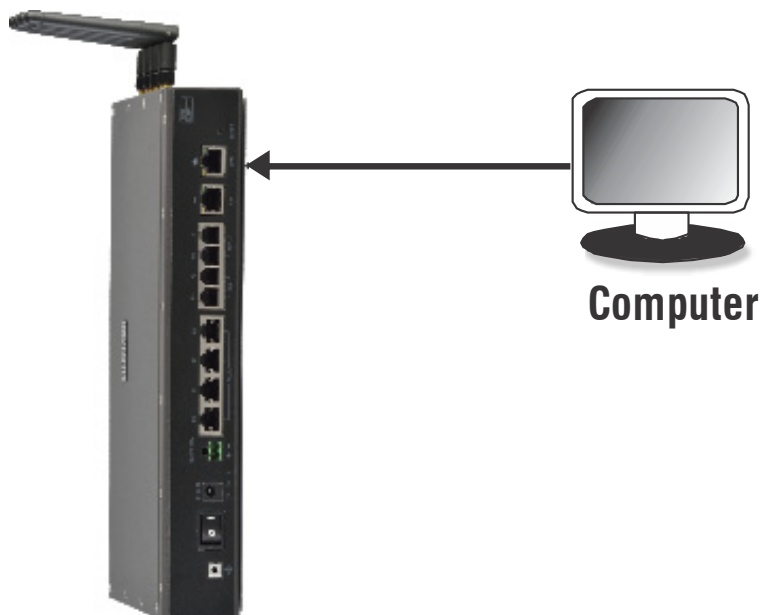
You may connect a standalone computer to SETU VGRX or grab any computer connected in the same LAN as SETU VGRX.



- *Connect a standalone computer to SETU VGRX, when installing the system for the first time. You may connect it to the LAN after you have finished installation and configuration of the system.*
- *If the computer for accessing Jeeves is connected in a LAN Switch and the WAN Port of SETU VGRX is connected behind a NAT router, make sure that both the LAN and WAN connections are in different Subnets.*

To connect a standalone computer,

- Plug one end of the Ethernet cable supplied with the system into the LAN Port of SETU VGRX. Plug the other end into the LAN Port of the computer.



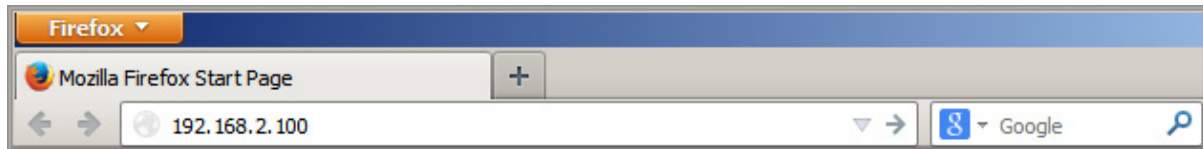
- Make sure the IP Address of the computer and the LAN Port of SETU VGRX do not conflict, and that both are in the same Subnet.

The default IP Address of the LAN Port of SETU VGRX is: **192.168.2.100**

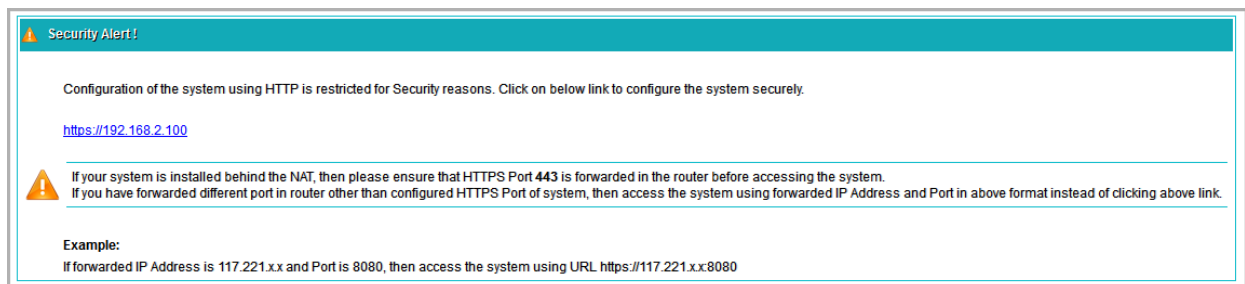
The default Subnet Mask of the LAN Port of SETU VGRX is: **255.255.255.000**

Change the Subnet of the computer, if necessary.

- Make sure a Web-browser, either Internet Explorer version 7 or later or Mozilla Firefox version 3.5 or later, is installed on the computer.
- Open the browser on the computer.
- In the address bar of the browser, enter the default IP address of the LAN Port: **192.168.2.100**.



- You will be redirected to the HTTPS protocol for security reasons.



- Click the <https://192.168.2.100> link.
- The **Login** page will open.
- In **Login Password**, enter **1234**, the default SE Password.
- Click the **Login** button.



- You will be prompted to change the default SE Password.

### Password Change

**Login through default password is not allowed. Change the password to login.**

Current Password


New Password

Confirm New Password

**Note :**

Password must follow following requirements:

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

 Submit

- In **Current Password**, enter the default SE Password.
- Enter the **New Password**. All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ' , Double Quote " and **Space**) and digits 0 to 9 are allowed. The new password must be:
  - a minimum of 6 characters to a maximum of 16 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit**. You will be re-directed to the Login page again.
- In **Login Password**, enter the new password.



*As this password is meant for restricting access to the SE mode, we strongly recommend you to:*

- *Keep the password secret.*
- *Select a complex password that cannot be easily guessed.*
- *Change the password regularly. See "[Login Password](#)" for instructions.*

On successful login, the **Home** page of Jeeves opens.

The left navigation bar displays the links **Basic Settings**, **Advanced Settings**, **Maintenance** and **Status**.



**Basic Settings** break down the complexities of configuration and are sufficient to get your system into operation.

**Advanced Settings** enable you to configure the advanced features and facilities of SETU VGRX.

**Maintenance** allows you to carry out system maintenance and monitoring activities like uploading/upgrading firmware and configuration, system debug, system restart.

**Status** allows you to view the system details and the status of all the ports.

You may now configure the Basic Settings of SETU VGRX.

If you need to change the IP Address and the Subnet Mask of the LAN Port and the WAN Port of SETU VGRX, you may do so by dialing System Commands from the telephone connected to the FXS Port. It is also possible to view the current IP Address and Subnet Mask of the LAN Port and the WAN Port by dialing System Commands from the telephone connected to the FXS Port. For instructions, see "[System Commands](#)" in the *Appendix*.






The Basic Settings enable you to configure SETU VGRX for basic functions. You will be able to operate and use the system efficiently, when you configure Basic Settings.


To configure Basic Settings,

- Click the **Basic Settings** link.

The links to the different basic parameters appear on the left navigation bar.



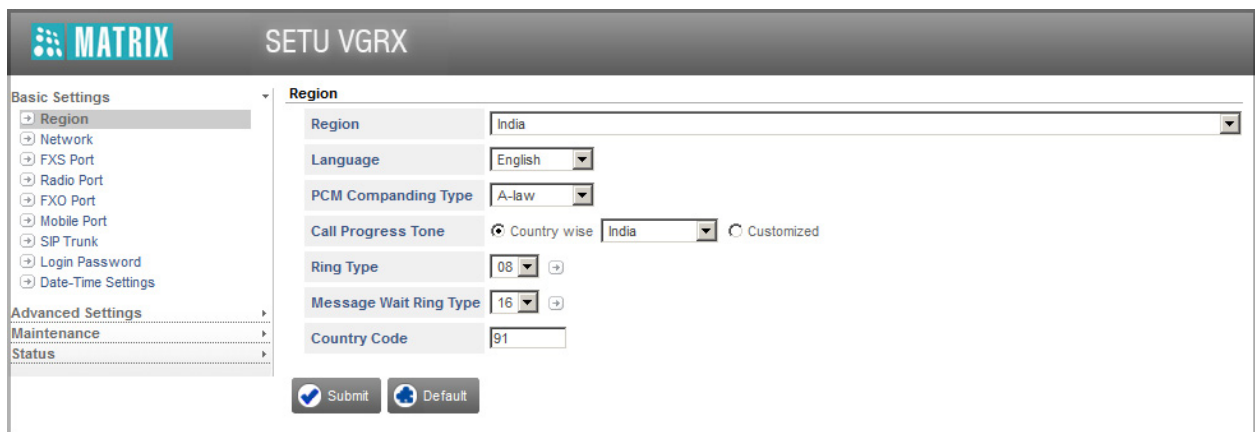
- Click the sub link of the required parameter: **Region, Network, FXS Port, Radio Port, FXO Port, Mobile Port, SIP Trunk, Login Password** and **Date-Time Settings**.
- The selected parameter page opens.
  - Click **Expand**  to expand a link and display all parameters under the link.
  - Click **Collapse**  to collapse a link and hide all parameters under the link.
  - Click **Settings**  to configure / edit the settings of a parameter further.
  - Click the **Submit** button to save changes made on the page.
  - Click the **Default** button to assign factory set values to all the parameters on the page.

- Click the **Add** button to add a new record.
- Click the **Delete** button to delete a record.
- Click the **Close** button to exit a window.
- Click **Logout**  to end the login session and exit Jeeves. You will return to the login page of Jeeves.
- Set the parameters on the page to the desired values and click **Submit** to save.

## Region

To configure Region and other region specific parameters,

- Click the **Region** link.



## Region

- In **Region**, click the name of the country where SETU VGRX is installed. Default: India.

When you change Region, an alert message will appear on the screen **“Changing Region shall assign default values to all parameters of the system. Do you want to continue?”** Click OK. All country specific parameters will be assigned default values. See [“Default Region Table”](#) in the Appendix for country specific default values.

## Language

- In **Language**, click the language in which you want the pages of the GUI, Jeeves, to be presented.

SETU VGRX can display the pages of the GUI, Jeeves, in English, Italian, Spanish, French, German, and Portuguese. Default: English.

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

## PCM Companding Type

- If required, you may change the **PCM Companding Type**—A-law or  $\mu$ -law—set automatically by SETU VGRX according to the Region you have selected. Default: A-law (for India).

## Call Progress Tones

- By default, **Country wise** option is selected for the **Call Progress Tone**. SETU VGRX supports country specific Call Progress Tone Generation (CPTG) to simulate the same tones of the local PSTN to which it is connected. The Call Progress Tones supported by SETU VGRX for different countries is presented in the Appendix.
- To match the call progress tone of the country where SETU VGRX is installed, select the Country from the **Country wise** list. Default: India.

The screenshot displays the 'Region' configuration window. The 'Region' field is set to 'India'. The 'Language' dropdown is set to 'English'. The 'PCM Companding Type' dropdown is set to 'A-law'. The 'Call Progress Tone' section has the 'Country wise' radio button selected, and a dropdown menu is open showing a list of countries: India, Argentina, Australia, Brazil, Canada, China, Egypt, France, Germany, Greece, India (highlighted), Indonesia, Iran, Iraq, Israel, Italy, Japan, and Kenya. The 'Customized' radio button is unselected. The 'Ring Type' dropdown is set to '08' and the 'Message Wait Ring Type' dropdown is set to '16'. The 'Country Code' field contains '91'. At the bottom left, there are 'Submit' and 'Default' buttons.

- If you want to change the cadence of the Call Progress Tones as per your requirement, select the **Customized** option.

- To customize the Call Progress Tones cadence, click **Settings** .

Region

Region

India

Language

English


PCM Companding Type

A-law

Call Progress Tone


☐ Country wise

India

Customized 


Ring Type

08




Message Wait Ring Type


16













Country Code


91


 Submit


 Default

- The **Call Progress Tone Cadence Table** opens.

Tone Type	Frequency1 (Hz)	Operator	Frequency2 (Hz)	Cadence					
				ON Time1 (msec)	OFF Time1 (msec)	ON Time2 (msec)	OFF Time2 (msec)	ON Time3 (msec)	OFF Time3 (msec)
Dial Tone	400	* 	25	9999	0	0	0	0	0
Ring Back Tone	400	* 	25	400	200	400	2000	0	0
Busy Tone	400	No 	0	750	750	0	0	0	0
Error Tone 1	400	No 	0	250	250	0	0	0	0
Confirmation Tone	400	No 	0	100	100	0	0	0	0
Feature Tone/ Programming Tone	400	* 	25	100	900	0	0	0	0
Intrusion Tone	400	No 	0	150	4850	0	0	0	0
Error Tone 2	400	No 	0	1000	1000	0	0	0	0
Routing Tone	400	* 	25	100	1900	0	0	0	0
Stuttered Dial Tone	425	No 	0	100	100	100	100	1000	1000

 Submit

 Default

 Close

Configure the following parameters:


- Frequency1 (Hz):** Configure frequency1. The range of frequency1 is 300-1400 Hz for all tones.
- Frequency2 (Hz):** Configure frequency2. The range of frequency2 is 20-1400 Hz for all tones.
- Operator:** Operator parameter has three options:
  - No:** If No is programmed, Frequency2 will not be applicable.
  - \* (Modulation):** If '\*' (Modulation) is programmed, Frequency1 and Frequency2 will be used as modulation ( $F1 * F2$ ).

- 3) + (Addition):** If '+' (Addition) is programmed, Frequency1 and Frequency2 will be used as addition (F1 + F2).
- **Cadence:** Program Cadence ON Time1-OFF Time1, ON Time2-OFF Time2 and ON Time3-OFF Time-3 for all tones. Valid ON Time and OFF Time range for all tones is 0000-9999 msec.
  - Click **Submit**.
  - Close the window to return to the **Region** page.






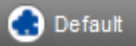
When you submit the page after changing the Region or PCM Companding Type or Call Progress Tone, an alert message will appear **“Submitting this page will restart the system. Do you want to continue?”** Click OK. SETU VGRX will restart and your changes will be saved.

## Ring Type

- If required, you may change the **Ring Type** set automatically by SETU VGRX for the Region you have selected. Default: 08 (India).
- To change the Ring Type, click **Settings** .


Region

Region	India	
Language	English	▼
PCM Companding Type	A-law	▼
Call Progress Tone	<input checked="" type="radio"/> Country wise India ▼ <input type="radio"/> Customized	
Ring Type	08 ▼	
Message Wait Ring Type	16 ▼	
Country Code	91	

The **Ring Type** table opens.

Ring Type							
Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, Korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	

 Close


The table presents you with the number of **Ring Types**, **1 to 16**, supported by the system, the **Ring Cadence** of each Ring Type, and the countries where each Ring Type is supported.

- Note the Ring Type number that you want to assign.
- Close the window to return to the **Region** page.
- In **Ring Type**, click the Ring Type number of your choice.

## Message Wait Ring Type

- This parameter is related to the [“Message Wait Indication on SIP Trunks”](#) feature. When you select *Message Wait Notification* type as *Ring* on the FXS Port, a Short, Fast ring is played to indicate the arrival of a new message.

If required, you may change the **Message Wait Ring Type**. By default, it is set to 16 for all Regions.

- To change the Message Wait Ring Type, click **Settings** .

Region

Region

India

Language

English

PCM Companding Type

A-law

Call Progress Tone

☒ Country wise

India

☐ Customized

Ring Type

08

+

Message Wait Ring Type

16

+

Country Code

91

Submit

Default

The **Ring Type** table opens. SETU VGRX supports 16 different Ring Types.

Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	

Close

- Note the Ring Type number that you want to assign.
- Close the window to return to the **Region** page.
- In **Message Wait Ring Type**, click the Ring Type number of your choice.

## Country Code

- If required you may change the **Country Code**, set automatically by SETU VGRX for the Region you have selected. Default: 91 (India).

If you have kept **Remove Country Code from CLI received** check box enabled in the System Parameters, the system will remove the Country Code configured here from the CLI received on the source port.

- Click the **Submit** button to save.

# Network Parameters

SETU VGRX may be installed typically, in a Public IP Network or in a Private network, behind a NAT Router.

When SETU VGRX is installed in a Public IP Network,

- the WAN Port of SETU VGRX is connected to a Broadband Router/Modem.
- Public IP is assigned to the WAN Port.

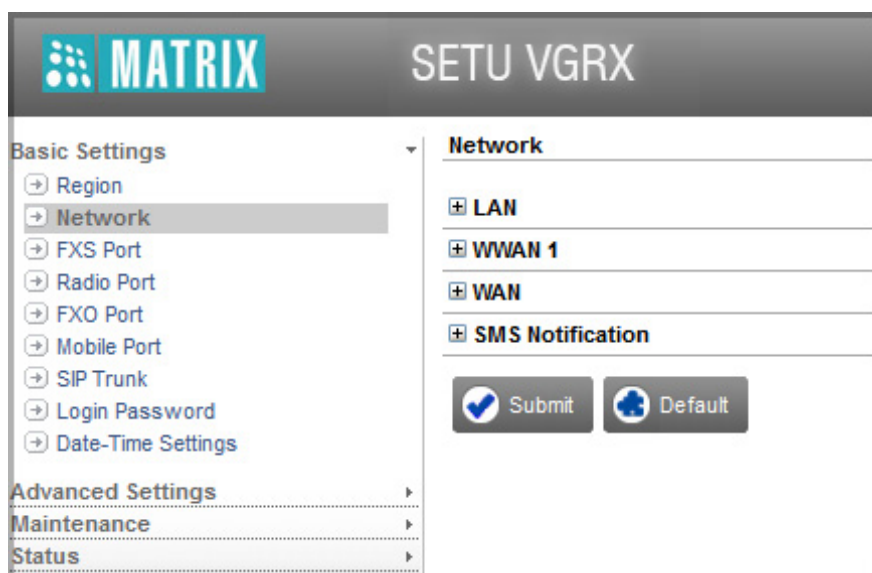
When SETU VGRX is installed in a Private Network, behind a NAT Router,

- the WAN Port of SETU VGRX is connected to the LAN Switch/Hub.
- Private IP is assigned to the WAN Port.

Depending on your installation scenario, configure the Network Port Parameters.

To configure Network parameters,

- Click the **Network** link. The Network Parameters page opens.



## LAN

- Click **LAN** to expand.

This is a detailed view of the LAN configuration section. It features two input fields: 'IP Address' and 'Subnet Mask'. The IP Address field is populated with the values 192, 168, 2, and 100, separated by dots. The Subnet Mask field is populated with the values 255, 255, 255, and 0, also separated by dots.

- In **IP Address**, the current IP Address of the LAN Port is displayed. Default: **192.168.2.100**
- In **Subnet Mask**, the current Subnet Mask of the LAN Port is displayed. Default: **255.255.255.000**

If required, you may change the LAN Port IP Address and Subnet Mask.



*When your SETU VGRX is installed in a Private Network, make sure the LAN Port and the WAN Port are connected in different subnets.*

## WWAN (Wireless WAN)

If you are using the Mobile Port 1 of SETU VGRX as your WAN interface, configure WWAN.



*SETU VGRX supports WWAN connection on Mobile Port 1 only.*

- To configure WAN interface on Mobile Port 1, click **WWAN 1** to expand.

Access Point	<input type="text"/>
Number to Dial	<input type="text"/>
User Name	<input type="text"/>
Password	<input type="password"/>
Fallback DNS Server Address	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Data Usage Allowed	<input type="text" value="999"/> <input type="text" value="GB"/>
Reset Data Usage Consumed on Scheduled Date	<input checked="" type="checkbox"/> Yes
Schedule Date	<input type="text" value="01"/>
Send SMS when 75% of Allowed Data Usage is Consumed	<input type="checkbox"/> Yes
When Allowed Data Usage is Consumed	<input type="text" value="No action"/>
Data Usage Consumed	<input type="text" value="0.00000 bytes"/> <input type="button" value="Reset Consumed Data"/>

- Enter the following information:
  - In **Access Point**, enter the access point provided by your Service Provider.
  - Enter the **Number to Dial** provided to you for the internet service by your Service Provider.
  - Enter the **User Name** provided to you for accessing the internet service by your Service Provider.
  - In **Password**, enter the authentication password for the User ID provided by your Service Provider.
  - When we configure the Mobile Port as the WAN interface, the system fetches the DNS Server Address of the Service Provider for connectivity. In **Fallback DNS Server Address**, enter the DNS Server Address that you want the system to use, when the DNS Server Address of the Service Provider fails to provide connectivity.
  - In **Data Usage Allowed**, enter the data usage as per the scheme provided by your Service Provider. Default: 999 GB.

- Clear the **Reset Data Usage Consumed on Scheduled Date** check box, if you do not want SETU VGRX to reset the data usage consumed on a particular date. Default: Enabled i.e. the system will automatically reset the value of data usage consumed on a Scheduled Date.
- In **Schedule Date**, configure the date on which you want the system to reset the data usage consumed. Default: 01.
- Select the **Send SMS when 75% of Allowed Data Usage is Consumed** check box, if you want SETU VGRX to send SMS to the pre-configured numbers, when 75% of the allowed data usage is consumed. Default: Disabled.

If you have enabled this parameter, make sure you have configured the mobile numbers in **Send SMS if 75% of Allowed Data Usage is Consumed** under [“SMS Notification”](#).



**SMS facility will not be supported, if a CDMA module is installed in your SETU VGRX.**

- You may select one of the following actions to be taken by SETU VGRX, **When Allowed Data Usage is Consumed**.
  - No action
  - Disconnect
  - Send SMS
  - Send SMS and Disconnect

Default: No action.

If you select **Send SMS** or **Send SMS and Disconnect** option, make sure you have configured the mobile numbers in **Send SMS if Allowed Data Usage is Consumed** under [“SMS Notification”](#).



**SMS facility will not be supported, if a CDMA module is installed in your SETU VGRX.**

- In **Data Usage Consumed**, total Data consumed on a Mobile Port is displayed.
- Click the **Reset Consumed Data** button, to reset the Data Usage Consumed manually.

## WAN

- Click **WAN** to expand.



## Connection Type

- Click **Connection Type**.

Connection Type

Connection Type ☐ DHCP ☐ PPPoE ☒ Static

Static IP

IP Address	192	168	1	100
Subnet Mask	255	255	255	0
Gateway	192	168	1	254

- Select the network connection type, that is, the IP Addressing Scheme used by your network to assign the IP address to the WAN Port: Static, DHCP, PPPoE. Default: Static.
- Static:** If your network uses Static IP addressing, select Static and configure the following parameters.
  - In **IP Address**, enter the IP Address you obtained from your Network Administrator for the WAN Port of SETU VGRX. Make sure that the IP Address does not conflict with that of any other device on the LAN. Default: 192.168.1.100
  - In **Subnet Mask**, enter the Subnet Mask you obtained from your Network Administrator for the WAN Port. Default: 255.255.255.0
  - In **Default Gateway**, enter the IP Address of the Router's LAN Interface as the Default Gateway IP Address.
- DHCP:** Whenever SETU VGRX is restarted, the DHCP server will dynamically assign an IP Address, Subnet Mask and Gateway Address to the WAN Port, if your network uses DHCP Addressing. You have to configure the Domain Name Server (DNS) Address only, if not already provided by your Internet Service Provider.
- PPPoE:** If your network uses PPPoE addressing, the PPPoE server will automatically assign an IP Address, Subnet Mask and Gateway Address to the WAN Port of SETU VGRX. You need to configure the following parameters provided by your Internet Service Provider:
  - In **PPPoE User ID**, enter the User Name provided by the Internet Service Provider. The User ID may be a maximum of 64 characters.
  - In **PPPoE Password**, enter the User Password provided by the Internet Service Provider. The password may be a maximum of 64 characters.
  - In **PPPoE Service Name**, enter the Service Name, if provided by your Internet Service Provider. The Service Name may consist of a maximum of 64 characters. If Service Name is not provided, leave this field blank.

## DNS Setting

Configure the Domain Name Server (DNS) settings as provided by your Internet Service Provider. You may consult your Network Administrator in this regard.

- Click **DNS Setting**.

- Select **DNS Server** as **Automatic** or **Static** according to the Connection Type (IP Addressing scheme) used by the network.
- Select **Static** if:
  - your network uses Static IP Addressing.
  - your network uses DHCP or PPPoE, but the DHCP/PPPoE server does not provide DNS Address automatically.

If your network does not assign DNS Address automatically, set DNS Address Assignment as **Static**. In the **DNS Address**, enter the DNS Server Address. Enter **DNS Domain Name**, if provided to you by your Network Administrator.

- Select **Automatic** if:
  - your network uses DHCP or PPPoE IP Addressing.
  - the DHCP/PPPoE server of your network assigns the DNS Address automatically.

## Dynamic DNS (DynDNS.org)

Dynamic DNS (DDNS) is a service that maps internet domain names to IP addresses. DDNS Service Provider provides the host name/domain name to the internet devices and also embeds DDNS client in the internet device. By doing so, whenever a new IP Address is assigned to the internet host, the DDNS client running in the internet host updates its new IP address in the Dynamic DNS server.

When the WAN Port of SETU VGRX is assigned a dynamic IP, its new IP Address needs to be updated regularly with the various devices or networks which utilise the WAN Port settings to function. Dynamic DNS resolves this by mapping a domain name to the WAN Port IP Address, which SETU VGRX can update in the Dynamic DNS Server.

Once the IP Address of the system is updated in the DNS server, any caller on the IP network can reach the system by dialing the host name/domain of the system.

SETU VGRX supports Dynamic DNS Server client of the Service Provider Dynamic DNS.org. To use this service, you must first register with DynDNS.org and then do the following:

- Click **Dynamic DNS (DynDNS.org)**.

- Select the **Dynamic DNS Enable** check box.
- Enter the **User Name** you created on DynDNS.org. The name can be of maximum 40 characters.
- Enter the **Password** you created for the User Name on DynDNS.org. The password can be of maximum 24 characters.
- Enter the **Host Name** you created on the DynDNS.org here. The Host Name can be of maximum 40 characters.

## VLAN/CoS

If SETU VGRX is connected in a VLAN, configure the **VLAN/CoS**. This parameter enables the SETU VGRX to add VLAN header to the packets generated by it. The VLAN header consists of the VLAN ID (12-bit) and Class of Service (CoS, 3-bit) for prioritization of traffic<sup>1</sup>.

- Click **VLAN/CoS (Layer 2)**.

- Select the **VLAN/CoS** check box to enable VLAN ID tagging on all packets generated by the system. Default: Disabled.
- Enter the **VLAN ID** that you have assigned to the VLAN in which the SETU VGRX is connected. The valid range for this is 0-4094. Default: 1.
- For **SIP CoS**, define the CoS (priority) bits which will be added in all SIP packets. The range of CoS bits is from 0 to 7. Default: 3
- For **RTP CoS**, define the CoS (priority) bits which will be added in all RTP packets. The range of CoS bits is from 0 to 7. Default: 6.

## QoS (Layer 3)

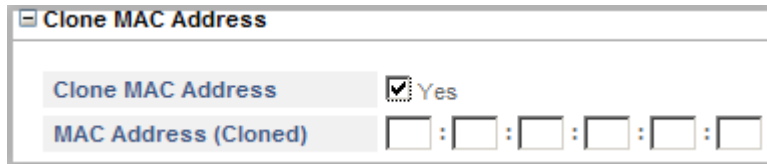
- Click **QoS (Layer 3)**.

1. 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.

- SETU VGRX will send all SIP messages using SIP QoS setting, enter the **SIP DiffServe/ ToS** as per your requirement. The valid range is from 00-63, Default: 26.
- SETU VGRX will send all the RTP packets with RTP QoS setting, enter the **RTP DiffServe/ ToS** as per your requirement. The valid range is from 00-63, Default: 46.

## Clone MAC Address

- Click **Clone MAC Address**.



- If you want to clone the MAC address, select the **Clone MAC Address** check box.

In **MAC Address (Cloned)**, enter the desired MAC address you want to clone in hexadecimal format, e.g. 00:50:c2:55:b0:10.

## SMS Notification

SETU VGRX supports Notification via Short Message Service (SMS) to inform you of the following conditions and events:

- When the WAN Port or WWAN Port link is down.
- Whenever there is a change in the IP Address of the system.
- SIM Balance Inquiry.
- Call Minutes Consumed.
- Data Usage Consumed.



**SMS facility will not be supported, if a CDMA module is installed in your SETU VGRX.**

To use SMS Notification,

- Click **SMS Notification**.

**SMS Notification**

**Send SMS if WAN Port link is down** ☐ Yes

**Send SMS if WWAN Port link is down** ☐ Yes

**Send SMS if IP Address of System is changed** ☐ Yes

**Send SMS if 75% of Allowed Data Usage is Consumed**

**To**

**SMS Text**

**Using**

Note: use [a] for Data Usage Allowed, [c] for Data Usage Consumed and [xx] for Used WWAN Port.

**Send SMS if Allowed Data Usage is Consumed**

**To**

**SMS Text**

**Using**

Note: use [a] for Data Usage Allowed and [xx] for Used WWAN Port.

- To request SMS Notification for **WAN Port Link** Status, select the **Send SMS if WAN Port link is down** check box.

**Send SMS if WAN Port link is down** ☒ Yes

**To**

**SMS Text**

**Using**

- In the **To** fields, you may enter up to 3 Mobile Numbers to which the SMS should be sent. The numbers can be a maximum of 24 characters. The characters allowed are 0-9, \*, # and +.
- In **SMS Text**, enter the text you want to be sent in the Notification. The text length can be a maximum of 80 characters.
- In the **Using** list, select the Mobile Port number which the system should use to send the SMS.
- Click **Submit** to save.
- Similarly, configure the parameters for:
  - **Send SMS if WWAN Port link is down**
  - **Send SMS if IP Address of System is changed**
- If you want SETU VGRX to **Send SMS if 75% of Allowed Data Usage is consumed**, configure the following parameters.

Send SMS if 75% of Allowed Data Usage is Consumed		
To	<input type="text"/>	
SMS Text	<input type="text" value="Update from Matrix-SETUVGRX : Data Usage of [xx] is 75% consumed ([c]/[a])."/>	
Using	<input type="text" value="Any Mobile Port"/>	
Note: use [a] for Data Usage Allowed, [c] for Data Usage Consumed and [xx] for Used WWAN Port.		

- In the **To** fields, you may enter up to 3 Mobile Numbers to which the SMS should be sent. The numbers can be a maximum of 24 characters. The characters allowed are 0-9, \*, # and +.
- In **SMS Text**, enter the text you want to be sent in the Notification. The text length can be a maximum of 80 characters.
- In the **Using** list, select the Mobile Port number which the system should use to send the SMS.
- Click **Submit** to save.
- If you want SETU VGRX to **Send SMS if Allowed Data Usage is consumed**, configure the following parameters.

Send SMS if Allowed Data Usage is Consumed		
To	<input type="text"/>	
SMS Text	<input type="text" value="Update from Matrix-SETUVGRX : Data Usage [a] for [xx] is consumed."/>	
Using	<input type="text" value="Any Mobile Port"/>	
Note: use [a] for Data Usage Allowed and [xx] for Used WWAN Port.		

- In the **To** fields, you may enter up to 3 Mobile Numbers to which the SMS should be sent. The numbers can be a maximum of 24 characters. The characters allowed are 0-9, \*, # and +.
- In **SMS Text**, enter the text you want to be sent in the Notification. The text length can be a maximum of 80 characters.
- In the **Using** list, select the Mobile Port number which the system should use to send the SMS.
- Click **Submit** to save.

To request SMS Notification of *SIM Balance Inquiry*, see [“SIM Balance Inquiry”](#) under [“Mobile Port”](#).

To request SMS Notification for *Call Minutes Consumed*, see [“Call Minutes”](#) under [“Mobile Port”](#).

When you finish configuring all the Network parameters as per your requirement,

- Click **Submit**.
- You will get this message **“Ongoing calls would be disconnected. Do you want to submit this page?”**
- Click **Yes** to save your settings.

## Restoring Default LAN IP Address

You can restore the Default LAN IP Address using the Reset Button. To do so,

- Press the Reset button for more than four seconds.
- Release the Reset button.

The LAN IP Address will be restored to default, **192.168.2.100**



- *If you press the Reset button for less than four seconds, SETU VGRX will restart.*
- *Along with the LAN IP Address, a few other parameters will also be set to default. See [“Restoring Default Settings using the Reset button”](#) for details.*

# FXS Port

SETU VGRX supports two FXS Ports. You can connect a PBX or any standard telephone instrument to the FXS Port.

To configure the parameters of the FXS Port,

- Click the **Basic Settings** link to expand.
- Click the **FXS Port** link.

Port	Enable	Name	Number	CLI Type	Outgoing Call Routing
<a href="#">FXS-1</a>	<input checked="" type="checkbox"/>		2001	FSK V.23	Route calls using port determined by <a href="#">Destination Number based Routing table</a>
<a href="#">FXS-2</a>	<input checked="" type="checkbox"/>		2002	FSK V.23	Route calls using port determined by <a href="#">Destination Number based Routing table</a>

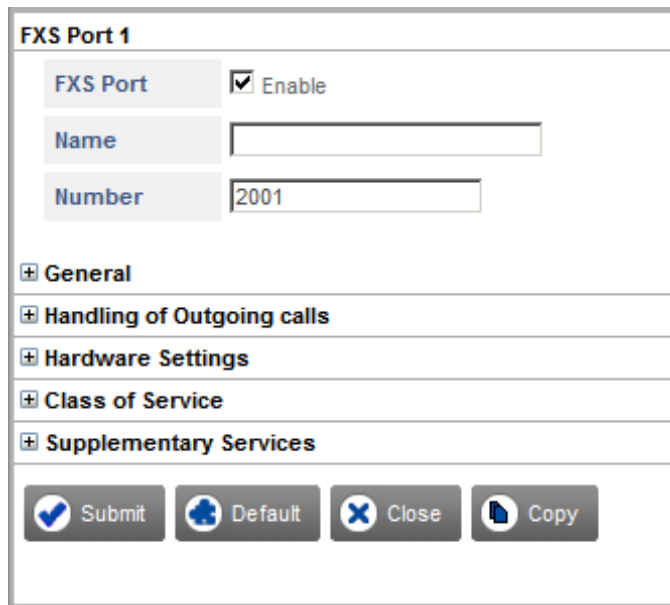
The FXS Port page displays the following parameters:

- **Port:** Displays the FXS Port numbers. To configure the FXS Port parameters, click on the desired FXS Port number link.
- **Enable:** Keep the **FXS Ports** enabled. Clear the FXS Port **Enable** check box, only if you do not want to use the Port.
- **Name:** You may assign a Name to the FXS Port, for identification.
- **Number:** You can assign a Number to the FXS Port, for identification.
- **CLI Type:** Displays the type of CLI selected for the FXS Port.
- **Outgoing Call Routing:** Displays the Outgoing Call Routing Method selected for the FXS Port.

To configure the FXS Port parameters,

- Click **FXS-1**.

The **FXS Port 1** window opens.



- Keep the **FXS Port** check box enabled.

Clear the **FXS Port Enable** check box, only when you do not want to use this FXS Port. Default: Enabled.

- You may assign a **Name** to the FXS Port. When a call is made through this port, the name you assigned will appear on the phone of the called party, if it supports CLI display.

The name you assign may consist of a maximum of 24 characters. Default: Blank.

- You can assign a **Number** to the FXS Port. When a call is made through this port, the number you assigned will appear on the phone of the called party, if it supports CLI display.

The number you assign to the FXS Port can have a maximum of 16 characters. Valid characters are 0 to 9, \*, # and +. Default: 2001.

## General

- Click **General**.

General	
CLI Type	FSK V.23
Answer Signaling	Battery Reversal
Disconnect Signaling	Battery Reversal
Flash Timer	600 msec
Message Wait Notification	LED Lamp (HV)
Call Pick-up Group	1
Automatic Number Translation(ANT) for Calling Number	<input type="checkbox"/> Apply

- Select the appropriate **CLI Type**, according to the CLI Type supported by the telephone instrument/PBX connected to the FXS Port.

SETU VGRX supports three signaling protocols for CLI on the FXS Port — DTMF, FSK V.23 and FSK BellCore. Default: FSK V.23

- Select the appropriate **Answer Signaling** Type on the FXS Port.

An Answer Signal is a signal generated on the FXS Port to indicate that the called party has answered (call maturity).

- Select **None**, if no answer signaling is to be generated on the FXS Port.
- Select **Battery Reversal**, if answer signaling is to be generated in the form of Battery Reversal on the FXS Port.

Default: Battery Reversal

- Select the appropriate **Disconnect Signaling** Type on the FXS Port.

A Disconnect Signal is the signal generated on the FXS Port to indicate that the called party has disconnected the call.

- Select **None**, if no signaling is to be generated on the FXS Port for call disconnection.
- Select **Battery Reversal**, if call disconnection is to be signaled in the form of Battery Reversal.
- Select **Open Loop Disconnect**, if call disconnection is to be signaled in the form of Open Loop Disconnect signal. If you select this option, you must configure the Open Loop Disconnect Timer.
  - Set the duration of **Open Loop Disconnect Timer** as per your requirement. Valid range is 001 to 999 msec. Default: 500 msec.

Default: Battery Reversal.

- Set the duration of the **Flash Timer**. This is the time for which Flash will be detected on the FXS Port. SETU VGRX uses this event to activate various features like Call Hold, Call Transfer. Valid range is 100 to 900 msec. Default: 600 msec.
- If you have subscribed to *Message Wait Indication* for the voicemail service from your ITSP, and have selected this FXS Port as the destination for receiving Message Wait Indication, you may select the desired type of **Message Wait Notification** from the following options.
  - Select **Stuttered Dial Tone**, if you want new message indication in the form of a stuttered dial tone, whenever the user picks up the phone connected to the FXS Port.
  - Select **LED Lamp (HV)**, if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using High Voltage.
  - Select **Ring**, if you want the arrival of a new message to be indicated by the *Message Wait Ring* (a Short, Fast ring).

You can select a different Ring Type to indicate message wait. For instructions, see [“Message Wait Ring Type”](#).

You can also set the duration for which the ring is to be played (Ring Timer), the number of times the ring is to be played (Ring Count) and the interval between rings (Ring Interval). For instructions, see [“Message Wait”](#) in [“System Parameters”](#).

- Select **LED Lamp (FSK)**, if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using FSK CLI.
- Select **Stuttered Dial Tone + LED Lamp (HV)**, if you want new message indication on the LED Lamp using High Voltage as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.
- Select **Stuttered Dial Tone + LED Lamp (FSK)**, if you want new message indication on the LED Lamp using FSK CLI as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

Default: LED Lamp (HV)

Whenever a new message arrives in the Mailbox of the SIP Trunk, SETU VGRX gives notification to this (destination) FXS Port according to the type of *Message Wait Notification* you select.

To know more about this feature and the Notification options, see [“Message Wait Indication on SIP Trunks”](#).

- If you want to allow the Call Pick-up feature on this FXS Port, assign the FXS Port to a **Call Pick-up Group**.

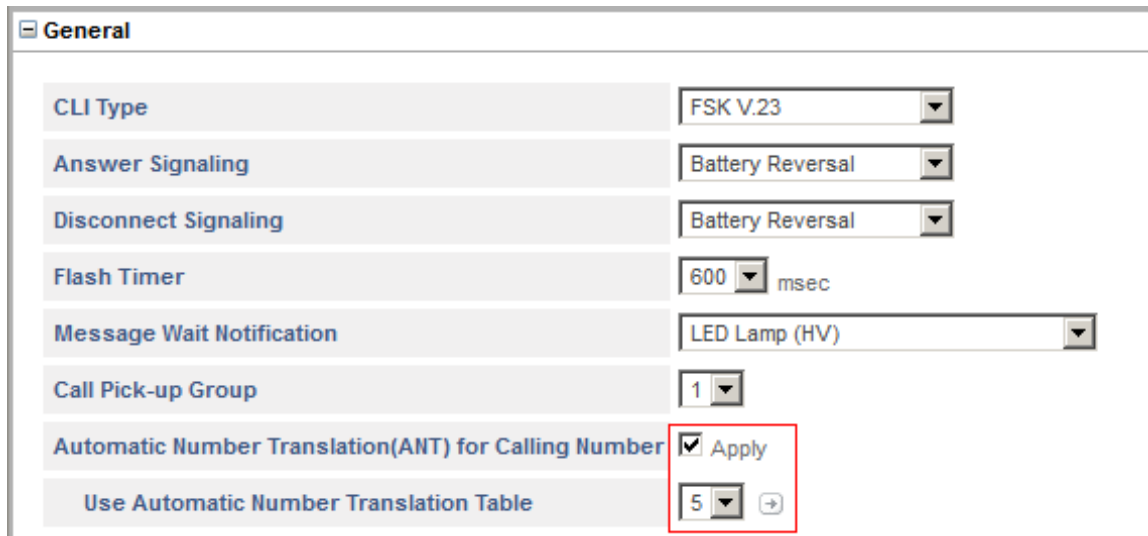
Call Pick-up feature allows the FXS Port user to answer calls that are ringing on any FXS Port in the same Call Pick-up Group, by dialing an access code. To know more about this feature, see [“Call Pick-up”](#).

By default, all the FXS Ports are assigned to group 1.

Make sure that the Call Pick-up feature is enabled in the *Class of Service* of all the FXS Ports in the group. See [“Class of Service”](#).

'0' is used to de-assign an FXS Port from a Call Pick-Up Group.

- You can apply Automatic Number Translation (ANT) logic on CLI number when call is to be placed on the FXS Port.
- To apply ANT logic on the Calling Numbers, select the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.



The screenshot shows a configuration window titled "General". It contains several settings for FXS ports:

- CLI Type: FSK V.23
- Answer Signaling: Battery Reversal
- Disconnect Signaling: Battery Reversal
- Flash Timer: 600 msec
- Message Wait Notification: LED Lamp (HV)
- Call Pick-up Group: 1
- Automatic Number Translation(ANT) for Calling Number: ☒ Apply
- Use Automatic Number Translation Table: 5

The "Apply" checkbox and the "5" dropdown are highlighted with a red box.

- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

If you have not configured the Automatic Number Translation Table,

- Click **Settings** . The **Automatic Number Translation Table** window opens.




1 2 3 4 5 6 7 8

**Automatic Number Translation Table - 5**

Index	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	

**Examples of Number Pattern**

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' to every 7-digit dialed number.

 Submit
 Default
 Close

- You may configure the default Automatic Number Translation Table or any other Table. See [“Automatic Number Translation \(ANT\)”](#) to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Click **Submit**.

## Handling of Outgoing Calls

- Click **Handling of Outgoing Calls**.

- If you do not want to route calls through this port, select the **Block all calls through this FXS Port** check box. Default: Disabled.

### Destination Port Determination

- In **Select Destination Port for routing calls**, select the method to be used for determining the Destination Port for routing calls from the FXS Port. You may select any one of these options:
  - Fixed
  - on the basis of Destination Number

Default: on the basis of Destination Number

Read further for instructions on selecting and configuring each of these destination port determination methods.



*If the destination number to be dialed out is an IP Address, SETU VGRX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. See ["IP Dialing"](#) for more details.*

#### Fixed

In this method, outgoing calls made from the FXS Port are routed to a Fixed Destination Port, irrespective of the number dialed from the FXS Port.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.



- Click **Settings**  .

Handling of Outgoing calls

Block all calls through this FXS Port

☐ Yes



Select Destination Port for routing calls

Fixed  

Allowed-Denied Logic

☐ Apply

Dial Plan

1  


First Digit Wait Timer

7 Seconds


Inter Digit Wait Timer

5 Seconds


End Of Dialing Digit

# 


Minimum Number of digits that must be dialed by the caller

02 

Maximum Number of digits that can be dialed by the caller


24 

Subscriber Type

Gateway 


The **Destination Port/Group for FXS Port** window opens.

Destination Port/Group for FXS Port

Edit	Routing Group	Fallback Routing Group	CLI Number on FXS Port
	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

 Close

The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to edit the default Routing Group options, click **Settings**  .

The **Edit Selective Port/Group for FXS Port** window opens.


- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.
 

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports*, *Radio Ports* and *SIP Trunks*.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default:1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
  - Click **Submit** to save changes. The **Edit** window closes.
  - The entry you edited appears in the **Destination Port/Group for FXS Port** window.
  - Close the **Destination Port/Group for FXS Port** window to return to the Handling of Outgoing Calls window.

### On the basis of Destination Number

In this method, outgoing calls made from the FXS Port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SETU VGRX will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.


- Click **Settings**  .

Handling of Outgoing calls


Block all calls through this FXS Port

☐ Yes

Select Destination Port for routing calls

On the basis of Destination Number 


If no match found in the Destination Number Table

Route calls to Fixed Port 

Allowed-Denied Logic

☐ Apply

Dial Plan

1 


First Digit Wait Timer

7 Seconds


Inter Digit Wait Timer

5 Seconds


End Of Dialing Digit

# 


Minimum Number of digits that must be dialed by the caller

02 



Maximum Number of digits that can be dialed by the caller

24 

Subscriber Type

Gateway 

The **FXS Port - Destination Port Determination - Destination Number Based** table window opens.

<input type="checkbox"/>	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
<input type="checkbox"/>		2001	FXS Port 1 - 1 (Ascending)	None	Received Calling Party
<input type="checkbox"/>		2002	FXS Port 2 - 2 (Ascending)	None	Received Calling Party


Total Records : 2


1


Testing


Enter the destination number to know which entry would be selected for routing

Search

 Add

 Delete

 Close

- The table displays 2 default entries, which you can edit as per your requirement.
- Click **Settings**  to edit the first entry in the table.

- The **Edit Entry** window opens.

**Edit Entry**

Destination Number
2001

CLI Number to be sent on Destination Port
Received Calling Party

**Routing Group**

☒ FXS Port
1 to 1 in Ascending order

☐ FXS Group
1

☐ FXO Port
1 to 1 in Ascending order

☐ FXO Group
1

☐ Mobile Port
1 to 1 in Ascending order

☐ Mobile Group
1

☐ SIP Trunk
1 to 1 in Ascending order

☐ SIP Group
1

☐ Radio Port
1 to 1 in Ascending order

**Fallback Routing Group**
☐ Apply

☒ FXS Port
1 to 1 in Ascending order

☒ FXS Group
1

☒ FXO Port
1 to 1 in Ascending order

☒ FXO Group
1

☒ Mobile Port
1 to 1 in Ascending order

☒ Mobile Group
1

☒ SIP Trunk
1 to 1 in Ascending order

☒ SIP Group
1

☒ Radio Port
1 to 1 in Ascending order

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + “Wildcard Characters”). Valid characters are 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

## Wildcard Characters

SETU VGRX supports following characters.

Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*	When * is configured in a number string, it will not be considered as End of Dialing.
+	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
[ - ]	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
[ , ]	Comma within a bracket is used as a separator between the groups of numbers.


[ ^ ]	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
T (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default:1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.
 

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports*, *Radio Ports* and *SIP Trunks*.
  - To create a group of *not-sequential FXS Ports* as members,
    - Select a **FXS Group**.
    - Select **FXS Group** number. Default:1.
    - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic [“Group”](#) under *Advanced Settings*.
    - Similarly, you can create a group of *not-sequential FXO Ports*, *Mobile Ports* and *SIP Trunks*.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Edit Entry** window closes.
- The entry you edited appears in the **FXS Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to edit another entry in this table.
- To delete an entry, select the check box and click the **Delete** button.



If there are multiple entries in the **Destination Number Based** table, to search a particular entry in the table, under **Testing** enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.


- The table can have a maximum of 100 entries.
- To add a new entry, click **Add**. The **Add Entry** window opens.

- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Destination Number Based** Table from Advanced Settings. For instructions, see [“Destination Port Determination”](#) under Advanced Settings.

- Select a method for routing outgoing calls, if no match is found in the Destination Number Table.

In **If no match found in the Destination Number Table**, select the desired option for routing the call. You may select — Route calls to Fixed Port or Disconnect Call. Default: Route calls to Fixed Port.

If you select *Route calls to Fixed Port*, click **Settings**  to configure the Destination Port/Group for routing the call. For instructions, see [“Fixed”](#).

## Allowed - Denied Logic

You can apply the Allowed-Denied logic on the FXS Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the FXS Port.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the FXS Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SETU VGRX uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

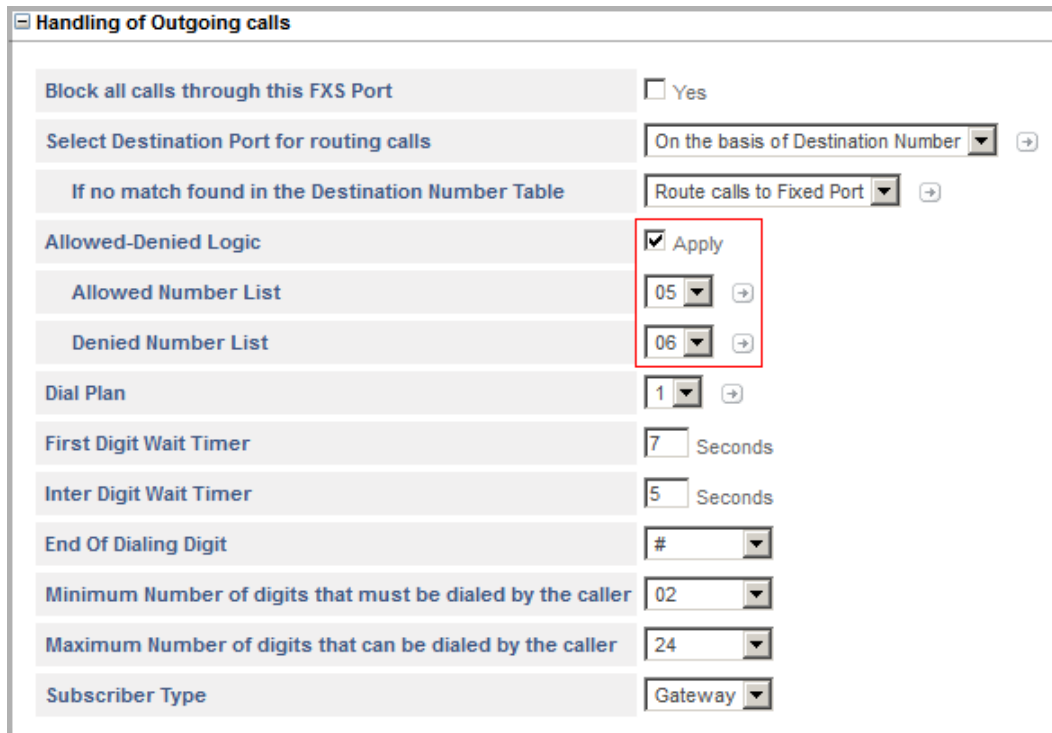
The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:

- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.

To apply Allowed - Denied Logic on the FXS Port,

- Select the **Allowed - Denied Logic** check box.




**Handling of Outgoing calls**


Block all calls through this FXS Port	<input type="checkbox"/> Yes
Select Destination Port for routing calls	On the basis of Destination Number
If no match found in the Destination Number Table	Route calls to Fixed Port
Allowed-Denied Logic	<input checked="" type="checkbox"/> Apply
Allowed Number List	05
Denied Number List	06
Dial Plan	1
First Digit Wait Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	#
Minimum Number of digits that must be dialed by the caller	02
Maximum Number of digits that can be dialed by the caller	24
Subscriber Type	Gateway

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the FXS Port. Default: 05.

If you have not configured the Allowed Number List,

- Click **Settings** . The Number Lists window opens.
- You may configure the default Allowed Number List 5 or any other list. See ["Number Lists"](#) to configure the allowed numbers.
- Click **Submit** to save the Allowed Number List and close the window.
- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the FXS Port. Default: 06.

If you have not configured the Denied Number List,

- Click **Settings** . The Number Lists window opens.
- You may configure the default Denied Number List 6 or any other list. See ["Number Lists"](#) to configure the restrict numbers.
- Click **Submit** to save the Denied Number List and close the window.

## Dial Plan

SETU VGRX supports 8 Dial Plans with total 64 entries in each table. The Dial Plan contains a series of digits and/or wildcard characters.

When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

- To configure the **Dial Plan Table**, click **Settings** ➔ .

The screenshot shows a configuration window titled "Handling of Outgoing calls". It contains several settings:

- Block all calls through this FXS Port**: ☐ Yes
- Select Destination Port for routing calls**: On the basis of Destination Number (dropdown) ➔
- If no match found in the Destination Number Table**: Route calls to Fixed Port (dropdown) ➔
- Allowed-Denied Logic**: ☐ Apply
- Dial Plan**: 1 (dropdown) ➔ (This dropdown is highlighted with a red box in the original image)
- First Digit Wait Timer**: 7 Seconds
- Inter Digit Wait Timer**: 5 Seconds
- End Of Dialing Digit**: # (dropdown)
- Minimum Number of digits that must be dialed by the caller**: 02 (dropdown)
- Maximum Number of digits that can be dialed by the caller**: 24 (dropdown)
- Subscriber Type**: Gateway (dropdown)

- The Dial Plan Table page opens in a new window.

1
2
3
4
5
6
7
8

### Dial Plan Table - 1

Index	Destination Number
01	
02	
03	
04	
05	
06	
07	
08	
09	
10	

#### Testing

Enter the destination number to know which entry would be selected for routing

- You may configure the default Dial Plan Table-1 or any other Table (from 2 to 8) for the FXS Port. You can store 64 Numbers at Index Numbers 01 to 64 respectively.
- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + [“Wildcard Characters”](#)). Valid characters: 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.  
  
See [“Dial Plan”](#) for more details.
- Click **Submit** to save the entries and close the window.
- Return to the Dial Plan parameter and assign the Dial Plan Table you configured.

## First Digit Wait Timer

**Handling of Outgoing calls**

Block all calls through this FXS Port	<input type="checkbox"/> Yes
Select Destination Port for routing calls	On the basis of Destination Number ▼ (+)
If no match found in the Destination Number Table	Route calls to Fixed Port ▼ (+)
Allowed-Denied Logic	<input type="checkbox"/> Apply
Dial Plan	1 ▼ (+)
First Digit Wait Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	# ▼
Minimum Number of digits that must be dialed by the caller	02 ▼
Maximum Number of digits that can be dialed by the caller	24 ▼
Subscriber Type	Gateway ▼

- Set the duration of the **First Digit Wait Timer**. This is the time in seconds for the which the system will wait for the user to dial the destination number. Valid range is 01 to 99 seconds. Default: 7 seconds.

## End-of-Dialing

**Handling of Outgoing calls**

Block all calls through this FXS Port	<input type="checkbox"/> Yes
Select Destination Port for routing calls	On the basis of Destination Number ▼ (+)
If no match found in the Destination Number Table	Route calls to Fixed Port ▼ (+)
Allowed-Denied Logic	<input type="checkbox"/> Apply
Dial Plan	1 ▼ (+)
First Digit Wait Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	# ▼
Minimum Number of digits that must be dialed by the caller	02 ▼
Maximum Number of digits that can be dialed by the caller	24 ▼
Subscriber Type	Gateway ▼

- Set the duration of the **Inter Digit Wait Timer**. This is the time for which you want the system to wait while receiving the digits dialed by the user, to consider it as end-of-dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 5 seconds.

- As **End of Dialing Digit**, select whether the system should consider # or \* as termination digit to detect end of dialing. Default: #
- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number. Valid range is 01 to 24 digits. Default: 2 digits.
- In **Maximum number of digits that can be dialed by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing. Valid range is 01 to 24 digits. Default: 24 digits.

When the caller dials a number, the system will match it with the above end-of-dialing indications and accept the one that matches first.

## Subscriber Type

- Select the **Subscriber Type** for SETU VGRX.

The screenshot shows the 'Handling of Outgoing calls' configuration window. The 'Subscriber Type' dropdown is highlighted with a red box and set to 'Gateway'. Other settings include:

- Block all calls through this FXS Port: ☐ Yes
- Select Destination Port for routing calls: On the basis of Destination Number
- If no match found in the Destination Number Table: Route calls to Fixed Port
- Allowed-Denied Logic: ☐ Apply
- Dial Plan: 1
- First Digit Wait Timer: 7 Seconds
- Inter Digit Wait Timer: 5 Seconds
- End Of Dialing Digit: #
- Minimum Number of digits that must be dialed by the caller: 02
- Maximum Number of digits that can be dialed by the caller: 24
- Subscriber Type: Gateway

When SETU VGRX is interfaced with a Service Provider — ITSP, the Matrix ETERNITY IP-PBX, or any other PBX— you can access the supplementary features supported by the Service Provider as well as the features of the SETU VGRX. These features can be accessed by dialing flash.

By selecting the Subscriber Type, you may choose to access the features of the service provider, or to primarily access the features of SETU VGRX.

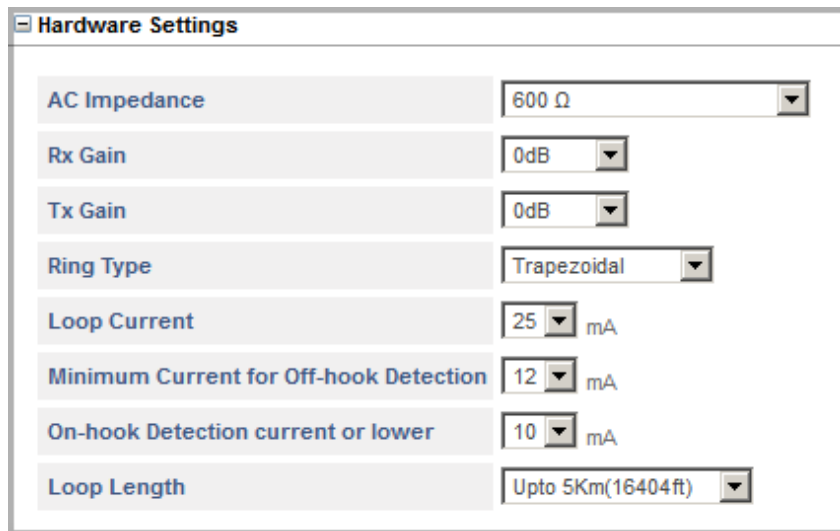
- Select **Network**, if you want to use the supplementary services supported by the PBX. When you set SETU VGRX in the Network mode, you can access the service provider features by dialing flash. You will not be able to access the local features of SETU VGRX.

- Select **Gateway**, if you want to use primarily the supplementary features of SETU VGRX. In the Gateway mode, you will also be able to access the supplementary services of the service provider which require dialing of Flash. To know more, see [“Supplementary Services of Service Provider”](#).

Default: Gateway.

## Hardware Settings

- Click **Hardware Settings**.



The screenshot shows a window titled "Hardware Settings" with the following parameters and their current values:

Parameter	Value
AC Impedance	600 $\Omega$
Rx Gain	0dB
Tx Gain	0dB
Ring Type	Trapezoidal
Loop Current	25 mA
Minimum Current for Off-hook Detection	12 mA
On-hook Detection current or lower	10 mA
Loop Length	Upto 5Km(16404ft)

- In **AC Impedance**, select the appropriate impedance according to the AC Impedance supported by the device connected to the FXS Port of SETU VGRX. The device may be a phone or a PBX.

You may select — 600  $\Omega$ , 900  $\Omega$ , 350  $\Omega$  + (1000  $\Omega$  || 0.21  $\mu$ F), 220  $\Omega$  + (820  $\Omega$  || 120 nF) or 270  $\Omega$  + (750  $\Omega$  || 150 nF). Default: 600  $\Omega$ .

- **Rx Gain** enables you to adjust the volume level of the speech received from the remote party. Select the Rx Gain accordingly. Default: 0dB.
- **Tx Gain** enables you to adjust the volume level of the speech transmitted to the remote party. Select the Tx Gain accordingly. Default: 0dB.
- Select the **Ring Type** to be generated on the FXS Port. You can select — Low Sinusoidal, Low Trapezoidal, Sinusoidal or Trapezoidal. Default: Trapezoidal.
- Select the **Loop Current** according to the *Loop Length* you select. You can select — 25 mA, 30 mA, 35 mA or 40 mA. Default: 25 mA.
- Select the **Minimum Current for Off-hook Detection (mA)** as per your requirement. You can select—10mA, 12mA, 14mA or 16mA. Default: 12 mA.
- Select the **On-hook Detection current or lower** as per your requirement. You can select — 10mA, 12mA, 14mA or 16mA. Default: 10 mA.

- Select the **Loop Length — Upto 5 Km (16404 ft) or Above 5 Km (16404 ft)** — depending on SETU VGRX's installation scenario. The Loop Length is the distance between the SETU VGRX and the telephone instrument connected to the FXS Port. Default: Upto 5 Km (16404 ft).

## Class of Service

- Click **Class of Service**.

**Class of Service**

Hotline	<input type="checkbox"/>	Call Waiting	<input type="checkbox"/>	Conference	<input type="checkbox"/>	Call Pick-up	<input type="checkbox"/>
Call Forward	<input type="checkbox"/>	Call Hold	<input type="checkbox"/>	Blind Transfer	<input type="checkbox"/>	Forced Release Radio Port	<input type="checkbox"/>
Do Not Disturb(DND)	<input type="checkbox"/>	Call Toggle	<input type="checkbox"/>	Attended Transfer	<input type="checkbox"/>		

**Note:** Vocoder on SIP must be same as system companding type (A-law /  $\mu$ -law) for adding SIP party to Conference i.e G.711 (A-law /  $\mu$ -law)

- Select the features of SETU VGRX that you want to allow in **Class of Service<sup>2</sup>** (CoS) of the FXS Port.

By default all the features are denied.

To allow a feature, select the respective check box.

To deny a feature, clear the respective check box.

## Supplementary Services

**Supplementary Services**

Call Waiting	<input type="checkbox"/> Enable
Do Not Disturb(DND)	<input type="checkbox"/> Enable
Call Forward-Unconditional	<input type="checkbox"/> Enable
Call Forward-Busy	<input type="checkbox"/> Enable
Call Forward-NoReply	<input type="checkbox"/> Enable
Hotline	<input type="checkbox"/> Enable

SETU VGRX offers the supplementary features — Call Waiting, Do Not Disturb (DND), Call Forward - Unconditional, Call Forward - Busy, Call Forward - No Reply and Hotline.

- Select the **Enable** check box for the features you want to use.
- By default all the features are disabled.



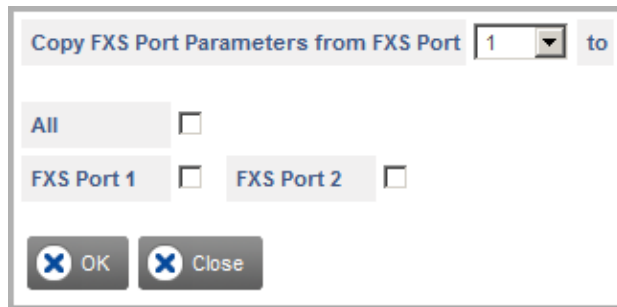
*To use any Supplementary feature, first make sure that you have enabled it in the Class of Service.*

2. Class of Service (CoS) defines the set of features that the phone connected to the FXS Port of SETU VGRX is allowed access to.

- If you have completed the configuration of FXS-1, click **Submit** to save settings.
- To configure the next FXS Port, click the FXS Port number tab and follow the same instructions as given earlier.

## Copy Port Parameters

- You can also copy the settings of a FXS Port to another FXS Port using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy FXS Port Parameters** window opens.



- In **Copy FXS Port Parameters from FXS Port**, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the ports, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the FXS Port you copied the settings to.

# Radio Port

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SETU VGRX supports four Radio Ports. The Radio Port acts as an interface between the system and the Radio devices such as Radio Phones, Radio Repeaters, to add two-way radio functionality in the SETU VGRX. In Two-way radio, the speech can be transmitted as well as received by the radio devices. Radio devices are also called Radio Transceivers /Combat-Net Radio devices. The Two-way radio works on High Frequency (HF), Very High Frequency (VHF) or Ultra High Frequency (UHF).

Generally, the Radio Transceivers remain in the receiving (Rx) mode, so that the broadcasted audio messages can be heard. To transmit speech, you must press the Press to Talk (PTT) button on the Radio Transceiver.

Additional interfaces are also supported on Radio Transceivers like Dial Pad, to dial out the DTMF digits as audio message. Such interfaces can be used where the Radio Transceivers are interfaced with the EPBX, which allows you to dial the DTMF digits.



*Matrix does not supply Radio Transceivers.*

## Direct Speech without System's Intervention

If you do not want system to intervene while transmitting speech between two Radio devices (with or without DTMF dialing), you must do the following configuration in the SETU VGRX.

On the Radio Port page, under **Handling of Outgoing calls**,

- In the **On Detecting Voice, route calls**, select **after collecting the digits**.
- In the **Select Destination Port for routing calls**, select **on the basis of Destination Number**.



*If you want to communicate between two radio devices directly by dialing their respective device number, do not configure such radio device number in the Radio - Destination Port Determination - Destination Number Based table.*

- In the **If no match found in the Destination Number Table**, select **Disconnect Call**.

See ["Handling of Outgoing Calls"](#) for configuration details.

## Configuring Radio Port Parameters

- Click the **Basic Settings** link to expand.
- Click the **Radio Port** link.

**SETU VGRX**

**Basic Settings**

- Region
- Network
- FXS Port
- Radio Port**
- FXO Port
- Mobile Port
- SIP Trunk
- Login Password
- Date-Time Settings

**Radio Port**

Port	Enable	Name	Number	Outgoing Call Routing
<a href="#">Radio-1</a>	<input checked="" type="checkbox"/>		2005	Route calls to FXS Port 1 - 2
<a href="#">Radio-2</a>	<input checked="" type="checkbox"/>		2006	Route calls to FXS Port 1 - 2
<a href="#">Radio-3</a>	<input checked="" type="checkbox"/>		2007	Route calls to FXS Port 1 - 2
<a href="#">Radio-4</a>	<input checked="" type="checkbox"/>		2008	Route calls to FXS Port 1 - 2

The Radio Port page displays the following:

- **Port:** Displays the Radio Port numbers. To configure the Radio Port parameters, click on the desired Radio Port number link.
- **Enable:** Keep the **Radio Ports** enabled. Clear the Radio Port **Enable** check box, only if you do not want to use the Port.
- **Name:** You may assign a Name to the Radio Port, for identification.
- **Number:** You can assign a Number to the Radio Port, for identification.
- **Outgoing Call Routing:** Displays the Outgoing Call Routing Method selected for the Radio Port.

To configure the Radio Port Parameters,

- Click **Radio-1**.

The **Radio Port 1** window opens.

**Radio Port 1**

Radio Port

☒ Enable

Name

Number

- Keep the **Radio Port** check box enabled.

Clear the **Radio Port Enable** check box, only when you do not want to use this Radio Port. Default: Enabled.

- You may assign a **Name** to the Radio Port. The name you assigned will appear on the phone of the called party, if it supports CLI display.

The name you assign may consist of a maximum of 24 characters. Default: Blank.

- You can assign a **Number** to the Radio Port. The number you assigned will appear on the phone of the called party, if it supports CLI display.

The number you assign to the Radio Port can have a maximum of 24 characters. Valid characters are 0 to 9, \*, # and +. Default: 2005.

## Handling of Outgoing Calls

- Click **Handling of Outgoing Calls**.

Block all calls through this Radio port	<input type="checkbox"/> Yes
On Detecting Voice, route call	using Fixed Destination Number ▼
Fixed Destination Number	<input type="text"/>
Select Destination Port for routing calls	Fixed ▼ ➕
Dial Plan	1 ▼ ➕
Allowed-Denied Logic	<input type="checkbox"/> Apply
Dial Tone Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	# ▼
Minimum Number of digits that must be dialed by the caller	02 ▼
Maximum Number of digits that can be dialed by the caller	16 ▼

- If you do not want to route calls through this port, select the **Block all calls through this Radio Port** check box. Default: Disabled.

## Destination Number Determination

- **On Detecting Voice** on the Radio Port, you may **route call**:
  - using Fixed Destination Number
  - or
  - after collecting the digits

## Route to the Fixed Destination Number

In this method, calls received on the Radio Port are routed to a fixed destination number configured for the Radio Port.

The screenshot shows the 'Handling of Outgoing calls' configuration window. The 'On Detecting Voice, route call' dropdown is set to 'using Fixed Destination Number'. The 'Fixed Destination Number' field is empty. The 'Select Destination Port for routing calls' dropdown is set to 'Fixed'. The 'Dial Plan' dropdown is set to '1'. A red box highlights the 'using Fixed Destination Number' dropdown and the 'Fixed Destination Number' field.

To apply this method, do the following:

- In the **On Detecting Voice, route call**, select **using Fixed Destination Number**.
- In the **Fixed Destination Number**, enter the desired destination number.

The Destination Number may consist of a maximum of 16 digits. Valid digits are 0 to 9, \*, # and (.) dot.  
Default: Blank.

- Click **Submit** to save the changes.

## Route After Collecting the Digits

In this method, the system plays dial tone to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.

The screenshot shows the 'Handling of Outgoing calls' configuration window. The 'On Detecting Voice, route call' dropdown is set to 'after collecting the digits'. The 'Select Destination Port for routing calls' dropdown is set to 'Fixed'. The 'Dial Plan' dropdown is set to '1'. The 'Allowed-Denied Logic' checkbox is unchecked. The 'Dial Tone Timer' is set to 7 seconds. The 'Inter Digit Wait Timer' is set to 5 seconds. The 'End Of Dialing Digit' dropdown is set to '#'. The 'Minimum Number of digits that must be dialed by the caller' dropdown is set to '02'. The 'Maximum Number of digits that can be dialed by the caller' dropdown is set to '16'. A red box highlights the 'after collecting the digits' dropdown and the 'Dial Plan' dropdown. Another red box highlights the 'Dial Tone Timer', 'Inter Digit Wait Timer', 'End Of Dialing Digit', 'Minimum Number of digits that must be dialed by the caller', and 'Maximum Number of digits that can be dialed by the caller' fields.

To apply this method, configure the following:

- In the **On Detecting Voice, route call**, select **after collecting the digits**.




Under **Settings**, the parameter **Time Between Consecutive PTT** will become uneditable and the parameter **PTT Count to Place a call** will be set to 1 and will also become uneditable.

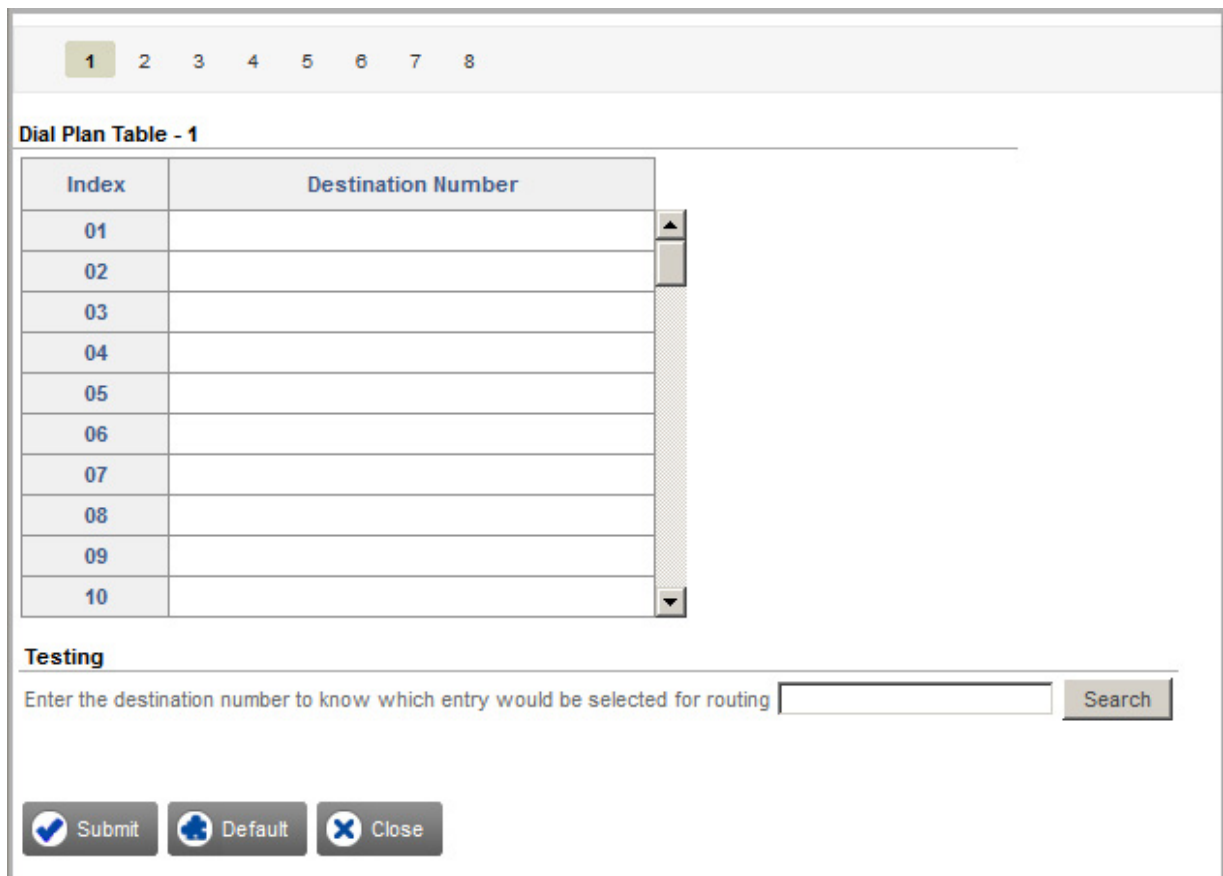
You may also configure the following related parameters:

#### Dial Plan

- SETU VGRX supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

- Click **Settings** .
- The Dial Plan Table page opens in a new window.



Index	Destination Number
01	
02	
03	
04	
05	
06	
07	
08	
09	
10	

**Testing**

Enter the destination number to know which entry would be selected for routing

- You may configure the default Dial Plan Table-1 or any other Table (from 2 to 8) for the Radio Port. You can store 64 Numbers at Index Numbers 01 to 64 respectively.

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + “[Wild card Characters](#)”). Valid characters: 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

See “[Dial Plan](#)” for more details.

## Wild card Characters

SETU VGRX supports following characters.

Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.
<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.
<b>[ ^ ]</b>	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
<b>T</b> (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

- Click **Submit** to save the entries and close the window.
- Return to the Dial Plan parameter and assign the Dial Plan Table you configured.

### Allowed - Denied Logic (Toll Control)

You can apply the Allowed-Denied logic on the Radio Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the Radio Port.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the Radio Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SETU VGRX uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

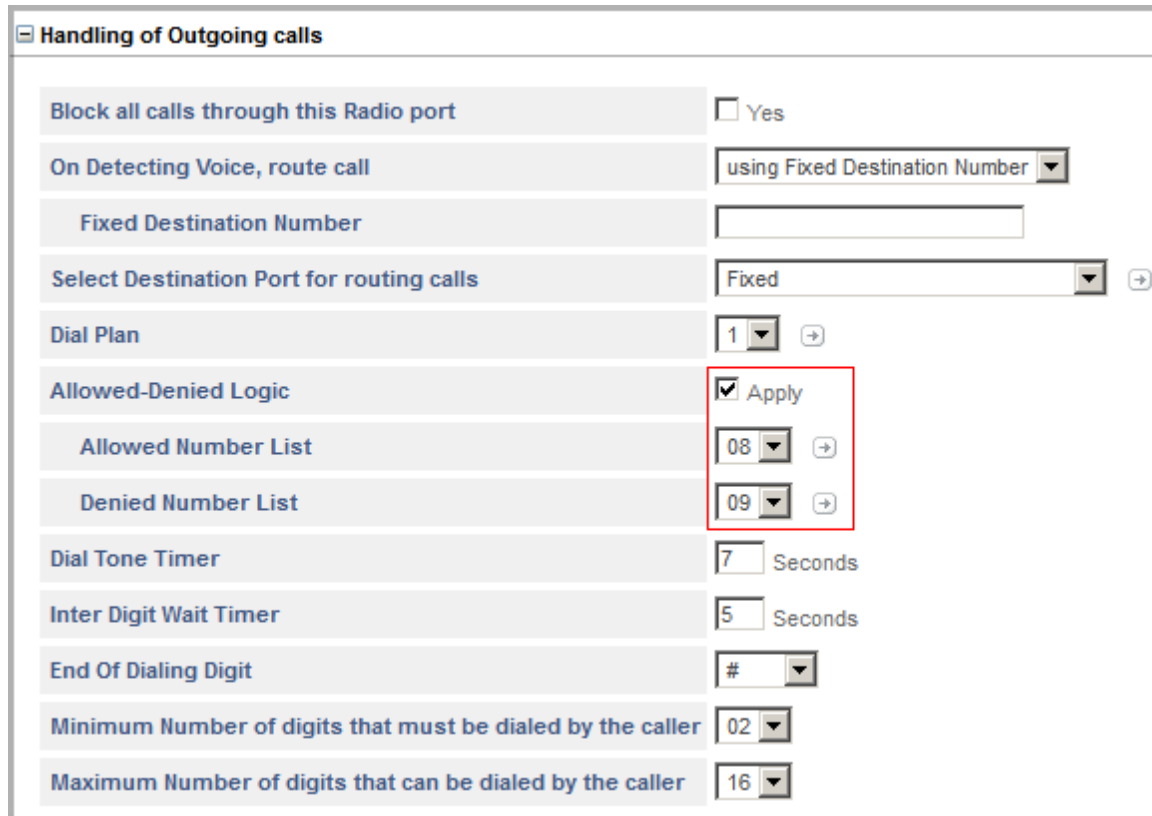
The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:

- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.

To apply Allowed - Denied Logic on the Radio Port,

- Select the **Allowed - Denied Logic** check box.



Handling of Outgoing calls	
Block all calls through this Radio port	<input type="checkbox"/> Yes
On Detecting Voice, route call	using Fixed Destination Number ▼
Fixed Destination Number	<input type="text"/>
Select Destination Port for routing calls	Fixed ▼ ➡
Dial Plan	1 ▼ ➡
Allowed-Denied Logic	<input checked="" type="checkbox"/> Apply
Allowed Number List	08 ▼ ➡
Denied Number List	09 ▼ ➡
Dial Tone Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	# ▼
Minimum Number of digits that must be dialed by the caller	02 ▼
Maximum Number of digits that can be dialed by the caller	16 ▼

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the Radio Port. Default: 08.

If you have not configured the Allowed Number List,

- Click **Settings** ➡. The Number Lists window opens.
- You may configure the default Allowed Number List 8 or any other list. See ["Number Lists"](#) to configure the allowed numbers.
- Click **Submit** to save the Allowed Number List and close the window.
- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the Radio Port. Default: 09.

If you have not configured the Denied Number List,

- Click **Settings** ➡. The Number Lists window opens.

- You may configure the default Denied Number List 9 or any other list. See [“Number Lists”](#) to configure the restrict numbers.
- Click **Submit** to save the Denied Number List and close the window.

### Dial Tone Timer

- In **Dial Tone Timer**, enter the time for which the system must play the dial tone on the Radio Port after the user presses the PTT button. Valid range: 01 to 99 seconds. Default: 7 seconds.

### End of Dialing

- You may configure the following options as End-of-Dialing indication on the Radio Port:

- In **Inter Digit Wait Timer**, enter the time for which the system must wait for the user to dial digits, to consider it as end-of-dialing. Valid range: 01 to 99 seconds. Default: 5 seconds.
- In **End of Dialing Digit**, select the termination digit on detecting which you want the system to consider the end of dialing. You may select # or \*. Default: #
- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits that is to be dialed by the user to route the call. Valid range: 01–24 digits. Default: 02 digits.



**Minimum number of digits that can be dialed by the caller** parameter will be applicable when:

- the Destination Port Determination method selected is On the basis of Destination Number and the dialed number is not found in the Destination Number Table.
- Or
- the dialed number is not found in the Dial Plan and the End of Dialing is detected.
- In **Maximum number of digits that can be dialed by the caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing. The valid range is 01 to 24 digits. Default: 16 digits.

When the caller dials a number, the system will match it with the end-of-dialing indications and accept the one that matches first.



- *SETU VGRX reloads the Inter Digit Wait Timer:*
  - *each time you dial a new digit till the termination digit is detected.*
  - *each time you dial a new digit till the entry is not matched in Dial Plan.*
  - *until you have dialed the maximum number of digits configured as End of Dialing.*

## Destination Port Determination

- In **Select Destination Port for routing calls**, select the method to be used for determining the Destination Port for routing calls from the Radio Port. You may select any one of these options:
  - Fixed
  - on the basis of Destination Number

Default: Fixed

Read further for instructions on selecting and configuring each of these destination port determination methods.



*If the destination number to be dialed out is an IP Address, SETU VGRX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. See ["IP Dialing"](#) for more details.*

### Fixed

In this method, outgoing calls made from the Radio Port are routed to a Fixed Destination Port, irrespective of the number dialed from the Radio Port.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.

- Click **Settings** .

Handling of Outgoing calls

Block all calls through this Radio port

☐ Yes

On Detecting Voice, route call

using Fixed Destination Number

Fixed Destination Number

Select Destination Port for routing calls

Fixed

Dial Plan

1

Allowed-Denied Logic

☐ Apply

Dial Tone Timer

7

Seconds

Inter Digit Wait Timer

5

Seconds

End Of Dialing Digit

#

Minimum Number of digits that must be dialed by the caller


02


Maximum Number of digits that can be dialed by the caller

16


The **Destination Port/Group for Radio Port** window opens.

Destination Port/Group for Radio Port

Edit	Routing Group	Fallback Routing Group	CLI Number on FXS Port
	FXS Port 1 - 2 (Ascending)	None	Received Calling Party

 Close

The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to edit the default Routing Group options, click **Settings** .

The **Edit Selective Port/Group for Radio Port** window opens.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.




*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default:1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports*, *Radio Ports* and *SIP Trunks*.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default:1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
  - Click **Submit** to save changes. The **Edit** window closes.
  - The entry you edited appears in the **Destination Port/Group for FXS Port** window.
  - Close the **Destination Port/Group for FXS Port** window to return to the Handling of Outgoing Calls.

#### **On the basis of Destination Number**

In this method, outgoing calls made from the Radio Port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SETU VGRX will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

- Click **Settings** .

Handling of Outgoing calls

Block all calls through this Radio port


☐ Yes

On Detecting Voice, route call


using Fixed Destination Number ▼

Fixed Destination Number


Select Destination Port for routing calls

On the basis of Destination Number ▼ 

If no match found in the Destination Number Table

Route calls to Fixed Port ▼ 

Dial Plan


1 ▼ 

Allowed-Denied Logic

☐ Apply

The **Radio - Destination Port Determination - Destination Number Based** table window opens.

Radio - Destination Port Determination - Destination Number Based


	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
---	------	--------------------	---------------	------------------------	---


Total Records : 0


Testing

Enter the destination number to know which entry would be selected for routing

Search

 Add

 Delete

 Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 100 entries.


- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wild card Characters"). Valid characters are 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.
- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*


- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports, Radio Ports and SIP Trunks.
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default:1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic "[Group](#)" under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **Radio - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.



*If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

- You can also edit the Routing Group and Fallback Routing Group, if required. To do so,
  - To edit an entry in the table, under Edit, click **Settings**  .
  - The **Edit Entry** window opens.
  - Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
  - Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see "[Destination Port Determination](#)" under *Advanced Settings*.

## Settings

- Click **Settings**.

Settings	
Minimum Sample Value for Speech Detection	300
Minimum Speech Time for valid IC Call	300 msec
Minimum Silence Time for valid IC Call	200 msec
Time Between Consecutive PTT	4000 msec
PTT Count to Place a call	3
PTT Activation	On Speech Detection
VAD Threshold level for IC Call	-15 dBm
VAD Threshold level for PTT Activation	-20 dBm
Minimum Speech Time to activate PTT	400 msec
Minimum Silence Time to de-activate PTT	2000 msec
VAD Hang Timer	500 msec
Allow Call Disconnection using Access Code	<input type="checkbox"/> Yes
Play Error Tone after Routing/Speech	<input type="checkbox"/> Yes
Maximum Noise Suppression	25 dBm
VAD Noise Floor Threshold	-15 dBm
AGC Target Power In	-20 dBm
AGC Max Loss Limit In	-3 dB
AGC Max Gain Limit In	09 dB

- Enter **Minimum Sample Value for Speech Detection**. This is the minimum value for detecting speech on the Radio Port. Valid range: 100 to 900. Default: 300.
- Enter **Minimum Speech Time for valid IC Call**. This is the minimum time for which the valid speech packets will be detected by the system for a valid incoming call. Valid range: 100 to 9999 msec. Default: 300 msec.
- Enter **Minimum Silence Time for valid IC Call**. This is the time for which the system will accept silence before actual speech packets for a valid incoming call. Valid range: 100 to 9999 msec. Default: 200 msec.
- Enter the maximum **Time Between Consecutive PTT**. Valid range: 2000 to 6000 msec. Default: 4000 msec.



Under **Handling of Outgoing calls**, if you select **after collecting the digits** in **On Detecting Voice**, route call, the parameter **Time Between Consecutive PTT** will become uneditable.

- Enter the **PTT Count to Place a call**. This is the count after which the system will place the call. Valid range: 1 to 5. Default: 3.



Under **Handling of Outgoing calls**, if you select **after collecting the digits in On Detecting Voice**, route call, the parameter **PTT Count to Place a call** will be set to 1 and will become uneditable.

- Select the desired option for **PTT Activation**. You may select:
  - On Speech Detection
  - Using '\*' Key

Default: On Speech Detection

If you select On Speech Detection, the PTT activation/deactivation will be done automatically. It depends on the values of **Minimum Speech Time to activate PTT** and **Minimum Silence Time to deactivate PTT**.

If you select Using '\*' Key, the user must press the '\*' key to activate as well as deactivate PTT. It does not depend on the values of Minimum Speech Time to activate PTT and Minimum Silence Time to deactivate PTT.

- Select **VAD Threshold Level for IC Call**. This parameter defines the level below which the Radio Port would not validate the audio signal as valid speech packet for incoming Call. Valid range: 0, -1 to -96dBm. Default: -15dB.
- Select **VAD Threshold Level for PTT Activation**. This parameter defines the level below which the Radio Port would not validate the audio signal as valid speech packet for PTT Activation. Valid range: 0 to -96dBm. Default: -20dBm.
- Enter **Minimum Speech Time to activate PTT**. This is the time after which PTT will be activated and the speech will be transmitted. This timer ensures continuous communication, once the call matures. Valid range: 100 to 9999 msec. Default: 400 msec.
- Enter **Minimum Silence Time to deactivate PTT**. This is the duration of silence after which the system will deactivate PTT. Valid range: 100 to 9999 msec. Default: 2000 msec.
- Enter **VAD Hang Timer**. The system will wait for the duration of this timer, after the expiry of Minimum Silence Time for valid Outgoing call, before disconnecting the call. Default: 500 msec.
- Select the **Allow Call Disconnection using Access code** check box, if you want to enable the feature Disconnect Call using Access Code on the Radio Port. Default: Disabled. See ["Disconnecting a Call using Access Code"](#).



To use **Call Disconnection using Access code**, the Radio device must support dialing of DTMF digits.

- Select the **Play Error Tone after Routing/Speech** check box, if you want SETU VGRX to play error tone after routing the call or after disconnecting the speech. Default: Disabled.
- Enter **Maximum Noise Suppression**. This parameter identifies how much background noise can be suppressed. Valid range: 0 to 95 dBm. Default: 25 dBm.
- Enter **VAD Noise Floor Threshold**. This parameter specifies threshold above which VAD (Voice activity detection) decides that voice activity is present or below which VAD decides that only noise is present. Valid range: 0 to -96 dBm. Default: -15 dBm.

- Enter **AGC Target Power In**. This field identifies the target power level for speech output. Speech input power levels below or above this value will be raised or lowered as needed. Valid range: 0 to -20 dBm. Default: -20 dBm.
- Enter **AGC Max Loss Limit In**. This field identifies how much speech input can be reduced when the input is above the Target Power level. Valid range: 0 to -23 dBm. Default: -3 dBm.
- Enter **AGC Max Gain Limit In**. This field identifies how much the speech input can be increased when the input is below the Target Power level. Valid range: 0 to 23 dBm. Default: 09 dBm.

## DTMF Settings

- Click **DTMF Settings**.

- Select the appropriate **DTMF Generation** option. You may select:
  - Using System DSP
  - Using Radio DSP

Default: Using System DSP

- Select the appropriate **DTMF Generation ON Time** for the Radio Port. The DTMF On Time is the time for which the DTMF digit which is to be outdialed by the system remains On. Valid range is 050 to 250 msec. Default: 070 msec.
- Select the appropriate **DTMF Generation OFF Time** for the Radio Port. This is the time for which the system will wait before dialing the successive DTMF digits. Valid range is 050 to 250 msec. Default: 070 msec.
- Select the appropriate **DTMF Detection** option. You may select:
  - Using System DSP
  - Using Radio DSP

Default: Using System DSP
- If you have selected DTMF Detection option as *Using Radio DSP*, select the appropriate **DTMF Detection ON Time (Minimum)**. This parameter signifies the time period for which the DTMF signal should remain on in order to be detected. Valid range: 020 to 200 msec. Default: 020 msec.
- If you have selected DTMF Detection option as *Using Radio DSP*, select the appropriate **DTMF Detection OFF Time (Minimum)**. This parameter signifies the minimum off time between successive DTMF digits. Valid range: 020 to 200 msec. Default: 020 msec.

## Copy Port Parameters

- You can also copy the settings of a Radio Port to another Radio Port using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy Radio Port Parameters** window opens.



The screenshot shows a dialog box titled "Copy Radio Port Parameters from Radio Port". It features a dropdown menu with the number "1" selected, followed by the word "to". Below this, there are four checkboxes labeled "All", "Radio Port 1", "Radio Port 2", "Radio Port 3", and "Radio Port 4". The "All" checkbox is currently checked. At the bottom of the dialog, there are two buttons: "OK" with a checkmark icon and "Close" with an 'X' icon.

- In **Copy Radio Port Parameters from Radio Port**, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the Radio Ports, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the Radio Port you copied the settings to.

# FXO Port

SETU VGRX supports two FXO Ports, which may be either,

- interfaced with the analog trunk from CO and used to route incoming calls to FXS Port, Mobile Ports, Radio Ports and SIP Trunks.  
- or -
- interfaced with FXS Port of the PBX.

To configure the parameters of the FXO Port,

- Click the **Basic Settings** link to expand.
- Click the **FXO Port** link.

Port	Enable	Name	Status	CLI Type	Incoming Call Routing
<a href="#">FXO-1</a>	<input checked="" type="checkbox"/>		Not Connected	FSK V.23	Route calls to FXS Port 1 - 2
<a href="#">FXO-2</a>	<input checked="" type="checkbox"/>		Not Connected	FSK V.23	Route calls to FXS Port 1 - 2

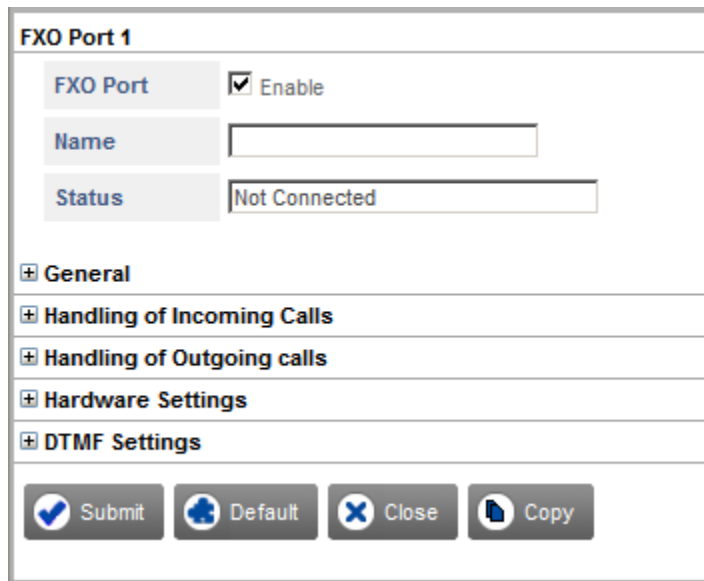
The FXO Port page displays the following parameters:

- **Port:** Displays the FXO Port numbers. To configure the FXO Port parameters, click on the desired FXO Port number link.
- **Enable:** Keep the **FXO Ports** enabled. Clear the FXO Port **Enable** check box, only if you do not want to use the Port.
- **Name:** You may assign a Name to the FXO Port, for identification.
- **Status:** Displays whether the FXO Port is connected or not.
- **CLI Type:** Displays the type of CLI selected for the FXO Port.
- **Incoming Call Routing:** Displays the Incoming Call Routing Method selected for the FXO Port.

To configure the FXO Port parameters,

- Click **FXO-1**.

The **FXO Port 1** window opens.



**FXO Port 1**

**FXO Port** ☒ Enable

**Name**

**Status**





+ **General**

+ **Handling of Incoming Calls**

+ **Handling of Outgoing calls**

+ **Hardware Settings**

+ **DTMF Settings**

 Submit  Default  Close  Copy

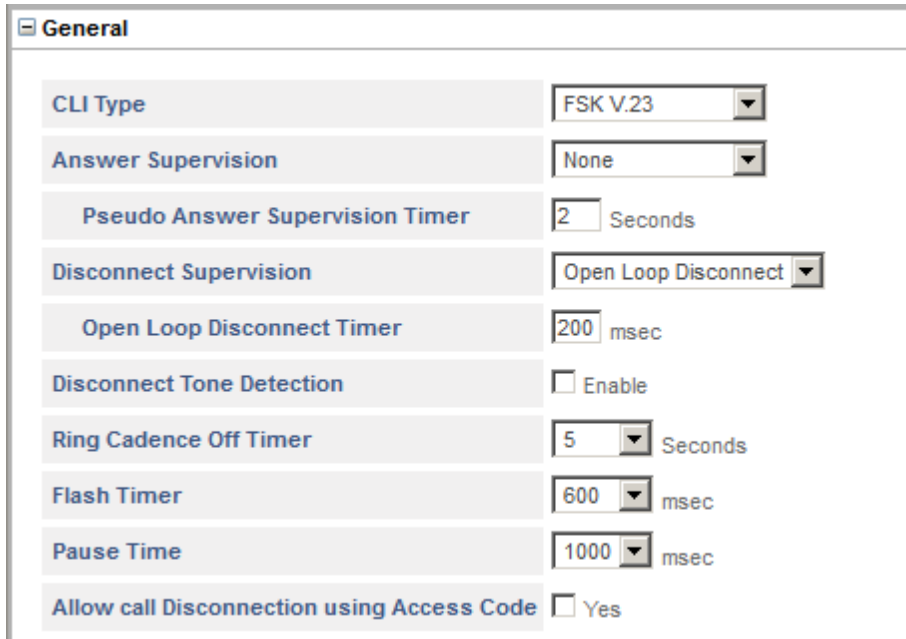
- Keep the **FXO Port** check box enabled.

Clear the **FXO Port Enable** check box only when you do not want to use this FXO Port. Default: Enabled.

- You can assign a **Name** to the FXO Port, which will be displayed to the called party, if the called party telephone instrument supports CLI display. The Name can be a maximum of 24 characters. Default: Blank
- **Status** displays whether the CO line is connected to the FXO Port or not.

## General

- Click **General**.



General	
CLI Type	FSK V.23
Answer Supervision	None
Pseudo Answer Supervision Timer	2 Seconds
Disconnect Supervision	Open Loop Disconnect
Open Loop Disconnect Timer	200 msec
Disconnect Tone Detection	<input type="checkbox"/> Enable
Ring Cadence Off Timer	5 Seconds
Flash Timer	600 msec
Pause Time	1000 msec
Allow call Disconnection using Access Code	<input type="checkbox"/> Yes

- Select the **CLI Type** — DTMF-India, DTMF-ETSI, DTMF-Denmark, DTMF-Brazil, DTMF-Any, FSK V.23 or FSK Bellcore — according to the CLI type supported by your CO network. Default: FSK V.23
- SETU VGRX uses **Answer Supervision** signaling to indicate that the call made through the FXO Port has been answered by the remote party.

Select the appropriate **Answer Supervision** — None or Battery Reversal — supported by your CO network.

- Select **None**, when no signaling is available from the CO network. The call will be considered as matured on the expiry of the *Pseudo Answer Supervision Timer*.
- Set the duration of the **Pseudo Answer Supervision Timer** as per your requirement. Valid range is 00 to 99 seconds. Default: 2 seconds.
- If you select **Battery Reversal**, SETU VGRX will consider the call as matured only if reversal of polarity is detected on the FXO Port. Pseudo Answer Supervision Timer will not be applicable in this case.

Default: None.



*When an outgoing call is made from the FXO Port and Answer Supervision is set to Battery Reversal, the system will disconnect the call if it does not mature within 120 seconds.*

- **Disconnect Supervision** is the signal given by the CO network to detect far end disconnection.

Whenever a call (incoming or outgoing) on the FXO Port is disconnected by the remote party, the CO network will send disconnect signal. SETU VGRX will detect this signal and release the FXO Port.

Select the appropriate **Disconnect Supervision** — None, Battery Reversal or Open Loop Disconnect — according to the type supported by your CO network.

- Select **None** when no signaling is available from the CO network.
- Select **Battery Reversal** when call disconnection is signaled in the form of reversed polarity.

When the call is disconnected by the remote user, the reversed polarity is signalled and the FXO Port is released (free). The caller will get an error tone.

- Select **Open Loop Disconnect** when call disconnection is signaled in the form of an Open Loop signal. The system will check Open Loop Disconnect signal for the time configured for the *Open Loop Disconnect Timer*.

Only if, the Open Loop signal is detected continuously for the time configured in *Open Loop Disconnect Timer*, it will be considered as a valid Open Loop signal for releasing the port.

If you select this option, you must configure the Open Loop Disconnect Timer.

- Set the duration of the **Open Loop Disconnect Timer** as per your requirement. Valid range is 001 to 999 msec. Default: 200 msec.

Default: Open Loop Disconnect.

- Disconnect Tone Detection is used by the system to release the FXO Port, when the remote party goes On-hook or disconnects the call. The tone detection is applicable for both incoming and outgoing calls from the FXO Port.

Select the **Disconnect Tone Detection** check box, if you want the system to detect Call Disconnect Tone sent by the CO network on the FXO Port.

- Select the **Disconnect Tone Type** that you want the system to use as the Call Disconnect Tone on the FXO Port. Default: Disconnect Tone 1

If required, you can customise the frequencies and cadences of the Disconnect Tone as per your requirement. For instructions, see [“Disconnect Tone”](#).

- Set the duration of the **Ring Cadence Off Timer**, to set the OFF time for Ring cadence. During the incoming call on FXO Port, if the CO gives ring in which the Ring OFF period is quite long, the system will consider that the ring has been stopped, and will stop ringing the FXS port, even though the incoming call is still present.

To get accurate indication, the system supports Ring Cadence OFF timer on the FXO Port so that ring can continue even for incoming calls with long Ring OFF period. Valid range is 1 to 9 seconds. Default: 5 seconds.

- Set the duration of the **Flash Timer**. This is the time for which Flash will be generated on the FXO Port. SETU VGRX uses this event to activate various features — Call Hold, Call Transfer, etc. Default: 600 msec.
- Set the duration of the **Pause Timer** to add delay while a call is being made from the FXO Port. After the FXO Port goes Off-hook, SETU VGRX adds some delay before dialing out the number. During this time,

no digit is dialed by the system on the FXO Port. This is used when the exchange takes some time to detect that the FXO Port is Off-hook. Default: 1000 msec.

This timer will also be used while applying ANT logic, if you have configured Pause (^) in the Add Prefix column of ANT table. See [“Automatic Number Translation \(ANT\)”](#) for more details.

- Select the **Allow Call Disconnection using Access code** check box, if you want to enable the feature Disconnect Call using Access Code on the FXO Port. See [“Disconnecting a Call using Access Code”](#).
- Click **Submit** to save the settings.

## Handling of Incoming Calls

- Click **Handling of Incoming Calls**.

- If you do not want to route calls received on this FXO Port, select the **Block all calls received on this FXO Port** check box. Default: Disabled.

## Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods.
  - without any Destination Number
  - to the Fixed Destination Number

- on the basis of Calling Party Number
  - after Answering the Call and Collecting the Digits
- Default: without any Destination Number



*If the destination number to be dialed out is an IP Address, SETU VGRX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. See “IP Dialing” for more details.*

## Route Calls without any Destination Number

In this method, all calls received on the FXO Port are directly routed to the destination port, regardless of the Destination Number.

**Handling of Incoming Calls**

Block all calls received on this FXO port	<input type="checkbox"/> Yes
Route all incoming calls (with CLI)	without any Destination Number
Block Calls received without CLI on this FXO Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits

- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

## Route to the Fixed Destination Number

In this method, calls received on the FXO Port are routed to a fixed destination number, which you must configure for this port.

**Handling of Incoming Calls**

Block all calls received on this FXO port	<input type="checkbox"/> Yes
Route all incoming calls (with CLI)	to the Fixed Destination Number
Block Calls received without CLI on this FXO Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits

**Fixed Destination Number**

Fixed Destination Number

To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In **Fixed Destination Number**, enter the desired destination number.

The Destination Number may consist of a maximum of 24 digits. Valid digits are 0 to 9, \*, # and (.) dot.  
Default: Blank.


- Click **Submit** to save the changes.

## Route on the basis of Calling Party Number

In this method, a call received on the FXO Port is routed to a specific destination number, as per the calling party's number. For this, the calling party numbers and their corresponding destination numbers must be configured in the Calling Party Number Based Table.

Whenever there is an incoming call on the FXO Port, SETU VGRX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call will be routed to the destination number configured for that Calling Party Number.

To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **on the basis of Calling Party Number**.
- Click **Settings** .



Handling of Incoming Calls	
Block all calls received on this FXO port	<input type="checkbox"/> Yes
Route all incoming calls (with CLI)	on the basis of Calling Party Number 
If no match found in the Calling Party Number Table, route calls	after Answering the Call and Collecting the Digits 
Block Calls received without CLI on this FXO Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits 

The **FXO Port Destination Number Determination: Calling Number Based** Table opens.

Index	Calling Number	Destination Number
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		
013		
014		
015		
016		
017		

- Configure the **Calling Number Based** table for the FXO Port. You can enter upto 499 Calling Party Numbers and their corresponding Destination Numbers in this table.
- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.
- In **Destination Number**, enter a corresponding destination number for each calling party number. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, \*, # and (.) dot are allowed. Default: Blank.
- Click **Submit** to save your entries. Close the window to return to the FXO Port window.

You can also configure the **Calling Number Based** Table from *Advanced Settings* link. For instructions, see [“Destination Number Determination”](#) under *Advanced Settings*.

- For incoming calls with Calling Party Numbers that do not match with the Calling Party Number Table, you may select the destination number determination method.

In **If no match found in the Calling Party Number Table, route calls**, you may select either **to the Fixed Destination Number** or **after Answering the Call and Collecting the Digits**. Default: after Answering the Call and Collecting the Digits.

## Route After Answering the Call and Collecting the Digits

In this method, the system answers the incoming call on the FXO Port and plays dial tone to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.

**Handling of Incoming Calls**

Block all calls received on this FXO port ☐ Yes

Route all incoming calls (with CLI) after Answering the Call and Collecting the Digits ▼

Block Calls received without CLI on this FXO Port ☐ Yes

Route all Incoming calls (without CLI) after Answering the Call and Collecting the Digits ▼

**Answering the Call and Collecting the Digits**

Prompt caller to enter PIN ☐ Yes

Dial Plan 1 ▼ ➡

First Digit Wait Timer 7 Seconds

Inter Digit Wait Timer 5 Seconds

End Of Dialing Digit # ▼

Minimum Number of digits that must be dialed by the caller 02 ▼

Maximum Number of digits that can be dialed by the caller 24 ▼

If No Digit dialed during First Digit Wait Timer Disconnect Call ▼

Allow making New Call using Access code ☐ Yes

Select Destination Port for routing calls Fixed ▼ ➡

Allowed-Denied Logic ☐ Apply

To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.


The related parameters appear under **Answering the Call and Collecting the Digits**.

- If you want to enable PIN Authentication on the FXO Port, select the **Prompt caller to enter PIN** check box.

If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for instructions, see [“PIN Authentication”](#) under *Advanced Settings*.

- SETU VGRX supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

- Click **Settings**  . The Dial Plan Table opens.
- Configure the numbers in the table. For detailed instructions, see [“Dial Plan”](#).
- Set the duration of the **First Digit Wait Timer**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds.
- Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 5 seconds.
- As **End of Dialing Digit**, select whether the system should consider # or \* as termination digit to detect end of dialing. Default: #
- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number. Valid range is 01 to 24 digits. Default: 02 digits.
- In **Maximum number of digits that can be dialed by the Caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing. Valid range is 01 to 24 digits. Default: 24 digits.

When the caller dials a number, the system will match it with the above end-of-dialing indications and accept the one that matches first.

- If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

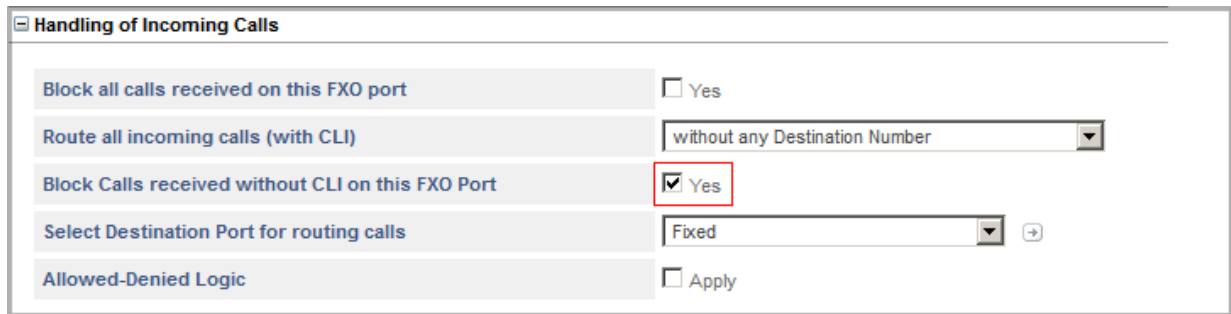
In **If No Digit dialed during First Digit Wait Timer**, you can select — **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

- If you select **Use Fixed Destination Number**, enter the desired destination number in **Fixed Destination Number**. The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, \*, # and (.) dot. Default: Blank.



- *The First Digit Wait Timer is loaded as soon as the system answers the call.*
- *When you dial the first digit, the First Digit Wait Timer is stopped and the system loads the Inter Digit Wait Timer.*
- *SETU VGRX reloads the Inter Digit Wait Timer:*
  - *each time you dial a new digit till the termination digit is detected.*
  - *each time you dial a new digit till the entry is not matched in Dial Plan.*
  - *until you have dialed the maximum number of digits configured as End of Dialing.*
- Select the **Allow making New Call using Access Code** check box, if you want to enable the feature Making New Call using Access Code on the FXO Port. See [“Making a New Call using Access Code”](#).
- Click **Submit** to save settings.

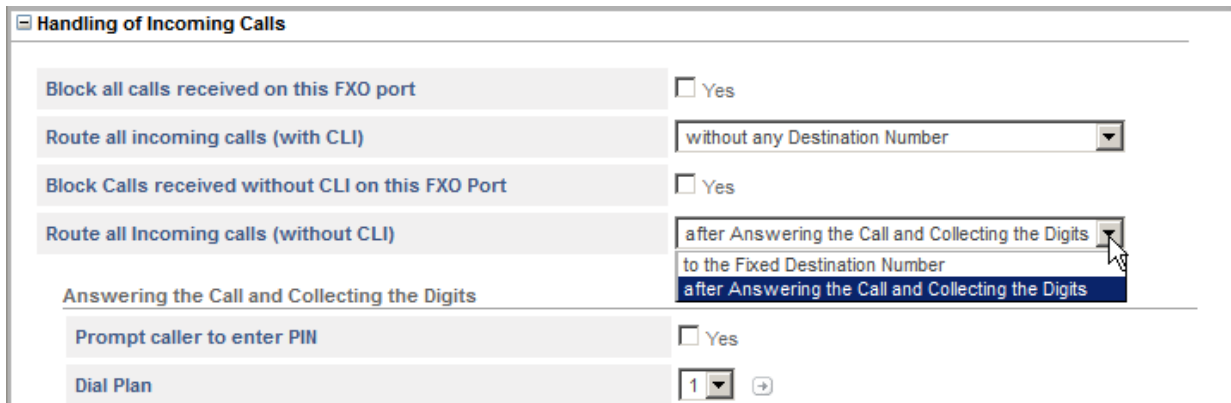
- If you do not want to route calls without CLI through this port, select the **Block Calls received without CLI on this FXO Port** check box.



**Handling of Incoming Calls**

Block all calls received on this FXO port	<input type="checkbox"/> Yes
Route all incoming calls (with CLI)	without any Destination Number
Block Calls received without CLI on this FXO Port	<input checked="" type="checkbox"/> Yes
Select Destination Port for routing calls	Fixed
Allowed-Denied Logic	<input type="checkbox"/> Apply

- To **Route all Incoming calls (without CLI)**, you may select from any of the following methods:
  - to the Fixed Destination Number, see ["Route to the Fixed Destination Number"](#).
  - after Answering the Call and Collecting the Digits, see ["Route After Answering the Call and Collecting the Digits"](#).
 Default: after Answering the Call and Collecting the Digits.



**Handling of Incoming Calls**

Block all calls received on this FXO port	<input type="checkbox"/> Yes
Route all incoming calls (with CLI)	without any Destination Number
Block Calls received without CLI on this FXO Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits
Answering the Call and Collecting the Digits	<input checked="" type="checkbox"/> Yes
Prompt caller to enter PIN	<input type="checkbox"/> Yes
Dial Plan	1

## Destination Port Determination

Select the Destination Port for routing calls for the FXO Port. You may select from any of the following options:

- Fixed
  - on the basis of Destination Number
  - on the basis of Calling Party Number
- Default: Fixed



*If the destination number to be dialed out is an IP Address, SETU VGRX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. See ["IP Dialing"](#) for more details.*

### Fixed

In this method, calls received on the FXO Port are routed to a Fixed Destination Port, irrespective of the number dialed on the FXO Port.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.

- Click **Settings**  .

Handling of Incoming Calls

Block all calls received on this FXO port

☐ Yes

Route all incoming calls (with CLI)

without any Destination Number

Block Calls received without CLI on this FXO Port

☐ Yes

Route all Incoming calls (without CLI)

to the Fixed Destination Number

Fixed Destination Number

Fixed Destination Number

Select Destination Port for routing calls


Fixed


Allowed-Denied Logic

☐ Apply


The **Destination Port/Group for FXO Port** window opens.

Destination Port/Group for FXO Port

Edit	Routing Group	Fallback Routing Group	CLI Number on FXS Port
	FXS Port 1 - 2 (Ascending)	None	Received Calling Party

 Close

The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to edit the default Routing Group options, click **Settings**  .

The **Edit Selective Port/Group for FXO Port** window opens.


- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.
 

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports*, *Radio Ports* and *SIP Trunks*.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default:1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for FXO Port** window.
- Close the **Destination Port/Group for FXO Port** window to return to the Handling of Incoming Calls window.

### On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SETU VGRX will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

- Click **Settings**  .

Handling of Incoming Calls

Block all calls received on this FXO port

☐ Yes

Route all incoming calls (with CLI)

without any Destination Number

Block Calls received without CLI on this FXO Port

☐ Yes



Route all Incoming calls (without CLI)

to the Fixed Destination Number

Fixed Destination Number

Fixed Destination Number

Select Destination Port for routing calls



On the basis of Destination Number  

Allowed-Denied Logic

☐ Apply

The **FXO Port - Destination Port Determination - Destination Number Based** table window opens.


FXO Port - Destination Port Determination - Destination Number Based


	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party


Total Records : 11

Testing

Enter the destination number to know which entry would be selected for routing

 Add

 Delete

 Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.

**Add Entry**

Destination Number

CLI Number to be sent on Destination Port

**Routing Group**

☐ FXS Port  to  in  order

☐ FXS Group

☐ FXO Port  to  in  order

☐ FXO Group

☐ Mobile Port  to  in  order

☐ Mobile Group

☒ SIP Trunk  to  in  order

☐ SIP Group

☐ Radio Port  to  in  order

**Fallback Routing Group** ☐ Apply

☐ FXS Port  to  in  order

☐ FXS Group

☐ FXO Port  to  in  order

☐ FXO Group

☐ Mobile Port  to  in  order

☐ Mobile Group

☐ SIP Trunk  to  in  order

☐ SIP Group

☐ Radio Port  to  in  order

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + “Wildcard Characters”). Valid characters are 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

## Wildcard Characters

SETU VGRX supports following characters.

Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.
<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.
<b>[ ^ ]</b>	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.


<b>T (letter T)</b>	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.
---------------------	---

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,
    - Select the desired **FXS Port** numbers as members. Default:1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.
 


Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports, Radio Ports and SIP Trunks.
  - To create a group of *not-sequential* **FXS Ports** as members,
    - Select a **FXS Group**.
    - Select **FXS Group** number. Default:1.
    - Click **Settings**  . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic "[Group](#)" under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
  - Click **Submit** to save changes. The **Add Entry** window closes.
  - The entry you added appears in the **FXO Port - Destination Port Determination - Destination Number Based** table.
  - Follow the same steps as above to add another entry to this table.
  - To delete an entry, select the check box and click the **Delete** button.



If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, under Edit, click **Settings**  .

**Edit Entry**

Destination Number: No Match Found

CLI Number to be sent on Destination Port: Received Calling Party

**Routing Group**

- ☐ FXS Port: 1 to 1 in Ascending order
- ☐ FXS Group: 1
- ☐ FXO Port: 1 to 1 in Ascending order
- ☐ FXO Group: 1
- ☐ Mobile Port: 1 to 1 in Ascending order
- ☐ Mobile Group: 1
- ☒ SIP Trunk: 1 to 1 in Ascending order
- ☐ SIP Group: 1
- ☐ Radio Port: 1 to 1 in Ascending order

**Fallback Routing Group** ☐ Apply

- ☐ FXS Port: 1 to 1 in Ascending order
- ☐ FXS Group: 1
- ☐ FXO Port: 1 to 1 in Ascending order
- ☐ FXO Group: 1
- ☐ Mobile Port: 1 to 1 in Ascending order
- ☐ Mobile Group: 1
- ☐ SIP Trunk: 1 to 1 in Ascending order
- ☐ SIP Group: 1
- ☐ Radio Port: 1 to 1 in Ascending order


- The **Edit Entry** window opens.
- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Port Determination”](#) under *Advanced Settings*.

### On the basis of Calling Party Number

In this method, incoming calls on the FXO Port are routed to a specific port as per the calling party's number.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.
- Click **Settings**  .

Handling of Incoming Calls

Block all calls received on this FXO port

☐ Yes

Route all incoming calls (with CLI)

without any Destination Number

Block Calls received without CLI on this FXO Port

☐ Yes


Route all Incoming calls (without CLI)

to the Fixed Destination Number

Fixed Destination Number

Fixed Destination Number






Select Destination Port for routing calls

On the basis of Calling Party Number 

Allowed-Denied Logic

☐ Apply

The **FXO Port - Destination Port Determination - Calling Number Based** table window opens.

FXO Port - Destination Port Determination - Calling Number Based					
	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party
Total Records : 1		1			
<div><div> Add</div><div> Delete</div><div> Close</div></div>					

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.

**Add Entry**

Calling Number

CLI Number to be sent on Destination Port Received Calling Party ▼

**Routing Group**

☐ FXS Port 1 ▼ to 1 ▼ in Ascending ▼ order

☐ FXS Group 1 ▼

☐ FXO Port 1 ▼ to 1 ▼ in Ascending ▼ order

☐ FXO Group 1 ▼

☐ Mobile Port 1 ▼ to 1 ▼ in Ascending ▼ order

☐ Mobile Group 1 ▼

☒ SIP Trunk 1 ▼ to 1 ▼ in Ascending ▼ order

☐ SIP Group 1 ▼

☐ Radio Port 1 ▼ to 1 ▼ in Ascending ▼ order

**Fallback Routing Group** ☐ Apply

☐ FXS Port 1 ▼ to 1 ▼ in Ascending ▼ order

☐ FXS Group 1 ▼

☐ FXO Port 1 ▼ to 1 ▼ in Ascending ▼ order

☐ FXO Group 1 ▼

☐ Mobile Port 1 ▼ to 1 ▼ in Ascending ▼ order

☐ Mobile Group 1 ▼

☐ SIP Trunk 1 ▼ to 1 ▼ in Ascending ▼ order

☐ SIP Group 1 ▼

☐ Radio Port 1 ▼ to 1 ▼ in Ascending ▼ order


- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, \*, #, (dot). Default: Blank.
- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.




*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports, Radio Ports and SIP Trunks.
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default:1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic "[Group](#)" under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **FXO Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- For the No Match Found entry in the table, under Edit, click **Settings** .
- The **Edit Entry** window opens.

<b>Edit Entry</b>	
<b>Calling Number</b>	No Match Found
<b>CLI Number to be sent on Destination Port</b>	Received Calling Party ▼
<b>Routing Group</b>	
<input type="radio"/> FXS Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> FXS Group	1 ▼
<input type="radio"/> FXO Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> FXO Group	1 ▼
<input type="radio"/> Mobile Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> Mobile Group	1 ▼
<input checked="" type="radio"/> SIP Trunk	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> SIP Group	1 ▼
<input type="radio"/> Radio Port	1 ▼ to 1 ▼ in Ascending ▼ order
<b>Fallback Routing Group</b> <input type="checkbox"/> Apply	
<input type="radio"/> FXS Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> FXS Group	1 ▼
<input type="radio"/> FXO Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> FXO Group	1 ▼
<input type="radio"/> Mobile Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> Mobile Group	1 ▼
<input type="radio"/> SIP Trunk	1 ▼ to 1 ▼ in Ascending ▼ order
<input type="radio"/> SIP Group	1 ▼
<input type="radio"/> Radio Port	1 ▼ to 1 ▼ in Ascending ▼ order
<input checked="" type="button" value="Submit"/> <input checked="" type="button" value="Close"/>	

- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
  - Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Port Determination”](#) under *Advanced Settings*.

## Allowed - Denied Logic

You can apply the Allowed-Denied logic on the FXO Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed by the caller on the FXO Port.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed by the caller on the FXO Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SETU VGRX uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:


- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number

To apply Allowed - Denied Logic on the FXO Port,

- Select the **Allowed - Denied Logic** check box.


- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the FXO Port. Default: 01

If you have not configured the Allowed Number List,

- Click **Settings** . The Number Lists window opens.

- You may configure the default Allowed Number List or any other list. See [“Number Lists”](#) for instructions.
- Click **Submit** to save the Allowed Number List and close the window.
- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the FXO Port. Default: 02.

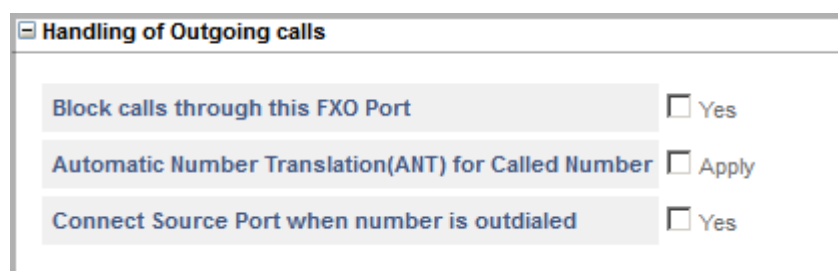
If you have not configured the Denied Number List,

- Click **Settings** . The Number Lists window opens.
- You may configure the default Denied Number List or any other list. See [“Number Lists”](#) for instructions.
- Click **Submit** to save the Denied Number List and close the window.

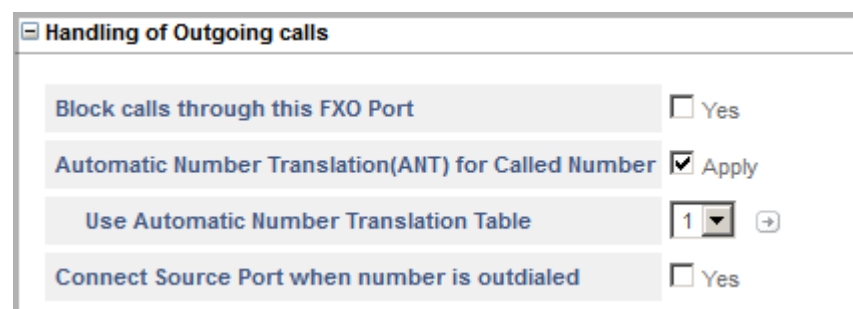
## Handling of Outgoing Calls

When a FXO Port is determined as the destination port, the numbers dialed from this port constitute outgoing calls.

- Click **Handling of Outgoing Calls**.



- If you do not want to route outgoing calls through this port, select the **Block calls through this FXO Port Yes** check box.
- You can apply **Automatic Number Translation logic** on outgoing calls made from the FXO Port.
- To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.



- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the FXO Port. Default: Table 1.

If you have not configured the Automatic Number Translation Table,

- Click **Settings**  . The **Automatic Number Translation Table** window opens.




1
2
3
4
5
6
7
8

**Automatic Number Translation Table - 1**

Index	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	

**Examples of Number Pattern**

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' to every 7-digit dialed number.

 Submit
 Default
 Close

- You may configure the default Automatic Number Translation Table or any other Table. See [“Automatic Number Translation \(ANT\)”](#) to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Click **Submit** to apply List.
- Select the **Connect Source Port when number is outdialed** check box to enable. This will connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature. Default: Disabled.

In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, i.e. the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call — called party is busy, not responding, not reachable or is rejecting the call.

- Click **Submit** to save.



If you enable **Connect Source Port when number is outdiald**, [“Making a New Call using Access Code”](#) feature will not be allowed to the users.

## Hardware Settings

- Click **Hardware Settings**.

Hardware Settings	
AC Termination Impedance	600 Ω
CO Termination	None
CO Line Type	None
Rx Gain	+0dB
Tx Gain	+0dB
On-Hook Speed	< 0.5 msec
Off-Hook Speed	8 msec
Current Limiting	<input type="checkbox"/> Yes
Minimum Loop Current	10 mA
TIP-RING Voltage	3.5 volts
Ringer Impedance	High
Ringer Threshold	13.5 - 16.5 Vrms

- In **AC Termination Impedance**, select the appropriate Impedance of the FXO Port as per the AC Termination Impedance supported by your CO Network. Default: 600 Ω.
- In **CO Termination**, select the appropriate line impedance match. This would depend on the region where SETU VGRX is deployed. Default: None.

This parameter allows you to increase near-end echo cancellation on the FXO Port. Near-end echo is primarily caused due to the mismatch between AC Termination Impedance (presented by FXO Port of SETU VGRX to the line) and CO Termination (Impedance presented by the Central Office to the line) as well as the transmit and receive signal path.

By correcting the line impedance mismatch, you can increase near-end echo cancellation. This is done by selecting the AC Termination Impedance, CO Termination and a Line Type that closely models the line connecting the FXO Port of SETU VGRX to the Central Office.

- In **CO Line Type**, select a line type that closely models the line connecting SETU VGRX to the Central Office. You may select a specific EIA line model (EIA-0 to EIA-7) or a specific wire gauge with its length (2000 ft. 22/24/26awg). Default: None.



You are recommended to conduct the **AC Impedance Test** for the line connected to the FXO Port to determine the most appropriate values for the AC Termination Impedance, CO Termination and the CO Line Type. For more information see the topic [“AC Impedance Test \(FXO\)”](#).

- **Rx Gain** enables you to adjust the volume level of the speech received from the remote party. Select the Rx Gain accordingly. Default: +0dB.
- **Tx Gain** enables you to adjust the volume level of the speech transmitted to the remote party. Select the Tx Gain accordingly. Default: +0dB.
- In **On-Hook Speed**, set the time period required for the line-side device (DAA) to go On-hook.

It is the time duration between On-hook bit clearance until loop current equals zero. You can select — <0.5, 3 or 26. Default: <0.5 msec.

- In **OFF- Hook Speed**, set the time that would be required by the line transients to settle. Only after this time period, the transmission or reception can take place. You can select — 512, 128, 64 or 8. Default: 8 msec.
- If you want to limit the loop current, select the **Current Limiting** check box. The Loop Current will be limited to a maximum of 60mA. Default: Disabled.
- Set the **Minimum Loop Current** required by the line-side device (DAA) to operate. You can select — 10, 12, 14 or 16 — as per your requirement. Default: 10 mA.
- Set the **TIP-RING Voltage (Volts)** to adjust the TIP/Ring Voltage on the line side. You can select — 3.1, 3.2, 3.35 or 3.5. Default: 3.5 volts.

In countries where Low voltage is required, use lower TIP/RING voltage. Adjust the values of the TIP-RING Voltage to match your country requirements.

- Set the **Ringer Impedance** — High or Synthesized — for the FXO Port according to your country-specific requirement.

**High** signifies 20 MΩ Ringer Impedance. This is the default Ringer Impedance provided on the line side by the DAA module of the FXO Port. The DAA Module can provide higher impedance when **Synthesized** impedance is selected.

Some countries like Poland, South Africa and Slovenia require higher ring impedance which is achieved by the DAA module, when Ringer Impedance is set to Synthesized impedance. Default: High.

- Set the **Ringer Threshold** to the desired value. This parameter defines the level below which the FXO Port would not validate the Ring signal and the level above which it would validate the Ring signal. Set Ringer Threshold — 13.5 - 16.5, 19.35 - 23.65 or 40.5 - 49.5. Default: 13.5 - 16.5 Vrms.
- Click **Submit** to save.

## DTMF Settings

- Click **DTMF Settings** to expand.

**DTMF Settings**

Dial Type: DTMF

DTMF Generation ON Time: 70 msec

DTMF Generation OFF Time: 70 msec

- Select the appropriate **Dial Type** option as supported by your CO Network. You can select — Pulse or DTMF. Default: DTMF.
- If you have selected **Dial Type** as **Pulse**, you must set the **Pulse Dial Ratio** as per your country. You can select — 33:67, 40:60 or 50:50. Default: 33:67.
- Select the appropriate **DTMF Generation ON Time** for the FXO Port. This is the time for which the DTMF digit which is to be outdialed remains ON. Valid range is 50 to 200 msec. Default: 70 msec.
- Select the appropriate **DTMF Generation OFF Time** for the FXO Port. This is the time for which the system should wait before dialing the successive DTMF digits so that the CO Network can detect the dialed digits. Valid range is 50 to 200 msec. Default: 70 msec.
- To configure the next FXO Port, click the FXO Port number and follow the same instructions as given earlier.

## Copy Port Parameters

- You can also copy the settings of a FXO Port to another FXO Port using the **Copy** button. To do this,
- Click the **Copy** button. The **Copy FXO Port Parameters** window opens.

Copy FXO Port Parameters from FXO Port: 1 to

All: ☐

FXO Port 1: ☐ FXO Port 2: ☐

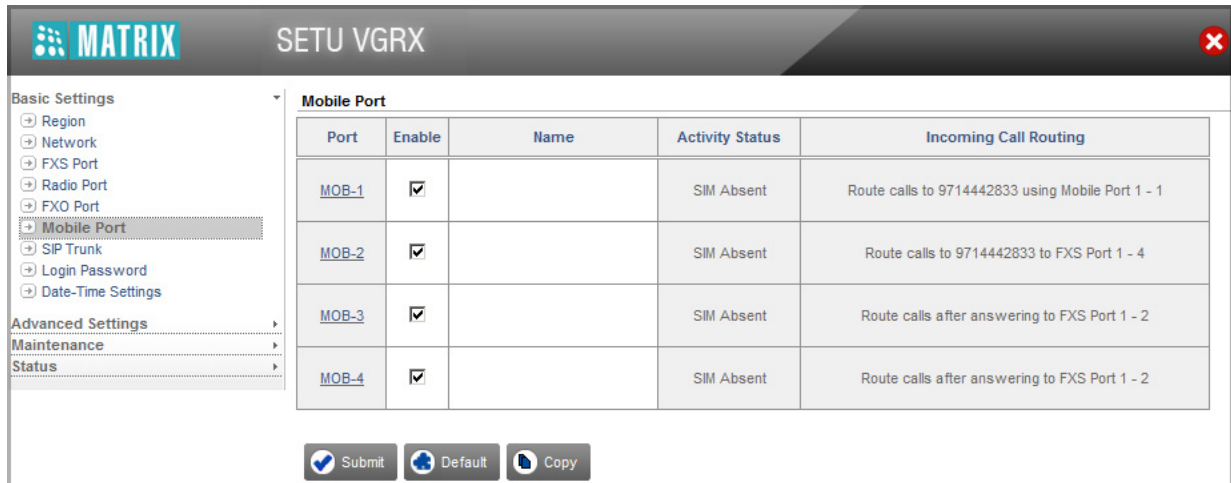
OK Close

- In **Copy FXO Port Parameters from FXO Port**, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the FXO Ports, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the FXO Port you copied the settings to.

# Mobile Port

To configure a Mobile Port,

- Click the **Basic Settings** link to expand.
- Click the **Mobile Port** link.



The screenshot shows the MATRIX SETU VGRX interface. On the left is a sidebar with a tree view containing 'Basic Settings' (expanded), 'Advanced Settings', 'Maintenance', and 'Status'. Under 'Basic Settings', there are links for Region, Network, FXS Port, Radio Port, FXO Port, Mobile Port (selected), SIP Trunk, Login Password, and Date-Time Settings. The main area is titled 'Mobile Port' and contains a table with the following data:

Port	Enable	Name	Activity Status	Incoming Call Routing
<a href="#">MOB-1</a>	<input checked="" type="checkbox"/>		SIM Absent	Route calls to 9714442833 using Mobile Port 1 - 1
<a href="#">MOB-2</a>	<input checked="" type="checkbox"/>		SIM Absent	Route calls to 9714442833 to FXS Port 1 - 4
<a href="#">MOB-3</a>	<input checked="" type="checkbox"/>		SIM Absent	Route calls after answering to FXS Port 1 - 2
<a href="#">MOB-4</a>	<input checked="" type="checkbox"/>		SIM Absent	Route calls after answering to FXS Port 1 - 2

At the bottom of the table are three buttons: 'Submit' (with a checkmark icon), 'Default' (with a circular arrow icon), and 'Copy' (with a document icon).

The Mobile Port page displays the following parameters:

- **Port:** Displays the Mobile Port numbers. To configure the Mobile Port parameters, click on the desired Mobile Port number link.
- **Enable:** Keep the **Mobile Ports** enabled. Clear the Mobile Port **Enable** check box, only if you do not want to use the Port.
- **Name:** You may assign a Name to the Mobile Port, for identification.
- **Activity Status:** Displays the port activity status.
- **Incoming Call Routing:** Displays the Incoming Call Routing Method selected for the Mobile Port.

To configure the **Mobile Port**,

- Click **MOB-1**.

The **Mobile Port-1** window opens.

**Mobile Port - 1**

**Mobile Port** ☒ Enable

**Name**

**Activity Status** Network Present

+ **General**

+ **Handling of Incoming Calls**

+ **Handling of Outgoing calls**

+ **Call Minutes**

+ **SIM Balance Inquiry**

+ **SIM Recharge**

+ **DTMF Settings**

**Note:**  
Following parameters/features are not applicable for CDMA engine:  
SIM PIN, Band Selection, Preferred Network Mode, Network Selection, SMSC.  
CLIR, Route Calls returned Unconnected to Original Caller, SMS, SIM Balance Inquiry, SIM Recharge.

This note will be displayed in the Jeeves only when CDMA module is installed in your SETU VGRX.

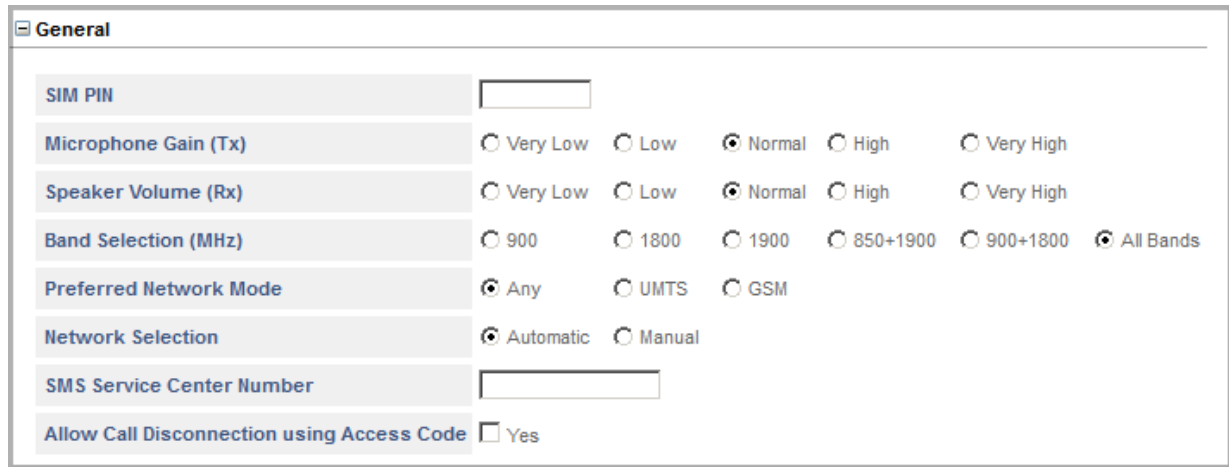
- Keep the **Mobile Port** enabled.

Clear this check box, only if you do not want to use this port. Default: Enabled.

- You can assign a **Name** to the Mobile Port. The Name can be a maximum of 24 characters. Default: Blank.
- **Activity Status** displays the port activity status — Module Initialization, SIM PUK Required, SIM PIN Wrong, SIM Absent, SIM Present, Network Absent, Network Present.

## General

- Click **General**.



General	
SIM PIN	<input type="text"/>
Microphone Gain (Tx)	<input type="radio"/> Very Low <input type="radio"/> Low <input checked="" type="radio"/> Normal <input type="radio"/> High <input type="radio"/> Very High
Speaker Volume (Rx)	<input type="radio"/> Very Low <input type="radio"/> Low <input checked="" type="radio"/> Normal <input type="radio"/> High <input type="radio"/> Very High
Band Selection (MHz)	<input type="radio"/> 900 <input type="radio"/> 1800 <input type="radio"/> 1900 <input type="radio"/> 850+1900 <input type="radio"/> 900+1800 <input checked="" type="radio"/> All Bands
Preferred Network Mode	<input checked="" type="radio"/> Any <input type="radio"/> UMTS <input type="radio"/> GSM
Network Selection	<input checked="" type="radio"/> Automatic <input type="radio"/> Manual
SMS Service Center Number	<input type="text"/>
Allow Call Disconnection using Access Code	<input type="checkbox"/> Yes

- If you have enabled SIM PIN protection on the SIM Card, in **SIM PIN**, enter the same SIM PIN value (4 to 8 digits). Default: Blank.



- Do not configure the **SIM PIN**, if your SETU VGRX has a CDMA module.*
- SIM PIN is not set to default or does not change, if SETU VGRX is set to default or when you upgrade/downgrade firmware.*
- You can adjust the **Microphone Gain (Tx)** of the Mobile Port to improve the audibility of transmitting speech from SETU VGRX. Select the desired Tx Gain — **Very Low, Low, Normal, High** and **Very High**. Default: Normal.
- You can adjust the **Speaker Volume (Rx)** of the Mobile Port to improve the audibility of incoming speech. Select the desired Rx Gain — **Very Low, Low, Normal, High** or **Very High**. Default: Normal.
- The Frequency **Band Selection (MHz)** supported by the GSM networks varies from country to country.

You can select the Frequency Band used by the GSM Service Provider(s) in the country where SETU VGRX is installed. The supported bands are — 900, 1800, 1900, 850+1900, 900+1800 and All bands. Default: All Bands.

For instance, select 850 + 1900 GSM frequency band for countries which support both 850 and 1900 MHz frequencies for GSM network. Similarly, set 900 + 1800 frequency band for countries that support 900 or 1800 GSM frequency band.



**Band Selection** is not applicable, if a CDMA module is installed in your SETU VGRX.

- If your SETU VGRX has a UMTS Mobile Port, the SIM Card of this port will get registered with either GSM (2G) or UMTS (3G) network, whichever is available. You can select the Network with which the SIM should be registered by setting the **Preferred Network Mode**.

If the SIM you have installed in the UMTS Mobile Port supports both GSM and UMTS services, but you want the SIM to get registered with one of these networks, you can restrict the SIM registration with a particular network by setting the Preferred Network Mode. You may select:

- **Any:** Select this option, when you want the SIM to get registered with the UMTS (3G) network when available and to shift automatically to the GSM (2G) network when UMTS (3G) becomes unavailable.
- **UMTS:** Select this option, when you want the SIM to get registered with UMTS (3G) network only.
- **GSM:** Select this option, when you want the SIM to get registered with GSM (2G) network only.

Default: Any.

If your Mobile Port supports GSM only, do not change the default value of this parameter.



**Preferred Network Mode** is not applicable, if a CDMA module is installed in your SETU VGRX.

- Select the desired **Network Selection** mode — **Automatic** or **Manual**. Default: Automatic.



You must select **Automatic** as the **Network Selection** mode, if a CDMA module is installed in your SETU VGRX. CDMA module does not support Manual mode of Network Selection.

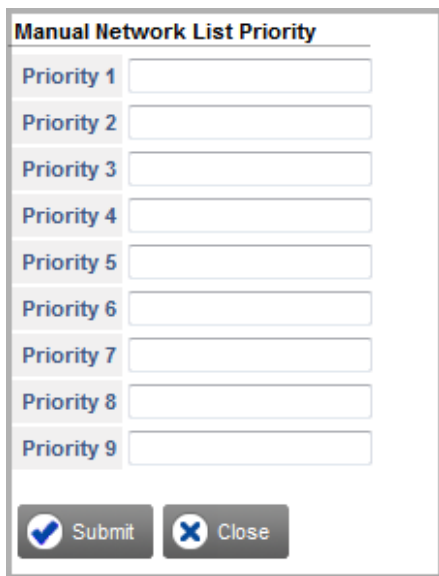
- Select **Automatic** when you want the Mobile Port to automatically locate and register with the Network supported by the SIM card. At each power ON, the SIM in the Mobile Port will automatically register with the Network.
- Select **Manual** when you want the Mobile Port to select the network operator according to the priority set by you. You also need to configure the list of network operators as per your preference.

To apply **Manual** Network Selection,

- Select **Manual** and click **Settings** ➔ .

The screenshot shows the 'General' settings window. The 'Network Selection' option is set to 'Manual', and the 'Settings' button next to it is highlighted with a red box. Other settings visible include 'SIM PIN', 'Microphone Gain (Tx)', 'Speaker Volume (Rx)', 'Band Selection (MHz)', 'Preferred Network Mode', 'SMS Service Center Number', and 'Allow Call Disconnection using Access Code'.

- The **Manual Network List Priority** window opens.



- In the Priority levels, **Priority1** to **Priority 9**, enter the Network Operator Codes with which you want the SIM to register, as per your preference.

The Network Operator Code may consist of a minimum of 5 digits and a maximum of 8 digits.  
Default: Blank.

- Click **Submit** and then close the **Manual Network List Priority** window.
- Enter the **SMS Service Center Number**, as provided to you by your service provider. This number is used by the system while sending the SMS notifications.



**SMS Service Center Number** is not applicable, if a CDMA module is installed in your SETU VGRX.

- Select the **Allow Call Disconnection using Access code** check box to enable the *Disconnect Call using Access Code* feature on the Mobile Port. For further details, see [“Disconnecting a Call using Access Code”](#).
- Click **Submit** to save.

## Handling of Incoming Calls

- Click **Handling of Incoming Calls**.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port ☐ Yes

Route all Incoming calls (with CLI) after Answering the Call and Collecting the Digits ▼

Block Calls received without CLI on this Mobile Port ☐ Yes

Route all Incoming calls (without CLI) after Answering the Call and Collecting the Digits ▼

**Answering the call and collecting the digits**

Prompt caller to enter PIN ☐ Yes

Dial Plan 1 ▼ ➔

First Digit Wait Timer 7 Seconds

Inter Digit Wait Timer 5 Seconds

End Of Dialing Digit # ▼

Minimum Number of digits that must be dialed by the caller 02 ▼

Maximum Number of digits that can be dialed by the caller 24 ▼

If No Digit dialed during First Digit Wait Timer Disconnect Call ▼

Allow making New Call using Access code ☐ Yes

Select Destination Port for routing calls Fixed ▼ ➔

Allowed-Denied Logic ☐ Apply

Reject Calls from Blacklisted Callers ☐ Apply

- If you do not want to route calls received on this Mobile Port, select **Block all calls received on this Mobile Port** check box. Default: Disabled.

## Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods.
  - without any Destination Number
  - to the Fixed Destination Number
  - on the basis of Calling Party Number
  - after Answering the Call and Collecting the DigitsDefault: after Answering the Call and Collecting the Digits

**Handling of Incoming Calls**

Block all calls received on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (with CLI)	after Answering the Call and Collecting the Digits
Block Calls received without CLI on this Mobile Port	
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits

Read further for instructions on selecting and configuring each of these destination number determination methods.



*If the destination number to be dialed out is an IP Address, SETU VGRX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. See “[IP Dialing](#)” for more details.*

### Route Calls without any Destination Number

In this method, all calls received on the Mobile Port are directly routed to the destination port, irrespective of the Destination Number.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (with CLI)	without any Destination Number
Block Calls received without CLI on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits

- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

### Route to the Fixed Destination Number

In this method, calls received on the Mobile Port are routed to a fixed destination number, which you must configure for this port.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (with CLI)	to the Fixed Destination Number
Block Calls received without CLI on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits

Fixed Destination Number

Fixed Destination Number

To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In **Fixed Destination Number**, enter the desired destination number.

The Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, \*, # and (.) dot are allowed. Default: Blank.


- Click **Submit** to save the changes.

### Route on the basis of Calling Party Number

In this method, a call received on the Mobile Port is routed to a specific destination number, as per the calling party's number. For this, the calling party numbers and their corresponding destination numbers must be configured in the Calling Party Number Based Table.

Whenever there is an incoming call on the Mobile Port, SETU VGRX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call will be routed to the destination number configured for that Calling Party Number.

To apply this method, do the following:


- In **Route all incoming calls (with CLI)**, select **on the basis of Calling Party Number**.
- Click **Settings** .

Handling of Incoming Calls


Block all calls received on this Mobile Port

☐ Yes

Route all Incoming calls (with CLI)

on the basis of Calling Party Number 


If no match found in the Calling Party Number Table, route calls

after Answering the Call and Collecting the Digits 

Block Calls received without CLI on this Mobile Port

☐ Yes

Route all Incoming calls (without CLI)

after Answering the Call and Collecting the Digits 

The **Mobile Port Destination Number Determination: Calling Number Based** Table opens.

1-100101-200201-300301-400401-499

Mobile Port - Destination Number Determination: Calling Number Based

Index	Calling Number	Destination Number	Allow Callback?
001			<input type="checkbox"/>
002			<input type="checkbox"/>
003			<input type="checkbox"/>
004			<input type="checkbox"/>
005			<input type="checkbox"/>
006			<input type="checkbox"/>
007			<input type="checkbox"/>
008			<input type="checkbox"/>
009			<input type="checkbox"/>
010			<input type="checkbox"/>
011			<input type="checkbox"/>
012			<input type="checkbox"/>

☒ Submit

☐ Default All

☐ Close

- Configure the **Calling Number Based** table for the Mobile Port. You can enter upto 499 Calling Party Numbers and their corresponding Destination Numbers in this table.
- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.
- In **Destination Number**, enter a corresponding destination number for each calling party number. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, \*, # and (.) dot are allowed. Default: Blank.
- Click **Submit** to save your entries. Close the window to return to the Mobile Port window.

You can also configure the **Calling Number Based** Table from *Advanced Settings* link. For instructions, see ["Destination Number Determination"](#) under *Advanced Settings*.

- For incoming calls with Calling Party Numbers that do not match with the Calling Party Number Table, you may select the destination number determination method.

In **If no match found in the Calling Party Number Table, route calls**, you may select either **to the Fixed Destination Number** or **after Answering the Call and Collecting the Digits**. Default: after Answering the Call and Collecting the Digits.

## Route After Answering the Call and Collecting the Digits

In this method, the system answers the incoming call on the Mobile Port and plays dial tone to the caller, allowing the caller to dial the desired number. The number dialed by the caller is considered as the destination number.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (with CLI)	after Answering the Call and Collecting the Digits
Block Calls received without CLI on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits

**Answering the call and collecting the digits**

Prompt caller to enter PIN	<input type="checkbox"/> Yes
Dial Plan	1 →
First Digit Wait Timer	7 Seconds
Inter Digit Wait Timer	5 Seconds
End Of Dialing Digit	#
Minimum Number of digits that must be dialed by the caller	02
Maximum Number of digits that can be dialed by the caller	24
If No Digit dialed during First Digit Wait Timer	Disconnect Call
Allow making New Call using Access code	<input type="checkbox"/> Yes

To apply this method, do the following:

- In **Route all incoming calls (with CLI)**, select **after Answering the Call and Collecting the Digits**.

The related parameters appear under **Answering the Call and Collecting the Digits**.

- If you want to enable PIN Authentication on the Mobile Port, select the **Prompt caller to enter PIN** check box.

If you enable this check box, you must also configure the PIN Authentication Table. To know more about this feature and for detail instructions, see [“PIN Authentication”](#) under *Advanced Settings*.

- SETU VGRX supports 8 Dial Plans with total 64 entries in each table. When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then route the call as per the Destination Port Selection method configured.

Select the **Dial Plan** table number you configured for this port. If you have not configured the Dial Plan table you may do so now,

- Click **Settings** . The Dial Plan Table opens.
- Configure the numbers in the table. For detailed instructions, see [“Dial Plan”](#).

- Set the duration of the **First Digit Wait Timer (FDWT)**. This is the duration for which you want the system to wait for the caller to dial the destination number after the dial tone. Valid range is 01 to 99 seconds. Default: 7 seconds.
- Set the duration of the **Inter Digit Wait Timer**. This is the duration for which you want the system to wait while receiving the digits dialed by the caller to consider it as End of Dialing. You may change this timer, if required. Valid range is 01 to 99 seconds. Default: 5 seconds.
- As **End of Dialing Digit**, select whether the system should consider # or \* as termination digit to detect end of dialing. Default: #
- In **Minimum number of digits that can be dialed by the caller**, select the minimum number of digits to be dialed by the user for the system to consider it as a valid number. Valid range is 01 to 24 digits. Default: 02 digits.
- In **Maximum number of digits that can be dialed by the Caller**, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing. Valid range is 01 to 24 digits. Default: 24 digits.

When the caller dials a number, the system will match it with the above end-of-dialing indications and accept the one that matches first.

- If the caller fails to dial the number during the First Digit Wait Timer, you can either have the system disconnect the call or route the call to a fixed destination number.

In **If No Digit dialed during First Digit Wait Timer (FDWT)**, you can select — **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

- If you select **Use Fixed Destination Number**, enter the desired destination number in **Fixed Destination Number**. The Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, \*, # and (.) dot are allowed. Default: Blank.



- *The First Digit Wait Timer is loaded as soon as the system answers the call.*
- *When you dial the first digit, the First Digit Wait Timer is stopped and the system loads the Inter Digit Wait Timer.*
- *SETU VGRX reloads the Inter Digit Wait Timer:*
  - *each time you dial a new digit till the termination digit is detected.*
  - *each time you dial a new digit till the entry is not matched in Dial Plan.*
  - *until you have dialed the maximum number of digits configured as End of Dialing.*
- Select the **Allow making New Call using Access Code** check box, if you want to enable the feature *Making New Call using Access Code* on the Mobile Port. See [“Making a New Call using Access Code”](#).
- Click **Submit** to save settings.

- If you do not want to route calls without CLI through this port, select the **Block Calls received without CLI on this Mobile Port** check box.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (with CLI)	after Answering the Call and Collecting the Digits
Block Calls received without CLI on this Mobile Port	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	after Answering the Call and Collecting the Digits
<b>Answering the call and collecting the digits</b>	
Prompt caller to enter PIN	<input type="checkbox"/> Yes
Dial Plan	1

- To **Route all Incoming calls (without CLI)**, you may select from any of the following methods:
    - to the Fixed Destination Number, see ["Route to the Fixed Destination Number"](#).
    - after Answering the Call and Collecting the Digits, see ["Route After Answering the Call and Collecting the Digits"](#).
- Default: after Answering the Call and Collecting the Digits.

## Destination Port Determination

Select the Destination Port for routing calls for the Mobile Port. You may select from any of the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

Default: Fixed.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port ☐ Yes

Route all Incoming calls (with CLI) after Answering the Call and Collecting the Digits

Block Calls received without CLI on this Mobile Port ☐ Yes

Route all Incoming calls (without CLI) after Answering the Call and Collecting the Digits

**Answering the call and collecting the digits**

Prompt caller to enter PIN ☐ Yes

Dial Plan 1

First Digit Wait Timer 7 Seconds

Inter Digit Wait Timer 5 Seconds

End Of Dialing Digit #

Minimum Number of digits that must be dialed by the caller 02

Maximum Number of digits that can be dialed by the caller 24

If No Digit dialed during First Digit Wait Timer Disconnect Call

Allow making New Call using Access code ☐ Yes

Select Destination Port for routing calls Fixed

Allowed-Denied Logic

Reject Calls from Blacklisted Callers ☐ Apply

Read the description and follow the instructions for each of these destination port selection methods given below.

## Fixed

In this method, calls received on the Mobile Port are routed to a Fixed Destination Port, irrespective of the number dialed on the Mobile Port.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **Fixed** option.

- Click **Settings** ➡ .

Handling of Incoming Calls

Block all calls received on this Mobile Port

☐ Yes

Route all Incoming calls (with CLI)

after Answering the Call and Collecting the Digits

Block Calls received without CLI on this Mobile Port

☐ Yes

Route all Incoming calls (without CLI)

after Answering the Call and Collecting the Digits

Answering the call and collecting the digits

Prompt caller to enter PIN

☐ Yes

Dial Plan

1

First Digit Wait Timer

7

Seconds

Inter Digit Wait Timer

5

Seconds

End Of Dialing Digit

#

Minimum Number of digits that must be dialed by the caller

02

Maximum Number of digits that can be dialed by the caller

24

If No Digit dialed during First Digit Wait Timer

Disconnect Call

Allow making New Call using Access code

☐ Yes

Select Destination Port for routing calls

Fixed

Allowed-Denied Logic

☐ Apply

Reject Calls from Blacklisted Callers

☐ Apply

The **Destination Port/Group for Mobile Port** window opens.

Destination Port/Group for Mobile Port

Edit	Routing Group	Fallback Routing Group	CLI Number on FXS Port
➡	FXS Port 1 - 2 (Ascending)	None	Received Calling Party

✕

Close

The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to edit the default Routing Group options, click **Settings** ➡ .

The **Edit Selective Port/Group for Mobile Port** window opens.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.




*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports*, *Radio Ports* and *SIP Trunks*.
- To create a group of *not-sequential FXS Ports* as members,
  - Select a **FXS Group**.

- Select **FXS Group** number. Default: 1.
- Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for Mobile Port** window.
- Close the **Destination Port/Group for Mobile Port** window to return to the Handling of Incoming Calls window.

### On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SETU VGRX will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

- Click **Settings** ➡ .

Handling of Incoming Calls

Block all calls received on this Mobile Port

☐ Yes

Route all Incoming calls (with CLI)

after Answering the Call and Collecting the Digits

Block Calls received without CLI on this Mobile Port

☐ Yes

Route all Incoming calls (without CLI)

after Answering the Call and Collecting the Digits

Answering the call and collecting the digits

Prompt caller to enter PIN

☐ Yes

Dial Plan

1

First Digit Wait Timer

7 Seconds

Inter Digit Wait Timer

5 Seconds

End Of Dialing Digit

#

Minimum Number of digits that must be dialed by the caller

02

Maximum Number of digits that can be dialed by the caller

24

If No Digit dialed during First Digit Wait Timer

Disconnect Call

Allow making New Call using Access code

☐ Yes

Select Destination Port for routing calls

On the basis of Destination Number

Allowed-Denied Logic

☐ Apply

Reject Calls from Blacklisted Callers

☐ Apply

The **Mobile Port - Destination Port Determination - Destination Number Based** table window opens.

Mobile Port - Destination Port Determination - Destination Number Based

	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
	➡	No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1      1

Testing

Enter the destination number to know which entry would be selected for routing

Search

➕ Add

ⓧ Delete

✕ Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + “Wildcard Characters”). Valid characters are 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

## Wildcard Characters


SETU VGRX supports following characters.

Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.
<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.
<b>[ ^ ]</b>	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
<b>T</b> (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*


- Create the **Routing Group**.
  - To create a group of *sequential* **FXS Ports** as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.  
  
Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports, Radio Ports and SIP Trunks.
  - To create a group of *not-sequential* **FXS Ports** as members,
    - Select a **FXS Group**.
    - Select **FXS Group** number. Default: 1.
    - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **Mobile Port - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

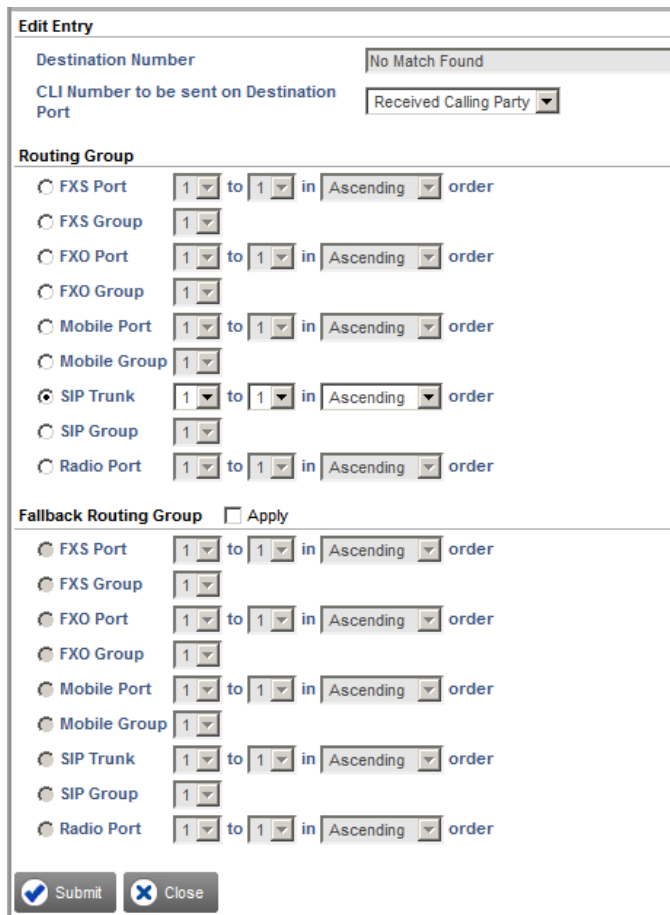


*If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- under Edit, click **Settings** .
- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Port Determination”](#) under *Advanced Settings*.

### On the basis of Calling Party Number

In this method, incoming calls on the Mobile Port are routed to a specific port as per the calling party's number.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.

- Click **Settings**  .

Handling of Incoming Calls

Block all calls received on this Mobile Port

☐ Yes

Route all Incoming calls (with CLI)

after Answering the Call and Collecting the Digits

Block Calls received without CLI on this Mobile Port

☐ Yes

Route all Incoming calls (without CLI)

after Answering the Call and Collecting the Digits

Answering the call and collecting the digits

Prompt caller to enter PIN

☐ Yes

Dial Plan

1

First Digit Wait Timer

7

Seconds

Inter Digit Wait Timer

5

Seconds

End Of Dialing Digit

#

Minimum Number of digits that must be dialed by the caller

02

Maximum Number of digits that can be dialed by the caller

24

If No Digit dialed during First Digit Wait Timer

Disconnect Call

Allow making New Call using Access code

☐ Yes

Select Destination Port for routing calls

On the basis of Calling Party Number

Allowed-Denied Logic



☐ Apply

Reject Calls from Blacklisted Callers


☐ Apply


The **Mobile Port - Destination Port Determination - Calling Number Based** table window opens.


Mobile Port - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	SIP Trunk 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 11

 Add

 Delete

 Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.

**Add Entry**

Calling Number

CLI Number to be sent on Destination Port

**Routing Group**

☐ FXS Port  to  in  order

☐ FXS Group

☐ FXO Port  to  in  order

☐ FXO Group

☐ Mobile Port  to  in  order

☐ Mobile Group

☒ SIP Trunk  to  in  order

☐ SIP Group

☐ Radio Port  to  in  order

**Fallback Routing Group** ☐ Apply

☐ FXS Port  to  in  order

☐ FXS Group

☐ FXO Port  to  in  order

☐ FXO Group

☐ Mobile Port  to  in  order

☐ Mobile Group

☐ SIP Trunk  to  in  order

☐ SIP Group

☐ Radio Port  to  in  order


- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, \*, #, +. Default: Blank.
- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports, Radio Ports and SIP Trunks.
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default: 1.
  - Click **Settings**  . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic "[Group](#)" under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **Mobile Port - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, SIP Trunk 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- under Edit, click **Settings**  .

- The **Edit Entry** window opens.

- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Port Determination”](#) under *Advanced Settings*.

## Allowed - Denied Logic

You can apply the Allowed-Denied logic on the Mobile Port (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed by the caller.
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed by the caller on the Mobile Port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SETU VGRX uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:

- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number

To apply Allowed - Denied Logic on the Mobile Port,

- Select the **Allowed - Denied Logic** check box.

**Handling of Incoming Calls**

Block all calls received on this Mobile Port ☐ Yes

Route all Incoming calls (with CLI) after Answering the Call and Collecting the Digits ▼

Block Calls received without CLI on this Mobile Port ☐ Yes

Route all Incoming calls (without CLI) after Answering the Call and Collecting the Digits ▼

**Answering the call and collecting the digits**

Prompt caller to enter PIN ☐ Yes

Dial Plan 1 ▼ ➕

First Digit Wait Timer 7 Seconds

Inter Digit Wait Timer 5 Seconds

End Of Dialing Digit # ▼

Minimum Number of digits that must be dialed by the caller 02 ▼

Maximum Number of digits that can be dialed by the caller 24 ▼

If No Digit dialed during First Digit Wait Timer Disconnect Call ▼

Allow making New Call using Access code ☐ Yes

Select Destination Port for routing calls Fixed ▼ ➕

Allowed-Denied Logic ☒ Apply


Allowed Number List 01 ▼ ➕

Denied Number List 02 ▼ ➕


Reject Calls from Blacklisted Callers ☐ Apply

- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the Mobile Port. Default: 01

If you have not configured the Allowed Number List,

- Click **Settings** . The Number Lists window opens.
- You may configure the default Allowed Number List or any other list. See [“Number Lists”](#) to configure the allowed numbers.
- Click **Submit** to save the Allowed Number List and close the window.
- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the Mobile Port. Default: 02.

If you have not configured the Denied Number List,

- Click **Settings** . The Number Lists window opens.
- You may configure the default Denied Number List or any other list. See [“Number Lists”](#) to configure the restricted numbers.
- Click **Submit** to save the Denied Number List and close the window.

## Black Listed Callers

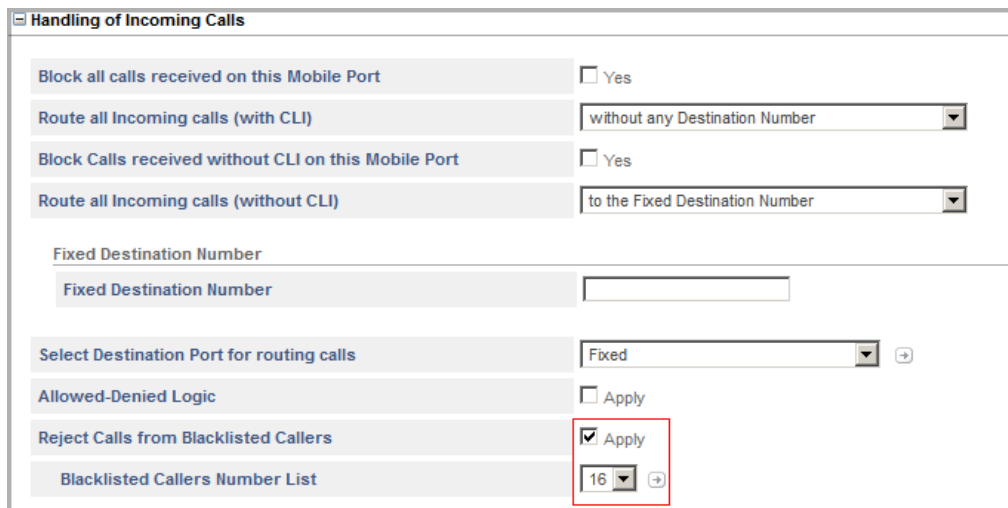
With the Black Listed Callers feature you can block incoming calls from specific numbers on the Mobile Port. Thus all incoming calls from the numbers you have 'blacklisted' will be automatically rejected by SETU VGRX.

To apply Black Listed Callers on Mobile Port,

- Select the **Reject Calls from Blacklisted Callers** check box.
- In the **Blacklisted Callers Number List**, select the list number you have configured with the numbers of unwanted callers. Default:16

If you have not configured the Blacklisted Callers Number List,

- Click **Settings** .



The Number Lists window opens.



The screenshot shows the 'Number Lists' window. At the top, there are tabs for different ranges: 1-4, 5-8, 9-12, 13-16 (selected), 17-20, and 21-24. Below the tabs is a table with the following structure:

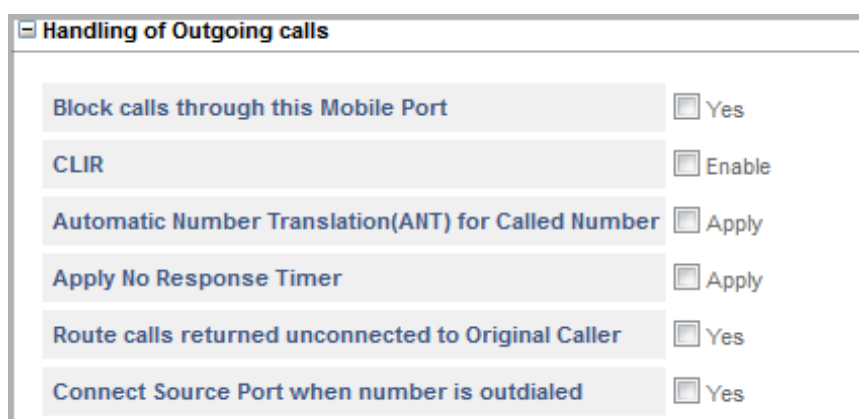
Location	List 13	List 14	List 15	List 16
01				
02				
03				
04				
05				
06				
07				
08				
09				
10				
11				
12				

At the bottom of the window, there are three buttons: 'Submit' (with a checkmark icon), 'Default' (with a plus icon), and 'Close' (with an X icon).

- You may configure the default Blacklisted Callers Number List or any other list. See [“Number Lists”](#) to configure the numbers of unwanted callers.
- Click **Submit** and close the window.

## Handling of Outgoing Calls

When Mobile Port is determined as the destination port, numbers dialed from this port constitute outgoing calls.



The screenshot shows the 'Handling of Outgoing calls' window. It contains the following settings:

Block calls through this Mobile Port	<input type="checkbox"/> Yes
CLIR	<input type="checkbox"/> Enable
Automatic Number Translation(ANT) for Called Number	<input type="checkbox"/> Apply
Apply No Response Timer	<input type="checkbox"/> Apply
Route calls returned unconnected to Original Caller	<input type="checkbox"/> Yes
Connect Source Port when number is outdialed	<input type="checkbox"/> Yes

- If you do not want to route outgoing calls through this Mobile Port, select the **Block calls through this Mobile Port** check box. Default: Disabled.
- By default, the CLI of the Mobile Port is sent to the called party when outgoing calls are made using the Mobile Port. If you do not want to send CLI, enable the **CLIR** check box. Default: Disabled.



**CLIR** is not applicable, if a CDMA module is installed in your SETU VGRX.

- You can apply **Automatic Number Translation logic** on outgoing calls made from the Mobile Port.
- To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.

Handling of Outgoing calls	
Block calls through this Mobile Port	<input type="checkbox"/> Yes
CLIR	<input type="checkbox"/> Enable
Automatic Number Translation(ANT) for Called Number	<input checked="" type="checkbox"/> Apply
Use Automatic Number Translation Table	1 [dropdown arrow] [right arrow]
Pause Timer	2 [dropdown arrow] Seconds
Apply No Response Timer	<input type="checkbox"/> Apply
Route calls returned unconnected to Original Caller	<input type="checkbox"/> Yes
Connect Source Port when number is outdialed	<input type="checkbox"/> Yes

- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Mobile Port. Default: Table 1.

If you have not configured the Automatic Number Translation Table,

- Click **Settings**  . The **Automatic Number Translation Table** window opens.

1
2
3
4
5
6
7
8

**Automatic Number Translation Table - 1**

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	

**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' to every 7-digit dialed number.

- You may configure the default Automatic Number Translation Table or any other Table. See [“Automatic Number Translation \(ANT\)”](#) to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.
- Select the **Apply No Response Timer** check box. The system will route the call through a Fallback Routing Group or Port, if a response other than—Alert, Connect, Busy, No Reply, Disconnect with cause as Busy or No Reply—is received from the network within the specified time period. Default: Disabled.
- Set the duration of the **No Response Timer**, if you have enabled the *Apply No Response Timer* option. It is the time for which SETU VGRX will wait for the valid response from the network for any request. If no valid response is received before the expiry of this timer, SETU VGRX will fallback to alternate Routing Group or Port for further processing of the call. Valid range is 01 to 99 seconds. Default: 10 seconds.



To apply Fallback logic on the Mobile Port, make sure you have enabled Fallback Routing Group under [“Destination Port Determination”](#).

- Enable **Route calls returned unconnected to Original Caller**, if you want SETU VGRX to route outgoing calls made from this port that return unconnected back to the original caller.

If you enable this feature, when an outgoing call is made using this port, and the Called Party is found busy or does not respond, SETU VGRX stores the number of the calling party, the number of the called party and this port (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

If the called party returns the call before the expiry of this Timer, SETU VGRX checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see [“System Parameters”](#).



**Route calls returned unconnected to Original Caller** is not applicable, if a CDMA module is installed in your SETU VGRX.

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, enable the **Connect Source Port when number is outdialed** check box. Default: Disabled.

In all Destination Number Determination methods except *After Answering the Call and Collecting the Digits*, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

- Click **Submit** to save.




If you enable **Connect Source Port when number is outdialed**, *“Making a New Call using Access Code”* feature will not be allowed to the users.


## Call Minutes

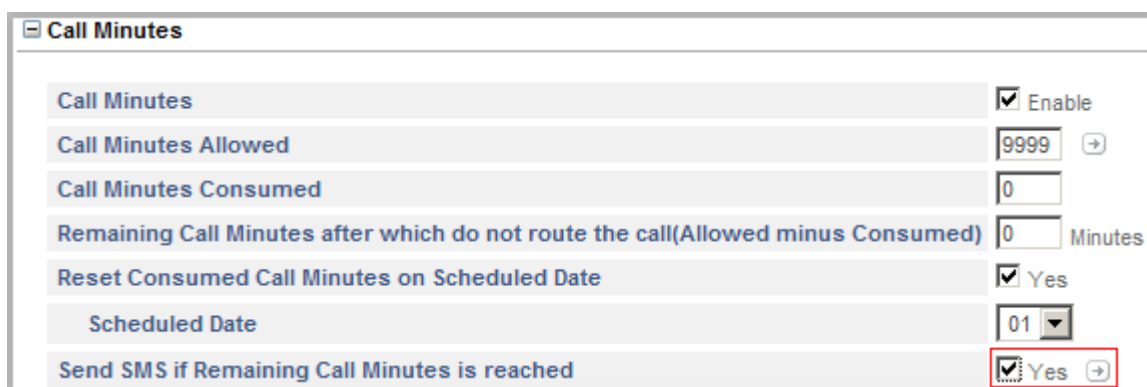
Mobile Service Providers offer different tariff schemes to their subscribers. For example, mobile service providers in India offer first 500 Minutes free, CUG calling, first 500 minutes calling at 30 paise. SETU VGRX allows you to take advantage of these tariff schemes by configuring Call Minutes on the Mobile Port.

- Click **Call Minutes**.

- Select the **Call Minutes** check box to apply this feature.
- The **Call Minutes Allowed** displays the minutes allotted to the Mobile Port for making outgoing calls. Valid range is 0000 to 9999. Default: 9999.
- To change the value of Call Minutes Allowed,
  - Click **Settings**  .

- The **Call Minutes** window opens.
  - In **Call Minutes**, enter the desired value.
  - Click the **Set Minutes** button. The value entered appears in the **Call Minutes Allowed**.
  - If you want to add minutes to the existing value,
    - In **Call Minutes**, enter the desired value you want to add.
    - Click the **Add Minutes** button. The value you entered gets added to the existing minutes and appears in the **Call Minutes Allowed**.
- Close the window to return to the main page.

- The **Call Minutes Allowed** displays the Call Minutes set/added by you.
- The minutes consumed are displayed in **Call Minutes Consumed**.
- You may block the outgoing calls from the Mobile Port after certain call minutes have been consumed. To do so, configure the **Remaining Call Minutes after which do not route the call (Allowed minus Consumed)**. The system will block the outgoing calls made from this port when the call minutes you configured are remaining. Valid range is 000 to 999. Default: 000.
- Select the **Reset Consumed Call Minutes on Scheduled Date** check box, if you want the system to automatically reset the value of Call Minutes on a scheduled date. In the **Scheduled Date**, select the desired date.
- If you want the system to send SMS notification for the Remaining Call Minutes after which the call will not be routed,
  - Select the **Send SMS if Remaining Call Minutes is reached** check box.
  - Click **Settings** .



The **Send SMS if Remaining Call Minutes is reached** window opens.



- In the **To** fields, you may enter up to 3 different Mobile Numbers to which the SMS should be sent. The numbers can be a maximum of 24 digits. The digits allowed are 0-9, \*, # and +.
- In **SMS Text**, enter the text you want to be sent in the SMS Notification. The text length can be a maximum of 80 characters.
- In the **Using** list, select the Mobile Port number which the system should use to send the SMS.

- Click **Submit** to save and close the window to return to the main page.



**SMS** facility will not be supported, if a CDMA module is installed in your SETU VGRX.

## SIM Balance and Recharge

SETU VGRX supports Balance Inquiry and Recharging of the SIM Card installed in its Mobile Ports<sup>3</sup>.



You will not be able to send request for **SIM Balance** inquiry and **SIM Recharge**, if a CDMA module is installed in your SETU VGRX.

To be able to use this feature, first collect the following information from your Network Operator:

- **Balance Inquiry Number:** This is the number provided by the Network Operator to the subscribers to check Balance. Different Network Operators have different numbers. For example, the Balance Inquiry number of Vodafone is **\*141#**.
- **Recharging Service Number:** This is the number provided by the Network Operators to their subscribers for Recharging Service. Different Network Operators have different numbers for Recharging Service. For example, the Recharging Service Number of Vodafone is **\*140\***.

## SIM Balance Inquiry

To check the SIM Balance using Jeeves,

- Click **SIM Balance Inquiry**.

- In **Balance Inquiry Number**, enter the number provided to you by the Network Operator to check Balance. The number can be a maximum of 16 digits. The digits allowed are 0-9, \* and #.
- Click **Submit**.
- To run the query, click the **Balance Inquiry** button.

3. SETU VGRX supports Unstructured Supplementary Service Data (USSD), the standard for transmitting information over CSM signaling channels and a commonly used method to query the available balance and other similar information in pre-paid GSM services.

- If you want the system to check the SIM Balance at fixed intervals, select the **Balance Inquiry on Scheduled Basis** check box and configure the following:

- In **Schedule Time - at**, enter the day and the time when you want the system to make the Balance query.
- Select the **Balance Inquiry on every Power ON of the system** check box, if you want the system to make the SIM Balance query at every Power ON.
- The response received from the mobile network (including possible error messages) will be displayed under **USSD Reply**.

## SIM Recharge

To Recharge the SIM,

- Click **SIM Recharge**.

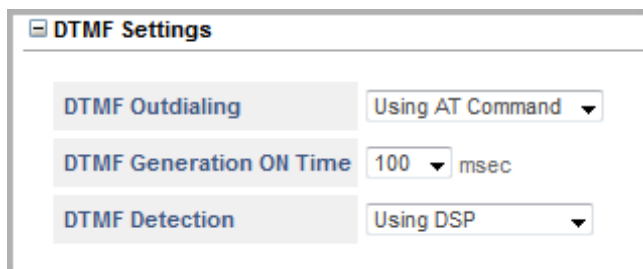
- In **Recharge Number**, enter the number provided to you by the Network Operator to recharge the SIM. The number can be a maximum of 8 digits. The digits allowed are 0-9, \* and #.
- Click **Submit**.
- Click the **Recharge** button.
- A new window opens. In **Enter Recharge PIN Number**, enter the number printed on the Recharge Voucher.

- Click **OK**.
- The response received from the mobile network (including possible error messages) will be displayed under **USSD-Reply**.

## DTMF Settings

To configure the DTMF Settings for the Mobile Port,

- Click **DTMF Settings**.



- Select the desired **DTMF Outdialing** option as **Inband** or **Using AT Command**. Default: Using AT Command.
- Select the appropriate **DTMF Generation ON Time** for the Mobile Port, if you select *DTMF Outdialing* as *Using AT Command*. This parameter determines the time for which the DTMF digit will remain ON, while being out dialed by the system. You may select 100 msec or 200 msec. Default: 100 msec.
- Select the appropriate **DTMF Detection** option. You may select **Using DSP** or **Using GSM Engine**. Default: Using DSP.

If you have selected DTMF Detection option as *Using GSM Engine*,

- Select the appropriate **DTMF Detection Minimum ON Duration**. Valid range is 20 to 100 msec. Default: 30 msec.



*If a CDMA module is installed in your SETU VGRX, select DTMF Detection as **Using GSM Engine** and set DTMF Detection Minimum ON Duration as **20 msec**.*

## Copy Port Parameters

- You can also copy the settings of a Mobile Port to another Mobile Port using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy Mobile Port Parameters** window opens.



Copy Mobile Port Parameters from Mobile Port 1 to

All ☐

Mobile Port 1 ☐ Mobile Port 2 ☐ Mobile Port 3 ☐ Mobile Port 4 ☐

OK Close

- In **Copy Mobile Port Parameters from Mobile Port**, select the number of the port you want to copy settings *From*. Select the check box of the respective port numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the ports, select the **All** check box.
- Click the **OK** button.

Once you have copied the settings, you can again edit the specific parameters of the Mobile Port you copied the settings to.

# SIP Trunk

SETU VGRX supports 9 SIP Trunks. You can register all SIP Trunks with the same ITSP or with different ITSPs. These SIP Trunks may be configured as Proxy or Peer-to-Peer (non-proxy).

- Click the **Basic Settings** link to expand.
- Click the **SIP Trunk** link.

Trunk	Enable	Name	Status	SIP ID	SIP Registration	SIP Network Profile	Incoming Call Routing
<a href="#">SIP-1</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-2</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-3</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-4</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-5</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-6</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-7</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-8</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2
<a href="#">SIP-9</a>	<input type="checkbox"/>		Disabled		<input checked="" type="checkbox"/>	<a href="#">Network Profile 1</a>	Route calls to number received in INVITE message to FXS Port 1 - 2

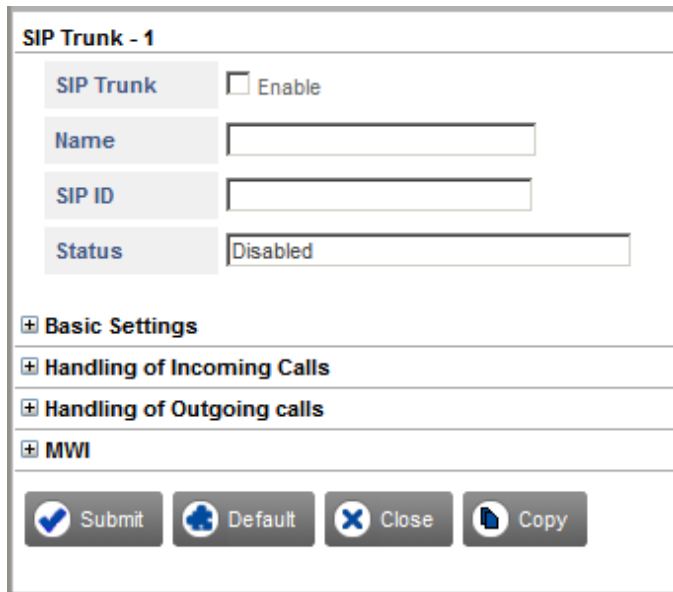
The SIP Trunk page displays the following:

- **Trunk:** Displays the SIP Trunk numbers. To configure the SIP Trunk parameters, click on the desired SIP Trunk number link.
- **Enable:** Click the check box to enable the desired SIP Trunk.
- **Name:** You may assign a Name to the SIP Trunk, for identification.
- **Status:** Displays the status of the SIP Trunk.
- **SIP ID:** Displays the SIP ID assigned to the SIP Trunk.
- **SIP Registration:** By default, SIP Registration check box is enabled for all SIP Trunks. You may clear the check box, if you want to disable the registration.
- **SIP Network Profile:** Displays the Network Profile assigned to the SIP Trunk. To configure the Network Profile, click on the Network Profile link.
- **Incoming Call Routing:** Displays the Incoming Call Routing Method selected for the SIP Trunk.

To configure the **SIP Trunk 1** parameters,

- Click **SIP-1**.

The **SIP Trunk-1** window opens.



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

Calls will not be routed through this trunk, if disabled.

- You can assign a **Name** to the SIP Trunk for identification. The Name can be a maximum of 24 characters. Default: Blank.

Assign the name of the ITSP with which the trunk is registered, or any other name of your choice. The name will appear on the display of the remote party's phone, when a call is made through this SIP Trunk.

- In **SIP ID**, the SIP ID that you assign under "[Basic Settings](#)" is displayed.
- **Status** displays the status of the SIP Trunk.

## Basic Settings

- Click **Basic Settings**.

Basic Settings	
SIP ID	<input type="text"/>
Authentication ID	<input type="text"/>
Authentication Password	<input type="text"/>
SIP Network Profile	Network Profile 1
SIP Registration	<input checked="" type="checkbox"/> Enable
Maximum Calls	8
FAX Protocol	<input checked="" type="radio"/> T.38 <input type="radio"/> Pass Through
Allow Call Disconnection using Access code	<input type="checkbox"/> Yes

- In **SIP ID**, enter the SIP ID provided by your ITSP. For example, if the SIP URI provided by the ITSP is 12345@abc.com, enter 12345. Default: Blank.

The SIP ID is the number which remote parties will use to call this SIP Trunk.

The SIP ID may be a number or text consisting of a maximum of 40 characters.

- Enter the **Authentication ID** (User ID) provided by your ITSP. Default: Blank.
- Enter the **Authentication Password** provided by your ITSP. Default: Blank.
- In **SIP Network Profile**, you can either select the default **Network Profile 1** or **Add New Network Profile**.
  - Click **Settings** to configure the parameters of the selected Network Profile. For instructions, see [“SIP Network Profile”](#).

You can also configure the SIP Network Profile from *Advanced Settings*.

- Keep the **SIP Registration** check box enabled. Default: Enabled.

SETU VGRX will send the REGISTER message to Registrar Proxy or Outbound Proxy as applicable.

Clear the check box, if you want to disable the registration.

- In **Maximum Calls**, select the number of simultaneous calls you want to allow on this SIP Trunk. Default: 8.

The number of simultaneous SIP calls depend on the number of simultaneous calls allowed by the ITSP with whom you have subscribed this SIP Trunk.

SETU VGRX supports 8 simultaneous calls. Ask your ITSP the number of simultaneous SIP calls supported on this SIP Trunk.

- Select the desired **Fax Protocol**, to send and receive the Fax over IP.

- **T.38:** If you select this option, the device you are sending the fax to, must also support this protocol.
- **Pass Through:** Select this option, if you need to send fax over G.711. The device you are sending fax to must also use G.711.

Default: T.38.

- Select the **Allow Call Disconnection using Access code** check box, if you want to enable the feature *Disconnect Call using Access Code* on the SIP Trunk. Default: Disabled. To know more about the feature, see [“Disconnecting a Call using Access Code”](#).

## Handling of Incoming Calls

- Click **Handling of Incoming Calls**.

- Keep the **Block all calls received on this SIP Trunk** check box disabled.

Select this check box, only if you do not want to route calls received on this SIP Trunk.

- By default, SETU VGRX identifies the Called Party Number for routing the incoming call on the SIP Trunk further, by the number received in the **Request-URI** of the INVITE message.

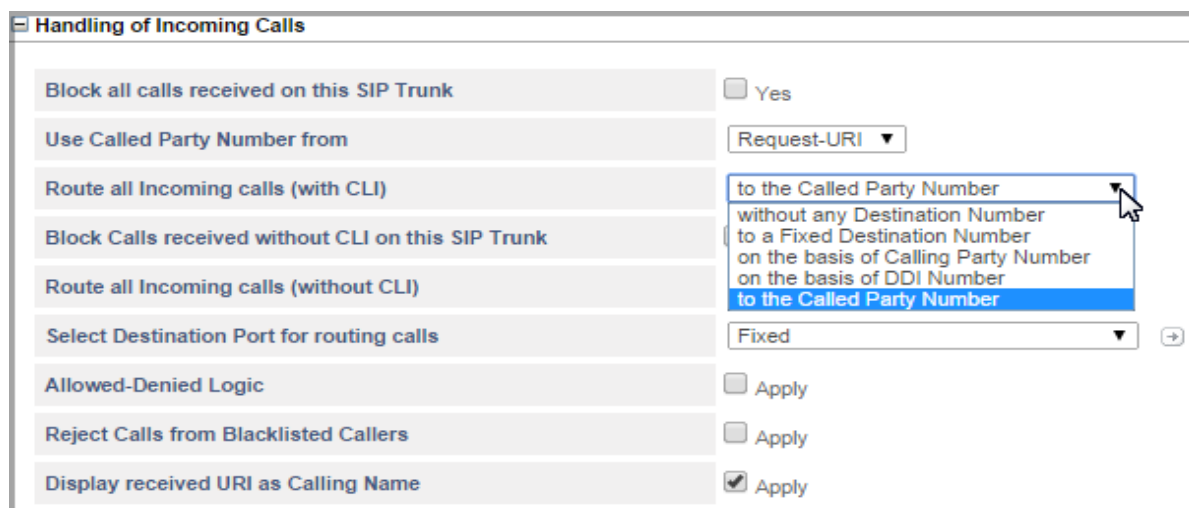
If you want the system to identify the Called Party Number from the 'To Field' of the INVITE message, in the **Use Called Party Number From** parameter, select the **To Field** option.

## Destination Number Determination

Select the desired destination number determination method for routing incoming calls *with* and *without* CLI.

- To **Route all Incoming calls (with CLI)**, you may select from any of the following methods:
  - without any Destination Number
  - to a Fixed Destination Number
  - on the basis of Calling Party Number
  - on the basis of DDI Number
  - to the Called Party Number

Default: to the Called Party Number



**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	Fixed ▼ ➡
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

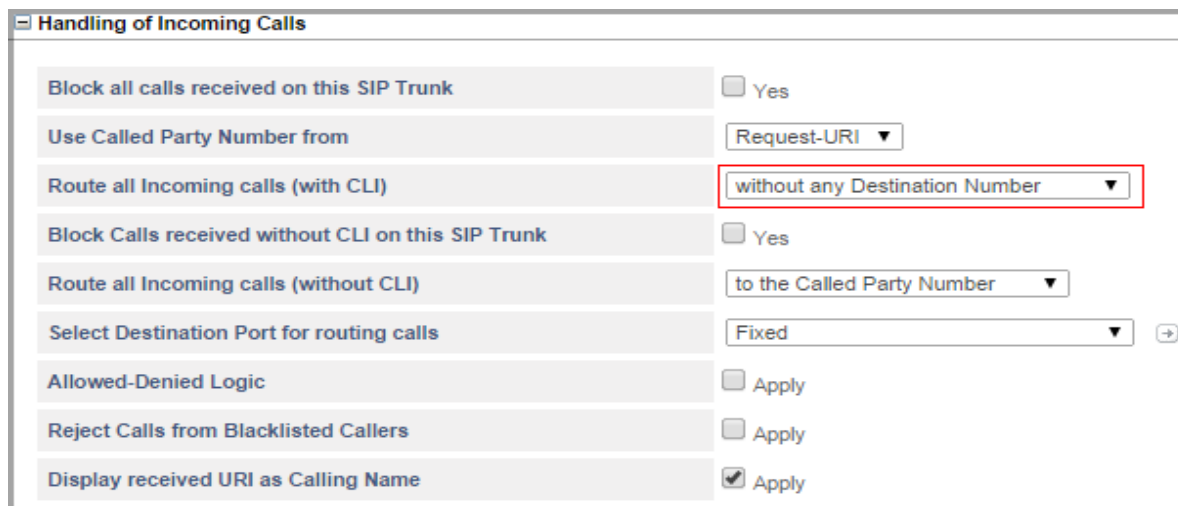
Read further for instructions on selecting and configuring each of these destination number determination methods.



If the destination number to be dialed out is an IP Address, SETU VGRX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. See [“IP Dialing”](#) for more details.

### Route Calls without any Destination Number

In this method, all calls received on the SIP Trunk are directly routed to the destination port, irrespective of the Destination Number.



**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	without any Destination Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	Fixed ▼ ➡
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

- To apply this method, in **Route all incoming calls (with CLI)**, select **without any Destination Number**.

## Route to a Fixed Destination Number

In this method, calls received on the SIP Trunk are routed to a fixed destination number, which is configured for the SIP Trunk.

**Handling of Incoming Calls**

Block all calls received on this SIP Trunk ☐ Yes

Use Called Party Number from Request-URI ▼

Route all Incoming calls (with CLI) to a Fixed Destination Number ▼

Block Calls received without CLI on this SIP Trunk ☐ Yes

Route all Incoming calls (without CLI) to the Called Party Number ▼

**Fixed Destination Number**

Fixed Destination Number

Select Destination Port for routing calls Fixed ▼ ➔

Allowed-Denied Logic ☐ Apply

Reject Calls from Blacklisted Callers ☐ Apply

Display received URI as Calling Name ☒ Apply

To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **to the Fixed Destination Number**.
- In **Fixed Destination Number**, enter the desired destination number. The Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, \*, # and (.) dot are allowed. Default: Blank.
- Click **Submit** to save your settings.

## Route on the basis of Calling Party Number

In this method, a call received on the SIP Trunk is routed to a specific number, as per the calling party's number. You must configure the calling party numbers in the *Calling Party Number Based Table*.

When there is an incoming call on the SIP Trunk, SETU VGRX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination number configured for that Calling Party Number.

To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of Calling Party Number**.

- Click **Settings**  .

Handling of Incoming Calls


Block all calls received on this SIP Trunk

☐ Yes

Use Called Party Number from

Request-URI ▼

Route all Incoming calls (with CLI)

on the basis of Calling Party Number ▼ 

If no match found in the Calling Party Number Table, route calls

to the Called Party Number ▼


Block Calls received without CLI on this SIP Trunk

☐ Yes

Route all Incoming calls (without CLI)

to the Called Party Number ▼

Select Destination Port for routing calls

Fixed ▼ 

Allowed-Denied Logic

☐ Apply

Reject Calls from Blacklisted Callers

☐ Apply

Display received URI as Calling Name


☒ Apply


- The **SIP Trunk- Destination Number Determination: Calling Number Based** Table window opens.


1-100 101-200 201-300 301-400 401-499

SIP Trunk - Destination Number Determination: Calling Number Based

Index	Calling Number	Destination Number
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		

 Submit

 Default All

 Close

- In **Calling Number**, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.
- In **Destination Number**, enter a corresponding destination number for each calling party number. Destination numbers may consist of a maximum of 24 characters. Digits 0 to 9, \*, # and (.) dot are allowed. Default: Blank.
- Click **Submit** to save your entries. Close the window to return to the main page.

You can also configure the **Calling Number Based** table from *Advanced Settings*. For instructions, see [“Destination Number Determination”](#) under *Advanced Settings*.

- Select a method for routing incoming calls with CLI that *do not match* with any entries in the Calling Party Number Based Table.

In **If no match found in the Calling Party Number Table, route calls**, select the desired method from the following options for processing the call:

- to a Fixed Destination Number
  - to the Called Party Number
  - on the basis of DDI Number
- Default: to the Called Party Number.

### Route on the basis of DDI Number

In this method, incoming calls on the SIP Trunk are routed to specific numbers as per the DDI number received in the SETUP message on the SIP Trunk.

To apply this method, do the following:

- In **Route all Incoming calls (with CLI)**, select **on the basis of DDI Number**.
- Click **Settings** ➔ .

The screenshot shows the 'Handling of Incoming Calls' configuration window. It contains several settings:

- Block all calls received on this SIP Trunk**: ☐ Yes
- Use Called Party Number from**: Request-URI ▼
- Route all Incoming calls (with CLI)**: on the basis of DDI Number ▼ (highlighted with a red box)
- Block Calls received without CLI on this SIP Trunk**: ☐ Yes
- Route all Incoming calls (without CLI)**: to the Called Party Number ▼
- Select Destination Port for routing calls**: Fixed ▼ ➔
- Allowed-Denied Logic**: ☐ Apply
- Reject Calls from Blacklisted Callers**: ☐ Apply
- Display received URI as Calling Name**: ☒ Apply

The **SIP Trunk - Destination Number Determination: DDI Number Based** Table opens.

DDI Number Generation

**SIP Trunk - Destination Number Determination: DDI Number Based**

Index	DDI Number	Destination Number	Reverse DDI	
			Apply	Reference ID
001			<input type="checkbox"/>	1
002			<input type="checkbox"/>	1
003			<input type="checkbox"/>	1
004			<input type="checkbox"/>	1
005			<input type="checkbox"/>	1
006			<input type="checkbox"/>	1
007			<input type="checkbox"/>	1
008			<input type="checkbox"/>	1
009			<input type="checkbox"/>	1
010			<input type="checkbox"/>	1

Submit

Default All

Close

- In **DDI Number**, enter the DDI Numbers allotted by your service provider.
- For each DDI Number, enter the corresponding destination number in **Destination Number**.
- To apply **Reverse DDI** for each number, select the respective **Apply** check boxes and select the **Reference ID** for the number. Default: Apply Reverse DDI is disabled and Reference ID is 1.
- Click **Submit** to save and close the window to return to the main page.

You can also configure the **DDI Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Number Determination”](#) under *Advanced Settings*.

### Route to the Called Party Number

In this method, a call received on the SIP Trunk is routed to a specific number depending upon the called party number received in the SETUP Message on the SIP Trunk.

- To apply this method, in **Route all incoming calls (with CLI)**, select **to the Called Party Number**.

**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	Fixed ▼ ➔
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

- If you do not want to route the incoming calls received without CLI, through this SIP Trunk, select **Block Calls received without CLI on this SIP Trunk** check box.
- To **Route all Incoming calls (without CLI)**, you may select from any of the following methods:
  - to a Fixed Destination Number, see ["Route to a Fixed Destination Number"](#).
  - on the basis of DDI Number, see ["Route on the basis of DDI Number"](#).
  - to the Called Party Number, see ["Route to the Called Party Number"](#).
 Default: to the Called Party Number.

**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number to a Fixed Destination Number to the Called Party Number on the basis of DDI Number ▼ ➔
Select Destination Port for routing calls	Fixed ▼ ➔
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply


## Destination Port Determination

For the SIP Trunk, select the Destination Port for routing calls from the following options:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

Default: Fixed.

**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	Fixed ▼ 
Allowed-Denied Logic	
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

Read the description and follow the instructions for each of these destination port selection methods given below.


### Fixed

In this method, calls received on the SIP Trunk are routed to a Fixed Destination Port, irrespective of the number dialed on the SIP Trunk.


To apply this method, do the following:


- In **Select Destination Port for routing calls**, select **Fixed** option.
- Click **Settings**  .

**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	Fixed ▼ 
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

The **Destination Port/Group for SIP Trunk** window opens.


































Edit	Routing Group	Fallback Routing Group	CLI Number on FXS Port
	FXS Port 1 - 2 (Ascending)	None	Received Calling Party

 Close

The default **Routing Group** and **Fallback Routing Groups** appear.

- If you wish to edit the default Routing Group options, click **Settings**  .

The **Edit Selective Port/Group for SIP Trunk** window opens.


Edit Selective Port/Group for SIP Trunk			
CLI Number to be sent on Destination Port			Received Calling Party 
<b>Routing Group</b>			
<input checked="" type="radio"/> FXS Port	1 	to 2 	in Ascending  order
<input type="radio"/> FXS Group	1 		
<input type="radio"/> FXO Port	1 	to 1 	in Ascending  order
<input type="radio"/> FXO Group	1 		
<input type="radio"/> Mobile Port	1 	to 1 	in Ascending  order
<input type="radio"/> Mobile Group	1 		
<input type="radio"/> Radio Port	1 	to 1 	in Ascending  order
<b>Fallback Routing Group</b> <input type="checkbox"/> Apply			
<input type="radio"/> FXS Port	1 	to 1 	in Ascending  order
<input type="radio"/> FXS Group	1 		
<input type="radio"/> FXO Port	1 	to 1 	in Ascending  order
<input type="radio"/> FXO Group	1 		
<input type="radio"/> Mobile Port	1 	to 1 	in Ascending  order
<input type="radio"/> Mobile Group	1 		
<input type="radio"/> Radio Port	1 	to 1 	in Ascending  order
 Submit		 Close	

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.

- Create the **Routing Group**.

- To create a group of *sequential FXS Ports* as members,
  - Select the desired **FXS Port** numbers as members. Default: 1.
  - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.  
  
Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
- Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports* and *Radio Ports*.
- To create a group of *not-sequential FXS Ports* as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default: 1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic [“Group”](#) under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential FXO Ports* and *Mobile Ports*.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Edit** window closes.
- The entry you edited appears in the **Destination Port/Group for SIP Trunk** window.
- Close the **Destination Port/Group for SIP Trunk** window to return to the main page.

### On the basis of Destination Number

In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SETU VGRX will match the called party number dialed by the caller with the entries of this table. If a match is found for the number in the table, the call is routed to the destination.

To apply this method, do the following:

- In **Select Destination Port for routing calls**, select **On the basis of Destination Number** option.

- Click **Settings**  .

Handling of Incoming Calls

Block all calls received on this SIP Trunk

☐ Yes

Use Called Party Number from

Request-URI ▼

Route all Incoming calls (with CLI)

to the Called Party Number ▼


Block Calls received without CLI on this SIP Trunk

☐ Yes

Route all Incoming calls (without CLI)

to the Called Party Number ▼

Select Destination Port for routing calls

On the basis of Destination Number ▼ 

Allowed-Denied Logic

☐ Apply

Reject Calls from Blacklisted Callers



☐ Apply

Display received URI as Calling Name

☒ Apply

The **SIP Trunk - Destination Port Determination - Destination Number Based** table window opens.


SIP Trunk - Destination Port Determination - Destination Number Based


	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	FXS Port 1 - 1 (Ascending)	None	Received Calling Party


Total Records : 1      1

Testing

Enter the destination number to know which entry would be selected for routing

 Add

 Delete

 Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 1000 entries.

**Add Entry**

Destination Number

CLI Number to be sent on Destination Port Received Calling Party

**Routing Group**

☒ FXS Port 1 to 1 in Ascending order

☐ FXS Group 1

☐ FXO Port 1 to 1 in Ascending order

☐ FXO Group 1

☐ Mobile Port 1 to 1 in Ascending order

☐ Mobile Group 1

☐ Radio Port 1 to 1 in Ascending order

**Fallback Routing Group** ☐ Apply

☐ FXS Port 1 to 1 in Ascending order

☐ FXS Group 1

☐ FXO Port 1 to 1 in Ascending order

☐ FXO Group 1

☐ Mobile Port 1 to 1 in Ascending order

☐ Mobile Group 1

☐ Radio Port 1 to 1 in Ascending order

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters"). Valid characters are 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

## Wildcard Characters

SETU VGRX supports following characters.


Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.
<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.

[ ^ ]	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
T (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*


- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.  
  
Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports* and *Radio Ports*.
  - To create a group of *not-sequential FXS Ports* as members,
    - Select a **FXS Group**.
    - Select **FXS Group** number. Default: 1.
    - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic [“Group”](#) under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential FXO Ports* and *Mobile Ports*.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **SIP Trunk - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.

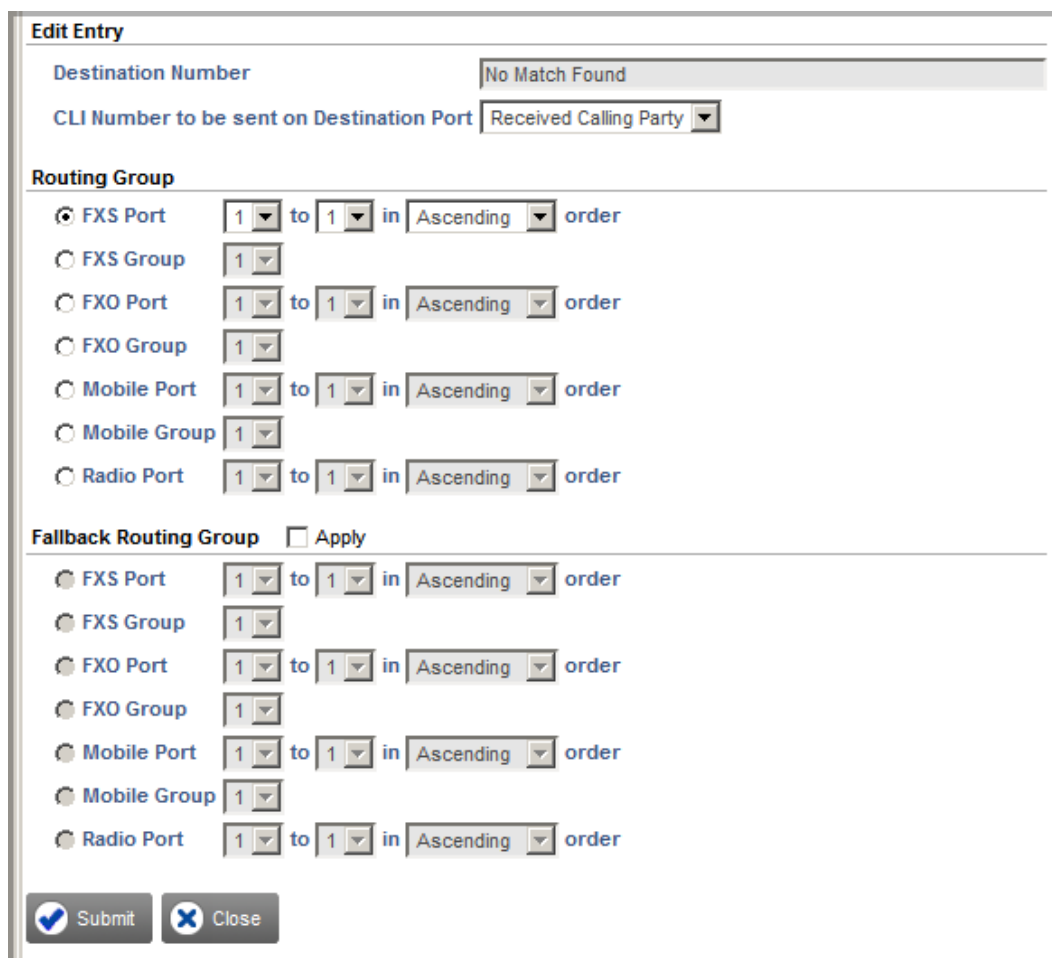


If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.

- By default, FXS Port 1 - 1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- under Edit, click **Settings** .
- The **Edit Entry** window opens.




- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

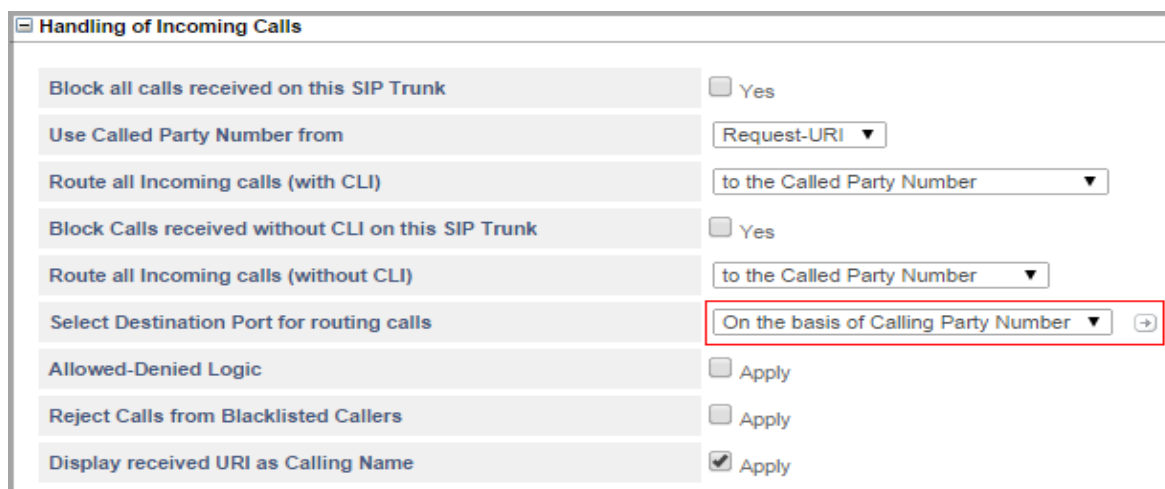
You can also configure the **Destination Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Port Determination”](#) under *Advanced Settings*.

## On the basis of Calling Party Number


In this method, incoming calls on the SIP Trunk are routed to a specific port as per the calling party's number.

To apply this method, do the following:






- In **Select Destination Port for routing calls**, select **On the basis of Calling Party Number** option.
- Click **Settings** .



**Handling of Incoming Calls**

Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	On the basis of Calling Party Number ▼ 
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input type="checkbox"/> Apply
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

The **SIP Trunk - Destination Port Determination - Calling Number Based** table window opens.

SIP Trunk - Destination Port Determination - Calling Number Based					
	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	FXS Port 1 - 1 (Ascending)	None	Received Calling Party
Total Records : 1		1			
	Add		Delete		Close

- To add a new entry, click **Add**. The **Add Entry** window opens. You can add upto 499 entries.

**Add Entry**

Calling Number

CLI Number to be sent on Destination Port

**Routing Group**

☒ **FXS Port**  to  in  order

☐ **FXS Group**

☐ **FXO Port**  to  in  order

☐ **FXO Group**

☐ **Mobile Port**  to  in  order

☐ **Mobile Group**

☐ **Radio Port**  to  in  order

**Fallback Routing Group** ☐ Apply

☐ **FXS Port**  to  in  order

☐ **FXS Group**

☐ **FXO Port**  to  in  order

☐ **FXO Group**

☐ **Mobile Port**  to  in  order

☐ **Mobile Group**

☐ **Radio Port**  to  in  order

☒ Submit ☐ Close


- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Valid digits are 0 to 9, \*, #, (dot). Default: Blank.
- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.

- Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports and Radio Ports.
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default: 1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
- Similarly, you can create a group of *not-sequential* FXO Ports and Mobile Ports.
- You may also create a **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **SIP Trunk - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, FXS Port 1 - 1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found).

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- under Edit, click **Settings** .

- The **Edit Entry** window opens.

**Edit Entry**

Calling Number

CLI Number to be sent on Destination Port

**Routing Group**

☒ FXS Port  to  in  order

☐ FXS Group

☐ FXO Port  to  in  order

☐ FXO Group

☐ Mobile Port  to  in  order

☐ Mobile Group

☐ Radio Port  to  in  order

**Fallback Routing Group** ☐ Apply

☐ FXS Port  to  in  order

☐ FXS Group

☐ FXO Port  to  in  order

☐ FXO Group

☐ Mobile Port  to  in  order

☐ Mobile Group

☐ Radio Port  to  in  order

- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

You can also configure the **Calling Number Based** Table from *Advanced Settings*. For instructions, see [“Destination Port Determination”](#) under *Advanced Settings*.

## Allowed - Denied Logic

You can apply the Allowed-Denied logic on the SIP Trunk (source port), if you want to allow or restrict the dialing of specific numbers. You can use this feature for Toll Control.

The Allowed-Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the SIP Trunk.

- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the SIP Trunk.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SETU VGRX uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if it:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

The system does not apply the Allowed-Denied Logic:

- When dialed number string matches with any Access Code.
- When dialed number string matches with any Emergency Number.
- When any one of the following is selected to Route all Incoming Calls (with CLI):
  - on the basis of Calling Party Number
  - to a Fixed Destination Number
  - on the basis of DDI Number

To apply Allowed - Denied Logic on the SIP Trunk,

- Select the **Allowed - Denied Logic** check box.


- In the **Allowed Number List**, select the list number you have configured with numbers you want to allow to be dialed out from the SIP Trunk. Default: 07

If you have not configured the Allowed Number List,

- Click **Settings** . The Number Lists window opens.
- You may configure the default Allowed Number List or any other list. See [“Number Lists”](#) to configure the allowed numbers.

- Click **Submit** to save the Allowed Number List and close the window.
- In the **Denied Number List**, select the list number you have configured with numbers you want to restrict to be dialed out from the SIP Trunk. Default: 08


If you have not configured the Denied Number List,

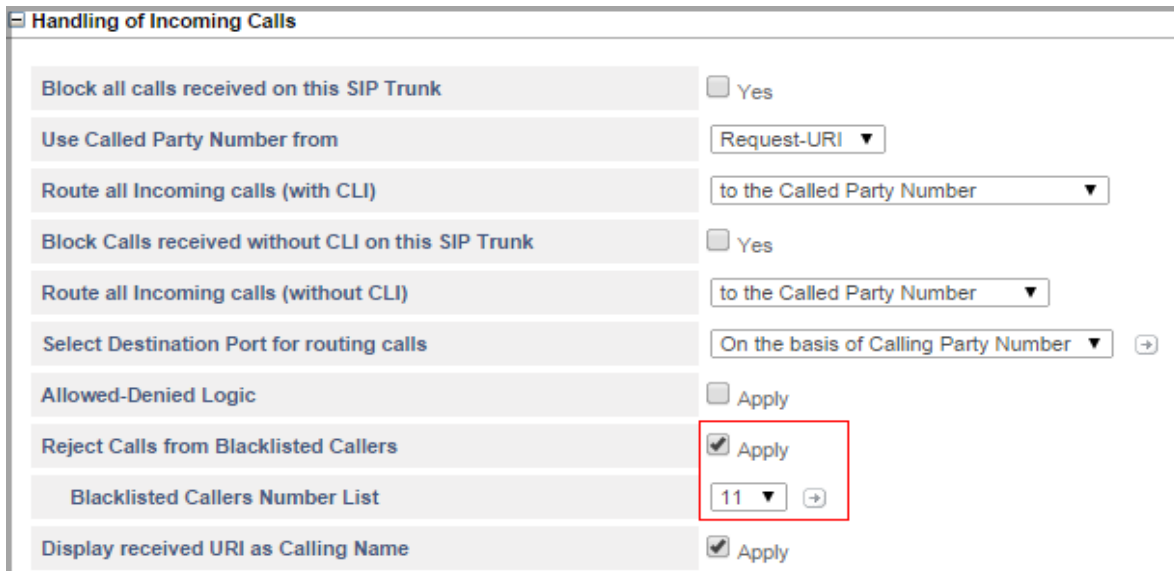
- Click **Settings** . The Number Lists window opens.
- You may configure the default Denied Number List or any other list. See ["Number Lists"](#) to configure the restricted numbers.
- Click **Submit** to save the Denied Number List and close the window.

## Black Listed Callers

With the Black Listed Callers feature, you can block incoming calls from specific addresses/numbers on SIP Trunks. Thus the incoming calls from the numbers you have blacklisted, will be rejected automatically by the SETU VGRX.

To apply Black Listed Callers on SIP Trunk,

- Select the **Reject Calls from Blacklisted Callers** check box.
- Configure the **Black Listed Callers** table. To do this,
- Click **Settings** .



Handling of Incoming Calls	
Block all calls received on this SIP Trunk	<input type="checkbox"/> Yes
Use Called Party Number from	Request-URI ▼
Route all Incoming calls (with CLI)	to the Called Party Number ▼
Block Calls received without CLI on this SIP Trunk	<input type="checkbox"/> Yes
Route all Incoming calls (without CLI)	to the Called Party Number ▼
Select Destination Port for routing calls	On the basis of Calling Party Number ▼ 
Allowed-Denied Logic	<input type="checkbox"/> Apply
Reject Calls from Blacklisted Callers	<input checked="" type="checkbox"/> Apply
Blacklisted Callers Number List	11 ▼ 
Display received URI as Calling Name	<input checked="" type="checkbox"/> Apply

- The Number List window opens.

1-4
5-8
**9-12**
13-16
17-20
21-24

**Number Lists**

Location	List 9	List 10	List 11	List 12
01				
02				
03				
04				
05				
06				
07				
08				
09				
10				
11				
12				

- By default, Number List 11 is assigned as Black Listed Callers List.
- Enter the numbers of unwanted callers in this list.
- Click **Submit** to save the entries and close the window to return to the main page.

## Display Received URI as Calling Name

- Keep the **Display Received URI as Calling Name** check box enabled.

Display received URI as Calling Name
☒ Apply

When Name is received in the "FROM" header for incoming call on the SIP Trunk, SETU VGRX will display name and received URI as calling name. When Name is not received, SETU VGRX will display only the received URI as calling name.

You may disable this check box, if you do not want the system to display the received URI as calling name on the SIP Trunk.

- Click **Submit**.

## Handling of Outgoing Calls

When a SIP Trunk is determined as the destination port, numbers dialed from this port constitute outgoing calls.

Handling of Outgoing calls	
Block calls through this SIP Trunk	<input type="checkbox"/> Yes
Route calls through this SIP Trunk without Registration	<input type="checkbox"/> Yes
CLIR	<input type="checkbox"/> Enable
SIP ID in "FROM" header of INVITE message	<input type="text" value="SIP ID configured"/>
"P-Asserted-Identity" header in INVITE message	<input type="text" value="Donot send"/>
Reverse DDI Reference ID	<input type="text" value="1"/>
Automatic Number Translation(ANT) for Called Number	<input type="checkbox"/> Apply
Automatic Number Translation(ANT) for Calling Number	<input type="checkbox"/> Apply
Route calls returned unconnected to Original Caller	<input type="checkbox"/> Yes
Connect Source Port when 183(Session Progress) is received on SIP	<input type="checkbox"/> Yes

- Click **Handling of Outgoing calls**.
- If you do not want to route outgoing calls through this SIP Trunk, select the **Block calls through this SIP Trunk** check box.
- To allow the users to make outgoing calls irrespective of whether the SIP Trunk has been successfully registered with the proxy or not, select the **Route Calls through this port without Registration** check box.

By default, the system does not allow outgoing calls to be made if the status of the SIP Trunk is 'not registered'.

- By default, the CLI of the SIP Trunk is sent to the called party when outgoing calls are made using the SIP Trunk. If you do not want to send CLI, enable the **CLIR** check box. Default: Disabled.
- SETU VGRX supports flexible options for sending **SIP ID in "FROM" header of INVITE message** during an outgoing call. You may select the desired option — SIP ID configured, Caller ID received on Source Port, Caller ID after applying Reverse DDI logic. Default: SIP ID configured
- SETU VGRX supports flexible options for sending **P-Asserted-Identity" header in INVITE message** during an outgoing call. You may select the desired option — Do not send, Send SIP ID configured, Send Caller ID received on Source Port, Send Caller ID after applying Reverse DDI logic, Send Fixed Number. Default: Do not send.

If you select *Send Caller ID after applying Reverse DDI logic*, SETU VGRX allows you to configure the desired option for **If no match found using Reverse DDI logic** — Send SIP ID configured, Send Caller ID received on Source Port, Send Fixed Number. Default: Do not send

If you select *Send Fixed Number* as an option for *"P-Asserted-Identity" header in INVITE message* or *If no match found using Reverse DDI logic*, you must configure the **Fixed Number**. The Fixed Number can be a maximum of 24 characters. Characters 0-9, +, \* and # are allowed. Default: Blank.



If you have enabled **CLIR** and "**P-Asserted-Identity**" header in **INVITE message** is configured other than **Do not send**, then SETU VGRX will add **Privacy = ID** header in the **INVITE message** during an outgoing call from the SIP Trunk.

- Select **Reverse DDI Reference ID**, if you have selected atleast one of the following:
  - *Caller ID after applying Reverse DDI logic* option as the **SIP ID in "FROM" header of INVITE message**.
  - *Send Caller ID after applying Reverse DDI logic* option as the "**P-Asserted-Identity**" header in **INVITE message**.


SETU VGRX will compare the Reference ID configured on the SIP Trunk with the one configured in the SIP Trunk - Destination Number Determination: DDI Number Based Table. If a match is found, SETU VGRX will send the corresponding DDI Number to the Called Party.

- You can apply **Automatic Number Translation logic** on outgoing calls made from the SIP Trunk.
  - To apply ANT logic on the Called Numbers, select the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.

Handling of Outgoing calls	
Block calls through this SIP Trunk	<input type="checkbox"/> Yes
Route calls through this SIP Trunk without Registration	<input type="checkbox"/> Yes
CLIR	<input type="checkbox"/> Enable
SIP ID in "FROM" header of INVITE message	SIP ID configured
"P-Asserted-Identity" header in INVITE message	Donot send
Reverse DDI Reference ID	1
Automatic Number Translation(ANT) for Called Number	<input checked="" type="checkbox"/> Apply
Use Automatic Number Translation Table	1
Pause Timer	2 Seconds
Automatic Number Translation(ANT) for Calling Number	<input type="checkbox"/> Apply
Route calls returned unconnected to Original Caller	<input type="checkbox"/> Yes
Connect Source Port when 183(Session Progress) is received on SIP	<input type="checkbox"/> Yes

- In **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Called Numbers. Default: Table 1.

If you have not configured the Automatic Number Translation Table,

- Click **Settings** . The Automatic Number Translation Table window opens.



1
2
3
4
5
6
7
8

**Automatic Number Translation Table - 1**

Index	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	

**Examples of Number Pattern**

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' to every 7-digit dialed number.

 Submit
 Default


- You may configure the default Automatic Number Translation Table or any other Table. See [“Automatic Number Translation \(ANT\)”](#) to configure the ANT Table.
- Click **Submit** to save the ANT Table and close the window.
- Return to ANT parameter and assign the ANT Table you configured.
- Click **Submit**.
- Set the duration of the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. Valid range is 1 to 9 seconds. Default: 2 seconds.

- To apply ANT logic on the Calling Numbers, select the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.

Handling of Outgoing calls	
Block calls through this SIP Trunk	<input type="checkbox"/> Yes
Route calls through this SIP Trunk without Registration	<input type="checkbox"/> Yes
CLIR	<input type="checkbox"/> Enable
SIP ID in "FROM" header of INVITE message	SIP ID configured
"P-Asserted-Identity" header in INVITE message	Donot send
Reverse DDI Reference ID	1
Automatic Number Translation(ANT) for Called Number	<input type="checkbox"/> Apply
Automatic Number Translation(ANT) for Calling Number	<input checked="" type="checkbox"/> Apply
Use Automatic Number Translation Table	5
Route calls returned unconnected to Original Caller	<input type="checkbox"/> Yes
Connect Source Port when 183(Session Progress) is received on SIP	<input type="checkbox"/> Yes

- In the **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the Calling Numbers. Default: Table 5.

If you have not configured the Automatic Number Translation Table,

- Click **Settings** . The Automatic Number Translation Table window opens.




1 2 3 4 5 6 7 8

**Automatic Number Translation Table - 5**

Index	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	

**Examples of Number Pattern**

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' to every 7-digit dialed number.

 Submit
 Default
 Close

- You may configure the default Automatic Number Translation Table 5 or any other Table. See [“Automatic Number Translation \(ANT\)”](#) to configure the ANT Table.
  - Click **Submit** to save the ANT Table and close the window.
  - Return to ANT parameter and assign the ANT Table you configured.
  - Click **Submit** to apply List.
- Select the **Route calls returned unconnected to Original Caller** check box, if you want SETU VGRX to route outgoing calls made from this Trunk that return unconnected back to the original caller.

If you enable this feature, when an outgoing call is made using this Trunk, and the Called Party is found busy or does not respond, SETU VGRX stores the number of the calling party, the number of the called party and this trunk (through which the outgoing call was made). A record of each such call is stored for the duration of the Unconnected Calls Record Delete Timer (configurable; default: 999 minutes).

If the called party returns the call before the expiry of this Timer, SETU VGRX checks whether *Apply RCOC only if the caller calls back on the same trunk from which the call was made* is enabled or not, and accordingly places the incoming call to the original calling party. To change the duration of this timer, delete records of such calls and enable/disable the *Apply RCOC only if the caller calls back on the same trunk from which the call was made* check box, see [“System Parameters”](#).

- To connect the Source Port with the Destination Port without waiting for the call on the Destination Port to mature, select the **Connect Source Port when 183 (Session Progress) is received on SIP** check box. Default: Disabled.

In all Destination Number Determination methods, the Source Port gets connected to the Destination Port only after the call has matured, that is, the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.



*If you enable **Connect Source Port when 183 (Session Progress) is received on SIP**, [“Making a New Call using Access Code”](#) feature will not be allowed to the users.*

- Click **Submit** to save the changes.

## Message Wait Indication (MWI)



*Message Wait Indication parameters are applicable only when SIP Trunk is configured as Proxy.*

- If you have subscribed for Message Wait Indication on the SIP Trunk for the voicemail service from your ITSP, click **MWI** and configure the following parameters:
  - Select the **Subscribe for MWI** check box. Default: Disabled.
  - In **Message Retrieval Number**, enter the number provided to you by your ITSP. This number is used for retrieval of voicemail on the SIP Trunk. The Message Retrieval Number may consist of a maximum of 24 characters. Valid range is 0 to 9, \* and # are allowed. Default: Blank.
  - Enter the **Authentication ID** (User ID) provided by your ITSP. Default: Blank.
  - Enter the **Authentication Password** provided by your ITSP. Default: Blank.
  - In **Send Message Notification on**, select the FXS Port on which Message Wait Indication is to be sent whenever there is a new message on the SIP Trunk. Default: FXS Port 1.

To know more about Message Wait Indication (MWI), see [“Message Wait Indication on SIP Trunks”](#).

- If you have completed the configuration of SIP Trunk 1, click **Submit** to save settings.

- Close the window.
- To configure the next SIP Trunk, click the desired SIP Trunk number and follow the same instructions as given above.

## Copy SIP Trunk Parameters

- You can also copy the settings of a SIP Trunk to another SIP Trunk using the **Copy** button. To do this,
  - Click the **Copy** button. The **Copy SIP Trunk Parameters** window opens.

Copy SIP Trunk Parameters from SIP Trunk 1 to

☐ All

☐ SIP Trunk 1 ☐ SIP Trunk 2 ☐ SIP Trunk 3 ☐ SIP Trunk 4

☐ SIP Trunk 5 ☐ SIP Trunk 6 ☐ SIP Trunk 7 ☐ SIP Trunk 8

☐ SIP Trunk 9

- In **Copy SIP Trunk Parameters from SIP Trunk**, select the number of the trunk you want to copy settings *From*. Select the check box of the respective trunk numbers you want to copy the settings *To*.
- If you want to copy the settings *To* all the trunks, select the **All** check box.
- Click the **OK** button.
- Once you have copied the settings, you can again edit the specific parameters of the SIP Trunk you copied the settings to.

# Login Password

---

You can configure SETU VGRX using Jeeves and by dialing commands from a telephone (only specific parameters).

## Login Password for Jeeves

To configure the system, you must log into the Jeeves using the Jeeves Password. The default Jeeves Password is 1234. However, you must change it for security reasons.

The Jeeves Password must fulfill the following requirements.

- It must not be less than 6 characters and can be of upto 16 characters.
- All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ' , Double Quote " and **Space**) and digits 0 to 9 are allowed.
- It must include atleast one upper-case, one lower-case, one number and one special character.

To provide additional security, if you enter a wrong password five times consecutively within 10 minutes, the system will block the source IP Address for 10 minutes. The notification (Warning) will be sent for this event to the SNMP Manager. See "[Simple Network Management Protocol \(SNMP\)](#)" for more details.

To change the Jeeves Password:

- Log into Jeeves.
- Click the **Basic Settings** link to expand.

- Click **Login Password**.

**Password Change**

Jeeves / FTP / Telnet

Current Password

New Password

Confirm New Password

Note :

Password must follow following requirements:

Minimum length must be 6 characters.

Password must include atleast 1 uppercase, 1 lowercase , 1 number and 1 special character.

Allowed characters are 0-9, a-z, A-Z, all special characters except %, =, #, +, &, \, <, >, " , ' and space.

Command

New Password

Confirm New Password

Submit

Under **Jeeves/FTP/Telnet**,

- Enter **Current Password**.
- Enter **New Password**. All ASCII characters (except Percentage %, Hash #, Equal to =, Plus +, And &, Backslash \, Less than <, Greater than >, Apostrophe ' , Double Quote " and **Space**) and digits 0 to 9 are allowed. The new password must be:
  - a minimum of 6 characters to a maximum of 16 characters.
  - include atleast one upper-case, one lower-case, one number and one special character.
- In **Confirm New Password**, re-enter the new password to confirm.
- Click **Submit** button to save your new password.



- *Password for Jeeves is case sensitive.*
- *When you default the system, Jeeves Password will not be set to default.*

## Login Password for System Commands

To configure the system by dialing System Commands, you must enter the SE mode using the Command Password. This Password must not be less than 4 digits and can be of upto 8 digits. Digits 0-9 are allowed. The default Command Password is **1234**. You may change this Password, if required.

To change the Command Password:

- Log into Jeeves.
- Click the **Basic Settings** link to expand.
- Click **Login Password**.

**Password Change**

Jeeves / FTP / Telnet

Current Password

New Password

Confirm New Password

Note :

Password must follow following requirements:

Minimum length must be 6 characters.


Password must include atleast 1 uppercase , 1 lowercase , 1 number and 1 special character.

Allowed characters are 0-9, a-z, A-Z, all special characters except %, =, #, +, &, \, <, >, ", ' and space.

Command

New Password

Confirm New Password

 Submit

Under **Command**,

- In **New Password**, enter the new password.
- In the **Confirm New Password**, type the new password again for confirmation.
- Click **Submit** to save.



*When you default the system, Command Password will not be set to default.*

## Forgot the Login Password?

If you have already changed the default Jeeves Password (1234) and are unable to recall or locate it, you can restore the default Jeeves Password using the Reset button.

To restore the default Jeeves Password,

- Press the Reset button for more than four seconds.

- Release the Reset button.



- *If you press the Reset button for less than four seconds, SETU VGRX will restart.*
- *When you restore the default Jeeves Password (1234), a few other parameters will also be set to default. See [“Restoring Default Settings using the Reset button”](#) for details.*

# Date-Time Settings

## Real Time Clock

SETU VGRX has a Real Time Clock (RTC) to store date and time. When you select the Region, the RTC parameters are set automatically.

However, the RTC can drift over a long period. So, you may check and reset the RTC values at regular intervals to correct this drift.

To set the Real Time Clock,

- Log into Jeeves.
- Click the **Basic Settings** link to expand.
- Click the **Date-Time Settings** link.
- The **Real Time Clock** parameters appear on your screen.

**Basic Settings**

- Region
- Network
- FXS Port
- Radio Port
- FXO Port
- Mobile Port
- SIP Trunk
- Login Password
- Date-Time Settings**

**Advanced Settings**

**Maintenance**

**Status**

**Real Time Clock (RTC)**

Current Date: 21-Jul-2015

Current Time (HH:MM:SS): 17:18:14

Current Day: Tuesday

Sync Date-Time with PC

**Daylight Saving Time**

DST Type: Disable

**SNTP Settings**

SNTP Server Address:

Time Zone: India(GMT+05:30)

Auto Date & Time Sync with SNTP During Power ON? ☐ Yes Sync Date-Time with SNTP Server

Submit

- Under **Real Time Clock (RTC)**, click **Settings** .

- A new window opens.

- Set the **Current Date** in date-month-year format.
- Set the **Current Time** in hours-minutes-seconds format.

The current day will be displayed automatically for the date and time you set.

- Close the window.
- Click **Submit** to save RTC settings.
- Click **Sync Date-Time with PC** button, if you want to sync the system's date and time with that of your PC.

## Daylight Saving Time

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<sup>4</sup> use it, though the start and end dates of DST vary by location and year.

SETU VGRX supports Daylight Saving Time adjustment to enable you to set the Date and Time<sup>5</sup> of SETU VGRX forward and backward according to the DST convention followed in your country.

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



*When SETU VGRX is set to default, your DST settings will remain unchanged.*

To configure DST,

- Click the **Basic Settings** link to expand.
- Click the **Date-Time Settings** link.
- Under **Daylight Saving Time**, do the following.

4. In most countries in Asia and Africa, and in certain countries of South America, DST is not observed.

5. SETU VGRX sets its Date and Time according to the **Time Zone** you selected, and synchronizes the time according to the **SNTP Server** you selected. See "[Region](#)".

- Select the **DST Type**. You may select **Auto** or **Custom**. If you do not want to apply DST select Disable. Default: Disabled.
- If you select **Auto**, you must select the **Region**. DST will be set automatically for the region you select.

The screenshot shows the 'Daylight Saving Time' configuration window. The 'DST Type' is set to 'Auto'. The 'Region' dropdown is open, showing a list of regions including Australia (Perth), Australia (Adelaide), Austria, Bahrain, Belgium, Brazil (Brasilia, Rio de Janeiro, Sao Paulo), Canada (St. John's), Canada (Halifax), Canada (Montreal, Ottawa, Toronto), Canada (Winnipeg), Canada (Calgary), Canada (Vancouver), Chile, Cuba, Denmark, Egypt, Finland, France, Germany, and Greece. The 'SNTP Settings' section is visible on the left, with fields for 'SNTP Server Address', 'Time Zone', and 'Auto Date & Time Sync with'. A 'Submit' button is at the bottom left.

- If you select **Custom**, you must configure the Time Offset and choose whether you want the DST to be applied by Day and Month or by Date and Month and define the DST Start and End time.

The screenshot shows the 'Daylight Saving Time' configuration window with 'DST Type' set to 'Custom'. The 'Time Offset (Minutes)' is set to 0. The 'Type' dropdown is set to 'Day-Month wise'. Below these settings is a table for defining the DST Start and End times.

	Ordinal	Day	Month	Time	
				Hours	Minutes
DST Start	1st	Sunday	January	00	00
DST End	1st	Sunday	January	00	00

- In the **Time Offset (Minutes)**, enter the time in minutes which the system 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.
  - or–
  - **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 selected the **Day-Month Wise** option, 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.

**Example:** If you are installing SETU VGRX in a country in the European Union, as per the European Summer Time, the DST would start on the Last Sunday in March and end on the Last Sunday in October each year. Clocks are advanced by one hour at 01:00 hours GMT at the start of DST and set back by one hour at 01:00 hours GMT when DST ends. Let us take the example of setting DST, if SETU VGRX were installed in Berlin, Germany. In the year 2011, the DST in Berlin starts on Sunday, 27 March at 02:00:00 hours and ends on Sunday 30 October at 03:00:00 hours. To set DST you must do the following:

1. Select the **DST Type** as **Custom**.
2. Set the **Time Offset** as 60 minutes.
3. Select the option **Date-Month Wise** as **Type**<sup>6</sup>.

---

6. You can also select Day-Month-wise as Type.

4. Configure the **DST Start** as follows:
  - Select **March** as the **Month**.
  - Select **27th** as the **Date**.
  - Set **Time** to 01:59:59
5. Now, go to the option **DST End**, and configure as follows.
  - Select **October** as the **Month**.
  - Select **30th** as the **Date**.
  - Set **Time** to 02:59:59.
6. Click **Submit** to save DST settings.

On Sunday 27 March at 01:59:59 the SETU VGRX will set the clock forward by 1 hour. On Sunday 30 October, SETU VGRX sets the clock back by 1 hour at 02:59:59.

## SNTP Settings

To use SNTP for synchronizing with the Real Time Clock,

- Click the **Basic Settings** link to expand.
- Click the **Date-Time Settings** link.
- Under **SNTP Settings**, do the following.

- In **SNTP Server Address**, enter the Time Server Address. The SNTP Server address can be of maximum 40 characters. Default: Blank.
- By default, the time zone for the country/region where SETU VGRX is installed is automatically selected when you select 'Region'. If required you may change the time zone by selecting the desired country/region from the **Time Zone** list. Default: India (GMT+05:30).
- If you want the system to synchronize date and time with the SNTP Server automatically at Power On, select the **Auto Date and Time Sync with SNTP during Power ON?** check box.

At every power ON, SETU VGRX will synchronize its date and time with the Time Server address you have entered as SNTP Server Address.

By default, Auto Date and Time Sync with SNTP during Power ON is disabled.

- To synchronize date and time of SETU VGRX with the SNTP Server whenever required, click the **Sync Date and Time Server with SNTP** button.
- Click **Submit** to save the changes.

## System Parameters

System Parameters are general parameters, related to features and facilities that are applied system-wide, such as system name, NAT and SIP related parameters, Distinctive Rings, Server Port, Management/Security, Certificate.

To change the settings of System Parameters,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **System Parameters** link. The **System Parameters** page opens.

The screenshot displays the MATRIX SETU VGRX web interface. The top header shows the MATRIX logo and the text 'SETU VGRX'. On the left, a navigation menu is visible with sections: 'Basic Settings', 'Advanced Settings' (expanded), 'Maintenance', and 'Status'. Under 'Advanced Settings', 'System Parameters' is selected and highlighted. The main content area is titled 'System Parameters' and contains a list of expandable sections: General, NAT, SIP, Distinctive Rings, Message Wait, Server Port, Management/Security, and Certificate. At the bottom of this section are two buttons: 'Submit' (with a checkmark icon) and 'Default' (with a reset icon).

## General Parameters

- Click **General** to expand and configure the following.

General	
System Name	<input type="text"/>
SIP Trunk for IP Dialing	SIP Group ▼ 1 ▼
Play Routing Tone	<input type="checkbox"/> Yes
Call Release Timer	<input checked="" type="checkbox"/> Apply
Release Timer	999 Minutes
VoIP Silence Detection	<input checked="" type="checkbox"/> Apply
VoIP Silence Disconnect Timer	999 Seconds
Radio Port Inactivity Timer	60 Seconds
Routing Group Busy Wait Timer	1 Seconds
Transfer Notification Timer	60 Seconds
Ring Timer	45 Seconds
DTMF Detection on FXS Port	Using SLIC module ▼
Error Tone Timer	7 Seconds
Error Tone Delay Timer	0 Seconds
Unconnected Calls Record Delete Timer	999 Minutes <input type="button" value="Clear Unconnected Call Records"/>
Replace '+' from the CLI	<input type="checkbox"/> Yes
Remove Country Code from CLI received	<input type="checkbox"/> Yes
Apply RCOC only if the caller calls back on the same trunk from which the call was made	<input type="checkbox"/> Yes

- System Name:** You can assign a name to SETU VGRX as System Name. This name can serve as an identifier, when there is more than one SETU VGRX connected in the same LAN network.

System Name can be of a maximum of 40 characters. Default: Blank.

- SIP Trunk for IP Dialing:** To use the IP Dialing feature, that is, to directly dial IP Addresses, you must select a **SIP Trunk** or **SIP Group** for routing the call to the IP Address. For example, if you configured SIP Trunk Group 3 for IP Dialing, you must select 3. See ["IP Dialing"](#) to know more about this feature.

The valid range for the SIP Trunk and SIP Group is 1 to 9. Default: SIP Group 1.

When you assign a SIP Trunk, make sure it is enabled and has the necessary configuration done. For instructions, see ["SIP Trunk"](#) under *Basic Settings*.

When you assign a SIP Group, you must also configure the SIP Group. See ["Group"](#) for instructions.

- Play Routing Tone:** Select this check box, to enable the routing tone. Default: Disabled. Routing Tone is played at the time of routing the call to the destination port. During an outgoing call, the routing tone indicates that the call is in progress.

- **Call Release Timer:** Select this check box, if you want the system to release the ports involved in a call after a definite time period. Default: Enabled.

This timer is loaded when a call gets matured and stops whenever a port involved in a call is released.

- **Release Timer:** Set the duration of the Release Timer. Valid range is 001 to 999 minutes. Default: 999 minutes.
- **VoIP Silence Detection:** Select this check box, if you want the system to disconnect the SIP call when continuous silence (no RTP Packets) is detected for the set time period. Default: Enabled.

This timer is loaded whenever silence is detected during an IP call. The IP call gets disconnected if continuous silence is detected after the expiry of this timer. This timer is applicable for all types of calls received or made through the SIP Trunks.

- **VoIP Silence Disconnect Timer:** Set the duration of the VoIP Silence Disconnect Timer. Valid range is 001 to 999 seconds. Default: 999 seconds.
- **Radio Port Inactivity Timer:** Set the duration of the Radio Port Inactivity Timer. Valid range is 15 to 999 seconds. Default: 60 seconds.

When the Radio Port is in speech with another port and if no activity is detected on the Radio Port for the set duration, SETU VGRX releases the Radio Port.

- **Routing Group Busy Wait Timer:** Set the duration of the Routing Group Busy Wait Timer. Valid range is 1 to 99 seconds. Default: 1 second.

It is the duration for which SETU VGRX searches for a free destination port in the Routing Group and the Fallback Routing Group to route and place the call. The Routing Group Busy Wait Timer is loaded when no destination port is free in both the Routing Group and the Fallback Routing Group.

- **Transfer Notification Timer (Seconds):** Set the duration of the Transfer Notification Timer. Valid range is 1 to 999 seconds. Default: 60 seconds.

It is the duration for which SETU VGRX will wait for notification of the status of a transferred call, whether transfer target is busy, has answered, has disconnected, etc. This timer is loaded as soon as a user performs a transfer activity and the user (transferor) is notified of the status of the transfer activity within this timer.

- **Ring Timer (Seconds):** Set the duration of the Ring Timer. Valid range is 1 to 99 seconds. Default: 45 seconds.

It is the duration for which SETU VGRX will play a ring on the FXS Port to indicate an incoming call. The Ring timer is loaded when the call is placed on the FXS Port, that is, either there is a ring event on the FXS Port or call waiting beeps in case the FXS Port is busy.

- **DTMF Detection on FXS Port:** Select an appropriate method to detect DTMF digits on the FXS Port. SETU VGRX can detect the DTMF digits either using SLIC Module or using DSP. Default: Using SLIC Module.
- **Error Tone Timer:** Set the duration of the Error Tone Timer. Valid range is 0 to 9 seconds. Default: 7 seconds.

It is the duration for which the system will play the Error Tone.

- **Error Tone Delay Timer:** Set the duration of the Error Tone Delay Timer. Valid range is 00 to 99 seconds. Default: 0 seconds.

It is the duration after which the system will play the Error Tone, if the call is disconnected during speech.

- **Unconnected Calls Record Delete Timer:** SETU VGRX offers a feature on the Mobile Port and SIP Trunks whereby outgoing calls made from these ports that return unconnected are routed to the original caller.

To use this feature on a Mobile Port or a SIP Trunk, you must enable **Route calls returned unconnected to Original Caller** under *Handling of Outgoing Calls* on the respective port.

When an outgoing call is made using the port on which this feature is enabled, and the Called Party is either busy or not responding, SETU VGRX stores the number of the Calling Party, the number of the Called Party, and the source port through which the outgoing call was made. A record of such call is stored for the duration of the *Unconnected Calls Record Delete Timer* (configurable; default: 999 minutes) in the system. If the called party calls back before the expiry of this timer, this incoming call is placed to the original calling party.

The records of 200 such Unconnected Calls are stored using FIFO method, and deleted on the expiry of the Record Delete Timer, or when the call returned by the called party is returned to the original caller and answered by the original caller.

By default, the Unconnected Calls Record Delete Timer is set to 999 minutes. If required, you may change, this timer to the desired duration.

You can also delete the records of unconnected calls any time, without waiting for this timer to expire. To do this, click the **Clear Unconnected Call Records** button.

- **Replace '+' from CLI received:** The GSM network presents the calling party number with the prefix '+' to the called party. However, not all equipments can present the calling party number containing '+'. SETU VGRX enables you to remove the prefix '+' and replace it with an appropriate number string, if required.

If you want the system to replace '+' in the CLI received, select this check box. Default: Disabled.

You may also program the number string with which '+' is to be replaced in the CLI. In **Replace '+' from CLI with the number string**, enter the number string with which you want to replace '+' received as prefix of the calling party number.

If you keep the number string field blank, SETU VGRX will remove the '+' sign from the CLI of the calling party and present the remaining digits to the Called Party.

For example:

The number string +919327237228 is received as CLI.

If '00' is configured as the replacement string, the CLI number would be presented as 00919327237228.

If no replacement string is configured (left blank), the CLI number would be presented as 919327237228.

- **Remove Country Code from CLI received:** You may remove country code from the CLI received on the source port, before presenting it on the destination port, if required.

If you want the system to remove country code from the CLI received, select this check box. Default: Disabled. Make sure you configure “Country Code” under *Region* in Basic Settings.

- **Apply RCOC only if the caller calls back on the same trunk from which the call was made:** Select this check box, if you want SETU VGRX to match the Trunk Port Parameters (Trunk Port Number and Type) of an incoming call with the entry in the RCOC table while applying RCOC logic on the “SIP Trunk” and “Mobile Port”. Default: Disabled.

If this check box is enabled, SETU VGRX will match Trunk port parameters of the incoming call with the entry stored in RCOC table. If a match is found, it will route the incoming call to original caller.

## NAT

- Click **NAT** to expand and configure the following.

- **STUN Server Address: Port:** STUN (Simple Traversal of UDP through NAT) server facilitates traversing through most NATs, except symmetric NATs. So, if your router has Symmetric NAT, do not configure STUN. If your SETU VGRX is located behind a NAT router that is other than symmetric, use STUN.

In **STUN Server Address: Port**, enter the STUN Server Address and the Listening Port of the STUN Server.

The STUN Server Address can have a maximum of 40 characters.

The valid range of the STUN Server Port is from 1024–65535. Default: 3478.

- **Use SIP Port Fetched using STUN:** Clear this check box, if your SETU VGRX is located behind the NAT router and you have forwarded the SIP listening port of the SETU VGRX in the router.

Keep the **SIP Port fetched using STUN** check box enabled, if you have *not* forwarded the SIP Listening Port in the router.



Make sure you configure the **NAT Type** as **STUN** under “Advanced” in the “SIP Network Profile”.

- **Router’s Public IP Address:** The Router’s public IP address specifies the public IP address of the NAT router behind which system is located. Default: Blank.

You need to configure this parameter only if the system is located behind the NAT router and a Static IP Address is assigned as Public IP Address of the Router.



Make sure you configure the **NAT Type** as **Router's IP Address** under **"Advanced"** in the **"SIP Network Profile"**.

- **UDP NAT Keep Alive:** When SETU VGRX 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.

Select the **UDP NAT Keep Alive** check box to enable. Default: Disabled.

- **Keep Alive Message:** Select the type of **Keep Alive Message** to be sent. You may select either REGISTER or NOTIFY. Default: NOTIFY.

As **Interval**, set the time period after which the system should send Keep Alive messages. This time period should be less than the NAT binding timer of the router. The valid range for the UDP NAT Keep Alive Interval is 001–999 seconds. Default: 120 seconds.

- **TCP NAT Keep Alive:** When SETU VGRX 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.

- Select the **TCP NAT Keep Alive** check box, if you want the system to send Keep Alive messages periodically to refresh the binding in the NAT router. Default: Disabled.

As **Interval**, set the time period after which the system should send Keep Alive messages. This time period should be less than the NAT binding timer of the router. The valid range for the TCP NAT Keep Alive Interval is 001–999 seconds. Default: 120 seconds.

- Click **Submit** to save changes.

## SIP

- Click **SIP** to expand and configure the following.

The screenshot shows a configuration window titled "SIP" with the following settings:

100rel/PRACK	<input type="checkbox"/> Enable
SIP over TCP	<input checked="" type="checkbox"/> Enable
SIP over TLS	<input checked="" type="checkbox"/> Enable
SIP UDP Port	5060
SIP TCP Port	5060
SIP TLS Port	5061
SIP INVITE Timer	30 Seconds
SIP Provisional Timer	180 Seconds
General Request Timer	20 Seconds

- **100rel/PRACK:** This parameter is to be configured if you want to support reliable transmission of (SIP) provisional responses.

Select the **100rel/PRACK Enable** check box, if you want the SETU VGRX to use 100rel SIP extension for reliable transmission of SIP provisional responses and to use PRACK (Provisional Acknowledgement). Default: Disabled.

- Select the **SIP Over TCP** check box, if you want SETU VGRX to receive SIP messages over TCP. Default: Enabled.

SETU VGRX supports transporting of SIP messages over User Datagram Protocol (UDP) as well as Transfer Control Protocol (TCP) connection. Despite the advantages that SIP Over TCP offers, it is more common to use UDP to transport the SIP messages.

Make sure that you have selected *TCP* as the *SIP Transport* option and have enabled the *TCP (Fallback to UDP)* check box. See [“Advanced”](#) in [“SIP Network Profile”](#).

- Select the **SIP Over TLS** check box, if you want SETU VGRX to receive SIP messages over TLS. Default: Enabled.

SETU VGRX supports transporting of SIP messages over TLS. TLS protects SIP signaling against loss of integrity, confidentiality and replay.

Make sure that you have selected *TLS* as the *SIP Transport* option. See [“Advanced”](#) in [“SIP Network Profile”](#).

- **SIP UDP Port:** Enter the SIP UDP Port. This is the port on which the SETU VGRX listens for SIP messages transported over UDP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1031–65534. Default: 5060.
- **SIP TCP Port:** Enter the SIP TCP Port. This is the port on which the SETU VGRX listens for SIP messages transported over TCP. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1031–65534. Default: 5060.
- **SIP TLS Port:** Enter the SIP TLS Port. This is the port on which the SETU VGRX listens for SIP messages transported over TLS. This port is also used as the source port for sending SIP messages to the remote peer. The valid range for this port is 1031–65534. Default: 5061.
- **SIP INVITE Timer:** Set the duration of the SIP INVITE Timer. Valid range is 10–200 seconds. Default: 30 seconds.

This is the time in seconds for which SETU VGRX 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 SETU VGRX terminates the call process and gives an error tone to the user.

- **SIP Provisional Timer:** Set the duration of the SIP Provisional Timer. Valid range is 10–200 seconds. Default: 180 seconds.

This is the time in seconds for which SETU VGRX 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 the expiry of the timer, the SETU VGRX terminates the call process and gives error tone to the user.

- **General Request Timer:** Set the duration of the General Request Timer. Valid range is 10–60 seconds. Default: 20 seconds.

This is the time in seconds for which the SETU VGRX waits for response for a transaction request. This timer starts on initiating a transaction and stops on the receipt of a response for the request. On expiry of the timer, the SETU VGRX clears the transaction.

- Click **Submit** to save changes.



*If you have made any changes in the NAT or SIP Parameters, all the current ongoing calls will be disconnected when you submit the page to save the changes.*

## Distinctive Rings

Distinctive Rings are the ringing patterns used for distinguishing between different types of call events.

The distinctive ringing pattern is selected according to Alert-Info header that is included in INVITE message. For example: Alert-Info <Bellcore-dr2>, or Alert-Info<http://.../Bellcore-dr2>. 'dr2' defines that ringing pattern number 2 will be played on the FXS Port. If the Alert-Info header is missing, the default ring tone will be played on the FXS Port.

- Click **Distinctive Rings** to expand and configure the following.

Distinctive Rings


☐

Index	Ring Text	Ring Type
1		4
2		4
3		4
4		4
5		4
6		4
7		4
8		4

- Select the **Distinctive Rings** check box, to enable.
- In **Ring Text**, enter the Ring Text that you expect to receive in the Alert-Info header of the INVITE message during an incoming call. The Ring Text can be a maximum of 24 characters.
- To assign the Ring Type, click on the **Ring Type** link.

The Ring Type window opens.

Ring Type							
Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, Korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	

 Close

The Ring Type numbers, Ring Cadence and the corresponding Supported Countries are displayed.

- Take a note of the Ring Type number you wish to configure for each Ring Text.
- Close the Ring Type window.
- Select the desired **Ring Type** number for each Ring Text.



Keep the *Distinctive Rings* check box disabled, if you want SETU VGRX to play country-specific Ring Type on the FXS Port. For details, see [“Region”](#).

- Click **Submit** to save changes.

## Message Wait

This parameter is related to the [“Message Wait Indication on SIP Trunks”](#) feature. If you have selected *Message Wait Notification* type as *Ring* on the FXS Port, you may also configure the following parameters related to Message Wait Ring.

- Click **Message Wait** to expand and configure the following.

Message Wait	
Message Wait Ring Timer	30 Seconds
Message Wait Ring Interval Timer	1 Minutes
Message Wait Ring Count	03

- Message Wait Ring Timer:** Set the Message Wait Ring Timer. Valid range is 01–60 seconds. Default 30 seconds.

It is the time duration for which the Message Wait Ring will be played on the FXS Port for Message Wait Notification.

- Message Wait Ring Interval Timer:** Set the Message Wait Ring Interval Timer. Valid range is 001–999 minutes. Default: 1 minute.

It is the time duration after which SETU VGRX will play the Message Wait Ring again on the FXS Port for the Message Wait Notification, if the previous ring was unanswered by the user.

- Message Wait Ring Count:** Select the Message Wait Ring Count. Valid range is 1–10. Default: 3.

For this number of times, the SETU VGRX will play the Message Wait Ring on the FXS Port, until it is answered by the FXS Port user.

## Server Port

- Click **Server Port** to expand and configure the following.

Server Port	
HTTP Web Server Port	80
HTTPS Web Server Port	443
FTP Server Port	21
Telnet Server Port	23

- HTTP Web Server Port:** SETU VGRX has an embedded web server called *Jeeves*, for system configuration. You can access *Jeeves* using HTTP.

Enter the HTTP Web Server Port. Valid range of this port is: 80, 1031-65535. By default, HTTP Web Server Port is 80.

- HTTPS Web Server Port:** You can access *Jeeves* of SETU VGRX using HTTPS.

Enter the HTTP Web Server Port. Valid range of this port is: 443, 1031-65535. By default, HTTPS Web Server Port is 443.

- FTP Server Port:** SETU VGRX has an embedded FTP server for Software Upgrade.

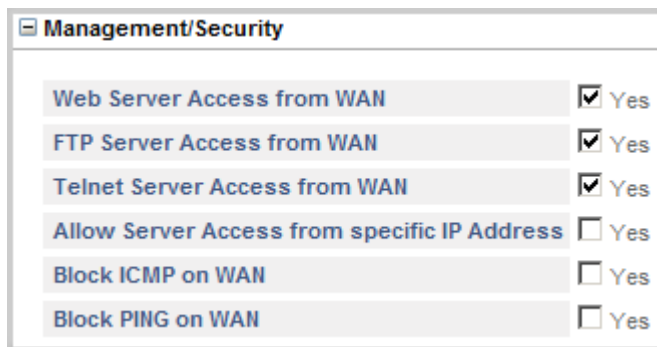
Enter the FTP Server Port. Valid range of this port is: 21, 1031-65535. By default, FTP Server Port is 21.

- **Telnet Server Port:** You can access SETU VGRX using Telnet.

Enter the Telnet Server Port. Valid range of this port is: 23, 1031-65535. By default, Telnet Server Port is 23.

## Management/Security

- Click **Management/Security** to expand and configure the following.



The screenshot shows a configuration window titled "Management/Security". It contains a list of settings, each with a label and a checkbox followed by the word "Yes".

Setting	Checkbox	Label
Web Server Access from WAN	<input checked="" type="checkbox"/>	Yes
FTP Server Access from WAN	<input checked="" type="checkbox"/>	Yes
Telnet Server Access from WAN	<input checked="" type="checkbox"/>	Yes
Allow Server Access from specific IP Address	<input type="checkbox"/>	Yes
Block ICMP on WAN	<input type="checkbox"/>	Yes
Block PING on WAN	<input type="checkbox"/>	Yes

- **Web Server Access from WAN:** Keep this check box enabled, if you want to allow users to access the system's Web Server (Jeeves) from the WAN Port.

You may clear this check box, if required. Default: Enabled.


- **FTP Server Access from WAN:** Keep this check box enabled, if you want to allow users to access the system's FTP Server from the WAN Port.

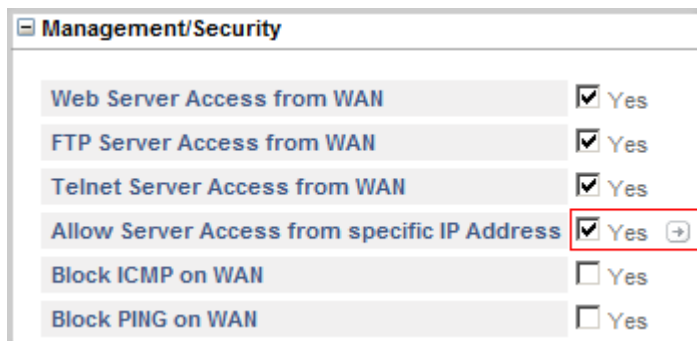
You may clear this check box, if required. Default: Enabled.

- **Telnet Server Access from WAN:** Keep this check box enabled, if you want to allow users to access the system using Telnet from the WAN Port.


You may clear this check box, if required. Default: Enabled.

- **Allow Server access from specific IP Address:** Enable this check box, if you want to allow users to access system from specific IP Addresses only. Default: Disabled.

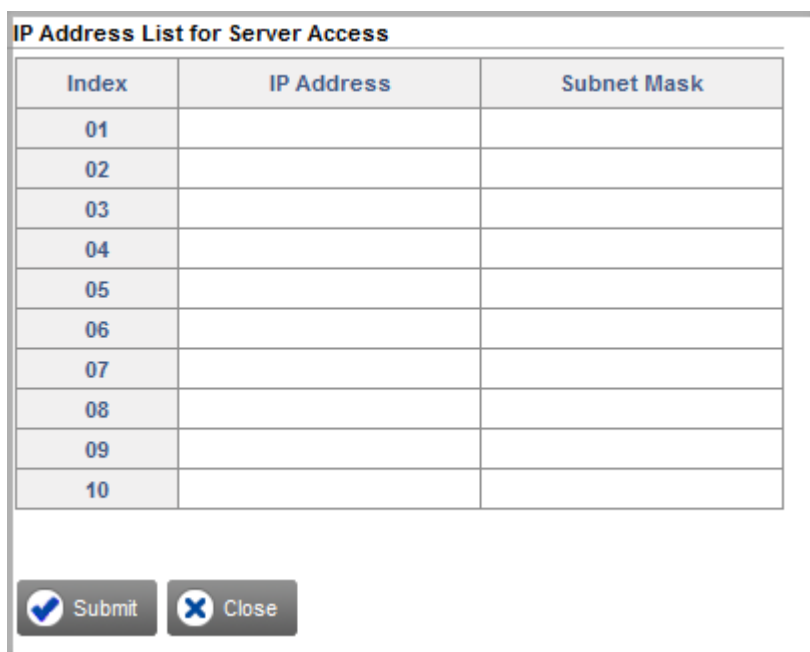
If you enable this parameter, you must configure the IP Address table for Server Access. To configure the IP Address table, Click **Settings**  .



The image shows a 'Management/Security' settings window. It contains several options with checkboxes and 'Yes' labels. The option 'Allow Server Access from specific IP Address' is checked, and its 'Yes' label is highlighted with a red box and a small arrow icon next to it.


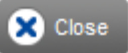
Setting	Value
Web Server Access from WAN	<input checked="" type="checkbox"/> Yes
FTP Server Access from WAN	<input checked="" type="checkbox"/> Yes
Telnet Server Access from WAN	<input checked="" type="checkbox"/> Yes
Allow Server Access from specific IP Address	<input checked="" type="checkbox"/> Yes 
Block ICMP on WAN	<input type="checkbox"/> Yes
Block PING on WAN	<input type="checkbox"/> Yes

The **IP Address List for Server Access** opens in a new window. You can store 10 entries in this table.



The image shows a window titled 'IP Address List for Server Access'. It contains a table with 10 rows and 3 columns: Index, IP Address, and Subnet Mask. Below the table are 'Submit' and 'Close' buttons.

Index	IP Address	Subnet Mask
01		
02		
03		
04		
05		
06		
07		
08		
09		
10		

- Enter the IP Addresses and their respective Subnet Mask in the table.
- Click Submit and close the window.

SETU VGRX will allow system access only to those users whose IP Address matches with the one configured in the IP Address List for Server Access.

- **Block ICMP on WAN:** Enable this check box, if you want the system to discard the ICMP packets received on WAN. Default: Disabled.
- **Block PING on WAN:** Enable this check box, if you want the system to discard the PING request received on WAN. Default: Disabled.

Blocking of PING on WAN will prevent your network from being pinged or detected by other Internet users and acquire your IP Address.



**Block PING on WAN** will not be applicable, if you have enabled **Block ICMP on WAN**.

## Certificate

- Click **Certificate** to expand and select the certificate for each of the following.

Certificate	
Local Certificate for TLS	DefaultServerCert_Setu ▼
Local Certificate for WebServer	DefaultServerCert_Setu ▼
Local Certificate for Firmware Upgrade	DefaultServerCert_Setu ▼
Local Certificate for Configuration Upgrade	DefaultServerCert_Setu ▼

- In **Local Certificate for TLS**, select the certificate to be used by the system for TLS.
- In **Local Certificate for WebServer**, select the certificate to be used by the system for accessing the WebServer.
- In **Local Certificate for Firmware Upgrade**, select the certificate to be used by the system for Firmware Upgrade.
- In **Local Certificate for Configuration Upgrade**, select the certificate to be used by the system for Configuration Upgrade.

To create and Upload /Download Certificates, see "[Certificate Manager](#)".

# Dial Plan

---

SETU VGRX supports 8 Dial Plans with total 64 entries in each table. The Dial Plan contains a series of digits and/or wildcard characters.

When a user dials a number, it is compared with the Destination Number configured in the Dial Plan. If a match is found, the system routes the call immediately without waiting for End of Dialing and if a match is not found, the system will wait for the End of Dialing and then routes the call as per the Destination Port Selection method configured.

Dial Plan will be applied on the **FXS Port**, when the Destination Port Selection method is **Fixed**.

Dial Plan will be applied on the **Radio Port** when,

- the Destination Number Selection method used for routing the call is **After collecting the digits**.
- and
- the Destination Port Selection method is **Fixed**.

Dial Plan will be applied on the—**FXO Port**, **Mobile Port** and **SIP Trunk**—when,

- the Destination Number Selection method used for routing the call is **Answering the call and collecting the digits**.
- and
- the Destination Port Selection method is either **Fixed** or **Calling Number Based**.

## Configuring Dial Plan Table

- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **Dial Plan** link.

The screenshot shows the SETU VGRX web interface. On the left, a sidebar menu lists various settings, with 'Dial Plan' selected under 'Advanced Settings'. The main content area displays the 'Dial Plan Table - 1', which is a table with two columns: 'Index' and 'Destination Number'. The table contains 15 rows, with the first row (Index 01) highlighted. Below the table, there is a 'Testing' section with a text input field for entering a destination number and a 'Search' button. At the bottom of the interface, there are three buttons: 'Submit', 'Default', and 'Copy'.

The Dial Plan Table allows you to configure up to 64 entries. Each entry is stored against an Index number.

For each entry,

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + “Wildcard Characters”). Valid characters: 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.
- Click **Submit** to save.



*If there are multiple entries in the Dial Plan table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

## Wildcard Characters

SETU VGRX supports following characters.

Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.

<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.
<b>[ ^ ]</b>	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
<b>T (letter T)</b>	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

Refer the following table to understand how a Dial Plan can be configured.

<b>Dial Plan Entry</b>	<b>Description</b>
1XX	Allows you to dial any number in a range from 100 to 199.
[2-5]XX	Allows you to dial any 3 digit number in a range from 200-599.
[2,3,8]XX	Allows you to dial any 3 digit number in the range from 200-299, 300-399, 800-899.
[2-9]XXXXXX	Allows you to dial any 7 digit number in the range from 2000000-9999999.
23[^2]1	Allows you to dial a 4 digit number: 2301, 2311, 2331, 2341, 2351, 2361, 2371, 2381, 2391.
2630[500-550]	Allows you to dial a 7 digit number in the range from 2630500-2630550.
[^6-7]X	Allows you to dial a 2 digit number in the range from 00 to 99 except the numbers from 60 to 79.
1234	Allows you to dial 1234 number only.
011T	Allows you to dial any number starting with 011. The number must be of minimum 3 digits and maximum digits must be as configured for the port.

# Number Lists

---

A Number List is a data structure that constitutes digit and character strings which must be configured for the system to support the features described in the following.

SETU VGRX offers as many as 24 number lists. Each number list can store up to 64 entries of a maximum of 24 characters each.

You need to configure number lists for the features described in the following. By default, each of these features is assigned particular number lists. You may retain the number list assigned by default, or configure another number list and assign this list to the feature.

## Allowed - Denied Logic

You can apply the Allowed-Denied logic on a source port—FXS, Radio, FXO, Mobile, SIP—if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

The Allowed-Denied logic makes use of two Number lists:

- **Allowed Number List:** This is the list of numbers that can be dialed out from the source port.
- **Denied Number List:** This list contains the numbers that are to be restricted from being dialed out from the source port.

Both lists must be programmed separately for each port first and then assigned to the respective port.

When Allowed-Denied Logic is enabled on a source port, for each number dialed from the port, SETU VGRX uses the best-match-found logic to compare the dialed number with the Allowed Number list and the Denied Number list.

The number is allowed to be dialed, if the dialed number:

- matches with both lists.
- matches with Allowed Number list, but not with the Denied Number list.
- matches with neither the Allowed List nor the Denied List.

The number is denied, if it matches with the Denied Number list, but not with the Allowed Number list.

Allowed-Denied Number feature is not applicable in following cases:

- Destination number string matches with any Access Code.
- Destination number string matches with any Emergency Number.
- For Call Forward Number programmed.
- *Route all Incoming Calls (with CLI)* option selected is:
  - Fixed Destination Number
  - or -
  - on basis of Calling Party Number.

To apply this feature,

- you must configure the numbers you want to allow and restrict from being dialed out in the Allowed and Denied Number lists.

By default, the following number lists are assigned for Allowed Denied Logic for each port type:

Port Type	Default Allowed Numbers List	Default Denied Numbers List
FXS Ports	List 05	List 06
Radio Ports	List 08	List 09
FXO Ports	List 01	List 02
Mobile Ports	List 01	List 02
SIP Trunks	List 07	List 08

You may retain these lists or configure any other Number list from 01 to 24.

- enable **Allowed-Denied Logic** on the port type—FXS, Radio, FXO, Mobile, SIP—on which you want to apply this feature.
- configure the numbers you want to allow and the numbers you want to restrict in the default **Allowed Number List** and **Denied Number List** assigned to the port.

For instructions, see the following topics under *Basic Settings*:

[“Handling of Outgoing Calls” on “FXS Port”](#)

[“Handling of Outgoing Calls” on “Radio Port”](#)

[“Handling of Incoming Calls” on “FXO Port”](#)

[“Handling of Incoming Calls” on “Mobile Port”](#)

[“Handling of Incoming Calls” on “SIP Trunk”](#)

If you do not want to use the default Number Lists assigned to the ports, you may select a different List Number and configure it. In which case, you must select the List Number you configured as the Allowed Number List/Denied Number List for the port.

## Black Listed Callers

The Black Listed Callers feature enables you to block incoming calls from specific numbers and addresses on SIP Trunks and Mobile Ports. You can apply this feature on a Source Port only.

To use this feature,

- you must configure the numbers of unwanted callers in a Number List.



*Make sure you have configured the entire SIP URI (for example: 12345@abc.com) of the unwanted callers in the Blacklisted Callers Number List.*

- enable the **Reject Calls from Blacklisted Callers** check box on the SIP Trunks and Mobile Ports on which you want to apply this feature.
- select the Number List you configured as **Black Listed Callers List**.

For instructions, see the following topics under *Basic Settings*:

[“Handling of Incoming Calls” on “SIP Trunk”](#)

[“Handling of Incoming Calls” on “Mobile Port”](#)

Make a list of numbers that you want to black list. Configure these numbers in a Number List. By default, Number List 16 is assigned as the Black Listed Callers List for the Mobile Ports and Number List 11 is assigned as the Black Listed Callers List for the SIP Trunks.

You may retain this list and configure all the numbers you want to black list in this list or you may configure different number lists for different ports and assign the lists to the ports.

Now, whenever there is an incoming call on the SIP Trunk or Mobile Port you have applied this feature, the SETU VGRX will match the number with the Blacklisted Callers' Number list you have assigned. If the number matches with any of the numbers you have blacklisted, the system will reject the call.



*Each number string in the List can have a maximum of 24 characters. If the callers' number exceeds 24 characters, the first 24 characters of the number will be checked. If the first 24 characters of the callers' number match perfectly with any of the numbers programmed in Blacklisted Callers List, the call will be rejected.*

## Call Detail Record Filters

SETU VGRX enables you to generate reports of Call Detail Records using different filters. You can generate Call Detail Record report of calls made to specific numbers (Called Party Numbers) and calls received from specific numbers (Calling Party Numbers).

When you want to filter calls by Called Party and Calling Party Numbers, you must configure a Number list for each of these.

To generate Call Detail Records using Called Party and Calling Party Numbers as filters,

- make a list of Called Party Numbers and another list of Calling Party Numbers.
- configure a Number List with the Called Party Numbers and another Number List with the Calling Party Numbers.

By default, Number list 01 is assigned for both Called Party and Calling Party numbers. You may retain this list and configure Called Party and Calling Party numbers in this list, or you may retain this for Called Party Numbers and configure another list number for Calling Party numbers. In which case you must assign the list you configured to the respective filter.

- assign the Called Party Number list you configured to the CDR filter **Called Party Number Matching with Number List**.
- assign the Calling Party Number list you configured to the CDR filter **Calling Party Number Matching with Number List**.

For instructions, see [“Call Detail Record”](#).

## Configuring Number Lists

You must determine the purpose for which the list is required and accordingly prepare them.

To configure Number lists,

- Log into Jeeves.

- Click the **Advanced Settings** link.
- Click the **Number List** link.

1-4
5-8
9-12
13-16
17-20
21-24

**Number Lists**

Location	List 1	List 2	List 3	List 4
01	0			
02	1			
03	2			
04	3			
05	4			
06	5			
07	6			
08	7			
09	8			
10	9			
11	*			
12	#			
13	+			
14	a			
15	b			
16	c			

✓ Submit
⚙ Default

- List 1 to 4 appear on the page. To select another List number, click the tab on the top of the table.
- Select the list number you want to configure.
- Enter the numbers strings in each list.
- Click **Submit** to save entries.
- Assign the list to the respective features for which you configured them on the various port types.

For example, if you configured Number List 22 with black listed numbers for the Black Listed Callers feature on SIP Trunk 2,

- Under **Basic Settings**, click **SIP Trunks**.
- Click the **SIP 2** tab.
- Under Handling of Incoming Calls, select the **Reject Calls from Blacklisted Callers** check box.
- In **Blacklisted Callers Number List**, select **22**.
- Click **Submit**.

You can also configure Number lists on the respective port—FXS Port, FXO Port, Radio Port, SIP Trunk, Mobile Port under the “[Basic Settings](#)”.

# Automatic Number Translation (ANT)

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Automatic Number Translation (ANT) is used to modify the number string—entire number or part thereof—into the desired number string as per your requirement. ANT is useful when you need to modify the Called/Calling number, before the system routes the call further.

For example, in India the PSTN requires you to dial the prefix 00 for calling international numbers, whereas the ITSP you have subscribed the SIP Trunk with, restricts the dialing of the prefix 00. If you dial this prefix, your call will be rejected by the ITSP. The ANT Table will enable you to modify the Number string as per your requirement so that the calls routed through the SIP Trunk are not rejected.

The Automatic Number Translation feature can be applied on all the SIP Trunks, Mobile Ports, FXO Ports and the FXS Ports.

Automatic Number Translation makes use of Automatic Number Translation Table. The ANT Table consists of three columns:

- **Number:** In this column, enter the numbers that you want the system to modify.
- **Strip Digit:** In this column, enter the number of digit(s) to be stripped off by the system from the Called/Calling number string. If you do not want any digits to be stripped, enter '0'.
- **Add Prefix:** In this column, enter the digit(s) which are to be added as prefix to the Called/Calling number string by the system before routing it further.

To apply this feature on the desired port,

- on a piece of paper make a table, in the first column note down the numbers that need to be modified. In the second column enter the number of digits you want the system to strip off (if required), and in the third column, enter the number you want the system to add as prefix (if required).
- configure the **Automatic Number Translation Table**. You can configure upto 8 different ANT Tables.
- enable **Automatic Number Translation (ANT) for Called Number** and/or **Automatic Number Translation (ANT) for Calling Number** on the respective ports/trunks, on which you want to apply this feature.
- assign the **Automatic Number Translation Table** you configured.
- configure the **Pause Timer**, if applicable.

For instructions, see [“General”](#) under the [“FXS Port”](#), [“Handling of Outgoing Calls”](#) under the [“FXO Port”](#), [“Handling of Outgoing Calls”](#) under the [“SIP Trunk”](#), and [“Handling of Outgoing Calls”](#) under the [“Mobile Port”](#).

Now, whenever there is a call on/from the Port on which you have applied this feature, SETU VGRX will match the Called/Calling number with the Number configured in the Automatic Number Translation Table using the best match found logic.

- If a match is found, the system will check whether and how many digits to strip off. It will strip off digits according to the number you have entered in the Strip Digit column. If '0' is configured in the Strip Digit column, it will check the Add Prefix column. If configured, the system will add that prefix. If no prefix is configured, the system will route the same number string further.

If ~ (Wait for Answer) is configured in the Add Prefix column, the system will wait for the call to mature. Similarly, if ^ (Pause) is configured in the Add Prefix column, the system will wait for the Pause timer and then route the call further.

- If no match is found for the Called/Calling number in the ANT Table, the system will route the number string, without modifying it.



*Automatic Number Translation feature will not be applied when Emergency Numbers are dialed.*

Automatic Number Translation also forms the basis of Multi-Stage Dialing. Using of Calling Card for making international calls is the most common example of Multi-Stage Dialing.

While using a Calling Card, you have to dial the digits in the following sequence:

1. Dial the number for using the Calling Card, for example, 160223.
2. After the call is matured, dial the PIN number printed on the Calling Card, for example, 113212.
3. At last, dial the international number you want to call. For example, 0014162357896.

Thus, you will have to dial the Calling Card number and the PIN number every time before dialing the international number. To avoid repetitive dialing of these fixed digits for making a call, you can configure the ANT table as under.

- In **Number**, configure '00', the prefix for international numbers.
- In **Add Prefix**, configure the Calling Card server number and the PIN Number.

As the system must wait for the Calling Card server to answer before dialing the PIN, you must configure Wait for Answer (~) between the Calling Card server number and the PIN number.

You must also insert a delay by configuring the Pause Timer (^) after the PIN number.

- Keep Strip Digit as 00.
- The Automatic Number Translation table would look like this:

Index	Number	Strip Digit	Add Prefix
1	00	00	160223~113212^
2			
3			
4			
5			
6			
:			
24			

- When the Automatic Number Translation table is configured, the user must simply dial the destination number, say, 0014125126508.

- The system matches the Called number with the Number configured in the ANT table. The number matches with the entry '00' stored in the table.
- The system dials the Add Prefix number string 160223 (number of the calling card server). It waits for the calling card server to answer the call.
- When the call is matured, i.e. the calling card server has answered the call, the system dials the PIN number 113212 and waits for the Pause Timer before dialing the destination number.

Thus, the user can directly dial the desired destination number and the system dials the rest using the ANT table.

## Configuring Automatic Number Translation Table

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Automatic Number Translation (ANT)** link.

1
2
3
4
5
6
7
8

**Automatic Number Translation Table - 1**

Index	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	

**Examples of Number Pattern**

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' to every 7-digit dialed number.

The Automatic Number Translation Table page will open. In this table, you can store as many as 24 Numbers at Index Numbers 01 to 24.

- In **Number**, enter the Called/Calling numbers that need to be modified. You can enter maximum 24 digits. Digits 0-9, #, \*, + and \$ are allowed. Default: Blank.

To configure a range of numbers you can use the character \$. Here, \$ is any number from 0 to 9.

For example, if you want SETU VGRX to add prefix '1' to all 10 digit numbers dialed by the user, configure Number as \$\$\$\$\$\$\$\$, Strip Digit as 0 and Add Prefix as 1. Now, when the user dials any number between the range of 0000000000 to 9999999999, say 4161231234, the system will add prefix 1 to it and dials out the number as 14161231234.

- In **Strip Digit**, enter the number of digits you want the system to strip off from the Called/Calling Number. You can configure from 00-24. Default: 00.
- In **Add Prefix**, enter the number string(s) that you want the system to add as prefix to the Called/Calling Number. You can enter maximum 24 characters. Characters 0-9, \*, #, +, ~ (Wait for Answer), ^ (Pause) are allowed. Default: Blank.
- Click **Submit** to save your entries.

# SIP Network Profile

You can either edit the settings of the default **Network Profile 1** or add a new profile as per your requirement.

To add a new profile,

- Click the **Advanced Settings** link.
- Click the **SIP Profile** link to expand.
- Click the **SIP Network Profile** link.
- By default, **SIP Network Profile 1** page opens.

The screenshot shows the Matrix SETU VGRX web interface. On the left is a sidebar menu with 'Basic Settings' and 'Advanced Settings'. Under 'Advanced Settings', 'SIP Profile' is expanded, and 'SIP Network Profile' is selected. The main content area is titled 'Network Profile 1' and 'SIP Network Profile 1'. It contains several configuration fields: 'SIP Network Profile' with an 'Enable' checkbox checked; 'Name' with the value 'Network Profile 1'; 'Status' with the value 'Enabled'; 'Mode' with radio buttons for 'Proxy' and 'Peer-to-Peer' (selected); 'Allowed IP Address for Incoming SIP Message' with a dropdown menu set to 'As per Peer-to-Peer table'; and 'Digest Authentication' with an 'Apply' checkbox. Below these fields are expandable sections for 'Codec', 'VoIP Profile', and 'Advanced'. At the bottom are three buttons: 'Submit' (with a checkmark icon), 'Default' (with a plus icon), and 'Add New Profile' (with a plus icon).

- Configure the following parameters:
  - Keep the **SIP Network Profile** check box enabled to use this profile.  
Clear this check box, if you do not want to use this profile.
  - Assign a **Name** to the Network Profile for identification. The Name can be a maximum of 24 characters.
  - **Status** displays the status of this Network Profile.
  - Select the **Mode** according to the type of your installation. You can select either **Proxy** or **Peer-to-Peer**. Default: Peer-to-Peer.

- Select the desired option in Allowed IP Address for Incoming SIP Message. See [“Allowed IP Address for Incoming SIP Message”](#) for more details.
- You may also enable [“Digest Authentication”](#) if required.

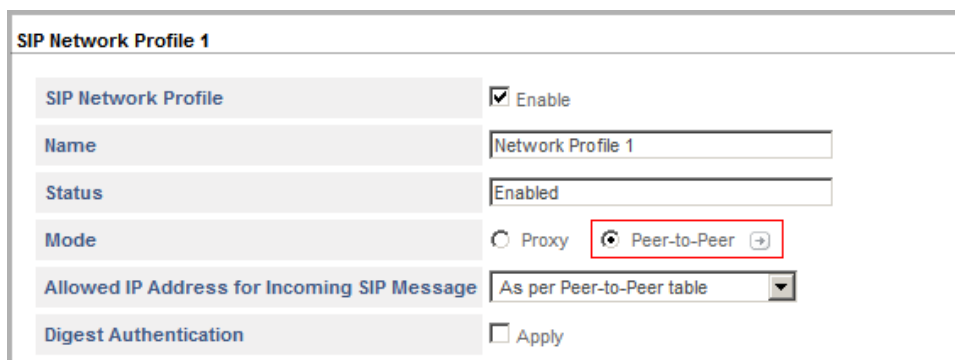
## SIP Trunk Mode: Peer-to-Peer

If you select SIP Trunk Mode as **Peer-to-Peer**, you must configure:



- the Peer-to-Peer Dialing table.
- Trusted IP Address Table, if you select *As per Trusted IP Address table* option in the *Allowed IP Address for Incoming SIP Message*.
- Codec
- VoIP Profile
- Advanced

By default **Peer-to-Peer** is selected as the SIP Trunk **Mode**. To use the SIP Trunk for Peer-to-Peer calls, you must configure the Peer-to-Peer Table.

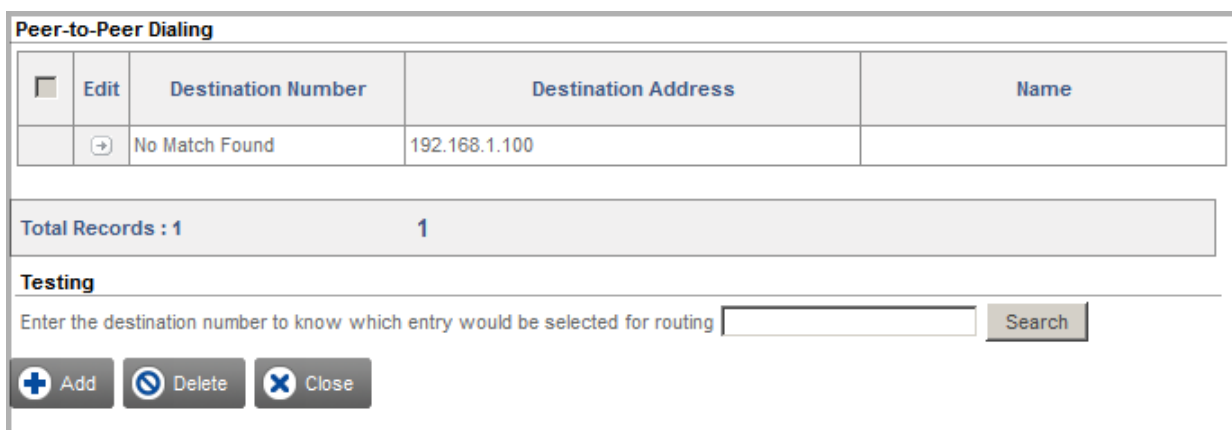
- Click **Settings** .





**SIP Network Profile 1**

SIP Network Profile	<input checked="" type="checkbox"/> Enable
Name	Network Profile 1
Status	Enabled
Mode	<input type="radio"/> Proxy <input checked="" type="radio"/> Peer-to-Peer 
Allowed IP Address for Incoming SIP Message	As per Peer-to-Peer table 
Digest Authentication	<input type="checkbox"/> Apply

The **Peer-to-Peer Dialing** table opens.






**Peer-to-Peer Dialing**

	Edit	Destination Number	Destination Address	Name
		No Match Found	192.168.1.100	

Total Records : 1      1

**Testing**

Enter the destination number to know which entry would be selected for routing

 Add  Delete  Close

- You can add maximum 500 entries. Each entry in the table consists of the Destination Number, Destination Address and Name. For detailed instructions, see [“Peer-to-Peer Dialing”](#).
- Click **Close** to return to the **SIP Network Profile 1** window.

## Allowed IP Address for Incoming SIP Message

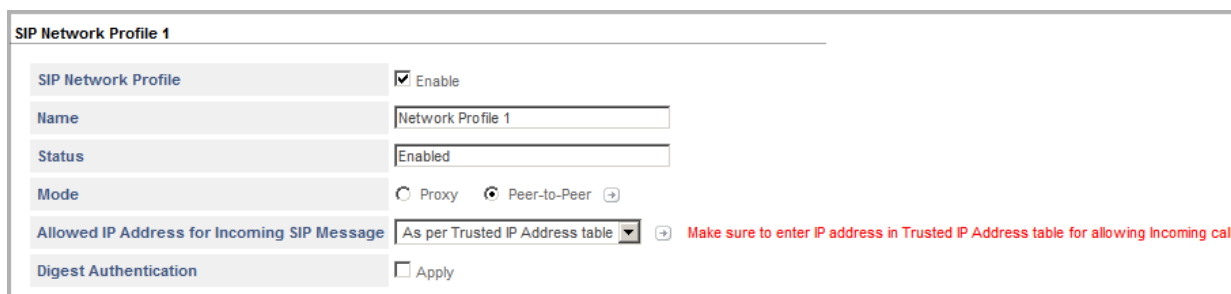
If you have selected the SIP Trunk **Mode** as **Peer-to-Peer**, you must select the desired option in **Allowed IP Address for Incoming SIP Message** from the following.

- **As per Trusted IP Address table:** If you select this option, the system matches the IP Address: Port received in the INVITE message (Source IP address from the Network layer and Source Port from the Transport layer) with the entries configured in the Trusted IP Address Table. If a match is found, the call will be routed to the desired destination. Else the call will be rejected.

You must configure the **Trusted IP Address Table** to receive incoming calls on this SIP Trunk. If you do not configure this table, incoming calls on this SIP Trunk will be rejected.

You can configure maximum 10 entries in the Trusted IP Address table.

To do so, click **Settings**  .





SIP Network Profile 1

SIP Network Profile ☒ Enable

Name

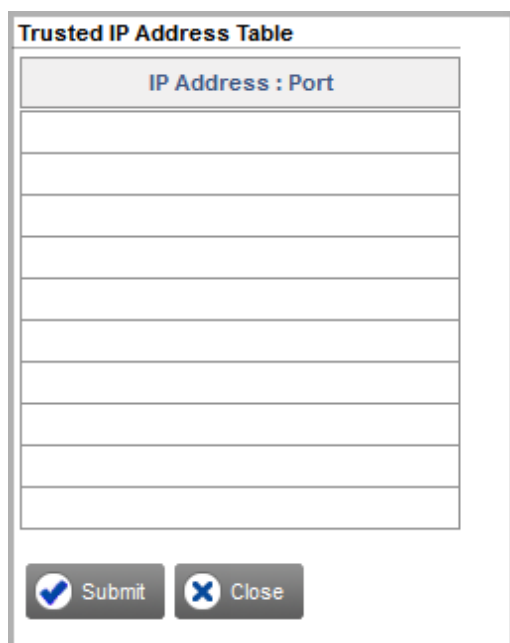
Status

Mode ☐ Proxy ☒ Peer-to-Peer 

Allowed IP Address for Incoming SIP Message   **Make sure to enter IP address in Trusted IP Address table for allowing Incoming call**



Digest Authentication ☐ Apply

The **Trusted IP Address Table** opens in a new window.



**Trusted IP Address Table**

IP Address : Port

 Submit  Close

- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls on this SIP Trunk. You can configure maximum 21 characters. Allowed characters are **0-9**, **dot** (.), **colon** (:).

Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit** and close the window.
- **As per Peer-to-Peer table:** If you select this option, the system matches the **IP Address: Port** received in the INVITE message (Source IP address from the Network layer and Source Port from the Transport layer) with the Destination Address configured in the Peer to Peer table. If a match is found, then the call will be routed to the desired destination. Else the call will be rejected.
- **Any:** If you select this option, **Digest Authentication** will be enabled automatically. The system will allow incoming calls only after the callers authenticate themselves with the correct credentials—User ID and Password. The system matches the User ID and Password entered by the callers with the entries stored in the Digest Authentication table. If a match is found, the call will be routed to the desired destination. Else the call will be rejected.

Default: **As per Peer-to-Peer table**.

## Digest Authentication

If you have selected the SIP Trunk **Mode** as **Peer-to-Peer**,

- you may also enable the **Digest Authentication**, if you set *Allowed IP Address for Incoming SIP Message* to *As per Trusted IP Address table* or *As per Peer to Peer table*. Digest Authentication is enabled automatically, if you set *Allowed IP Address for Incoming SIP Message* to *Any*.

If you enable Digest Authentication, you must also configure the Digest Authentication Table. Incoming calls on this SIP Trunk will be allowed only after the callers authenticate themselves with their User ID and Password. Default: Disabled. See [“Digest Authentication”](#) for more details.

## SIP Trunk Mode: Proxy

If you have selected the SIP Trunk **Mode** as **Proxy**, you must configure the following.

- Registrar Settings
- Redundancy Settings
- Codec
- VoIP Profile
- Advanced
- Timers

**Network Profile 1**

**SIP Network Profile 1**

<b>SIP Network Profile</b>	<input checked="" type="checkbox"/> Enable
<b>Name</b>	<input type="text" value="Network Profile 1"/>
<b>Status</b>	<input type="text" value="Enabled"/>
<b>Mode</b>	<input checked="" type="radio"/> Proxy <input type="radio"/> Peer-to-Peer

+ Registrar Settings

+ Redundancy Settings

+ Codec

+ VoIP Profile

+ Advanced

+ Timers

Submit

Default

Add New Profile

## Registrar Settings

If you have selected the SIP Trunk Mode as **Proxy**, you must configure the Registrar Settings.

- Click **Registrar Settings** to expand.

**Registrar Settings**

<b>Registrar Server Address:Port</b>	<input type="text" value=""/> : <input type="text" value="5060"/>
<b>Outbound Proxy</b>	<input type="checkbox"/> Enable
<b>Add 'rinstance' in REGISTER</b>	<input checked="" type="checkbox"/> Yes
<b>Allowed IP Address for Incoming SIP Message</b>	As per Trusted IP Address table ▼
<b>Digest Authentication</b>	<input type="checkbox"/> Apply
<b>Check SIP ID for Incoming SIP Message</b>	<input checked="" type="checkbox"/> Yes
<b>Check Proxy Address for Incoming SIP Message</b>	<input checked="" type="checkbox"/> Yes
<b>Check Proxy Port for Incoming SIP Message</b>	<input checked="" type="checkbox"/> Yes
<b>DNS SRV</b>	<input type="checkbox"/> Enable

- In **Registrar Server Address: Port**, enter the Registrar Server Address and the Registrar Server's listening port for SIP messages. The registrar server address may be an IP address or a domain. The Registrar Server Address can be of maximum 64 characters. Valid range is 1025 to 65534. Default: 5060.

- If your Service Provider uses outbound proxy for handling voice calls, select the **Outbound Proxy** check box. Default: Disabled.
- In **Outbound Proxy Server Address: Port**, enter the Outbound Proxy Server's IP Address and the Outbound Proxy Server's Listening Port for SIP. The Outbound Proxy Server Address may be of maximum 64 characters. Valid range is 1025 to 65534. Default: 5060.
- To add 'rinstance' in REGISTER message, keep the **Add 'rinstance' in REGISTER** check box enabled.

'rinstance' is any random value which can be used by SETU VGRX to fetch its own contact binding, that is, to know the Registration Expiry Timer assigned by the server.

When you enable 'rinstance' in Register, SETU VGRX will generate any random value of 'rinstance' and include it in the REGISTER message. The system will use the registration expiry timer of that contact binding.

- By default, the **Allowed IP Address for Incoming SIP Message** is set to **As per Trusted IP Address table** for Proxy SIP Trunk and is non-programmable. You must configure the **Trusted IP Address** table to allow incoming calls from specific IP addresses on this SIP Trunk.

Trusted IP Address table stores upto 13 entries, from which last three entries are uneditable. The last three entries in the table will display the *Registrar Server Address:Port* or *Outbound Proxy Address:Port* and *Fallback Registrar Server Address:Port1* and 2 or *Fallback Outbound Proxy Server Address:Port1* and 2, if configured for this SIP Trunk. If you do not configure the Trusted IP Address table, incoming calls will be allowed from the last three IP Addresses only.

To configure Trusted IP Address table, click **Settings**  .

The **Trusted IP Address Table** opens in a new window.

IP Address : Port

- Enter the **IP Address** and the corresponding **Port** from which you want to allow incoming calls on this SIP Trunk. You can configure maximum 21 characters. Allowed characters are **0-9**, **dot** (.), **colon** (:).

Do not configure the port, if you want to allow incoming calls from all the ports for a particular IP Address.

- Click **Submit** and close the window.
- If you want to allow incoming calls on this SIP Trunk only after the callers authenticate themselves with their User ID and Password, enable **Digest Authentication**. Default: Disabled.

If you enable Digest Authentication, you must also configure the Digest Authentication Table. See ["Digest Authentication"](#) for more details.

- Keep the **Check SIP ID for Incoming SIP Message** check box enabled, if you want SETU VGRX to validate the SIP ID during an incoming call. Default: Enabled.
- Keep the **Check Proxy Address for Incoming SIP Message** check box enabled, if you want SETU VGRX to validate the Proxy Address during an incoming call. Default: Enabled.
- Keep the **Check Proxy Port for Incoming SIP Message** check box enabled, if you want SETU VGRX to validate the Proxy Port during an incoming call. Default: Enabled.
- If you want the system to send DNS SRV query to the configured domain server, enable **DNS SRV**. When disabled, the system will send DNS A query to the configured domain server. Default: Disabled.



If you enable DNS SRV, Fallback Server logic will not be applicable.

## Redundancy Settings

If you have selected the SIP Trunk Mode as **Proxy**, configure the Redundancy Settings.

- Click **Redundancy Settings** to expand.

Redundancy Settings	
Fallback Server	<input type="checkbox"/> Yes
Fallback Event	503 or No Response
No Response Timer	20 Seconds
Registration Behavior	Register with only one Server
Switch Registration to Alternate Server on Fallback	<input checked="" type="checkbox"/> Yes
Load Balancing	Last Call Active

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

If you have enabled Fallback Server and Outbound Proxy is disabled,

- In the **Fallback Registrar Server Address 1: Port** and **Fallback Registrar Server Address 2: Port**, enter 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 65534. Default: 5060.

If you have enabled Fallback Server and Outbound Proxy is enabled,

- In **Fallback Outbound Proxy Server Address 1: Port** and **Fallback Outbound Proxy Server Address 2: Port**, enter 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 65534. Default: 5060.
- In **Fallback Event**, select the event on occurrence of which SETU VGRX should fallback to an alternate Registrar/Outbound Proxy Server, if available.
  - No Response
  - 503 or No Response
  - 5xx or No ResponseDefault: 503 or No Response

In case, the Fallback Server does not respond and the call is not routed to the destination port, the call will be routed to another port type as per the Routing/Fallback Routing Group configured for the SIP Trunk.

- Set the duration of the **No Response Timer**. This timer defines the time period for which SETU VGRX will wait for the response from the server for any request. If no valid response is received before the expiry of this timer, SETU VGRX 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.



*If the SIP General Request Timer configured in the System Parameters is less than the No Response Timer, then SETU VGRX will fallback to alternate Registrar/Outbound Proxy Server or Routing Group/Fallback Routing Group on the expiry of the SIP General Request Timer and the No Response Timer will stop.*

- In the **Registration Behavior**, select the desired option:
  - Register with all Servers
  - Register with only one Server

If you select **Register with only one Server**, SETU VGRX will get registered with the Registrar/Outbound Proxy Server. If registration with the Registrar/Outbound Proxy Server fails, it will get registered with Fallback Registrar/Outbound Proxy Server 1 or Fallback Registrar/Outbound Proxy Server 2 respectively for further processing of call.

If you select **Register with all Servers**, SETU VGRX will get registered with Registrar/Outbound Proxy Server as well as Fallback Registrar/Outbound Proxy Servers. It will not apply Fallback logic even if *Fallback Server* is enabled.



*The **Registration Behavior** will be applicable only if, **SIP Registration** is enabled for the SIP Trunk.*

- Keep the **Switch Registration to Alternate Server on Fallback** check box enabled. SETU VGRX will get unregistered with the current server and will register with the alternate server, if fallback occurs while sending the INVITE message.



*The **Switch Registration to Alternate Server on Fallback** will be applicable only if, **SIP Registration** is enabled for the SIP Trunk and **Registration Behavior** is set as **Register with only one Server**.*

- Select the desired option for **Load Balancing** from the following:
  - **Last Call Active:** Each new call will be processed through the Registrar/Outbound Proxy Server through which the last active call has been processed.  
  
For example, if the last call has been processed by Fallback Registrar/Outbound Proxy Server 2, the new call will also be processed through Fallback Registrar/Outbound Proxy Server 2 only.
  - **First Active:** Each new call will be processed through the first active Registrar/Outbound Proxy Server only.
  - **Cyclic:** Each new call will be processed through the next active Registrar/Outbound Proxy Server.

For example, if the last call has been processed by Fallback Registrar/Outbound Proxy Server 1, the new call will be processed through Fallback Registrar/Outbound Proxy Server 2 and the subsequent new call will be processed through the Registrar/Outbound Proxy Server 1.

Default: Last Call Active.

## Codec Profile

You must configure the Codec Profile, whether you have selected the SIP Trunk Mode as **Proxy** or **Peer-to-Peer**.

- Click **Codec** to expand.

The screenshot shows a window titled "Codec" with two main sections: "Available Codecs Profiles" and "Selected Codecs Profiles".

**Available Codecs Profiles:** This section is currently empty.

**Selected Codecs Profiles:** This section contains a list of five codec profiles:

- G.729 - 20msec - Silencesupp=off
- G.723 - 30msec - Silencesupp=off
- GSM FR - 20msec - Silencesupp=off
- G.711 (u-law) - 20msec - Silencesupp=off
- G.711 (A-law) - 20msec - Silencesupp=off

Between the two lists are two arrows: a right-pointing arrow (to move from available to selected) and a left-pointing arrow (to move from selected to available).

Below the lists are four configuration options:


- Silence Suppression in SDP for G.711 codec:** A dropdown menu set to "Send using Silence Suppression attribute".
- Comfort Noise (CN):** A checkbox labeled "Yes" (unchecked).
- Include pttime header in SDP:** A checkbox labeled "Yes" (unchecked).
- Send Re-INVITE when multiple codec is received in 200(O.K):** A checkbox labeled "Yes" (checked).

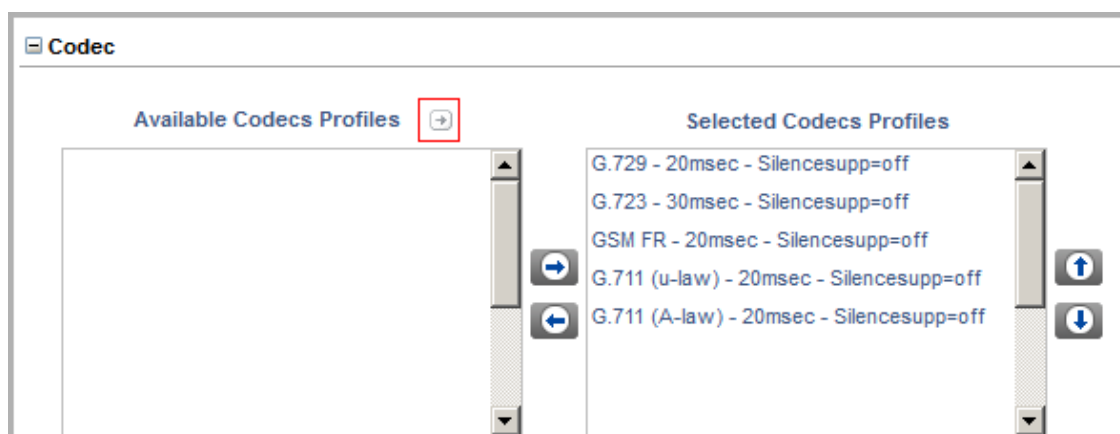
Codecs are used to compress the data in RTP packets to enable quick transmission. It also decompresses the received data.

The codec profiles supported by SETU VGRX appears in the **Selected Codecs Profiles** list in the following order of preference:

1. G.729 - 20msec - Silencesupp=off
2. G.723 - 30msec - Silencesupp=off
3. GSM FR - 20msec - Silencesupp=off
4. G.711 (u-law) - 20msec - Silencesupp=off
5. G.711 (A-law) - 20msec - Silencesupp=off

- You can change the order of preference by moving the desired Codecs up or down the list. To move a Codec up or down the list, do the following:
  - In the **Selected Codecs Profiles** list, click the Codec you want to move.
  - Click the UP/DOWN ARROW to move the Codec to the desired position in the list.
- To remove a Codec from the **Selected Codecs** list, click the Codec you want to remove, and then click the LEFT ARROW. The Codec is moved to the **Available Codecs Profiles** list.
- To move a Codec from the **Available Codecs Profiles** list to the **Selected Codecs Profiles** list, click the Codec you want to move, and then click the RIGHT ARROW.

- You can edit any existing Codec profile or add a new profile. To do so, click **Settings**  .



- The **Codec Profile** window opens. For detailed instructions, see [“Codec Profile”](#).
- Select the desired **Silence Suppression in SDP for G.711 codec** option. SETU VGRX suppresses the *Silence* packets and allows only the *Voice* packets to pass through.

This is used to deactivate certain processes during non-speech section of an audio session to avoid unnecessary coding/ transmission of silence packets in VoIP application. Hence, it results in saving on computation and network bandwidth.

You can select either *Send using VAD attribute* or *Send using Silence Suppression attribute*.

If you select *Send using VAD attribute*, SETU VGRX will send *VAD=NO* in the SDP offer / answer exchanges.

If you select *Send using Silence Suppression attribute*, SETU VGRX will send *Silence Suppression=OFF* in the SDP offer / answer exchanges.

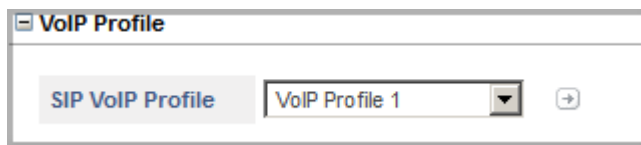
Default: Send using Silence Suppression attribute.


- Select the **Comfort Noise (CN)** check box, if you want SETU VGRX to negotiate the Comfort Noise received in the SDP body with the remote peer. Default: Disabled.
- Select the **Includeptime header in SDP** check box, if you want SETU VGRX to addptime header in the SDP offer / answer exchanges. Default: Disabled.
- Clear the **Send Re-INVITE when multiple codec is received in 200 (OK)** check box, if you do not want SETU VGRX to send Re-INVITE message and use only the first codec from the multiple codecs received in 200 (OK). Default: Enabled.

## VoIP Profile

You must configure the VoIP Profile, whether you have selected the SIP Trunk Mode as **Proxy** or **Peer-to-Peer**.

- Click **VoIP Profile**.



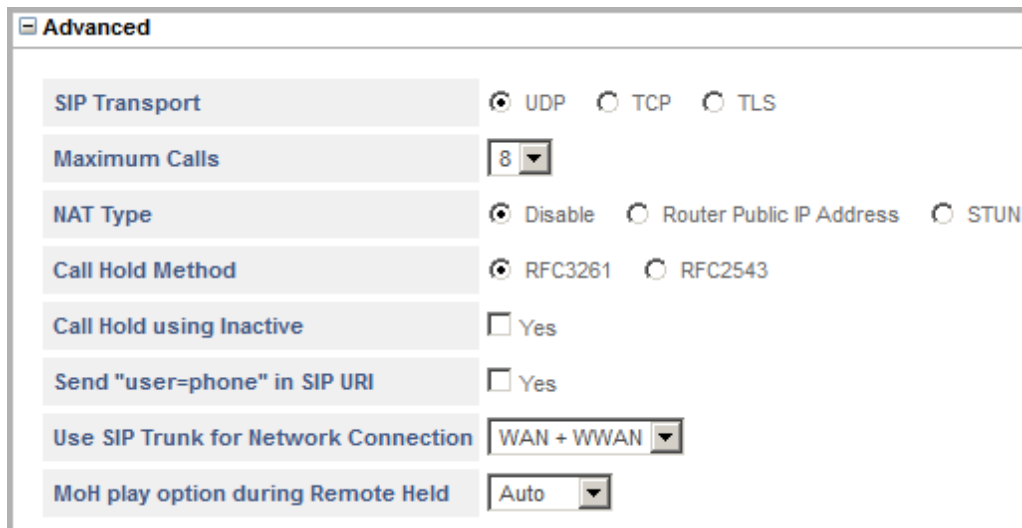
- In **SIP VoIP Profile**, you can either select the default **VoIP Profile 1** or **Add New VoIP Profile** option.
- Click **Settings**  to configure the parameters of the selected VoIP Profile.

For detailed instructions, see [“SIP VoIP Profile”](#).

## Advanced

You must configure Advanced settings, whether you have selected the SIP Trunk Mode as **Proxy** or **Peer-to-Peer**.

- Click **Advanced**.



- Select the default **SIP Transport** for outgoing SIP messages from the following options:
  - **UDP**: Outgoing messages are transported using UDP.
  - **TCP**: Outgoing messages are transported using TCP.
    - **TCP Fallback to UDP**: When TCP is used for outgoing messages, you may select this checkbox. If the TCP connection fails, the system will attempt to send the outgoing message again over UDP.
  - **TLS**: Outgoing messages are transported using TLS.

Default: UDP



To use TCP or TCP Fallback to UDP, you must enable **SIP over TCP** in the [“System Parameters”](#).

To use TLS, you must enable **SIP over TLS** in the [“System Parameters”](#).

- In **Maximum Calls**, configure the number of simultaneously calls you want to allow on the SIP Trunk. Default: 8.

The number of simultaneous SIP calls depends on the number of simultaneous calls allowed by the ITSP. SETU VGRX supports maximum 8 simultaneous calls.

- When the system is installed behind a NAT Router, select specific NAT traversal mechanism to be used as **NAT Type**. Default: Disabled.

- Select **Router Public IP Address**, if your SETU VGRX is located behind the NAT router (any type).

Make sure you disable Outbound Proxy on the SIP Trunk and you have configured the same IP Address under NAT settings in the [“System Parameters”](#) page.

- Select **STUN**, if your system is located behind the NAT router other than Symmetric.

Make sure you disable Outbound Proxy on the SIP Trunk and you have configured the STUN Server Address and port under NAT settings in the [“System Parameters”](#) page.

- In **Call Hold Method**, select the desired option — RFC 2543 or RFC 3261— that is compatible with your ITSP proxy server / remote peer. Default: RFC 3261
- Select the **Call Hold using Inactive** check box, if you want the system to send ‘*a=inactive*’ message instead of ‘*a=sendonly*’ message on the SIP Trunk, when the user puts the call on hold. Default: Disabled.
- Select the **Send "user=phone" in SIP URI** check box, if you want SETU VGRX to add user=phone in the Request URI / From / To header of the INVITE message. Default: Disabled.

SETU VGRX will send user=phone in SIP URI, only if the SIP ID is numeric.

- Select the desired option to **Use SIP Trunk for Network Connection**. You may select:
  - WAN
  - WWAN
  - WAN+WWAN

Default: WAN+WWAN

- Select the desired **MoH play option during Remote Held**. You may select:
  - Auto
  - Local
  - Remote

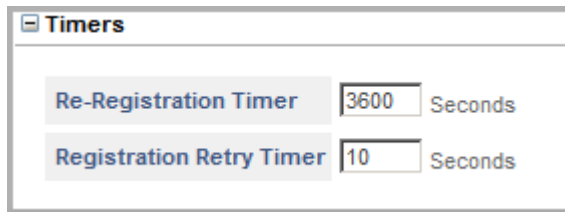
Default: Auto

- Click **Submit** to save the changes.

## Timers

If you have selected the SIP Trunk Mode as **Proxy**, configure the Timers.

- Click **Timers**.



The screenshot shows a window titled "Timers" with two configuration rows. The first row is "Re-Registration Timer" with a text input field containing "3600" and the unit "Seconds". The second row is "Registration Retry Timer" with a text input field containing "10" and the unit "Seconds".

- Set the duration of the **Re-registration Timer**. This is the time period after which the SETU VGRX will send registration request to maintain registration binding with the Registrar Server. Valid range is 00001 to 65535 seconds. Default: 3600 seconds.
- Set the duration of the **Registration Retry Timer**. When a registration attempt fails, SETU VGRX will resend registration request to the Registrar Server after the expiry of the Re-registration Timer. Valid range is 00001 to 65535. Default: 10 seconds.



*The Re-registration and Registration Retry timers will be applicable only if, **SIP Registration** is enabled for the **SIP Trunk**.*

To add a new profile, click the **Add New Profile** button at the bottom of the page. **SIP Network Profile 2** page opens. Follow the same instructions as given earlier to configure it.

# SIP VoIP Profile

You can either edit the settings of the default **VoIP Profile 1** or add a new profile as per your requirement.

To add a new profile,

- Click the **Advanced Settings** link.
- Click the **SIP Profile** link to expand.
- Click the **SIP VoIP Profile** link.
- By default, **VoIP Profile 1** page opens.

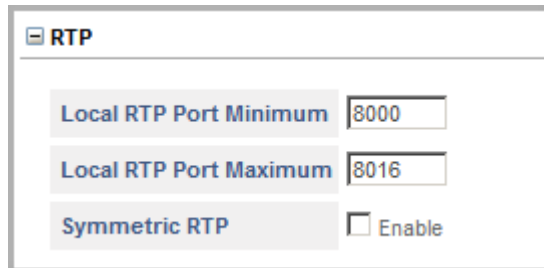
The screenshot shows the Matrix SETU VGRX web interface. On the left is a sidebar with a tree view of settings. Under 'Advanced Settings', the 'SIP Profile' section is expanded, and 'SIP VoIP Profile' is selected. The main content area is titled 'VoIP Profile 1'. It contains a section 'SIP VoIP Profile 1' with a checkbox labeled 'SIP VoIP Profile' and 'Enable' (checked). Below this is a text field labeled 'Name' with the value 'VoIP Profile 1'. Further down are several expandable sections: RTP, SRTP, DTMF, FAX, T.38 FAX Parameters, and Pass-Through FAX Parameters. At the bottom of the main area are three buttons: 'Submit' (with a checkmark icon), 'Default' (with a circular arrow icon), and 'Add New Profile' (with a plus icon).

- Configure the following parameters:
  - Keep the **SIP VoIP Profile** check box enabled to use this profile.  
Clear this check box, if you do not want to use this profile.
  - Assign a **Name** to the VoIP Profile for identification. The Name can be a maximum of 24 characters.

## RTP

**RTP Port** is the port on which the SETU VGRX listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer.

- Click **RTP** to expand.



**RTP**

Local RTP Port Minimum

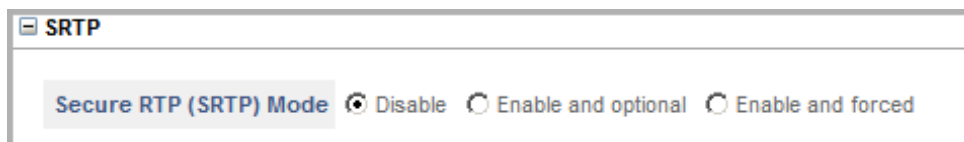
Local RTP Port Maximum

Symmetric RTP ☐ Enable

- **Local RTP Port Minimum:** Enter the desired minimum RTP Port Number. Valid range is 1032 to 65535. Default: 8000.
- **Local RTP Port Maximum:** Enter the desired maximum RTP Port Number. Valid range is 1032 to 65535. Default: 8720.
- **Symmetric RTP:** Enable the **Symmetric RTP** check box, if you want the system to send RTP packets to the original IP address and Port from where RTP packets are received. Default: Disabled.

## SRTP

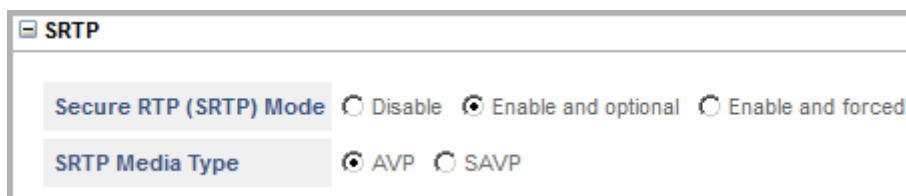
- Click **SRTP** to expand.



**SRTP**

Secure RTP (SRTP) Mode ☒ Disable ☐ Enable and optional ☐ Enable and forced

- For secure conversations over SIP, SETU VGRX supports the **Secure RTP (SRTP)** mode. Select the desired option as per your requirement:
  - **Disable:** Select this option, if you want SETU VGRX to use normal RTP instead of SRTP for transporting the speech packets.
  - **Enable and optional:** Select this option, if you want SETU VGRX to use SRTP for transporting the speech packets. If the remote user does not support SRTP, normal RTP will be used.
    - If you select this option, you must configure the **SRTP Media Type**. You may select *AVP* or *SAVP*. Default: AVP.



**SRTP**

Secure RTP (SRTP) Mode ☐ Disable ☒ Enable and optional ☐ Enable and forced

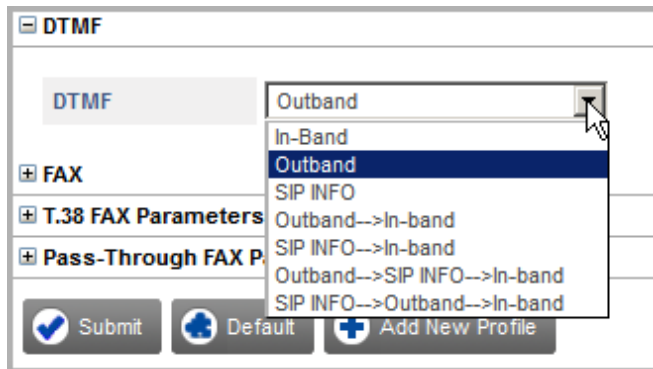
SRTP Media Type ☒ AVP ☐ SAVP

- **Enable and forced:** Select this option, if you want SETU VGRX to use only SRTP (SAVP) for transporting the speech packets. If the remote user does not support SRTP, SETU VGRX will reject the incoming calls from such users and will drop the outgoing calls made to such users.

Default: Disabled.

## DTMF

- Click **DTMF** to expand.



- Select the appropriate **DTMF** sending/ receiving mechanism that is compatible with the DTMF sending/ receiving mechanism of your ITSP or remote peer. SETU VGRX supports:
  - **In-band:** System will send and detect digits in In-band only.
  - **Outband:** System will send and detect digits in Outband events only.
  - **SIP INFO:** System will send and detect digits in SIP INFO message only.
  - **Outband-->In-band:** System will send and detect digits in Outband, if negotiated in offer/ answer else it will use In-band.
  - **SIP INFO-->In-band:** System will send and detect digits in SIP INFO, if negotiated in offer/ answer else it will use In-band.
  - **Outband-->SIP INFO-->In-band:** System will send and detect digits in Outband or SIP INFO, if negotiated in offer/ answer else it will use In-band. If both Outband and SIP INFO are negotiated, Outband will have priority over SIP INFO.
  - **SIP INFO-->Outband-->In-band:** System will send and detect digits in SIP INFO or Outband, if negotiated in offer/ answer else it will use In-band. If both SIP INFO and Outband are negotiated, SIP INFO will have priority over Outband. Default: Outband.

Default: Outband.

## FAX

- Click **FAX** to expand.

**FAX**

Convert FAX call to Speech call when FAX is complete ☐ Yes

Use FAX Protocol configured for Outgoing FAX ☐ Yes

- Select the **Convert FAX call to Speech call when FAX is complete** check box, if you want SETU VGRX to convert the fax call to a speech (voice) call after the fax complete event is received. Default: Disabled.
- Select the **Use FAX Protocol configured for Outgoing FAX** check box, if you want SETU VGRX to use the Fax Protocol configured for outgoing fax for the SIP Trunk and not the one that is received in REINVITE message from the remote end. Default: Disabled.

## T.38 FAX Parameters

- Click **T.38 FAX Parameters** to expand.

**T.38 FAX Parameters**

Type ☒ UDPTL ☐ RTP

Version

- Select the **Type** of Fax Protocol— UDPTL or RTP — that is compatible with your ITSP proxy server/ remote peer. Default: UDPTL.
- Select a compatible **Version**— 0, 1, 2 — as supported by your ITSP proxy server/ remote peer for the T.38 Fax Type. Default: 0.

## Pass-through FAX Parameters

- Click **Pass-through FAX Parameters** to expand.

**Pass Through FAX Parameters**

Passthrough FAX Codec

- Select an appropriate **Passthrough FAX Codec** — G.711(μ-law) or G.711 (A-law) — that is compatible with your ITSP proxy server/remote peer. Default: G.711 (μ-law).

To add a new profile, click the **Add New Profile** button at the bottom of the page. The **VoIP Profile 2** page opens. Follow the same instructions as given earlier to configure it.

# Codec Profile

You can either edit the settings of the existing **Codec Profiles** (1 to 5) or add a new profile as per your requirement.

To add a new profile,

- Click the **Advanced Settings** link.
- Click the **SIP Profile** link to expand.
- Click the **Codec Profile** link.

The **Codec Profile** page opens.

Codec Profile	Enable	Codec	p-time	Silence Suppression
<a href="#">1</a>	<input checked="" type="checkbox"/>	G.729	20	<input type="checkbox"/>
<a href="#">2</a>	<input checked="" type="checkbox"/>	G.723 (6.3 Kbps)	30	<input type="checkbox"/>
<a href="#">3</a>	<input checked="" type="checkbox"/>	GSM FR	20	<input type="checkbox"/>
<a href="#">4</a>	<input checked="" type="checkbox"/>	G.711 (u-law)	20	<input type="checkbox"/>
<a href="#">5</a>	<input checked="" type="checkbox"/>	G.711 (A-law)	20	<input type="checkbox"/>

- The **Codec Profile** page displays the following:
  - **Codec Profile:** It displays the Codec Profile numbers. To configure the Codec Profile Parameters, click on the desired Codec Profile number link.
  - **Enable:** Keep this check box enabled to use the Codec Profile.
  - **Codec:** It displays the Codec assigned to each of the Codec Profiles.
  - **p-time:** It displays the p-time selected for each of the Codec Profiles.
  - **Silence Suppression:** It displays whether the Silence Suppression for the respective Codec Profile is enabled or not.

- Click the **Add New Profile** button at the bottom of the page to add a new profile.

A new Codec Profile page opens.

**Codec Profile 6**

Enable ☒ Yes

Codec G.711 (A-law)

p-time 20

Silence Suppression ☐ Enable

- By default, the Codec Profile is enabled.

Clear this check box, if you do not want to use this profile.

- The **Codecs** are fixed for the **Codec Profile 1** to **Codec Profile 5**. You can assign the desired **Codec** — G. 729, G.723, GSM FR, G.711 (u - law), G.711 (A - law) — to a new Codec Profile. Default: G.711 (A-law).
- If you select G.723 codec, you must also select the **Bit Rate** — 5.3 Kbps or 6.3 Kbps. Default: 6.3kbps.

**Codec Profile 6**

Enable ☒ Yes

Codec G.723

p-time 30

Silence Suppression ☐ Enable

Bit Rate 6.3 Kbps

When G.723 is negotiated, the selected Bit Rate will be applied only when sending the RTP packets. While receiving the RTP packets from the remote end, both the Bit Rates of G.723 will be accepted.

- Select the desired **p-time** value. You can select from the following:
  - 10
  - 20
  - 30
  - 40

Default: 20



- *For the codec G.723, SETU VGRX will use p-time value as 30 only.*
- *You may select only 20 or 40 as p-time value for the codec GSM FR. Also Silence Suppression will not be applicable for the codec GSM FR.*
- *For Passthrough FAX, SETU VGRX will use the default ptime value 20 only.*
- For the codecs — G.729, G.723 and G.711 (u - Law) — enable the **Silence Suppression** check box, if you want SETU VGRX to suppress the Silence packets and allow only the Voice packets to pass through.  
Default: Disabled.

Similarly, you can add other Codec Profiles. You may also edit the existing **Codec Profiles** (1 to 5), if required. Click the desired Codec Profile number link to do so and follow the same instructions as given earlier.

# Destination Number Determination

---

The process of routing calls originated on FXS Ports, FXO Ports, Radio Ports, Mobile Ports and SIP Trunks to the destination port in SETU VGRX takes place in two steps:

- Determination of Destination Number
- Determination of Destination Port

SETU VGRX supports different methods of determining the destination number for the calls originated on the FXO Ports, Radio Ports, Mobile Ports and SIP Trunks.

## Destination Number Determination on Radio Ports

For Radio Ports, the system supports the following methods for Destination Number Determination:

- using Fixed Destination Number
- after collecting the digits

## Destination Number Determination on SIP Trunks

For SIP Trunks, the system supports the following methods for Destination Number Determination:

- without any Destination Number
- to a Fixed Destination Number
- on the basis of Calling Party Number
- on the basis of DDI Number
- to the Called Party Number

To apply Destination Number Determination **on the basis of Calling Party Number**, you must configure the **Destination Number Determination: SIP-Calling Number Based** table. When there is an incoming call on the SIP Trunk, SETU VGRX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination port.

To apply Destination Number Determination **on the basis of DDI Number**, you must configure the **Destination Number Determination: SIP-DDI Number Based** table. When there is an incoming call on the SIP Trunk, SETU VGRX will match the DDI Number received in the SIP INVITE message with the entries of the DDI Number Based Table. If a match is found, the call is routed to the destination port.

## Destination Number Determination on Mobile Ports

For Mobile Ports, the system supports the following methods for Destination Number Determination:

- without any Destination Number
- to the Fixed Destination Number
- on the basis of Calling Party Number
- after Answering the Call and Collecting the Digits

To apply Destination Number Determination **on the basis of Calling Party Number**, you must configure the **Destination Number Determination: Mobile-Calling Number Based** table. When there is an incoming call on the Mobile Port, SETU VGRX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination port.

## Destination Number Determination on FXO Ports

For FXO Ports, the system supports the following methods for Destination Number Determination:

- without any Destination Number
- to the Fixed Destination Number
- on the basis of Calling Party Number
- after Answering the Call and Collecting the Digits

To apply Destination Number Determination **on the basis of Calling Party Number**, you must configure the **Destination Number Determination: FXO-Calling Number Based** table. When there is an incoming call on the FXO Port, SETU VGRX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call is routed to the destination port.

## Configuring SIP-Calling Number Based Table

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Destination Number Determination** link to expand.
- Click the **SIP - Calling Number Based** link. The **SIP Trunk - Destination Number Determination: Calling Number Based** page opens.

1-100101-200201-300301-400401-499

**SIP Trunk - Destination Number Determination: Calling Number Based**

Index	Calling Number	Destination Number
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		
013		
014		
015		

Submit

Default All

Configure following parameters in this table:

- In **Calling Number**, enter the calling party numbers. Calling numbers may consist of a maximum of 24 characters. All ASCII characters are allowed. Default: Blank.
- In **Destination Number**, enter a corresponding destination number for each calling party number. Destination numbers may consist of a maximum of 24 characters. Characters 0-9, \*, # and dot (.) are allowed. Default: Blank.
- Click **Submit** to save the entries.
- Click **Default All** to clear all the entries.
- To configure the table for the FXO Port, click the **FXO-Calling Number Based** link.
- To configure the table for the Mobile Port, click the **Mobile-Calling Number Based** link.
- **Mobile-Calling Number Based** Table has an additional parameter — **Allow Callback?**. Select the check boxes of the calling numbers for which you want to set callback. When there is an incoming call from this calling number, the system disconnects the call and will automatically initiate the call to the calling number within 2 to 5 seconds.

## Configuring SIP-DDI Number Based Table

To configure the DDI Number Based Table,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Destination Number Determination** link.
- Click the **SIP-DDI Number Based** link. The **SIP Trunk - Destination Number Determination: DDI Number Based** page opens.

DDI Number Generation

SIP Trunk - Destination Number Determination: DDI Number Based

Index	DDI Number	Destination Number	Reverse DDI	
			Apply	Reference ID
001			<input type="checkbox"/>	1
002			<input type="checkbox"/>	1
003			<input type="checkbox"/>	1
004			<input type="checkbox"/>	1
005			<input type="checkbox"/>	1
006			<input type="checkbox"/>	1
007			<input type="checkbox"/>	1
008			<input type="checkbox"/>	1
009			<input type="checkbox"/>	1
010			<input type="checkbox"/>	1
011			<input type="checkbox"/>	1
012			<input type="checkbox"/>	1

☒ Submit
 ☒ Default All

- There are two ways to generate the DDI Numbers:
  - Using the **DDI Number Generation** Button to automatically generate the DDI Number Table.
  - OR**
  - Entering each DDI Number manually.
- If you want to generate DDI Numbers automatically, click the **DDI Number Generation** button and configure the following parameters:

DDI Numbers Generation

Total DDI Numbers

Enter Start Index Number

Enter Start DDI Number

Enter Start Destination Number

Apply Reverse DDI (for all DDI Numbers) ☐

Enter Reverse DDI Reference ID (for all DDI Numbers)

☒ Apply
 ☒ Close

- Total DDI Numbers:** The DDI numbers are allotted by the service provider. You must enter the total number of DDI numbers you want to generate in the DDI Number Based table. You can generate upto 100 numbers. Default: 10

- **Enter Start Index Number:** Enter the desired Index Number from where you want to start the DDI Number generation. Default: 1
- **Enter Start DDI Number:** Enter the start DDI Number. DDI Number can be of maximum 24 characters. Characters 0-9, +, \* and # are allowed.
- **Enter Start Destination Number:** Each DDI Number can be assigned a corresponding destination number. Enter the Start Destination Number corresponding to the Start DDI Number. Destination Number can be 24 characters long. Characters 0 to 9, # and \* are allowed.
- **Apply Reverse DDI (for all DDI Numbers):** When the user makes a call from the assigned DDI number, this number will be displayed to the called party. Select the check box to apply Reverse DDI logic on all DDI Numbers.
- Click **Apply** button to generate the table. The DDI numbers generated will appear in the DDI Number Based Table.

DDI Number Generation

**SIP Trunk - Destination Number Determination: DDI Number Based**

Index	DDI Number	Destination Number	Reverse DDI	
			Apply	Reference ID
001	2630555	2001	<input checked="" type="checkbox"/>	1
002	2630556	2002	<input checked="" type="checkbox"/>	1
003	2630557	2003	<input checked="" type="checkbox"/>	1
004	2630558	2004	<input checked="" type="checkbox"/>	1
005	2630559	2005	<input checked="" type="checkbox"/>	1
006	2630560	2006	<input checked="" type="checkbox"/>	1
007	2630561	2007	<input checked="" type="checkbox"/>	1
008	2630562	2008	<input checked="" type="checkbox"/>	1
009	2630563	2009	<input checked="" type="checkbox"/>	1
010	2630564	2010	<input checked="" type="checkbox"/>	1
011			<input type="checkbox"/>	1
012			<input type="checkbox"/>	1

☒ Submit
 ☐ Default All

- You can also edit the generated numbers, if required.
- If you want to generate DDI Numbers manually,
  - Enter each DDI Number and its corresponding Destination Number against the desired Index in the table.
- To apply **Reverse DDI** logic on the DDI Number,

- Select the **Apply** check box.
- Select the corresponding **Reference ID**.

For detailed instruction for generating DDI Numbers manually, see [“Route on the basis of DDI Number”](#) under SIP Trunks.

- Click **Submit** to save your entries.
- Click **Default All** to clear all the entries.

# Destination Port Determination

---

The process of routing calls originated on FXS Ports, Radio Ports, FXO Ports, Mobile Ports and SIP Trunks to the destination port in SETU VGRX takes place in two steps:

- Determination of Destination Number
- Determination of Destination Port

SETU VGRX supports different methods of determining the destination port for the calls originated on FXS Ports, Radio Ports, FXO Ports, Mobile Ports and SIP Trunks.

## Destination Port Determination on FXS Ports

For FXS Port, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **FXS Port - Destination Port Determination - Destination Number Based** table.

## Destination Port Determination on Radio Ports

For Radio Port, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **Radio - Destination Port Determination - Destination Number Based** table.

## Destination Port Determination on FXO Ports

For FXO Ports, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **FXO Port - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **FXO Port - Destination Port Determination - Destination Number Based** table.

## Destination Port Determination on Mobile Ports

For Mobile Port, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number

To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **Mobile Port - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **Mobile Port - Destination Port Determination - Destination Number Based** table.

## Destination Port Determination on SIP Trunks

For SIP Trunks, the system supports the following methods for Destination Port Determination:

- Fixed
- On the basis of Destination Number
- On the basis of Calling Party Number



To apply Destination Port Determination **on the basis of Calling Party Number**, you must configure the **SIP Trunk - Destination Port Determination - Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **SIP Trunk - Destination Port Determination - Destination Number Based** table.


## Configuring SIP Trunk - Calling Number Based Table


- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Destination Port Determination** link to expand.
- Click the **SIP - Calling Number Based** link. The **SIP Trunk - Destination Port Determination - Calling Number Based** page opens.

SIP Trunk - Destination Port Determination - Calling Number Based

	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	FXS Port 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 11

 Add

 Delete

- To add an entry, click **Add**. A new window opens.

**Add Entry**

Calling Number

CLI Number to be sent on Destination Port Received Calling Party ▼

**Routing Group**

☒ **FXS Port** 1 ▼ to 1 ▼ in Ascending ▼ order

☐ **FXS Group** 1 ▼

☐ **FXO Port** 1 ▼ to 1 ▼ in Ascending ▼ order

☐ **FXO Group** 1 ▼

☐ **Mobile Port** 1 ▼ to 1 ▼ in Ascending ▼ order

☐ **Mobile Group** 1 ▼

☐ **Radio Port** 1 ▼ to 1 ▼ in Ascending ▼ order

**Fallback Routing Group** ☐ Apply

☐ **FXS Port** 1 ▼ to 1 ▼ in Ascending ▼ order

☐ **FXS Group** 1 ▼

☐ **FXO Port** 1 ▼ to 1 ▼ in Ascending ▼ order

☐ **FXO Group** 1 ▼

☐ **Mobile Port** 1 ▼ to 1 ▼ in Ascending ▼ order

☐ **Mobile Group** 1 ▼

☐ **Radio Port** 1 ▼ to 1 ▼ in Ascending ▼ order

Configure the following parameters:

- In **Calling Number**, enter the number (max. 24 characters) from which you expect calls to be received. Characters 0-9, \*, # and dot (.) are allowed. Default: Blank.
- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.




*CLI to be sent on Destination Port is applicable when FXS Port/ Group is determined as the Destination Port.*


- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.



Similarly, you can create a group of *sequential* FXO Ports, Mobile Ports, Radio Ports and SIP Trunks.

- To create a group of *not-sequential* **FXS Ports** as members,
  - Select a **FXS Group**.
  - Select **FXS Group** number. Default: 1.
  - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic “[Group](#)” under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential* FXO Ports, Mobile Ports and SIP Trunks.
- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **SIP Trunk - Destination Port Determination - Calling Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.
- By default, FXS Port 1-1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found) in the **SIP Trunk - Destination Port Determination - Calling Number Based** table.


To change the default Routing Group and to create the Fallback Routing Group for the No Match Found entry,


- Under Edit, click **Settings** .

SIP Trunk - Destination Port Determination - Calling Number Based

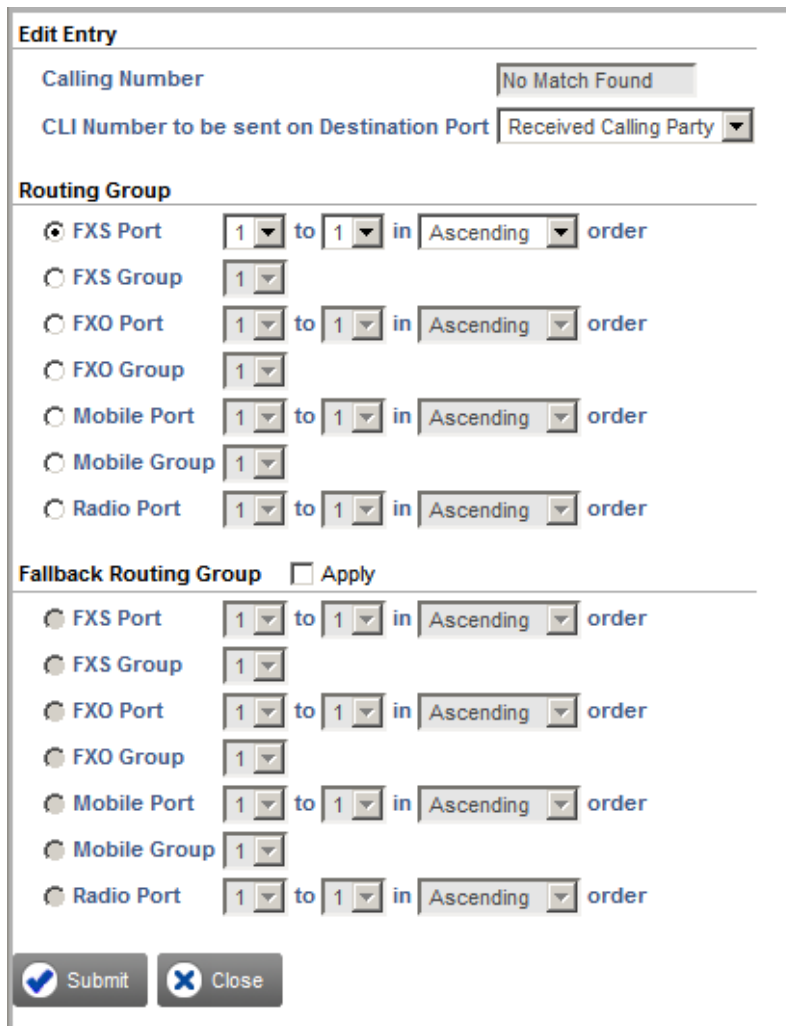
	Edit	Calling Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	FXS Port 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 11

 Add

 Delete

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/editing entries.

Similarly, you can:

- configure the **FXO Port - Destination Port Determination - Calling Number Based** table by clicking the **FXO - Calling Number Based** link.
- and
- configure the **Mobile Port - Destination Port Determination - Calling Number Based** table by clicking the **Mobile - Calling Number Based** link.

## Configuring SIP Trunk - Destination Number Based Table

- Log into Jeeves.

- Click the **Advanced Settings** link.
- Click the **Destination Port Determination** link to expand.
- Click the **SIP - Destination Number Based** link.

The **SIP Trunk - Destination Port Determination - Destination Number Based** page opens.

SIP Trunk - Destination Port Determination - Destination Number Based					
<input type="checkbox"/>	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
<input type="checkbox"/>		No Match Found	FXS Port 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1      1

**Testing**

Enter the destination number to know which entry would be selected for routing

- To add an entry, click **Add**. A new window opens.

**Add Entry**

Destination Number

CLI Number to be sent on Destination Port

**Routing Group**

☒ **FXS Port**  to  in  order

☐ **FXS Group**

☐ **FXO Port**  to  in  order

☐ **FXO Group**

☐ **Mobile Port**  to  in  order

☐ **Mobile Group**

☐ **Radio Port**  to  in  order

**Fallback Routing Group** ☐ Apply

☐ **FXS Port**  to  in  order

☐ **FXS Group**

☐ **FXO Port**  to  in  order

☐ **FXO Group**

☐ **Mobile Port**  to  in  order

☐ **Mobile Group**

☐ **Radio Port**  to  in  order

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + "Wildcard Characters"). Valid characters are 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

## Wildcard Characters

SETU VGRX supports following characters.


Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.
<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.
<b>[ ^ ]</b>	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
<b>T</b> (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

- Select the **CLI Number to be sent on Destination Port**. You can select Received Calling Party or Received Called Party. Default: Received Calling Party.



*CLI to be sent on Destination Port is applicable when FXS Port/ Group is determined as the Destination Port.*

- Create the **Routing Group**.
  - To create a group of *sequential FXS Ports* as members,
    - Select the desired **FXS Port** numbers as members. Default: 1.
    - In the **in - order** box, select the order in which the system should hunt for a free member FXS Port to route the call.

Select **Ascending** to start hunting from the first to the last member FXS Port. Select **Descending** to start hunting from the last to the first member FXS Port. Default: Ascending.
  - Similarly, you can create a group of *sequential FXO Ports*, *Mobile Ports*, *Radio Ports* and *SIP Trunks*.
  - To create a group of *not-sequential FXS Ports* as members,
    - Select a **FXS Group**.
    - Select **FXS Group** number. Default: 1.
    - Click **Settings** . The **FXS Port - Groups** window opens. Create the FXS Group. For detailed instructions on creating groups, see the topic [“Group”](#) under *Advanced Settings*.
  - Similarly, you can create a group of *not-sequential FXO Ports*, *Mobile Ports* and *SIP Trunks*.


- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier for creating *sequential* and *not-sequential* groups of FXS Ports.
- Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **SIP Trunk - Destination Port Determination - Destination Number Based** table.
- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click the **Delete** button.




*If there are multiple entries in the Destination Number Based table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

- By default, FXS Port 1 - 1 (Ascending) is assigned as the Routing Group, for routing calls from numbers that do not match with any of the destination numbers you configured (No Match Found) in the **SIP Trunk - Destination Port Determination - Destination Number Based** table.

To change the default Routing Group and to create the Fallback Routing Group for the No Match Found numbers entry,

- Under Edit, click **Settings** .

SIP Trunk - Destination Port Determination - Destination Number Based					
<input type="checkbox"/>	Edit	Destination Number	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
<input type="checkbox"/>		No Match Found	FXS Port 1 - 1 (Ascending)	None	Received Calling Party

**Total Records :** 1      1

**Testing**

Enter the destination number to know which entry would be selected for routing

- The **Edit Entry** window opens.

Edit Entry

Destination Number
No Match Found

CLI Number to be sent on Destination Port
Received Calling Party

Routing Group

☒ FXS Port
1 to 1 in Ascending order

☐ FXS Group
1

☐ FXO Port
1 to 1 in Ascending order

☐ FXO Group
1

☐ Mobile Port
1 to 1 in Ascending order

☐ Mobile Group
1

☐ Radio Port
1 to 1 in Ascending order

Fallback Routing Group
☐ Apply

☐ FXS Port
1 to 1 in Ascending order

☐ FXS Group
1

☐ FXO Port
1 to 1 in Ascending order

☐ FXO Group
1

☐ Mobile Port
1 to 1 in Ascending order

☐ Mobile Group
1

☐ Radio Port
1 to 1 in Ascending order

- Create the **Routing Group** and **Fallback Routing Group** as per your requirement.
- Click **Submit** and close the window.
- Close the window if you have finished adding/ editing entries.

Similarly, you can:

- configure the **FXS Port - Destination Port Determination - Destination Number Based** table by clicking the **FXS - Destination Number Based** link.
- and
- configure the **Radio - Destination Port Determination - Destination Number Based** table by clicking the **Radio - Destination Number Based** link.
- and
- configure the **FXO Port - Destination Port Determination - Destination Number Based** table by clicking the **FXO - Destination Number Based** link.
- and
- configure the **Mobile Port - Destination Port Determination - Destination Number Based** table by clicking the **Mobile - Destination Number Based** link.

# Group

SETU VGRX supports the following methods of determining the destination port for the calls originated on FXS Ports, Radio Ports, FXO Ports, Mobile Ports and SIP Trunks.

- Fixed
- on the basis of Destination Number
- on the basis of Calling Party Number



*The Destination Port Determination Method- on the basis of Calling Party Number is not applicable for the FXS Port and the Radio Port.*

A Routing Group may have *sequential* or *not-sequential* ports as members.

A Routing Group of *sequential* ports is to be formed when you select **FXS Port** or **Radio Port** or **FXO Port** or **Mobile Port** or **SIP Trunk** as the destination port.

A Routing Group of *not-sequential* ports is to be formed when you select **SIP - Group** or **Mobile - Group** or **FXO - Group** or **FXS - Group** as the destination port. The **SIP/Mobile/FXO/FXS Group** has members of the same port type, but not in a sequence. For example, a SIP Group can have only SIP Trunks as members and a Mobile Group can have only Mobile Ports as members.

## Configuring Groups

To create a Group,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Group** link to expand.
- Click **SIP Group** to create Groups of SIP Trunks.

SIP Trunk - Groups											
SIP Group Number	Member Selection Method	Member 1	Member 2	Member 3	Member 4	Member 5	Member 6	Member 7	Member 8	Member 9	
1	First Free	1		2		3		4		5	
2	First Free	1	None	None	None	None	None	None	None	None	
3	First Free	2	None	None	None	None	None	None	None	None	
4	First Free	3	None	None	None	None	None	None	None	None	
5	First Free	4	None	None	None	None	None	None	None	None	
6	First Free	5	None	None	None	None	None	None	None	None	
7	First Free	6	None	None	None	None	None	None	None	None	
8	First Free	7	None	None	None	None	None	None	None	None	
9	First Free	8	None	None	None	None	None	None	None	None	

You can create 9 Groups with 9 members in each group.

- Select a SIP Group Number from **1 to 9** and configure the member ports—**Member 1** to **Member 9**.

- For each **Member**, select a SIP Trunk number from **1 to 9**.

If you do not want any more members in a group, select **None**. For example, you want two members in a group, select the SIP Trunk numbers for member 1 and member 2. Select None for the remaining members in the group.

- Define the **Member Selection Method**. To route a call, the system checks availability of a free port. There are two options for port selection, namely:
  - **First Free:** The first port which is free will be used for routing the call each time. For example, SIP Group Number 1 has four members SIP Trunk 1 (Member 1), 2(Member 2), 3 (Member 3) and 6 (Member 4). For every incoming call, SETU VGRX will check the status of Member 1 first. If free, the call will be routed using this port else system will check status of Member 2 and so on.
  - **Rotation:** The first call will be routed through the first member port and the subsequent call through the next member port and so on. For example, SIP Group Number 2 has four members SIP Trunk 6 (Member 1), 7(Member 2), 8 (Member 3) and 9 (Member 4). For the first incoming call, SETU VGRX will check the status of Member 1 (SIP Trunk 6). If free, the call will be routed using this port else system will check status of Member 2 (SIP Trunk 7) and so on. For the next call, system will check the status of Member 2 (SIP Trunk 7). If free, call will be routed using this port else Member 3 (SIP Trunk 8) will be checked. Similarly, for the subsequent call the system will check the next member port in the group.

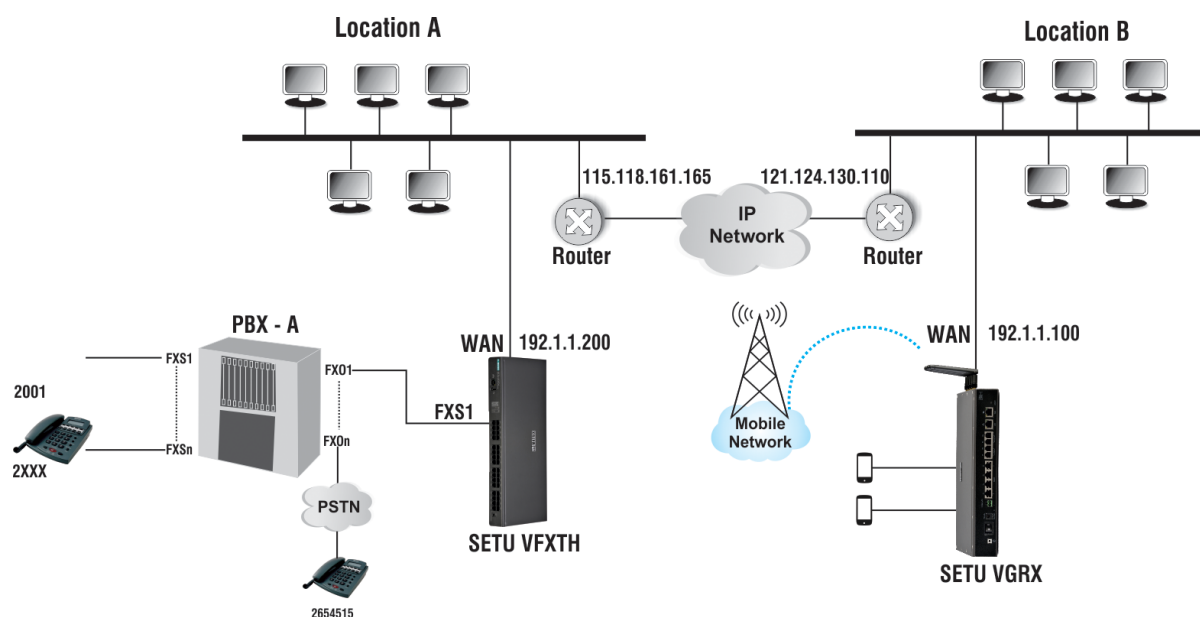
Default: **First Free**.

- Click **Submit** to save the group.
- Similarly, you can create FXO Groups, FXS Groups and Mobile Groups.
- To create Groups of Mobile Ports, click **Mobile Group**. You can create 4 Groups with 4 members in each group.
- To create Groups of FXO Ports, click **FXO Group**. You can create 4 Groups with 4 members in each group.
- To create Groups of FXS Ports, click **FXS Group**. You can create 4 Groups with 4 members in each group.

# Peer-to-Peer Dialing

Making an IP call without the intervention of a proxy server is called Peer-to-Peer Calling. As Peer-to-Peer calling does not require a proxy server, voice communication using this application can be done virtually free of cost. The major cost savings offered by this application makes it a very attractive mode of inter-branch or intra-office voice communication.

Let us understand how to use Peer-to-Peer Calling with the following illustration.



- Two offices are connected to the IP network.
- At Location A, a PBX (PBX A) and a Gateway (SETU VFXTH) is installed as shown above.
- SETU VGRX is installed at Location B.
- Peer-to-Peer calls can be made between the two locations with suitable configuration of SETU VGRX and the Gateway (SETU VFXTH).
- At **Location A**, you need to do the following configuration in SETU VFXTH:
  - Select a SIP Trunk to be used for this application and enable it. For example, SIP Trunk 1.
  - Set the **SIP Trunk Mode** of this trunk as **Peer-to-Peer**.
  - Keep the **SIP ID** of the SIP Trunk **blank**.



*In the Router, you must configure the same SIP and RTP Ports as configured in the SETU VFXTH. In other words, you must configure Port Forwarding for SIP and RTP on the Router.*

- By default, **Allowed IP Address for Incoming SIP Message** is set to **As per Peer to Peer table**. In the Peer to Peer table at Location A, you must configure the IP Address of the Router at Location B.
- Under **Handling of Incoming Calls** on the SIP Trunk, set the Incoming Call Routing option as **Route all incoming calls (with CLI) - to the Called Party Number**.

- For **SIP Trunk 1**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **FXS Port**.
- For **FXS Port**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **SIP Trunk 1** only.
- Now, configure the **Peer-to-Peer Table**.

In this example, you would have to configure the Peer-to-Peer table as follows:

- At Location A, in the **Destination Number**, enter the Number you want to dial to call the phone at Location B. In this case, 9898012345.
- For the number you entered, in the **Destination Address**, enter the IP Address of the Router connected at Location B. In this case, 121.124.130.110
- The Peer-to-Peer table you configure for SETU VFXTH at Location A would look like this:

Peer-to-Peer Dialing						
<input type="checkbox"/>	Edit	Destination Number	Minimum Digits	Maximum Digits	Destination Address	Name
<input type="checkbox"/>	➔	No Match Found	3	16		
<input type="checkbox"/>	➔	9898012345	3	16	121.124.130.110	Location B

Total Records : 2      1

+ Add
- Delete



*Instead of configuring the complete number string, you may configure only the prefix of the number to be dialed as follows, the system will place all calls that start with '9898' to the IP Address 121.124.130.110.*

Destination Number	Destination Address	Name
No Match Found		
9898	121.124.130.110	Location B

- At **Location B**, you need to do the following configuration in SETU VGRX:
  - Select a SIP Trunk to be used for this application and enable it. For example, SIP Trunk 1.
  - Keep the **SIP ID** of the SIP Trunk **blank**.
  - By default, **Network Profile 1** is selected as **SIP Network Profile** for SIP Trunk 1.
  - Set the **SIP Trunk Mode** as **Peer-to-Peer** for Network Profile 1.
  - By default, **Allowed IP Address for Incoming SIP Message** is set to **As per Peer to Peer table**. In the Peer to Peer table at Location B, you must configure the IP Address of the Router at Location A.



In the Router, you must configure the same SIP and RTP Ports as configured in the SETU VGRX. In other words, you must configure Port Forwarding for SIP and RTP on the Router.

- Under **Handling of Incoming Calls** on the SIP Trunk, set the Incoming Call Routing option as **Route all incoming calls (with CLI) - to the Called Party Number**.
- For **SIP Trunk 1**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **Mobile Port**.
- For **Mobile Port**, select the **Destination Port for Routing Calls** as **Fixed**, and create **Routing Group** as **SIP Trunk 1** only.

For instructions on configuring SIP Trunk parameters, see “[SIP Trunk](#)” under *Basic Settings*.

- Now, configure the **Peer-to-Peer Table**.

In this example, you would have to configure the Peer-to-Peer table as follows:

- At Location B, in **Destination Number**, enter the Number you want to dial to call the phone at Location A. In this case, 2001.
- For the number you entered in the **Destination Address**, enter the IP Address of the Router connected at Location A. In this case, 115.118.161.165
- The Peer-to-Peer table you configure for SETU VGRX at Location B would look like this:

Peer-to-Peer Dialing				
<input type="checkbox"/>	Edit	Destination Number	Destination Address	Name
<input type="checkbox"/>	➔	No Match Found	192.168.1.100	
<input type="checkbox"/>	➔	2001	115.118.161.165	

Total Records : 2

1

Testing

Enter the destination number to know which entry would be selected for routing

Search

- Configure PBX at location A such that calls received on the FXO Port of the PBX are routed to the FXS Port in sequential order, that is, calls to 2001 are routed to FXS 1 and so on. Similarly, when any FXS Port user dials a number starting with '9898', it should be routed using the FXO Port of the PBX to the FXS Port of the SETU VFXTH.
- When user 2001 of location A calls 9898012345, the call is routed using the FXO Port of the PBX to FXS Port of the SETU VFXTH. Further, it will be routed using the SIP Trunk of the SETU VFXTH to the IP address 121.124.130.110, as the system finds a matching entry for the dialed number in the Peer-to-Peer table.

- On receiving a call, the SETU VGRX at location B routes this call through the Mobile Port of the SETU VGRX to the Mobile user 9898012345.
- Similarly, when the Mobile user 9898012345 of location B calls 2001, the call is received on the SIP Trunk of the SETU VGRX and is placed to the IP address 115.118.161.165, as the system finds a matching entry for the dialed number in the Peer-to-Peer table.
- On receiving a call, the SETU VFXTH at Location A routes this call through the FXS Port of the SETU VFXTH to the FXO Port of the PBX, which is further routed to 2001.

## How to Configure

To use Peer-to-Peer calling, you must configure the related SIP Trunk parameters for the Peer-to-Peer application, namely: SIP Trunk Mode, Peer-to-Peer Table, SIP ID and Handling of Incoming Calls. For instructions, see “[SIP Trunk](#)” under *Basic Settings*.

The Peer-to-Peer table stores upto 500 entries. Each entry consists of the parameters —Destination Number, Destination Address and Name.

To configure the Peer-to-Peer Table,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Peer-to-Peer Dialing** link.
- The **Peer-to-Peer Dialing** table opens.

**MATRIX SETU VGRX**

**Peer-to-Peer Dialing**

Edit	Destination Number	Destination Address	Name
	No Match Found	192.168.1.100	

Total Records : 1      1

**Testing**

Enter the destination number to know which entry would be selected for routing

In the Peer-to-Peer table, the first entry is reserved for No Match Found.

- Click the **Add** button. The **Add Entry** window opens.

- In **Destination Number**, enter the number you expect the callers to dial. You may enter upto 64 characters (Digits + “Wildcard Characters”). Valid characters: 0 to 9, \*, #, X, T, Comma [,], Hyphen [-], Caret [^]. Default: Blank.

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

## Wildcard Characters

SETU VGRX supports following characters.

Character	Description
<b>X</b> (letter X)	X represents any single digit from 0 to 9.
<b>#</b>	When # is configured in a number string, it will not be considered as End of Dialing.
<b>*</b>	When * is configured in a number string, it will not be considered as End of Dialing.
<b>+</b>	+ (plus) can be configured as a first character of the Destination Number string in the <i>SIP Trunk-Destination Port Determination-Destination Number Based</i> table only.
<b>[ - ]</b>	Hyphen within the bracket, defines a range. Only digits 0-9 are allowed within a bracket.
<b>[ , ]</b>	Comma within a bracket is used as a separator between the groups of numbers.
<b>[ ^ ]</b>	Caret within a bracket is used to deny or restrict the number or range defined after the symbol. Only digits 0-9 are allowed after the caret.
<b>T</b> (letter T)	Character T can be configured only as a last character in a number string. When configured in a number string, the system waits for End of Dialing.

- In **Destination Address**, enter the domain name or IP Address to where the call is to be placed. The Destination Address may consists of upto 40 characters (maximum). Default: 192.168.1.100.

For example, if the peer-to-peer number to be dialed out is 1234@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 **Name**, enter a name to identify the number string you configured. It may be the name of your contact or any name you wish to assign to the number string. The name may consist of 24 characters (maximum). Default: Blank.

The name you configure here will not be used in SIP signaling.

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



*If there are multiple entries in the Peer to Peer table, to search a particular entry in the table, under Testing enter the desired number in the **Enter the destination number to know which entry would be selected for routing** search box.*

You can configure the Peer-to-Peer Table from the SIP Trunk page also.

# PIN Authentication

---

PIN Authentication is a necessary security feature to restrict access to the system and prevent possible misuse of the resources.

You can use PIN Authentication on the Source Port to establish the identity of callers before their call is processed by SETU VGRX.

PIN Authentication can be used on the Source Port only if the incoming call routing for the Source Port is set to ***Route calls After Answering the Call and Collecting the Digits***.

To be able to use PIN Authentication, this feature must be enabled on the Source Port and the PIN Authentication table must be configured.

The PIN Authentication table stores upto 500 PIN Numbers and their corresponding authentication Passwords.

When you enable PIN Authentication on the Source Port, SETU VGRX answers the incoming call on the port and prompts the caller to enter the PIN Number and the Password. It collects the digits dialed by the caller and matches it with the PIN Authentication table. When a match is found in the table, SETU VGRX authenticates the caller and allows the call to be processed.

If the digits dialed by the caller do not match with any entry in this table, SETU VGRX allows the caller to make two more attempts to dial a valid PIN Number and the Password. If the caller fails to dial the correct PIN Number and the Password, the system disconnects the call.

## Configuring PIN Authentication

To use this feature, you must enable PIN Authentication on the desired — FXO Ports, SIP Trunks, Mobile Ports — and configure the PIN Authentication Table.

To configure PIN Authentication table,

- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **PIN Authentication** link.

**PIN Authentication**

Index	PIN Number	PIN Password
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		
013		
014		

Submit Default All

- Now, configure the **PIN Authentication** table.
  - In **PIN Number**, enter the numbers with which callers will authenticate themselves. Default: Blank. The digits 0 to 9, \* and # are allowed in the PIN Numbers.



*The length of the PIN Number must not exceed four digits. If you enter a PIN Number that is less than 4 digits, the system will add leading zeros. The caller must also dial the PIN Number with the leading zeros to authenticate.*

- For each PIN Number you store, enter a password in **PIN Password**. The password can be of a maximum of four digits. The digits 0 to 9, \* and # allowed. Default: Blank.
- Click **Submit** to save the entries.
- You can enable PIN Authentication on the desired — FXO Ports and Mobile Ports.

To do so, first you need to select **After Answering the Call and Collecting the Digits** as the Incoming Call Routing option under *Handling of Incoming Calls*. Now, enable **Prompt caller to enter PIN** on the respective port.

Under “[Basic Settings](#)”, see “[FXO Port](#)” and “[Mobile Port](#)” for instructions.

# Digest Authentication

---

Digest Authentication is a challenge-based authentication service of SIP to authenticate the identity of the originator of SIP request in the INVITE message. The recipient of the request can ascertain whether or not the originator of the request is authorised to make the request. When the digest credentials of the originator — User Name and Password — in the INVITE message are authenticated and accepted by the recipient, the originator and the recipient are connected.

SETU VGRX supports Digest Authentication. The Digest Authentication feature works on the basis of the Digest Authentication Table in which the credentials — User Name and Password — of trusted/ authorised calling party SIP devices are stored. You must enable Digest Authentication on the SIP Trunk and configure the Digest Authentication table.

SETU VGRX will check the Digest Authentication table,

- when you enable this feature on a SIP Trunk.
- when SIP Trunk mode is Peer to Peer and **Allowed IP Address for Incoming SIP Message** is set to **Any**.

When you enable this feature on a SIP Trunk, for all incoming calls (SIP requests),

- SETU VGRX will challenge the identity of the calling party, that is, the SIP device initiating the request to send its digest credentials.
- When the calling party sends its credentials, SETU VGRX authenticates the credentials by matching it with its Digest Authentication Table.
- If a match is found, the calling party will be authenticated and the call will be allowed on the SIP Trunk.
- If no match is found, SETU VGRX will consider it as invalid authentication information and reject the call.

You may use Digest Authentication to:

- restrict access to SETU VGRX to specific callers.
- prevent unwanted or malicious calls.

## Configuring Digest Authentication

To use this feature, you must enable **Digest Authentication** on the desired SIP Trunk and configure the Digest Authentication Table.

You can configure the Digest Authentication Table also from the SIP Trunk parameters page of Jeeves.

To configure Digest Authentication table,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Digest Authentication** link.

The **Digest Authentication** Table page opens. You can configure up to 500 entries in this table. This Table is common for all SIP Trunks.

1-100
101-200
201-300
301-400
401-500

### Digest Authentication

Index	User ID	User Password
001		
002		
003		
004		
005		
006		
007		
008		
009		
010		
011		
012		
013		
014		

Submit
 

 Default All

- In the **User ID**, enter the user name assigned to the caller/calling device. SETU VGRX will use this User ID to match the digest credentials sent by the caller/calling devices when challenged.

Make sure the User ID you enter here and the User ID assigned at the *calling end* are the same. The User ID can be up to 40 characters long. Default: Blank.

- In the **User Password**, enter the password to authenticate the user ID. The password may consist of a maximum of 24 characters. Default: Blank.

Make sure the User Password you enter here and the User Password assigned at the calling end are the same.

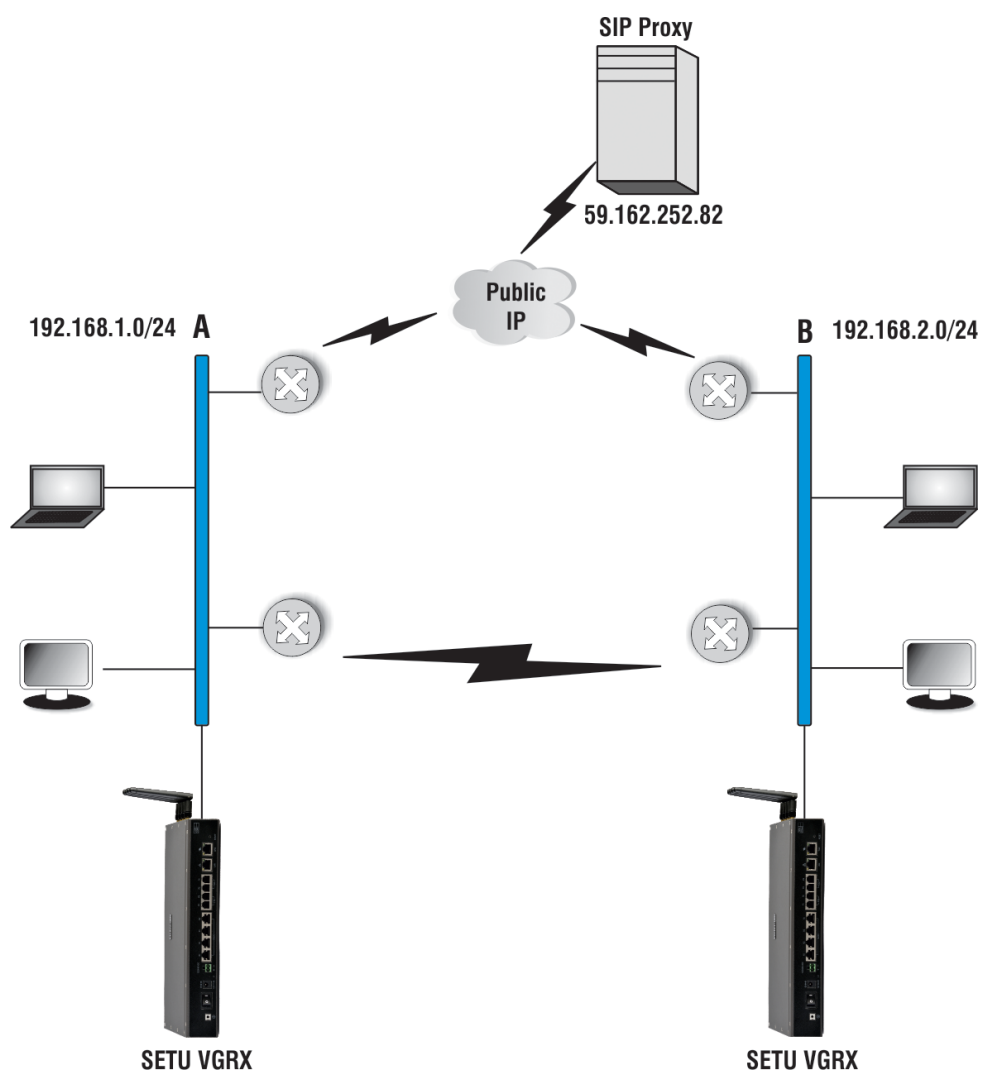
- Click **Submit** to save the entries.
- Make sure you also enable Digest Authentication on the desired SIP Trunk. For instructions, see [“SIP Trunk”](#) under *Basic Settings*.

# Static Routing

Static Routing Table is required when you have more than one router (gateway) in your network and you want SETU VGRX to send packets to multiple routers/gateways for different types of calls.

Static Routing Table helps route calls between point to point sites (connected through Multi Protocol Label Switching-MPLS, Frame Relay, etc.) and to public internet at the same time.

For example, two Local Area Networks, Network A and Network B, are connected through Frame Relay/ Multi Protocol Label Switching (MPLS) network to give access to local resources and also to make Peer-to-Peer calls. SETU VGRX is connected at both sites behind a router.



These sites are also connected to public IP network to:

- give internet access to local hosts.
- access DID service provided by ITSPs to make PSTN/ GSM calls over IP network.

Network A and Network B are in different subnets.

The Static Routing Table makes it possible to route different types of outgoing calls—Peer to Peer or Proxy—made to different subnets through different Gateways.

The Static Routing Table defines the appropriate Gateway Address (or Router's LAN Address) where the IP packets are to be sent.

In the Static Routing Table, you must configure:

- The address of the final Destination where the packets are to be sent.
- The Subnet Mask to be applied on the final destination address.
- The Gateway Address where the IP packets are to be sent.

When SETU VGRX sends packets, if the final destination IP Address and SETU VGRX are not in the same Subnet, the system will check the Static Routing Table.

If a perfect match is found, SETU VGRX will start sending the IP packets to the corresponding Gateway Address configured in the table.

If no match is found, SETU VGRX will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in "[Network Parameters](#)" the page.



- *The Static Routing Table is common for all SIP Trunks.*
- *The Static Routing Table is applicable only when the Network Connection is established through WAN.*

## Configuring Static Routing Table

The Static Routing Table must be configured at each location where SETU VGRX is installed. To configure the Static Routing Table,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Static Routing** link. The Static Routing Table page opens.

**Static Routing**

Index	Destination Address	Subnet Mask	Gateway Address
1			
2			
3			
4			
5			
6			
7			
8			

Submit Default

The Static Routing Table allows you to configure up to 8 entries. Each entry is stored against an Index number.

For each entry, you must configure the following:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.
- **Subnet Mask:** This is the mask to be applied on the destination address.
- **Gateway Address:** This is the IP address of the node where the IP packets are to be sent. Generally, it is the IP address of the LAN interface of the Router.

The Gateway Address must be in the same subnet as SETU VGRX.

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

As per the above example, the Static Routing Table of SETU VGRX at Location A should be configured as:

Index	Destination Address	Subnet Mask	Gateway Address
1	192.168.2.0	255.255.255.0	192.168.1.1
2			
:			
8			

- The Destination Address 192.168.2.0 specifies the network address of Location B.
- The Subnet Mask is the mask to be applied on the Destination address.
- The Gateway Address 192.168.1.1 specifies the LAN address of the Router A which connects location A and location B.

The IP address of the LAN interface of the router which connects Location A to the public internet should be configured as Default Gateway in the Network Parameters of SETU VGRX at Location A.

With the Static Routing Table configured, all calls made by SETU VGRX to 192.168.2.0/ 24 will be routed through the router which connects Location A to Location B. Whereas, all calls made by SETU VGRX to addresses other than 192.168.2.0/ 24 will be routed through the Default Gateway.

Similarly, configure the Static Routing Table in SETU VGRX at location B to enable calling from Location B to Location A.

# Network Connection

SETU VGRX offers connectivity to the IP Network over its WAN Port as well as over the Mobile Wireless WAN (Mobile Port 1).

For uninterrupted connectivity to the IP network and to minimize down time, you can connect SETU VGRX to the IP network over both WAN and Mobile Port 1. You can set the priority for each internet connection, ensuring that you have a fallback network connection each time the connection is down. You can also set SETU VGRX to monitor connectivity to the IP network and automatically switch to another, when one link is down.

For this, you need to configure the Network Connection settings.

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Network Connection** link.

The screenshot displays the 'SETU VGRX' web interface. On the left is a sidebar menu with categories: 'Basic Settings', 'Advanced Settings' (expanded), and 'Maintenance'. Under 'Advanced Settings', 'Network Connection' is selected. The main content area is titled 'Internet connection using' and contains the following sections:

- Internet connection using:** Two rows for 'Priority 1' and 'Priority 2'. 'Priority 1' is set to 'WAN Port' and 'Priority 2' is set to 'Mobile Port 1'. A note below states: '(Note: 2 priorities can not be same)'.
- Fallback Parameters:** Two checkboxes. 'Fallback to another network connection when connection is unavailable.' is unchecked. 'Switch back to priority 1 when connection is available.' is checked.
- Internet Connectivity Check:** Three fields. 'Server to check internet connectivity' is 'google.com'. 'Is to send DNS query?' is unchecked. 'Interval' is '120' seconds.

At the bottom of the main area are two buttons: 'Submit' (with a checkmark icon) and 'Default' (with a reset icon).

- Configure the network connection settings as per your preference and requirement.

## Internet Connection Using

- Under **Internet Connection using** options, by default **WAN Port** is set as **Priority1** and **Mobile Port 1** as **Priority 2**. You can change the **Priority 1** and **2** as per your requirement, from the different connection interfaces:
  - WAN Port
  - Mobile Port 1

You cannot select the same option for two priorities.

## Fallback Parameters

- Select the **Fallback to another network connection when connection is unavailable** check box, if you want the system to automatically switch over to the next Priority to re-establish the network connection, when the network connection set as Priority 1 fails. Default: Disabled.
- To have the system switch back to Priority 1 when the connection to Priority 1 network is restored, keep the **Switch back to priority 1 when connection is available** check box enabled.

## Internet Connectivity Check

To have the system monitor network connectivity, set **Internet Connectivity Check**:

- In **Server to check Internet Connectivity**, enter any Public IP Address / Domain Name like 'google.com'.  
  
The system will ping this programmed address to check internet connectivity regularly at fixed intervals.  
Default: google.com
- Select the **Is to send DNS query?** check box, if you want the system to check the internet connectivity by sending the DNS query. Default: Disabled.
- In the **Interval**, define the time interval at which you want the system to ping the Public IP Address / Domain Name to check the Internet Connectivity. Default: 120 seconds.
- Click **Submit** to save the changes.

With these parameters configured, the system pings the server at regular intervals (which you have configured) to monitor the status of the connection, whether the link is up or down.

The system re-establishes the link by selecting the next available connection set as the next Priority for Fallback. Thus, network connectivity is not hampered.

# Access Codes

---

Access Code is a string of digits dialed to use a feature. SETU VGRX users, can access the following features and facilities by dialing their Access Codes from a phone.

Feature/Function	Default Access Code
System Engineer (SE) Programming	#19 <sup>a</sup>
Call Waiting - Set/Cancel	#16
Do Not Disturb (DND) - Set/Cancel	#18
Hotline - Set/Cancel	#151
Hotline - Number	#152
Hotline - Timer	#153
Call Forward Unconditional - Set/Cancel	#131
Call Forward Unconditional - Number	#135
Call Forward Busy - Set/Cancel	#132
Call Forward Busy - Number	#136
Call Forward No-Reply - Set/Cancel	#133
Call Forward No-Reply - Number	#137
Call Forward No-Reply - Ring Timer	#139
Call Pick-up	#5
Call Hold/Retrieve	Flash <sup>a</sup>
Call Toggle (Call Split)	#2
Reject the Waiting Call and Speech with Current Call	#31
Ignore the Waiting Call and Speech with Current Call	#32
Accept the Waiting Call and Hold Current Call	#33
Accept the Waiting Call and Release Current Call	#34
Blind Transfer	#6
Attended Transfer	^
Conference	#8
Using Supplementary Services of Service Provider	#4
Using Voice Mail of the Service Provider	#7
Making a New Call	#91
Disconnect Call	#92
Forced Release Radio Port	#93

a. *Non-programmable.*

You can change the access codes assigned by default to the above features and facilities to suit your requirement.

## Configuring Access Codes

To change the default Access Codes assigned to the features and facilities,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Access Code** link.

Access Codes	
System Engineer(SE) Programming	#19
Call Waiting - Set/Cancel	#16
Do Not Disturb(DND) - Set/Cancel	#18
Hotline - Set/Cancel	#151
Hotline - Number	#152
Hotline - Timer	#153
Call Forward Unconditional - Set/Cancel	#131
Call Forward Unconditional - Number	#135
Call Forward Busy - Set/Cancel	#132
Call Forward Busy - Number	#136
Call Forward No-Reply - Set/Cancel	#133
Call Forward No-Reply - Number	#137
Call Forward No-Reply - Ring Timer	#139
Call Pick-up	#5
Call Hold/Retrieve	Flash
Call Toggle	#2
Reject the Waiting Call and Speech with Current Call	#31
Ignore the Waiting Call and Speech with Current Call	#32
Accept the Waiting Call and Hold Current Call	#33
Accept the Waiting Call and Release Current Call	#34
Blind Transfer	#6

- Change the default access code for the feature/facility, as required. Access Codes can be a maximum of 4 characters. Characters 0-9, \*, # and ^ are allowed.



*Do not configure Access Codes that may conflict with the Emergency Numbers.*

- Click **Submit** to save changes.

# Emergency Numbers

---

SETU VGRX supports the dialing of Emergency Numbers from all ports. Emergency numbers and their respective Routing Groups (through which they are to be routed) must be configured in the Emergency Number Table.

When you select “[Region](#)”, the system loads the Emergency Numbers used in the country you selected as Region, in the Emergency Number Table.

For each of these numbers loaded, the system assigns a default Routing Group to route the number. You may reassign the Routing Group, as appropriate.

You may also add numbers of emergency services as per your requirement and assign Routing Group for the numbers in the Emergency Number Table.

The Emergency Number Table stores up to 10 numbers.



- *For a few Regions, the system may not load default Emergency numbers in the Emergency Table. You may add the numbers as per your requirement.*
- *Emergency number Dialing will not work, if Mains power to SETU VGRX fails.*
- *Emergency number Dialing will not work, if CDMA module is installed in your SETU VGRX.*
- *Emergency Numbers have priority over Destination Number Table, PIN Number and Access Codes.*
- *The system does not apply End-of-Dialing when dialing Emergency Numbers.*
- *The system does not check Allowed-Denied Logic and Automatic Number Translation table when dialing an Emergency Number.*

SETU VGRX will dial out an emergency number only if — FXO Ports/ SIP Trunks/ Mobile Ports — included in the Routing Group for the emergency numbers are enabled.

Emergency Number can be dialed even when the option **Block all calls through this FXS Port** is enabled on the FXS Ports and **Block all calls through this Radio port** is enabled on the Radio Port.

SETU VGRX can dial out the numbers available in the Emergency Number Table even in the following situations:

- When SIM is absent
- When SIM is invalid
- When wrong SIM PIN is entered
- When SIM is blocked
- When GSM module is not registered



*Some countries do not allow dialing of Emergency Number without SIM. As per TEC standard, India allows dialing of Emergency Number without SIM.*

## Configuring Emergency Numbers

To configure the Emergency Number Table,

- Log into Jeeves.

- Click the **Advanced Settings** link.
- Click the **Emergency Number** link.

- To **Add** an Emergency Number in the table, click the **Add** button.
- To **Edit** an Emergency Number or a Routing Group, under Edit, click **Settings** .

A new window opens, to allow you to add/edit the entry.

- In the **Emergency Number**, enter the emergency number used in your country/region.




*Make sure that Access Codes you have configured do not conflict with the Emergency Numbers.*

- Create the **Routing Group**.
  - To create a group of *sequential FXO Ports* as members,
    - Select the desired **FXO Port** numbers as members. Default: 1.
    - In **in - order**, select the order in which the system should hunt for a free member FXO Port to route the call.

To start hunting from the first to the last member FXO Port, select **Ascending**.

To start hunting from the last to the first member FXO Port, select **Descending**.

Default: Ascending.

- To create a group of *not-sequential* **FXO Ports** as members,
  - Select a **FXO Group**.
  - Select **FXO Group** number. Default:1.
  - Click **Settings** . The **FXO Groups** window opens. Create the FXO Group. See “[Group](#)” for detailed instructions.
- Similarly, you can create groups of *sequential* and *not-sequential* SIP Trunks and Mobile Ports as members.
- Click **Submit** to save. Close the **Add Entry/Edit Entry** window. The entries you added appear on the Emergency Numbers page.

# Disconnect Tone

If Call Disconnection is signaled by your CO Network in the form of Disconnect Tone, configure **Disconnect Tone** parameters on the FXO Ports. You must enable *Disconnect Tone Detection* on the *FXO Port* and select the *Disconnect Tone Type*.

To enable the system to detect the Disconnect Tone accurately, you must configure the Cadence (ON-OFF time) and Frequency of the Disconnect Tone Type you selected, as supported by the CO network. To do this,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Disconnect Tone** link.

The Disconnect Tone Cadence Table opens.

Disconnect Tone	Frequency1 (Hz)	Operator	Frequency2 (Hz)	Cadence					
				ON Time1 (msec)	OFF Time1 (msec)	ON Time2 (msec)	OFF Time2 (msec)	ON Time3 (msec)	OFF Time3 (msec)
Disconnect Tone1	400	None	20	750	750	0	0	0	0
Disconnect Tone2	480	+	620	500	500	0	0	0	0
Disconnect Tone3	425	None	20	375	375	0	0	0	0
Disconnect Tone4	425	None	20	200	200	0	0	0	0

- For each Disconnect Tone, set the following parameters:
  - **Frequency 1 (Hz):** Frequency 1 is from 300 to 1400 Hz.
  - **Operator:** You may select Modulation (\*), Addition (+) or No operator.

If Modulation is selected, frequency 1 and frequency 2 will be used as modulation ( $F1 * F2$ ).

If Addition is selected, frequency 1 and frequency 2 will be used as addition ( $F1 + F2$ ).

If No Operator is selected, frequency 2 will not be applicable.

- **Frequency 2 (Hz):** Frequency 2 is from 20 to 1400 Hz. Select Frequency 2 if the Disconnect Tone supported by the CO network consists of Dual Frequency.
- **Cadence:** Configure the ON Time1-OFF Time1, ON Time2-OFF Time2 and ON Time3-OFF Time-3 for each Disconnect Tone. Valid ON Time and OFF Time range is 0000-9999 milliseconds.

When the system detects disconnect tone on the FXO Port and if it matches with the Frequency and Cadences you have set, the call will be disconnected and the FXO Port will be released.

- Click **Submit** to save.

# Prefix to Domain Name Conversion

---

Prefix to Domain Name Conversion is used when a user sets Call Forward or makes a Blind Transfer on SIP. This feature is applicable only when the destination port is SIP.

SETU VGRX supports multiple SIP Trunks and FXS Ports. When the FXS Port user dials a SIP number, SETU VGRX routes the call to the IP network using the SIP Trunk determined by the routing mechanism. The FXS Port user can dial only numbers, not domain names. Therefore, it becomes necessary to assign Prefix codes to domain names, which the FXS user can dial.

Now, it is necessary that the number string dialed by SETU VGRX is understood by the ITSP through which the call is routed. So, an appropriate Prefix Code is assigned to the Domain of the ITSP through which the calls are to be routed.

However, when the FXS Port user sets Call Forward or Blind Transfer a call, the Prefix Code and the number are sent to the calling party in the redirect message, without the domain name. So the calling party will not be able to reach the FXS user at the forwarded/transfer destination. The *Prefix to Domain Name Conversion* feature resolves this.

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



- Assume that SETU VGRX is configured to route calls made to the domain 'abc.com' from the FXS Port, through the SIP Trunk subscribed with the ITSP 'Pulver.com'.
- Since the FXS Port user cannot dial the domain name, a prefix code must be assigned to the domain name.
- The Prefix code, \*234 is assigned to the domain 'abc.com'.
- When the FXS Port user wants to dial the SIP ID 9874@abc.com, the user must dial \*234 followed by 9874.
- SETU VGRX determines that the called party is the subscriber of abc.com and converts \*2349874 to 9874@abc.com and routes the call to the desired destination through 'Pulver.com'.
- Now, given the above scenario, assume that the FXS Port user sets Call Forward to \*2349874 (that is, 9874@abc.com).
- When an external caller calls the FXS Port user, the caller will receive \*2349874 in the Redirect message.
- However, to reach the FXS user at the forwarded destination, the caller must have a domain name in the contact address. Since the caller will not recognize \*234 (the prefix code assigned to abc.com), the caller will not be able to reach the FXS user at the forwarded destination address.
- With Prefix to Domain Name Conversion, SETU VGRX converts this prefix code to the domain name abc.com (9874@abc.com), and sends it in the Redirect message to the external caller informing the caller of the new contact address.
- On receiving this information in the Redirect message, the external caller can call the new contact number, 9874@abc.com, and talk to the FXS user.

To configure Prefix to Domain Name Conversion,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Prefix to Domain Name Conversion** link.
- The Prefix to Domain Name Conversion table opens. You can store upto 64 entries in this table.

**Prefix-to-Domain Name Conversion**

Index	Prefix	Domain Name
01		
02		
03		
04		
05		
06		
07		
08		
09		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		

 Submit  Default

- In **Prefix**, enter the Prefix Code you want to assign to the Domain Server Names. The Prefix code must not exceed four digits. Valid digits are 0 to 9, \* and #
- For each Prefix Code you assigned, enter the corresponding **Domain Name**. The Domain Name may consist of a maximum of 40 characters.
- Click **Submit** to save the entries.

# Certificate Manager

SETU VGRX supports certification for TLS, Web Server, Firmware Upgrade, Configuration Upgrade.

SETU VGRX supports two types of Certificates: **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


- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Certificate Manager** link to expand.
- Click the **Generate** link.




- Click **Self Signed CA Certificate** to expand and configure the following parameters.

Self Signed CA Certificate

Country Name - 2 letter code (eg. IN)	<input type="text"/>
State or Province Name - full name	<input type="text"/>
Locality Name (eg, city)	<input type="text"/>
Organization Name (eg, company)	<input type="text"/>
Organizational Unit Name (eg, section)	<input type="text"/>
Common Name (eg, System's hostname/IP Addr.)	<input type="text"/>
Email Address (eg. me@myhost.mydomain)	<input type="text"/>

 Generate

 Download

- In **Country Name - 2 letter code (eg. 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 (eg. city)**, enter the name of your city.
- In **Organization Name (eg. company)**, enter the name of your organization where SETU VGRX is installed.
- In **Organizational Unit Name (eg. section)**, enter the name of the unit or section or domain of your organization, where SETU VGRX is installed.
- In **Common Name (eg. System's hostname/IP Addr.)**, enter your Server's (SETU VGRX) host name or IP Address. This Common Name serves as the distinguishing factor.
- In **Email Address (eg. 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** link. The CA Certificate you created appears in the **Root CA Certificate** table.

**Trusted Root CA**

No file selected.
(Valid format .cer, .crt & .pem)

**Root CA Certificates**

<input type="checkbox"/>	Issued To	Issued By	Expiration Date	Friendly Name
<input type="checkbox"/>	www.MatrixComSec.com	www.MatrixComSec.com	Dec 31 2036	SelfSignedCaCertificate

- 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**.

A sample Self-Signed CA Certificate is as under:

```
Certificate:
  Data:
    Version: 3 (0x2)
    Serial Number: 1 (0x1)
    Signature Algorithm: sha1WithRSAEncryption
    Issuer: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD.,
    OU=R&D, CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
    Validity
      Not Before: Aug 13 13:13:18 2013 GMT
      Not After : Dec 31 13:13:18 2036 GMT
    Subject: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD.,
    OU=R&D, CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
    Subject Public Key Info:
      Public Key Algorithm: rsaEncryption
      RSA Public Key: (2048 bit)
        Modulus (2048 bit):
          00:da:9e:27:ae:64:58:1d:88:d1:58:10:96:1d:42:
          cf:7a:cc:ef:07:ef:66:8c:93:1e:66:3b:15:07:60:
          ea:87:f0:72:a2:93:de:31:05:64:97:92:14:e9:31:
          47:3e:d2:dd:13:d3:06:d0:19:d4:f9:d6:b9:b6:f3:
          9a:0c:ec:bb:bd:eb:1e:b5:24:1a:30:a5:53:2f:d5:
          74:54:a9:10:fa:da:f1:39:05:3d:7d:09:cd:d6:d6:
          23:37:d1:c4:d7:a4:a7:34:22:70:66:4d:b0:65:f9:
          3b:bf:06:d0:1a:e8:97:e0:ef:c0:9e:ef:40:f1:c4:
          c9:e2:a7:7e:03:b6:72:00:fd:8c:02:c5:57:9c:57:
          fc:99:8c:36:22:9f:e9:7a:32:49:27:a5:11:21:3d:
          f9:e9:6f:d2:1f:88:65:a9:45:5a:99:e2:1a:51:cb:
          69:31:b1:dc:06:7b:ef:94:24:2e:c0:f9:f0:bd:25:
          67:6a:e5:e9:46:f7:e8:d7:6c:f5:5c:ed:dc:cd:7c:
          82:02:0f:7d:f7:fd:0b:66:d0:ee:24:e1:2b:64:97:
          58:27:3b:96:bd:dd:b4:ea:3f:51:f7:a5:2c:dd:c7:
          22:72:b9:3c:09:75:04:df:56:5b:af:f8:3d:fe:f0:
          50:3f:01:c9:8e:2a:3e:36:66:1f:fe:dd:87:84:99:
          11:7b
        Exponent: 65537 (0x10001)
    X509v3 extensions:
      X509v3 Basic Constraints:
        CA:FALSE
      Netscape Comment:
        OpenSSL Generated Certificate
      X509v3 Subject Key Identifier:
```

In the above Self-Signed CA Certificate:

- **C** = Country
- **ST** = State
- **L** = Location
- **O** = Organization
- **OU** = Organization Unit
- **CN** = Common Name
- **Issuer** represents the details of the CA issuing the Certificate. Here, the Organization itself is the CA (issuer), hence, the O, OU and CN of both Issuer and Subject is same.
- **Validity** represents the valid period of this certificate.
- **Subject** represents the credentials of the Server / User requesting for certification.

- **Public Key** represents the public key of the certificate.

## 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,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Certificate Manager** link to expand.
- Click the **Generate** link.
- Click **System Certificate** to expand and configure the following parameters.

System Certificate

Generate

☒ Self-Signed Certificate
 ☐ Certificate Signing Request (CSR)

Friendly Name

Country Name - 2 letter code (eg. IN)

State or Province Name - full name

Locality Name (eg, city)

Organization Name (eg, company)

Organizational Unit Name (eg, section)

Common Name (eg, System's hostname/IP Addr.)

Subject Alternate Name (eg. DNS:hostname,IP:ipaddr)

Email Address (eg. me@myhost.mydomain)

Validity upto

31

December

2036

Generate

- 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 (eg. 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 (eg. city)**, enter the name of your city.
- In **Organization Name (eg. company)**, enter the name of your organization where SETU VGRX is installed.
- In **Organizational Unit Name (eg. section)**, enter the name of the unit or section or domain of your organization, where SETU VGRX is installed.
- In **Common Name (eg. System's hostname/IP Addr.)**, enter your Server's (SETU VGRX) host name or IP Address. This Common Name serves as the distinguishing factor.
- In **Subject Alternate Name (eg. 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 the 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 Manager**, click the **Local Certificate** link to expand.

The generated certificate appears in the **Local Certificates** table.

Local Certificates

Upload Certificate

Browse...

No file selected.

(Valid format .cer, .crt & .pem)

Upload Private Key


Browse...

No file selected.

(Valid format .pem & .key)

Upload

Local Certificates

	Issued To	Issued By	Expiration Date	Friendly Name	Download
<input type="checkbox"/>	www.MatrixComSec.com	www.MatrixComSec.com	Dec 31 2036	DefaultServerCert_Setu	

Delete

- 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** .

A sample Self-Signed System Certificate is as under:

Certificate:

Data:

```

Version: 3 (0x2)
Serial Number: 2 (0x2)
Signature Algorithm: sha1WithRSAEncryption
Issuer: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD., OU=R&D,
CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
Validity
  Not Before: Aug 13 13:14:57 2013 GMT
  Not After : Dec 31 13:14:57 2036 GMT
Subject: C=IN, ST=Gujarat, L=Vadodara, O=MATRIX COMSEC PVT. LTD., OU=R&D,
CN=www.MatrixComSec.com/emailAddress=Support@MatrixComSec.com
Subject Public Key Info:
  Public Key Algorithm: rsaEncryption
  RSA Public Key: (2048 bit)
    Modulus (2048 bit):
      00:b5:29:61:26:35:db:d7:a8:fd:05:4d:ac:2d:6c:
      65:70:4d:42:fb:f6:1e:c8:18:bd:1c:c7:5a:92:b3:
      28:52:48:66:7c:0f:c8:35:6f:13:46:62:1e:23:44:
      b3:27:28:f5:8e:43:1a:e3:f6:7e:d5:8f:a9:73:8a:
      2c:34:1e:35:d0:c8:0c:b2:68:12:dc:1a:23:da:fe:
      02:af:88:4e:a1:7a:7f:a0:2b:ca:b9:72:5d:ac:3a:
      e3:9b:fd:0d:ab:0f:c3:57:a9:99:cd:2e:be:02:9c:
      60:0e:83:e8:69:2d:0f:95:79:52:87:66:9f:4a:10:
      09:db:4e:41:e2:f2:b4:86:cd:42:a9:55:6d:33:a3:
      60:67:fd:1d:3d:0e:8d:6a:53:77:e0:07:78:c9:c8:
      34:23:df:3d:94:02:41:e9:c4:2b:c8:04:10:ba:69:
      dc:d3:4c:85:39:09:a6:df:c4:1d:2d:80:2b:d8:f6:
      88:0a:c6:98:3f:85:34:19:c0:a5:fe:d9:f8:96:39:
      ec:cb:b7:c5:fa:84:e1:93:6d:82:7c:12:70:cf:67:
      5d:95:15:e9:1a:71:18:ad:f7:3f:09:1b:f5:0f:80:
      fb:9e:e9:96:54:91:59:39:6b:dd:5f:02:22:b9:c6:
      2a:60:e8:76:61:88:84:f1:e1:74:a1:17:12:66:98:
      6a:93
    Exponent: 65537 (0x10001)
X509v3 extensions:
  X509v3 Basic Constraints:
    CA:FALSE

```

In the above Self-Signed System Certificate,

- **Issuer** represents the details of the CA issuing the Certificate. Here, the Organization itself is the CA (issuer), hence, the O and CN of both Issuer and Subject is same.
- **Validity** represents the valid period of this certificate.

- **Subject** represents the credentials of the Server / User requesting for certification. Here, OU=R&D i.e. for whom the certificate is signed.
- **Public Key** represents the public key of the certificate.

## CA Signed Certificate

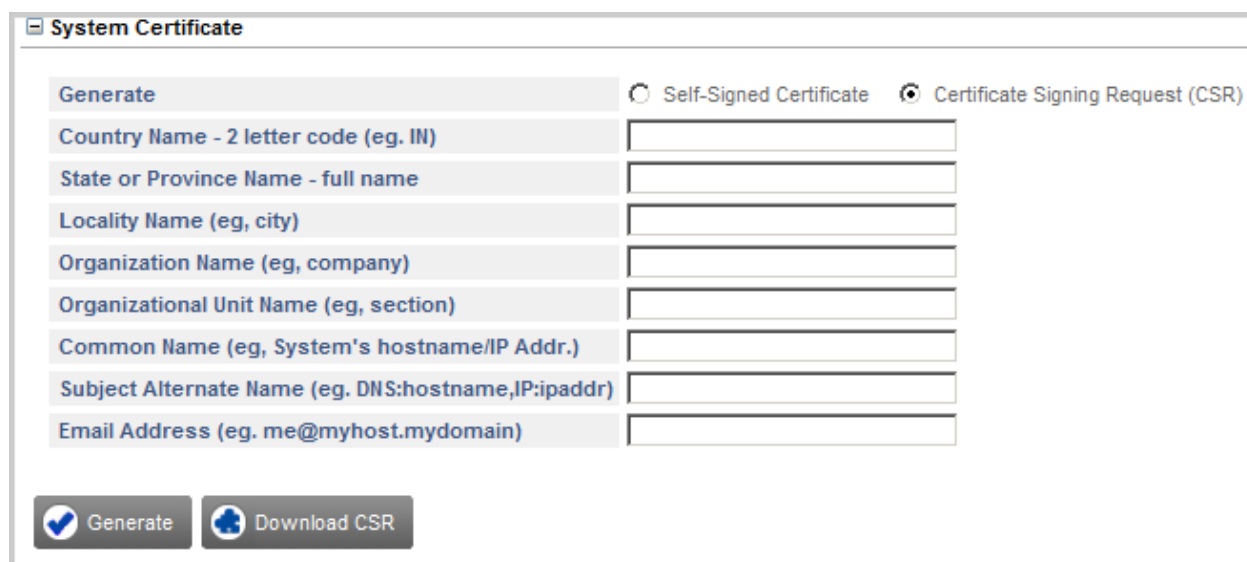
Certificate Authority (CA) is a trusted organization which creates and sells TLS Certificates to websites. *CA Signed Certificates* are the TLS Certificates which are created by such trusted CAs, signed and sold to any applicant. These certificates contains a public key and the identity of the owner; 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

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Certificate Manager** link to expand.
- Click the **Generate** link.
- Click **System Certificate** to expand and configure the following parameters.



**System Certificate**

**Generate** ☐ Self-Signed Certificate ☒ Certificate Signing Request (CSR)

Country Name - 2 letter code (eg. IN)

State or Province Name - full name

Locality Name (eg, city)

Organization Name (eg, company)

Organizational Unit Name (eg, section)

Common Name (eg, System's hostname/IP Addr.)

Subject Alternate Name (eg. DNS:hostname,IP:ipaddr)

Email Address (eg. me@myhost.mydomain)

☒ Generate ☐ Download CSR

- In **Generate**, select the type of certificate you want to create. You must select **Certificate Signing Request (CSR)**.
- In **Country Name - 2 letter code (eg. 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 (eg. city)**, enter the name of your city.
- In **Organization Name (eg. company)**, enter the name of your organization where SETU VGRX is installed.
- In **Organizational Unit Name (eg. section)**, enter the name of the unit or section or domain of your organization, where your SETU VGRX is installed.
- In **Common Name (eg. System's hostname/IP Addr.)**, enter your Server's (SETU VGRX) host name or IP Address. This Common Name serves as the distinguishing factor.
- In **Subject Alternate Name (eg. 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 (eg. 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.

The Certificate Signing Request (CSR) to be sent to any trusted CA, appears as under:

```
-----BEGIN CERTIFICATE REQUEST-----
MIIDLDCCAQCQAQAwgaoxCzAJBgNVBAYTAklOMRAwDgYDVQQIEwdHdWphcmFOMREw
DwYDVQQHEWhWYWRvZGFyYTEgMB4GA1UEChMXTUUFUUK1YIENPTVNFQyBQV1QuIExU
RC4xDDAKBgNVBAsUA1ImRDEdMBsGA1UEAxMUd3d3Lk1hdHJpeENvbVNIYy5jb20x
JzAlBgkqhkiG9w0BCQEWGfN1cHBvcnRATWFOcm14Q29tU2VjLmNvbTCCASIwDQYJ
KoZIhvcNAQEBBQADggEPADCCAQoCggEBAPutA1/cZcz/qZe3soIITiVpPI8PIZ6d
9RvInx4haqVob7M110dYwvN2rLmFod3ZtEu9dX645crC4NXn9pxKXmkp5iNdBVca
rm1qedZ63S1cR3m4YhL2dUc7DQ9T1GNTFpbXLR1A4sQk+nVwO+C+XU/jPlpqIR0sn
Idh2/eLWVOauRgY3qdGjPaN8ndq8xVieY+v1/XpLQa4Oyd6aP+xn+z4pSWK4YLeP
36/CRh5q4f3vfMpuQTfegxGA+UB1V3qPMSqI0jBr7r1jptDxlmwzXkwz5w1rovh8
ZNP+1sIYPyZ9zrZm+eyhxpSX8o09jCcEm/R816x6GHEER7UGdZR1HvUCAwEAAaA8
MDoGCSqGSIb3DQGEJDjEtmCswCQYDVROTBAlwADALBgNVHQ8EBAMCBwAwEQYDVRO
BAowCIIGTWFOcm14MA0GCSqGSIb3DQEBBQUAA4IBAQCQtMjNA13HAWYa9w1JGbKW
Yjoc/gbrhSUwgbR4Jh+13guInViTyJ5YDt9pLc8xzJe23MV2XDv4ImSSUSkRojcg
IpVTqNPgf91k50WmJHTIT0JJGEUXvzKE71V0kuf0XTelW0o81QYpjGn8GaSQqCDV
q746F0i84zwejY+/jL+pDMpczxvbnnotWg+wCkMXwkdAk0InqL+DuSTenuBEcW82
UFe0rqoMdt90XpS9YzpjIsotRYgTRNIFaBFF4LxQa1bYQ15pZ79MxWJIZQZTnqHf
MbwSosS/QM7ZjE147b13m9Lk69jdzfSAPmCW4AdulBe7PENGGI+MMzfAVyYSwdkw
-----END CERTIFICATE REQUEST-----
```

## 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:

- Log into Jeeves.
- Click the **Certificate Manager** link.
- Click the **Local Certificate** link.

**Local Certificates**

Upload Certificate

Browse...

No file selected.

(Valid format .cer, .crt & .pem)

Upload Private Key


Browse...

No file selected.

(Valid format .pem & .key)

Upload

**Local Certificates**

<input type="checkbox"/>	Issued To	Issued By	Expiration Date	Friendly Name	Download
<input type="checkbox"/>	www.MatrixComSec.com	www.MatrixComSec.com	Dec 31 2036	DefaultServerCert_Setu	

Delete

- 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** .

# Call Detail Record

---

SETU VGRX enables you to generate reports of Call Detail Records of calls using various filters such as:

- The port from which the calls originate (Source Port)
- The port on which the calls terminate (Destination Port)
- Calls made on particular dates
- Calls made at a particular time
- Calls of a certain duration
- Calls of certain Called Party Numbers
- Calls of certain Calling Party Numbers
- Calls made with PIN Authentication
- Calls made without PIN Authentication

You can set the different filters as required and generate Call Detail Record Report. The reports can be used for analyzing the call records for different purposes like cost savings, productivity enhancement, security and privacy.

The system stores records of matured calls only and it generates reports only of filters that are set. For example, if you have not enabled the filter for *Calls Originated from SIP Trunks*, the system will not generate report for calls originated from SIP Trunks.

SETU VGRX supports up to 2000 call record entries and these entries are stored using the First In First Out (FIFO) method.

Call records remain stored,

- when the system is set to default.
- when the firmware version is changed.

Call records can be cleared manually at any time.

## Configuring Call Detail Record Filters

- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **Call Detail Record (CDR)** link.

Filter	Apply Filter	From	To
Calls originated from FXS Ports	<input checked="" type="checkbox"/>	1	2
Calls originated from SIP Trunks	<input checked="" type="checkbox"/>	1	9
Calls originated from FXO Ports	<input checked="" type="checkbox"/>	1	2
Calls originated from Mobile Ports	<input checked="" type="checkbox"/>	1	4
Calls originated from Radio Ports	<input checked="" type="checkbox"/>	1	4
Calls terminated on FXS Ports	<input checked="" type="checkbox"/>	1	2
Calls terminated on SIP Trunks	<input checked="" type="checkbox"/>	1	9
Calls terminated on FXO Ports	<input checked="" type="checkbox"/>	1	2
Calls terminated on Mobile Ports	<input checked="" type="checkbox"/>	1	4
Calls terminated on Radio Ports	<input checked="" type="checkbox"/>	1	4
Calls Made From	<input checked="" type="checkbox"/>	01 Jul 2010	07 Apr 2016
Calls Made Between	<input checked="" type="checkbox"/>	00 : 00	23 : 59
Called Party Numbers Matching with Number List	<input checked="" type="checkbox"/>	01	
Calling Party Numbers Matching with Number List	<input checked="" type="checkbox"/>	01	
Call Duration equal to and greater than (HH:MM:SS)	<input checked="" type="checkbox"/>	00 : 00 : 00	
Calls without PIN Number	<input checked="" type="checkbox"/>		
Calls with PIN Number	<input checked="" type="checkbox"/>	0001	9999

Clear Call Records Download Call Records

Submit Default

## Setting Filters

- To set filters, click the **Filters** link under Call Detail Record.

By default, all the filters are enabled. You may disable the filter you do not want to use by clearing the related **Apply Filter** check box.

Some of these filters are enabled by default, you cannot disable them, but you can set them.

- Set the following filters as required:



*The filters you set are not applied on the downloaded report. The CSV and TXT files will contain all the records, without filters.*

- Calls originated from FXS Ports:** The system can generate a report of outgoing calls made from the FXS Ports of SETU VGRX. To generate report using this filter, set the range of the FXS Ports in the **From** and **To** fields.

You can also generate a report for a single FXS Port, by selecting the same port number in the **From** and **To** fields.

- Calls originated from SIP Trunks:** The system can generate a report of calls that were received on the SIP Trunks of SETU VGRX for further routing. To generate a report using this filter for a range of SIP Trunks, select the range of the SIP Trunks in the **From** and **To** fields.

You can also generate a report for a single SIP Trunk, by selecting the same trunk number in the **From** and **To** fields.

- **Calls originated from FXO Ports:** The system can generate a report of calls that were originated from the FXO Ports. Set the range of the FXO Ports in the **From** and **To** fields.

You can generate a report for a single FXO Port, by selecting the same port number in the **From** and **To** fields.

- **Calls originated from Mobile Ports:** The system can generate a report of calls that were originated from the Mobile Ports. Set the range of the Mobile Ports in the **From** and **To** fields.

You can generate a report for a single Mobile Port by selecting the same port number in the **From** and **To** fields.

- **Calls originated from Radio Ports:** The system can generate a report of outgoing calls made from the Radio Ports. Set the range of the Radio Ports in the **From** and **To** fields.

You can also generate a report for a single Radio Port, by selecting the same port number in the **From** and **To** fields.

- **Calls terminated on FXS Ports:** The system can generate a report of calls terminated on the FXS Ports. To generate a report using this filter, set the range of the FXS Ports in the **From** and **To** fields.

You can also generate a report for a single FXS Port, by selecting the same port number in the **From** and **To** fields.

- **Calls terminated on SIP Trunks:** The system can generate a report of calls terminated on the SIP Trunks. To generate a report using this filter for a range of SIP Trunks, set the range of the SIP Trunks in the **From** and **To** fields.

To generate a report for calls terminated on a single SIP Trunk, set the same trunk number in both fields.

- **Calls terminated on FXO Ports:** The system can generate a report of calls terminated on the FXO Ports. Set the range of the FXO Ports in the **From** and **To** fields.

You can generate report for a single FXO Port, by selecting the same port number in the **From** and **To** fields.

- **Calls terminated on Mobile Ports:** The system can generate report of calls terminated on the Mobile Ports. Set the range of ports in the **From** and **To** fields.

Set the same port number in both fields, if you want to generate report for calls terminated on a single Mobile Port.

- **Calls terminated on Radio Ports:** The system can generate report of calls terminated on the Radio Ports. Set the range of ports in the **From** and **To** fields.

Set the same port number in both fields, if you want to generate report for calls terminated on a single Radio Port.

- **Calls made from:** The system will generate report of calls made between particular dates. Enter the start date and end date in the corresponding **From** and **To** fields.

- **Calls made between:** The system will generate report of calls made between a particular time period. Enter the start time and end time in the corresponding **From** and **To** fields.
- **Called Party Number Matching with Number List:** Select the Number List you want to assign to this filter. Make sure that you also configure this Number List with the Called Party Numbers which you want the system to match. See [“Number Lists”](#) for instructions.
- **Calling Party Numbers Matching with Number List:** The system generates report for calls received from specific numbers.

Select a Number List you want to assign to this filter. Make sure that you also configure this Number List with the Calling Party Numbers which you want the system to match. See [“Number Lists”](#) for instructions.

- **Call Duration equal to and greater than (HH: MM: SS):** The system generates report for calls of a specific time duration. Select the call duration in HH: MM: SS format.
- **Calls without PIN Number:** The system will generate report for calls without PIN Authentication.
- **Calls with PIN Number:** The system will generate a report for calls that were made using PIN Authentication. You can generate report of calls of specific PIN Numbers also.

Enter the range of PIN Numbers in the **From** and **To** fields. PIN Numbers can be in the range of 0000 to 9999. The system will generate Report of all calls having PIN Numbers within the range you have set and display them under the ‘PIN Numbers’ column of the report.

If you want to generate report of a particular PIN Number, enter the same PIN Number in the **From** and **To** fields.

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

## Clear Call Records

- You can clear the call detail records any time you want, by clicking the **Clear Call Records** button.

When call records are cleared, the **From** field of the filter **Calls Made Between** will change to the date of clearing of the records.

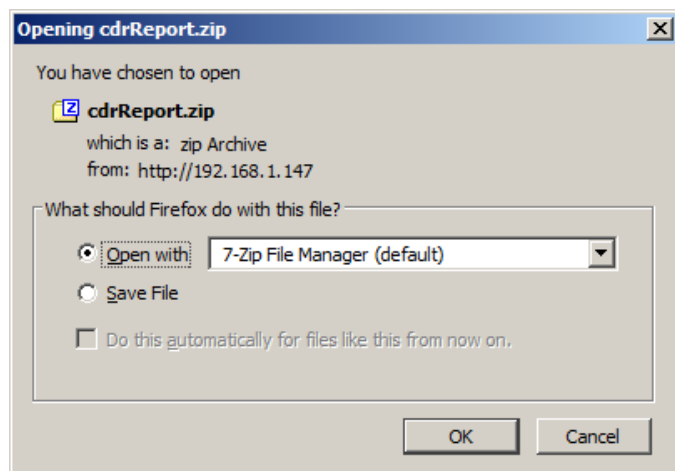
## Download Call Records

- If you want to open/ save Call Detail Record Report on your computer, click the **Download Call Records** button.



*If you are using Mozilla Firefox (version 3.5 recommended), set the Downloads option of your browser as **Always ask me where to save the files**.*

- You will get a prompt with the option to open the **cdrReport.zip** file or save the file to a location. Save the file on the local disk.



- Open the cdrReport.zip file from the location you saved. The zip file contains the CDR report in Excel and Text format.

## Printing Call Detail Record Report

- You can also print the Call Detail Record Report, if required.
- To print the CDR report in Excel format, open the file **CdrReport.csv**
- To print the CDR report in text format, open the file **CdrReport.txt**
- Print the file you opened. You may change the formatting of the text in the files before printing.



*The filters you set are not applied on the downloaded report. The CSV and TXT files will contain all the records, without filters.*

## Viewing Call Detail Report

- To view the report generated by the system for the filters you have set, click the **Report** link under Call Detail Record.

Sr. No.	Date	Start Time	Calling Number	Called Number	Duration (sec)	Source Port	Destination Port	Disconnected By
0001	27-Mar-2015	20:05	2005		00:00:14	RADIO-1	FXS-1	FXS-1
0002	27-Mar-2015	20:05	2001	2005	00:00:13	FXS-1	RADIO-1	FXS-1
0003	27-Mar-2015	20:06	2006		00:00:08	RADIO-2	FXS-1	FXS-1
0004	27-Mar-2015	20:06	2001	2006	00:00:12	FXS-1	RADIO-2	FXS-1
0005	27-Mar-2015	20:07	2007		00:00:09	RADIO-3	FXS-1	FXS-1
0006	27-Mar-2015	20:07	2001	2007	00:00:09	FXS-1	RADIO-3	FXS-1
0007	27-Mar-2015	20:07	2008		00:00:08	RADIO-4	FXS-1	FXS-1
0008	27-Mar-2015	20:08	2001	2008	00:00:09	FXS-1	RADIO-4	FXS-1
0009	28-Mar-2015	11:12	192.168.1.5	2356890	00:00:22	SIP-1	RADIO-4	SIP-1
0010	28-Mar-2015	11:13	2008		00:00:01	RADIO-4	FXS-1	FXS-1
0011	28-Mar-2015	11:14	2008		00:00:00	RADIO-4	SIP-1	SIP-1
0012	28-Mar-2015	11:14	2008	1234	00:00:09	RADIO-4	SIP-1	SIP-1
0013	28-Mar-2015	11:15	2008	1234	00:00:17	RADIO-4	FXO-1	FXO-1
0014	28-Mar-2015	11:17	2008	9924873632	00:00:17	RADIO-4	MOB-1	MOB-1
0015	28-Mar-2015	11:18	+918128680286	2356898	00:01:23	MOB-1	RADIO-4	MOB-1

Total Records : 17      1 2

- Call Detail Record Report generated as per the filters you set will appear in the following columns:
  - **Date:** Displays the date on which the calls are made.
  - **Start Time:** Displays the time at which the calls are made.
  - **Calling Number:** Displays the numbers from which the calls are received.
  - **Called Number:** Displays the numbers to which the calls are made.
  - **Duration:** Displays the duration of the calls.
  - **Source Port:** Displays whether the calls are originated from the SIP Trunks/FXO Ports/FXS Ports/Mobile Ports/Radio Ports.
  - **Destination Port:** Displays whether the calls are terminated on the SIP Trunks/FXO Ports/FXS Ports/Mobile Ports/Radio Ports.
  - **Disconnected By:** Displays the port that disconnected the call.
  - **Cause:** Displays the cause for call disconnection.
  - **PIN Number:** Displays the PIN Number dialed by the caller for making calls.
  - **Remarks:** Displays the type of call. A stands for Anonymous, U for Unanswered and N for Normal.
  - **By Port:** Displays the number of the FXS Port using the supplementary services.
  - **By Number:** Displays the number assigned to the FXS Port using the supplementary services.

The total number of records is displayed below the table.

On each page, 15 records are displayed. Click the page number at the bottom of the report to view the next 15 records.

The Alert message **No Calls to Display** will appear, if there are no records to be displayed.



---

SETU VGRX offers users the following telephony features, which they can access by dialing Access Codes.

- Call Hold
- Making a Second Call
- Call Toggle
- Call Transfer - Attended and Blind
- Call Forward
- Conference
- DND
- Call Waiting
- Hotline
- Supplementary Services of Service Providers
- Making a new call using Access Code
- Disconnecting a call using Access Code

You can change the default access codes assigned to these features and facilities as per your requirement. See [“Access Codes”](#).

In addition to these, SETU VGRX offers users IP Dialing and Fax over IP (FoIP). It supports *Remote Call Forward*, *Remote Held*, and *Remote Transfer* using SIP Signaling.

## Call Hold

Call Hold enables you to put an on-going conversation on hold, and call another person, or receive a call from another person. You can retrieve the call you put on hold, after the conversation with the other party has ended. You can also retrieve the call you put on hold in the middle of the conversation with the other party.

Call Hold is also used in the following features:

- Retrieve Held Call
- Make a Second Call
- Call Toggle
- Call Conference
- Call Transfer
- Call Waiting

## Configuring Call Hold



*This feature will work only if,*

- *Call Hold is enabled in the **Class of Service** of the FXS Port*
- *The **Subscriber Type** of the FXS Port is configured as **Gateway**.*

For instructions on enabling features in the Class of Service of the FXS Port and configuring Subscriber Type, see [“FXS Port”](#) under *Basic Settings*.

## How to use Call Hold

- You are in speech with party A.
- To put your call with A on hold, dial **Flash**.
- A is put on hold. You get feature tone for 7 seconds, followed by error tone.
- To retrieve the call you put on hold, dial **Flash** again during feature tone or during error tone.
- You will be in speech with A.



- *If you go On-hook during the feature tone, your call with party A will be disconnected.*
- *If you go On-hook during error tone, you will get ring back.*
- *If you go Off-hook when your phone rings, you will get connected with party A again.*

# Making a Second Call

---

You can make a second call, by putting the current call on hold.



To provide this feature to users, **Call Hold** must be enabled in the **Class of Service** of the FXS Port. For instructions, see [“FXS Port”](#) under Basic Settings.

## How to make a Second Call

- You are in speech with party A.
- You want to talk to party B.
- Dial **Flash** to put party A on Hold.
- You get feature tone.
- Dial the number of the desired party B.
- When party B answers the call, you are in speech with party B.



Making a second call feature can also be used with other features such as Call Transfer-Attended, Call Toggle and 3-Party Conference.

For example, after making a second call, you can toggle between the first and the second call using the Call Toggle access code; you can also conduct a Conference with both parties, by dialing the Conference access code.

# Call Toggle

---

Call Toggle (Call Split) allows you to have two simultaneous telephone conversations, talking to two persons alternately.

The parties for Call Toggle can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.

You must dial the Call Toggle access code to switch between the held call and the active call. You can use Call Toggle only when there are two held calls on the FXS Port.

## Configuring Call Toggle



*This feature will work only if, Call Toggle is enabled in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see [“FXS Port”](#) under Basic Settings.*

## Using Call Toggle

- You are in speech with A and you want to talk to B.
- Dial **Flash**, A is put on hold. You get feature tone.
- Dial the number of B. When B answers the call, you are in speech with B.
- To talk to A, dial **Flash-#2**.
- You are in speech with A and B is put on hold.
- To talk to B, dial **Flash-#2** again.
- You are in speech with B. A is put on hold.
- This way, you can talk alternately to A and B, by dialing **Flash-#2** again.



- *When toggling between calls, you can disconnect the call you are currently in speech with, by going On-hook.*
- *You can also use Call Toggle (Call Split) during a Conference call.*

# Call Transfer - Attended

---

Attended Call Transfer is when you transfer the call to the desired party after consulting the party and or obtaining their consent for transfer.

In the case of SIP to SIP Attended Transfer, SETU VGRX enables the transferor to know the result of the transfer activity, whether the call has been transferred successfully or not.

As soon as the transferror goes On-hook or dials the Attended Transfer access code (if configured) to transfer the call, SETU VGRX loads the Transfer Notification Timer.

SETU VGRX notify the transferor about the result of the transfer activity within this Timer. The Transfer Notification Timer is configurable.

This is how Attended Transfer works:

- A (transferor) is in speech with B. A wants to transfer B's (transferee) call to C (transfer target).
- A dials Flash (Call Hold access code) to put B on hold and then dials C's number.
- A goes On-hook (the default Access Code for attended call transfer) while the C's number is ringing or after speech with C.
- B is in speech with C.



- *Attended Call Transfer is not allowed, if you dial Blind Transfer Access code after putting the remote held call on hold.*
- *In Attended Call Transfer, the Transfer Notification Timer does not stop even if the transferor goes On-hook.*
- *The default Attended Transfer access code ^ (On-hook) can be changed as per user requirement. If an access code other than ^ (On-hook) is assigned to Attended Transfer, then the transferror (A, in this case) must dial the Attended Transfer access code after dialing the transfer target's number (C's number).*

## Configuring Attended Transfer

To use this feature, you must enable **Attended Transfer** and **Call Hold** in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see [“FXS Port”](#) under *Basic Settings*.

If required, you may change the duration of the **Transfer Notification Timer**. The range of this timer is 001 to 999 seconds. By default it is set to 60 seconds. To change the duration of this timer, go to [“System Parameters”](#).

## How to use Attended Transfer

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial **Flash** to put party A on Hold.
- Dial Party B's Number.
- When Party B answers the call, go On-hook to transfer.
- You can go On-hook even while Party B's number is ringing.

# Call Transfer - Blind

---

Blind Transfer is when you transfer the call to the desired party, without informing the party of the transfer.

SETU VGRX enables the transferor to know the result of the transfer activity; whether the call has been transferred successfully or not. As soon as the transferor dials the Blind Call Transfer access code, SETU VGRX loads the Transfer Notification Timer. This timer stops if the transferor goes On-hook. SETU VGRX indicates to the transferor the result of the transfer activity within this Timer. The Transfer Notification Timer is configurable.

This is how Blind Call Transfer works:

- A (transferor) is in speech with B (transferee). A wants to transfer B's (transferee) call to C (transfer target), without informing C.
- A dials Flash (or Call Hold access code) to put B on hold and then dials Blind Call Transfer access code.
- A dials the number of C and B's call is transferred to C.

Now, to know the result of the Blind call transfer, A should remain Off-hook after dialing C's number. One of the following results may occur:

- If transfer is successful, A (transferor) gets confirmation tone for the duration of the Confirmation Tone Timer, followed by error tone for the duration of the Error Tone Timer, followed by system standby.
- If the transfer target C is busy, A (transferor) gets busy tone, and speech is established between A and the transferee B.
- If no message is received during the Transfer Notification Timer, A (transferor) gets error tone for Error Tone Timer, followed by system standby.



*Blind Call Transfer is not allowed if you dial the Blind Transfer access code after putting the remote held call on hold.*

## Configuring Blind Transfer

To use this feature, you must enable **Blind Transfer** in the **Class of Service** of the FXS Port. **Call Hold** must also be enabled in the Class of Service. For instructions on enabling features in the Class of Service of the FXS Port, see [“FXS Port”](#) under *Basic Settings*.

If required, you may change the duration of the **Transfer Notification Timer**. The range of this timer is 001 to 999 seconds. By default it is set to 60 seconds. To change the duration of this timer, go to [“System Parameters”](#).

## How to use Blind Transfer

- You are in speech with party A.
- Party A wishes to speak to party B.
- Dial **Flash**
- Dial **#6** Blind Transfer Access Code.
- Dial the number of party B.
- On successful transfer, you will get confirmation tone.
- Party A will get connected with party B.
- Replace the handset of your phone.



*When you transfer a call, remain Off-hook, i.e. do not replace your handset, after dialing the transfer destination's number, until you know the result of the call transfer. One of the following results may occur:*

- If the transfer is successful, you will get confirmation tone, followed by error tone.*
- If the transfer target is busy, you will get busy tone, and you will be connected to the transferee (party A, in this case).*
- If the transfer has timed out, i.e. the Notification Timer has expired and no notification has been received from the transfer target (party B, in this case) you will get error tone, followed by system standby.*

# Call Forward

---

Call Forward feature enables you to answer incoming calls on your extension, even when you are away from your phone. SETU VGRX forwards incoming calls on the FXS Ports to any other number as per your requirement.

SETU VGRX supports the following Call Forward options:

- **Call Forward-Unconditional:** All incoming calls received on the FXS Port are forwarded to the desired destination number, automatically without waiting for a response from your phone.
- **Call Forward- Busy:** All incoming calls received on the FXS Port are forwarded to the destination number, when your phone is busy.
- **Call Forward- No Reply:** All incoming calls received on the FXS Port are forwarded to the destination number, when you do not answer your phone.

When you set Call Forward-No Reply, SETU VGRX waits for the duration of the **Call Forward-No Reply Timer** before forwarding the calls. This timer is configurable, and is set to 45 seconds by default.

If the phone does not answer before this timer expires, SETU VGRX considers it as No Reply and forwards it to the desired destination number.

You can set a different destination number for each Call Forward option.



- *Call Forward-Unconditional has priority over Call Forward-Busy and Call Forward-No Reply.*
- *Call Forward-When Busy has priority over Call Waiting feature in all conditions except in call mature state, that is, when the FXS Port is in speech with another port.*
- *If both the Do Not Disturb (DND) and Call Forward-Unconditional features are set, Call Forward-Unconditional will have priority over Do Not Disturb (DND).*
- *Do Not Disturb (DND) has priority over Call Forward-No Reply and Call Forward- Busy.*

## Configuring Call Forward

To provide this feature to users, you must enable **Call Forward** in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see [“FXS Port”](#) under *Basic Settings*.

## How to use Call Forward

You can set/cancel Call Forward from Jeeves and from the phone connected to the FXS Port.

To set or cancel Call Forward from Jeeves,

- Log into Jeeves.
- Click the **Basic Settings** link.
- Click the **FXS Port** link.

- Click the FXS Port number tab, on which you want to use this feature. The page of the selected port opens.
- On the FXS Port page, click **Supplementary Services** link to expand.

- Select desired Call Forward option—**Call Forward-Unconditional**, **Call Forward-Busy**, **Call Forward-No Reply** to enable Call Forward option. Default: Disabled.

To cancel the Call Forward option you selected, clear the respective check box.

- For the Call Forward option you select, enter the desired destination number (upto 40 characters) to which you want to forward your calls, in the corresponding **Number** field that appears.
- If you have selected **Call Forward- No Reply**, you may also change the duration of the no-reply **Ring Timer**. The range of the Call Forward No-Reply Ring Timer is 01 to 99 seconds. Default: 15 seconds.
- Click **Submit** to save.

To set or cancel Call Forward from the Phone,

- Lift handset of your phone.

#### Call Forward - Unconditional

- Dial **#131-1** to set.
- Dial **#131-0** to cancel.
- Dial **#135-Destination Number-End-of-Dialing** to configure destination number.  
Where,  
Destination number can be of maximum 24 digits.  
End of Dialing may be # or \* as configured in the system.

#### Call Forward - Busy

- Dial **#132-1** to set.

- Dial **#132-0** to cancel.
- Dial **#136-Destination Number-End-of-Dialing** to configure destination number.  
Where,  
Destination number can be of maximum 24 digits.  
End of Dialing may be # or \* as configured in the system.

#### **Call Forward - No Reply**

- Dial **#133-1** to set.
- Dial **#133-0** to cancel.
- Dial **#137-Destination Number-End-of-Dialing** to configure destination number.  
Where,  
Destination number can be of maximum 24 digits.  
End of Dialing may be # or \* as configured in the system.
- Dial **#139-Timer** to set the Call Forward No Reply Ring Timer.  
Where,  
Timer value is from 1 to 99 seconds.  
Default: 15 seconds.
- Replace the handset.

# Conference

---

Three-party Conference, also referred to as Three-Way Calling, is a telephone call, in which you can have two other persons participate in the call.

You can initiate a conference by calling the first person, and then put the first person on hold to call the second person. You can also include another person when you in the middle of speech with a person.

The parties to a conference can be:

- Two outgoing calls
- Two incoming calls
- One outgoing call and one incoming call.



- *A 3-Party Conference can be converted to Call Toggle by dialing Call Toggle (Call Split) access code **Flash-#2**. User will be connected to one of the parties and the other party goes on hold.*
- *During conference, if user dials Flash, user will be in speech with last held party and the other party will go on hold.*
- *Conference is not allowed to the FXS Port, which is already in conference.*
- *Conference can be initiated only from the FXS Ports. To add a SIP party to a conference the PCM Companding Type selected on the Region page and the Codec selected for the SIP Trunk should be same.*

## Configuring Conference

To provide this feature to users, you must enable **Conference** in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see [“FXS Port”](#) under *Basic Settings*.

## How to use Conference

- Lift handset.
- Dial the number of party A.
- When A answers the call, dial **Flash**.
- Party A is put on hold, and you will hear feature tone.
- Now, dial the number of party B.
- When B answers the call and you are in speech with B, dial **Flash-#8**.
- Three-way speech will be established between you, party A and party B.
- Replace the handset at the end of your conversation.
- The conference will be terminated.

# Call Waiting

---

When your phone is busy, the Call Waiting feature notifies you about another incoming call in the form of beeps.

The Call Waiting feature of SETU VGRX allows you to:

- reject the waiting call.
- ignore the waiting call.
- hold the current call and answer the waiting call.
- disconnect the current call and answer the waiting call.

To use any of these options, you may dial the respective Access Code.



- *Call Waiting feature has priority over Call Forward-When Busy in call mature state i.e. when the FXS Port is in speech with another port.*
- *If your SETU VGRX has a 3G module, disable Call Waiting in the SIM Card before inserting it in the Mobile Port to prevent current calls from being disconnected.*
- *Call Waiting feature does not apply:*
  - *If Call Waiting feature is disabled.*
  - *If Waiting Call is ignored.*
  - *If already one Waiting Call is present.*
  - *In the Programming Mode.*
  - *In Conference.*
  - *In Remote Held condition.*
  - *In Dial state, Routing state and in Disconnect state.*

## Configuring Call Waiting

To be able to hear Call Waiting Beeps as indication, you must enable **Call Waiting** in the **Class of Service** of the FXS Ports. By default, this feature is disabled. For instructions on enabling features in the Class of Service of the FXS Ports, see [“FXS Port”](#) under *Basic Settings*.

## How to use Call Waiting

You can set/cancel Call Waiting from Jeeves and from the phone connected to the FXS Port.

To set or cancel Call Waiting from Jeeves,

- Log into Jeeves.
- Click the **Basic Settings** link.,
- Click the **FXS Port** link.
- Click the FXS Number tab on which you want to use this feature. The page of the selected port opens.

- On the FXS Port page, click **Supplementary Services** link to expand.

Supplementary Services	
Call Waiting	<input checked="" type="checkbox"/> Enable
Do Not Disturb(DND)	<input type="checkbox"/> Enable
Call Forward-Unconditional	<input type="checkbox"/> Enable
Call Forward-Busy	<input type="checkbox"/> Enable
Call Forward-NoReply	<input type="checkbox"/> Enable
Hotline	<input type="checkbox"/> Enable

- To set Call Waiting, select the **Call Waiting Enable** check box. Default: Disabled.

To cancel, clear the check box.

- Click **Submit** to save.
- Log out of Jeeves.

To set or cancel Call Waiting from the Phone,

- Lift handset of your phone.
- Dial **#16-1** to set.
- Dial **#16-0** to cancel.
- Replace handset.

When Call Waiting is enabled and you are in speech with Party A, you will hear call waiting beeps indicating another incoming call from Party B. You may Reject, Ignore or Accept the Waiting Call by dialing the relevant Access Code.

- To *Reject the waiting call*, dial **Flash-#31**.

The beeps will stop, you will remain in speech with Party A.

OR

- To *Ignore the waiting call*, dial **Flash-#32**.

The beeps will continue, you will remain in speech with Party A.

OR

- To *Put current call on hold and accept the waiting call*, dial **Flash-#33**.

The beeps will stop, Party A will be put on hold and you will be in speech with Party B.

OR

- To *Disconnect current call and accept the waiting call*, dial **Flash-#34**.

The beeps will stop, Party A will be disconnected and you will be in speech with Party B.

# Message Wait Indication on SIP Trunks

---

SETU VGRX supports Message Wait Indication (MWI) on SIP Trunks for Voicemail service subscribed from ITSPs.

If you have subscribed Voicemail service from the ITSP of a SIP Trunk, you can subscribe Message Wait Indication for that SIP Trunk to get notification for new messages in your mailbox on the phone connected to an FXS Port.

You can retrieve messages from the phone connected to the FXS Port by dialing an access code. See [“How to Retrieve Messages”](#) at the end of this topic.

You can also view the status of MWI on the SIP Trunks Status page on Jeeves. See [“How to view Status of Message Wait Indication”](#) later in this topic.

To be able to use Message Wait Indication (MWI) for the voicemail service of the ITSP, you must do the following configuration on the SIP Trunk and FXS Port:

## On the SIP Trunk,

- Enable **Subscribe for MWI** check box.
- Configure the **Message Retrieval Number** provided to you by your ITSP. This number is used for retrieval of voicemail on the SIP Trunk. The Message Retrieval Number may consist of a maximum of 24 characters. Valid range is 0 to 9, \* and # are allowed. Default: Blank.
- Configure the **Authentication ID** and **Authentication Password** provided by your ITSP. Default: Blank.
- In **Send Message Notification on**, select the FXS Port on which Message Wait Indication is to be sent whenever there is a new message on the SIP Trunk. Default: FXS Port 1.
- Assign the FXS Port on which you want to receive MWI notification from the SIP Trunk, as the destination port to **Send MWI Notification on**. Whenever a new message arrives in the mailbox of the SIP Trunk, SETU VGRX gives notification to the FXS Port you have selected as destination, according to the type of Message Wait Notification you have selected on the FXS Port.
- If you have completed the configuration of SIP Trunk-1, click **Submit** to save settings.

## On the FXS Port selected as Destination for MWI,

- If you have subscribed to *Message Wait Indication* for the voicemail service from your ITSP, and have selected this FXS Port as the destination for receiving Message Wait Indication<sup>7</sup>, you may select the desired type of **Message Wait Notification** from the following options.
  - Select **Stuttered Dial Tone**, if you want new message indication in the form of a stuttered dial tone, whenever the user picks up the phone connected to the FXS Port.
  - Select **LED Lamp (HV)**, if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using High Voltage.
  - Select **Ring**, if you want the arrival of a new message to be indicated by the *Message Wait Ring* (a Short, Fast ring).

---

7. You have selected the number of this FXS Port for the **Send Message Notification on** parameter, under **MWI Parameters** you configured on the SIP Trunk.

You can select a different Ring Type to indicate message wait. For instructions, see [“Message Wait Ring Type”](#).

You can also set the duration for which the ring is to be played (Ring Timer), the number of times the ring is to be played (Ring Count) and the interval between rings (Ring Interval). For instructions, see [“Message Wait”](#) in [“System Parameters”](#).

- Select **LED Lamp (FSK)**, if the phone connected to the FXS Port is equipped with a 'Message Wait' lamp and you want new messages to be indicated on this LED lamp using FSK CLI.
- Select **Stuttered Dial Tone + LED Lamp (HV)**, if you want new message indication on the LED Lamp using High Voltage as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.
- Select **Stuttered Dial Tone + LED Lamp (FSK)**, if you want new message indication on the LED Lamp using FSK CLI as well as in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

Default: LED Lamp (HV)

Whenever a new message arrives in the Mailbox of the SIP Trunk, SETU VGRX gives notification to this (destination) FXS Port according to the type of *Message Wait Notification* you select.

## How to view Status of Message Wait Indication

You can view the status Message Wait Indication, that is, new messages, old messages, urgent old and new messages on the SIP Trunk Status page of Jeeves. For more information, in the *Status* chapter, under [“SIP Trunk”](#), see [“MWI Status”](#).

## How to Retrieve Messages

You can retrieve messages by dialing the access code for *Using Voice Mail of the Service Provider* **#7** (default) from the phone connected to the FXS port designated as destination for MWI. This access code is configurable. So, you can also change it to the desired value. For instructions, see [“Access Codes”](#).

Message retrieval works as follows:

- Dial the Using Voice Mail of the Service Provider Access Code, **#7** from the phone connected to SETU VGRX.
- SETU VGRX checks whether the FXS Port to which the phone is connected is configured as the destination for Message Wait Notification (**Send MWI Notification on**) for any SIP Trunk.
- On finding the FXS Port as the destination for Message Wait Notification for the SIP Trunk, SETU VGRX dials out the Message Wait Retrieval Number configured for the SIP Trunk.
- The FXS Port gets connected to the Voicemail server of the ITSP.
- You can follow the voice mail prompts to retrieve your messages.



If Message Wait Notification is enabled on more than one SIP Trunk and you have configured same FXS Port in '**Send MWI Notification on**', you are recommended not to use Access Code to retrieve the Voice Mail messages. Instead, do the following:

- Configure the Destination Number Based table. In the Destination Number Based table, enter the **Message Retrieval Number** as **Destination Number** and select the respective SIP Trunk port as **Routing Group**.
- Retrieve your messages, by dialing the Message Retrieval Number of that SIP Trunk from the FXS Port.
- You will be connected to the Voice Mail server of the ITSP of that SIP Trunk.

# Call Pick-up

---

Call Pick-up feature allows you to answer calls ringing on other extension phones connected to SETU VGRX from your own phone by dialing an access code (default: **#5**); without physically going over to the ringing phone to answer it.

As you can answer calls of your colleagues or co-workers without physically going to their desks, this feature ensures that all incoming calls are answered.

SETU VGRX stores the FXS Port number from where the call was answered as the *Destination port* in the Call Detail Record (CDR). To know more, see [“Call Detail Record”](#).



*You cannot pick up RCOC calls ringing on extensions in the same Call Pick-up group.*

For this feature to work,

- You must form a Call Pick-up group, consisting of the FXS Ports.

Calls ringing on a phone connected to the FXS Port within a group can be picked up from another phone in the same group by dialing the Call Pick-up access code (programmable; default: **#5**).

- You must enable **Call Pick-up** feature in the Class of Service of the **FXS Ports** included in the Call Pick-up group.

## Configuring Call Pick-up

To use this feature,

- assign the FXS Ports to a Call Pick-up Group. For instructions, see [“General”](#), under the [“FXS Port”](#) topic in the *Basic Settings*.
- enable **Call Pick-up** in the [“Class of Service”](#) of the [“FXS Port”](#) you have assigned to Call Pick-up groups.

You can also change the default access code for Call Pick-up, **#5**, to the desired value, if required. For instructions, see [“Access Codes”](#) in the *Advanced Settings*.

## How to use Call Pick-up

To pick up calls ringing on the phones in your Call Pick-up group,

- Lift the handset of your phone.
- Dial **#5**
- You are in speech with the caller.
- Talk.
- Replace the handset to disconnect.

# Do Not Disturb (DND)

---

If you do not want to receive any calls on your phone, you may set the Do Not Disturb (DND) feature on your phone. When you set DND, all incoming calls on your phone will be rejected. However, you can make the outgoing calls.



- If both the Do Not Disturb (DND) and Call Forward-Unconditional features are set, Call Forward-Unconditional will have priority over Do Not Disturb (DND).
- Do Not Disturb (DND) has priority over Call Forward-No Reply and Call Forward- Busy.

## Configuring DND

To provide this feature to users, you must enable **Do Not Disturb** in the **Class of Service** of the FXS Port. For instructions, see “[Class of Service](#)” under “[FXS Port](#)” in the *Basic Settings* chapter.

## How to use DND

You can set/cancel DND from Jeeves and from the phone connected to the FXS Port.

To set or cancel DND from Jeeves,

- Log into Jeeves.
- Click the **Basic Settings** link.
- Click the **FXS Port** link.
- Click the FXS Number tab, on which you want to use this feature. The page of the selected port opens.
- On the FXS Port page, click **Supplementary Services** link to expand.

Supplementary Services	
Call Waiting	<input checked="" type="checkbox"/> Enable
Do Not Disturb(DND)	<input type="checkbox"/> Enable
Call Forward-Unconditional	<input checked="" type="checkbox"/> Enable
Call Forward-Busy	<input checked="" type="checkbox"/> Enable
Call Forward-NoReply	<input checked="" type="checkbox"/> Enable
Hotline	<input checked="" type="checkbox"/> Enable

- To set DND, select the **Do Not Disturb (DND) Enable** check box. Default: Disabled.

To cancel, clear the check box.

- Click **Submit** to save.
- Log out of Jeeves.

To set or cancel DND from the Phone,

- Lift handset of your phone.
- Dial **#18-1** to set.
- Dial **#18-0** to cancel.
- Replace handset.

# Hotline

---

Using Hotline feature, you can connect immediately to a particular number after going Off-hook. You can use Hotline feature to dial the number, you call most frequently.

Let us understand how this feature works with an example,

- A is Sales Manager frequently dials the number of the B, the Sales Coordinator.
- Instead of dialing B's number each time, A sets Hotline and configures B's number as Hotline Number.
- A can also set the Hotline Timer, i.e. the time after which B's number should be dialed out. By default, the Hotline Timer is set to 5 seconds.
- Whenever A goes Off-hook, SETU VGRX plays feature tone and waits for the duration of the Hotline Timer.
- If A does not dial any digit before the expiry of this Timer, SETU VGRX dials out B's number.



*Allowed-denied number logic is applied for the Hotline number.*

## Configuring Hotline

To provide this feature to users, you must enable **Hotline** in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see "[FXS Port](#)" under *Basic Settings*.

## How to use Hotline

You can set/cancel Hotline from Jeeves and from the phone connected to the FXS Port.

To set or cancel Hotline from Jeeves,

- Log into Jeeves.
- Click the **Basic Settings** link.
- Click the **FXS Port** link.
- Click the FXS Number tab on which you want to use this feature. The page of the selected port opens.

- On the FXS Port page, click **Supplementary Services** link to expand.

Supplementary Services	
Call Waiting	<input type="checkbox"/> Enable
Do Not Disturb(DND)	<input type="checkbox"/> Enable
Call Forward-Unconditional	<input type="checkbox"/> Enable
Call Forward-Busy	<input type="checkbox"/> Enable
Call Forward-NoReply	<input type="checkbox"/> Enable
Hotline	<input checked="" type="checkbox"/> Enable
Number	<input type="text"/>
Timer	<input type="text" value="5"/>

- Select the **Hotline Enable** check box. Default: Disabled.  
To cancel, clear the check box.
- In **Number**, enter the Number to be dialed out from the port when you go Off-hook. Default: Blank.
- Set the **Timer** after which the Hotline Number is to be dialed out after you go Off-hook. The range of this timer is 1 to 9 seconds. Default: 5 seconds.
- Log out of Jeeves.

To set or cancel Hotline from the Phone,

- Lift handset of your phone.
- Dial **#151-1** to set.
- Dial **#151-0** to cancel.
- Dial **#152-Destination Number-End-of-Dialing** to configure Hotline Number.  
Where,  
Destination number can be of maximum 24 digits. Digits 0 to 9, \*, #, dot (.) are allowed.  
End of Dialing may be # or \* as configured in the system.
- Dial **#153-Timer** to set the Hotline Timer.  
Where,  
Timer value is from 1 to 9 seconds. Default: 5 seconds.
- Replace handset.

# Forced Release Radio Port

---

Forced Release Radio Port enables you to disconnect a busy Radio Port at will, and free the system resources (access to Radio Port) for yourself.

## Configuring Forced Release Radio Port



*This feature will work only if, **Forced Release Radio Port** is enabled in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see [“FXS Port”](#) under *Basic Settings*.*

## Using Forced Release Radio Port

- Radio Port 2005 is in speech with FXS Port 2001.
- You dial 2005 and get busy tone.
- Go On-hook and then go Off-hook, dial **#93**, followed by 2005 (that is the Radio Port number).
- The call between 2005 and 2001 is disconnected.
- Radio Port 2005 is now free and can be accessed.

# Supplementary Services of Service Provider

---

When the SETU VGRX is interfaced with a service provider server (ITSP, VoIP-PSTN adapter, the Matrix ETERNITY IP-PBX, or any other PBX) that supports supplementary services like Call Hold, Call Transfer, Call Waiting that require dialing of Flash<sup>8</sup>, you may choose to access the features of the service provider, or to access primarily the features of SETU VGRX.

If you want to use the features of the service provider, you must select **Network** as the *Subscriber Type* for the FXS Port of SETU VGRX.

If you want to use the features of SETU VGRX, you must select **Gateway** as the Subscriber Type for the FXS Port of SETU VGRX. However, you will be able to access the features of the service provider also in the Gateway mode, by dialing Flash, followed by the Access Code **#4**, for using Supplementary Services of Service Provider.

When the SETU VGRX is set in the Gateway mode, when you dial flash and the access code **#4** to access the supplementary services of the service provider, the active call will be put on hold and you will get a feature tone. SETU VGRX will start the internal timer, called the **Service Provider Access Code Wait Timer**. This timer has a duration of 10 seconds. Any activity performed on the FXS Port within this timer will be sent to the service provider server. Any activity performed on the FXS Port after the expiry of this timer, will be processed by SETU VGRX.

## Configuring Supplementary Services of Service Provider

To use Supplementary Services of Service Provider, configure the parameter **Subscriber Type** on the FXS Port as **Network** or as **Gateway**. For instructions on configuring Subscriber Type, see “[FXS Port](#)” under *Basic Settings*.

## How to use Supplementary Services of Service Provider

If SETU VGRX is set in the **Network Mode**,

- Dial **Flash** during speech.  
You will get feature tone played by your service provider.
- Dial feature access code provided by the service provider's server.

If SETU VGRX is set in the **Gateway Mode**,

- Dial **Flash** during speech. This will place the active call on hold and you will get a feature tone.
- Dial **#4**.  
You will be in Network mode for 10 seconds.
- Dial the feature access code provided by the service provider's server within 10 seconds.

---

8. To be able to use features of the service provider, SETU VGRX supports dialing of Flash during speech on SIP.

# Making a New Call using Access Code

---

This feature enables you to disconnect the current call and make a new call using SETU VGRX without getting disconnected from the system.

Let us understand this feature with an example:

- A Cyber-Cafe has installed SETU VGRX for providing international calling service to its customers.
- The Cyber-Cafe provides a number to call the SETU VGRX, a PIN Number and a Password to its customers for using this service.
- One of the customer, Mr. A has subscribed for this services.
- To make international calls, A must call SETU VGRX, dial the PIN Number and Password and then dial the international number. Thus, each time he wants to make a new call, he must repeat this process.
- Making a New Call feature eliminates repeated dialing of these numbers.
- After calling SETU VGRX, A must dial his PIN number and Password once and then dial the international number. At the end of the call, he can dial the Access Code for Making a New Call to remain connected to the system and can make another call.
- If the remote end disconnects the call during speech, SETU VGRX will play error tone for 4 seconds followed by dial tone. Mr. A can make a new call without dialing the feature Access Code.



- *This feature is applicable only on the Source Port and only when **After Answering the Call and Collecting the Digits** is selected as the option to Route all Incoming Calls (with CLI). Making New Call Access Code dialed by users will be ignored if any other option is selected to Route all Incoming Calls (with CLI).*
- *However, if you have enabled **Connect Source Port when number is outdialed** on the FXO Port or Mobile Port or have enabled **Connect Source Port when 183 (Session Progress) is received on SIP** on the SIP Trunk, you will not be able to provide this feature to the users.*

## Configuring Making a New Call

To provide this feature to users, you must enable **Allow making New Call using Access code** on the SIP Trunks, Mobile Ports and FXO Ports. For instructions, under [“Basic Settings”](#), see [“FXO Port”](#), [“SIP Trunk”](#), and [“Mobile Port”](#).

## How to make a New Call using Access Code

- When you are in speech during the current call.
- Dial **#91**.
- Disconnect the current call.
- Dial the new number you want to call.
- While in speech, dial **#91** again to make another call.

# Disconnecting a Call using Access Code

---

SETU VGRX enables you to disconnect a call using an access code. When the call disconnect access code is dialed, SETU VGRX releases the port engaged in the call.



*Radio Port user will be able to use this feature, only if the Radio devices supports dialing of DTMF digits.*

## Configuring Call Disconnection using Access Code

To provide this feature to users, you must enable **Allow Call Disconnection using Access code** on the SIP Trunks, FXO Ports and Mobile Ports. For instructions, under [“Basic Settings”](#), see [“FXO Port”](#), [“SIP Trunk”](#), and [“Mobile Port”](#).

## How to Disconnect a Call using Access Code

- Dial **#92**, when you are in speech or are at the end of the current call.

# IP Dialing

---

SETU VGRX supports direct dialing of IP Addresses from the source port. You must configure a SIP Trunk or a SIP Group for IP Dialing.

When a number is dialed out from the source port, SETU VGRX routes the call to the desired destination as per the routing mechanism configured for that port. However, when an IP Address is dialed from the source port of SETU VGRX, the system does not check the Destination Port Determination method configured for that port, instead it routes the dialed IP Address through the SIP Trunk or SIP Group configured for IP Dialing.

When dialing an IP Address, users must press \* key (star/asterisk) in place of. (dot/period) in the IP Address.

For example, to call the IP Address **192.167.100.1**, users must dial **192\*167\*100\*1** or **192\*167\*100\*001**

SETU VGRX interprets the \* dialed as a '.' (dot/period).

## Configuring IP Dialing

To provide this feature,

- select a SIP Trunk or a SIP Group through which the dialed IP Addresses are to be routed in the **SIP Trunk for IP Dialing**. By default, SIP Group 1 is selected for IP Dialing in the System Parameters. See [“General Parameters”](#) in the [“System Parameters”](#).

If you want to use a SIP Trunk group for IP Dialing, you must configure a SIP Group first. This Group is common for all port types. For instructions, see the topic [“Group”](#).

When you assign a SIP Trunk, make sure it is enabled and has the necessary configuration done. For instructions, see [“SIP Trunk”](#) under *Basic Settings*.



*If the SIP Trunk Group for IP Dialing is programmed as 'None', SETU VGRX will give error tone to the caller and the call will be rejected.*



---

## Firmware Upgrade

You can upgrade Firmware of SETU VGRX:

1. From a Provisioning Server
2. From a Personal Computer

### Firmware Upgrade from Provisioning Server

#### Auto Firmware Upgrade

Using Auto-Firmware Upgrade, SETU VGRX can automatically upgrade its firmware by downloading the firmware files stored at a central location: HTTP Server or HTTPS Server or Provisioning Server.

This feature is useful for ITSPs that have Provisioning Servers to store the firmware files. ITSPs can update the firmware of SETU VGRX provided to their customers from a centralized location without physically visiting the customer premises.



*For the **Auto Firmware Upgrade File**, contact Matrix Support Team.*

To perform Auto-Firmware Upgrade,

1. ITSPs must store the following Auto Firmware Upgrade files of SETU VGRX on the Provisioning Server.
  - matrix\_firmware.html file
  - SETU VGRX\_VwRx.y.z.Zip file
2. The following parameters must be configured in the SETU VGRX.
  - IP Address of the Provisioning Server.
  - Path of the Folder (containing the firmware files) on the Provisioning Server.
  - The protocol to be used: HTTP, HTTPS.
3. When SETU VGRX 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 SETU VGRX decide which firmware it should upgrade to.

- After SETU VGRX 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, SETU VGRX will upgrade its current firmware with the new firmware without the intervention or assistance of a technician.

The table below describes a few possible cases and the corresponding action taken by SETU VGRX.

Version-Revision of your SETU VGRX	Version- Revision in the matrix_firmware.html file received from the Provisioning Server	Action Taken by SETU VGRX
V1R5.1.0	V1R4.1.0	SETU VGRX will downgrade its current firmware with V1R4.1.0
	V1R5.1.0	SETU VGRX will discard the upgrade process as same Version/Revision is found.
	V1R6.1.0 and V1R7.1.0	SETU VGRX will upgrade its current firmware with V1R7.1.0
	V1R4.1.0 and V1R5.1.1	SETU VGRX will upgrade its current firmware with V1R5.1.1
	V2R2.1.0_V2R1.1.0, V2R1.1.0 and V1R8.1.0	Highest Version/Revision available is V2R2.1.0, however, V2R2.1.0 has a benchmark of V2R1.1.0. Therefore, SETU VGRX will first upgrade with V2R1.1.0 and then with V2R2.1.0.

To configure Auto Firmware Upgrade parameters,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **Firmware** link.

**Firmware**

Auto Firmware Upgrade

☐ Enable

Protocol for Auto Firmware Upgrade

☒ HTTP ☐ HTTPS

Server Address:Port

:

Firmware Folder Path

Upgrade Firmware Automatically at every Power ON

☐ Yes

Upgrade Firmware Automatically at Scheduled time

☐ Yes

Schedule Time

☐ Every  Minutes  
☐ Everyday at time  :   
☒ Every Month on Date  at time  :

Request Timeout

Seconds

Upgrade Firmware from Server

Upgrade Firmware from PC

Check Firmware Available On Server

☒ Submit

☐ Default

- Auto Firmware Upgrade:** Select the check box to enable Auto Firmware Upgrade. Default: Disabled.

- **Protocol for Auto Firmware Upgrade:** Select the protocol to be used by the Provisioning Server to upgrade the firmware of SETU VGRX. SETU VGRX generates file transfer request to the server according to the protocol you select. You may select **HTTP** or **HTTPS**. Default: HTTP.
- **Server Address: Port:** Enter the IP Address/Domain and Port of the Provisioning Server on which the firmware files of SETU VGRX are stored.

The Provisioning Server Address can also be obtained by SETU VGRX using DHCP (using Option 224). To fetch Provisioning Server Address using DHCP, keep the Server Address: Port field blank.

Make sure that you also set the *Connection Type* on the “[Network Parameters](#)” page to *DHCP*.

The default Port differs as per the protocol you select. For HTTP, the Default Port is 80 and for HTTPS, the Default Port is 443. You can also change the port as per your requirement. Valid Port Range: 80, 443, 1031 to 65534.

- **Firmware Folder Path:** Specify the path of the folder on the Provisioning Server where the firmware files are stored. Default: Blank.
- **Upgrade Firmware Automatically at Every Power ON:** Enable this check box, if you want SETU VGRX to check for updates in the firmware at each power ON.



- *At Power ON, if both Auto-Firmware upgrade and Auto-Configuration upgrade is enabled, Auto-Firmware upgrade has priority over Auto-Configuration upgrade.*
- *While upgrading itself, if SETU VGRX has to upgrade itself with the benchmark firmware first then it is recommended that you select **Upgrade Firmware Automatically at Every Power ON**.*
- **Upgrade Firmware Automatically at Scheduled Time:** Enable this check box, if you want SETU VGRX to check for updates in the firmware at a scheduled time. You may select any one of the following schedule options:
  - **Every XX minutes:** The minutes after which SETU VGRX should check for firmware updates.
  - **Everyday at HH:MM:** The time in **Hours(00-23)** and **Minutes(00-59)** when SETU VGRX should check for firmware updates everyday.
  - **Every Month on DD at HH:MM:** The **Date (01-31)** and Time in **Hours (00-23)** and **Minutes(00-59)** when SETU VGRX should check for firmware updates every month.



*If SETU VGRX has to upgrade itself with the benchmark firmware and you have selected **Upgrade Firmware Automatically at Scheduled Time**, SETU VGRX will first upgrade itself with the benchmark firmware. At the subsequent scheduled time, it will upgrade itself with the final firmware.*

- **Request Timeout:** Request Timeout is used when SETU VGRX tries to connect to the Provisioning Server for TCP/TLS binding. This timer specifies for how long SETU VGRX should wait for successful TCP/TLS binding.

Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

If SETU VGRX fails to connect to the Provisioning Server, it will make 10 attempts at a regular interval of 10 seconds between each attempt to establish the binding. Even then, if it is unable to establish the binding, it will abort the Auto upgrade process.

- Click **Submit** to save.
- To view the status of Auto-Firmware Upgrade from Jeeves, see “[Firmware](#)” under “[Status](#)” Chapter.

## Manual Firmware Upgrade

You can manually upgrade Firmware of SETU VGRX, whenever you want.

To manually upgrade firmware of SETU VGRX from server,

- Click the **Upgrade Firmware from Server** button on the Firmware page. SETU VGRX will automatically upgrade its firmware with the latest firmware available on the server.

## Checking Firmware Availability

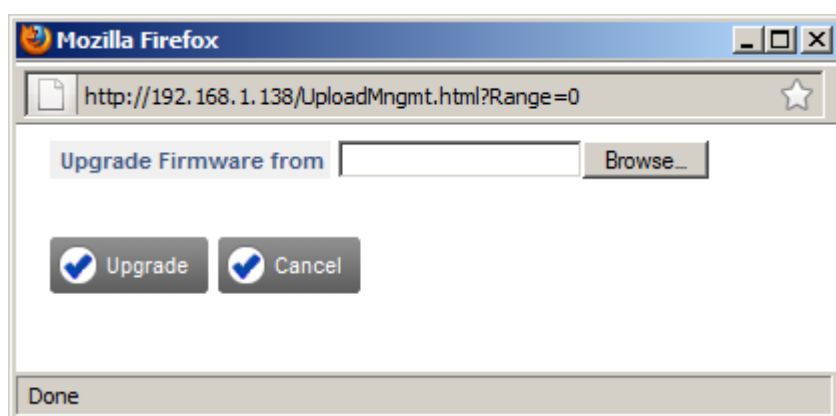
You can check the firmware files available on the server and then decide whether you want to upgrade SETU VGRX. Before upgrading Firmware from server, you can also choose the firmware with which you want to upgrade your SETU VGRX.

- To view the firmware files available on the Server, click the **Check Firmware Available on Server** button.
- A list of Firmware files available on the server appears in a new window.
- If you want to upgrade SETU VGRX with the desired Firmware, select the Firmware and click the **Submit** button.
- SETU VGRX will upgrade itself with the firmware you select.

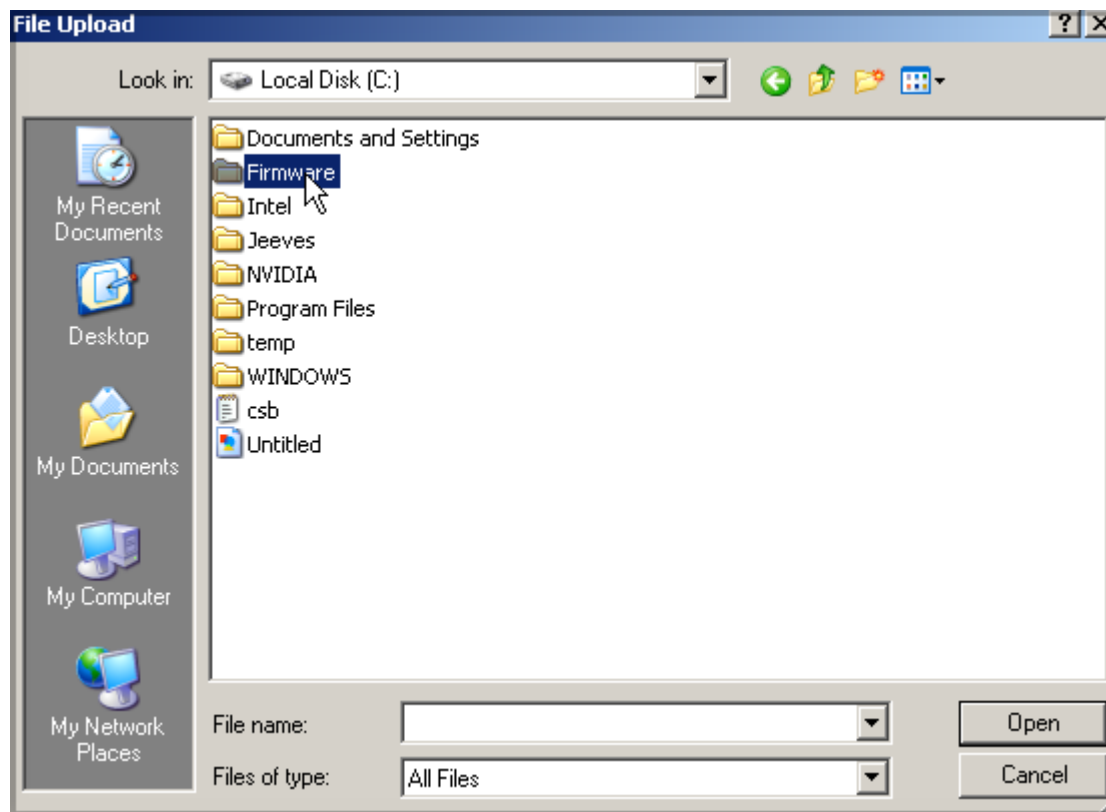
## Firmware Upgrade from Personal Computer

You can also upgrade firmware of SETU VGRX with the firmware files stored on your computer. To do so,

- Click the **Upgrade Firmware from PC** button. A new window - **Firmware Upgrade From** opens.
- Click the **Browse** button 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.



- Click the **Upgrade** button.

# Configuration Upgrade

---

You can upgrade Configuration of SETU VGRX:

1. From the Auto Configuration Server
2. From a Personal Computer

## Upgrading Configuration from the Auto Configuration Server

### Auto Configuration Upgrade

Using Auto-Configuration, SETU VGRX can automatically download the configuration files stored at a central location: Auto Configuration Server (ACS).

This feature is useful for ITSPs that have deployed a large number of SETU VGRX. ITSPs can store the configuration files of each SETU VGRX that they have provided to their customers on the Auto Configuration Server (ACS).



*For the **Auto Configuration File**, contact Matrix Support Team.*

To perform Auto Configuration,

1. Make sure that the configuration file of SETU VGRX is stored on the Auto-Configuration Server (ACS).
2. To ensure security, ITSP can encrypt the configuration file stored on the ACS. If the ITSP has encrypted the configuration file, the password to decrypt the file must be provided to you.
3. The following parameters must be configured in the SETU VGRX.
  - IP Address of the Auto Configuration Server (ACS).
  - 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, HTTPS.
4. When SETU VGRX installed at a customer site connects to the ITSP network, it will automatically download its configuration file stored on the Auto-Configuration Server (ACS), without the intervention or assistance of a technician.

To configure Auto Configuration parameters,

- Log into Jeeves.
- Click the **Maintenance** link.

- Click the **Configuration** link.

**Configuration**

**Auto Configuration Upgrade** ☒ Enable

**Protocol for Auto Configuration Upgrade** ☐ TFTP ☒ HTTP ☐ HTTPS

**Server Address:Port**  :

**Configuration Folder Path**

**Upgrade Configuration Automatically at every Power ON** ☐ Yes

**Upgrade Configuration Automatically at Scheduled time** ☐ Yes

**Schedule Time**

☒ Every  Minutes

☐ Everyday at time  :

☐ Every Month on Date  at time  :

**Request Timeout**  Seconds

**Password to Decrypt Configuration File**

- **Auto Configuration Upgrade:** By default, this check box is enabled. You may clear this check box, if required.
- **Protocol for Auto Configuration Upgrade:** Select the protocol used by the Auto Configuration Server to upgrade the configuration. SETU VGRX generates file transfer request to the Auto-Configuration Server according to the protocol you select. You may select **TFTP**, **HTTP** or **HTTPS**. Default: HTTP.
- **Server Address: Port:** Enter the IP Address/Domain and the Port of the Auto Configuration Server on which the configuration files of SETU VGRX are stored.

The Auto Configuration Server Address can also be obtained by SETU VGRX using DHCP (using Option 224). To fetch Auto Configuration Server Address using DHCP, keep the Server Address: Port field blank.

Make sure that you also set the *Connection Type* on the [“Network Parameters”](#) page to *DHCP*.

The default Port differs as per the protocol you select. For TFTP, the Default Port is 69. 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: 69, 80, 443, 1031 to 65534.

- **Configuration Folder Path:** Specify the path of the folder on the Auto Configuration Server where the configuration file is stored. Default: Blank.
- **Upgrade Configuration Automatically at Every Power ON:** Enable this check box, if you want SETU VGRX to check for updates in the configuration file at each Power ON.



*At Power ON, if both Auto-Firmware upgrade and Auto-Configuration upgrade is enabled, Auto-Firmware upgrade has priority over Auto-Configuration upgrade.*

- **Upgrade Configuration Automatically at Scheduled Time:** Enable this check box, if you want SETU VGRX to check for updates in the configuration at a scheduled time. You may select any one of the following schedule options:
  - **Every XX minutes:** The minutes after which SETU VGRX should check for configuration updates.
  - **Everyday at HH:MM:** The time in **Hours(00-23)** and **Minutes(00-59)** when SETU VGRX should check for configuration updates everyday.
  - **Every Month on DD at HH:MM:** The **Date (01-31)** and Time in **Hours (00-23)** and **Minutes(00-59)** when SETU VGRX should check for configuration updates every month.
- **Request Timeout:** Request Timeout is the time for which SETU VGRX will try to connect to the Auto Configuration Server for TCP/TLS binding using HTTP or HTTPS. This timer specifies for how long SETU VGRX should wait for successful TCP/TLS binding.

Enter the required time in seconds. The range of Request Timeout is 01-99 seconds. Default: 60 seconds.

If SETU VGRX fails to connect to the Auto-Configuration Server, it will make 10 attempts at a regular interval of 10 seconds to establish the binding. Even then, if it is unable to establish the binding, it will stop retrying and wait for next event of Auto-Configuration upgrade.

- **Password to Decrypt Configuration File:** Enter the Password as provided by your ITSP to decrypt the configuration file. During Auto-Configuration, if SETU VGRX receives an encrypted configuration file, it will decrypt the file using this password.

The password may consist of 40 characters (maximum). Default: Blank.



*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.
- To view the status of Auto-Configuration upgrade from Jeeves, see [“Configuration”](#) under [“Status”](#) Chapter.

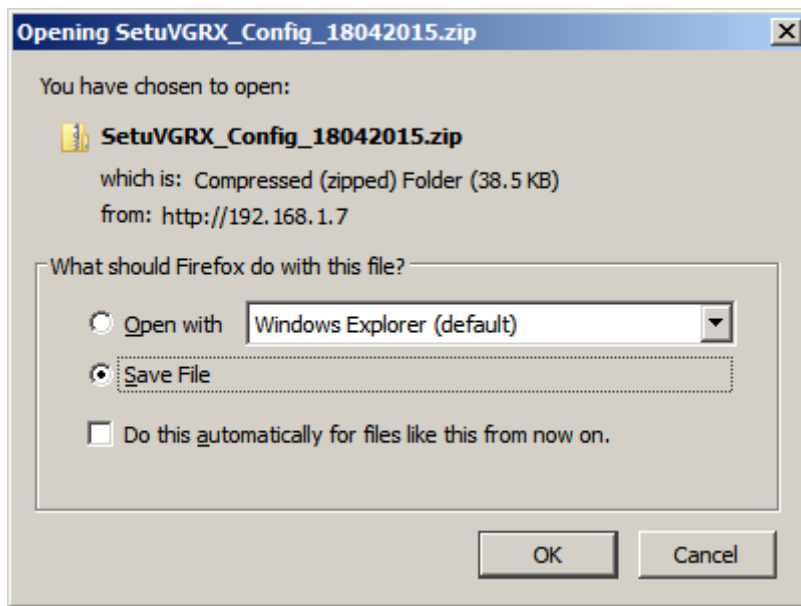
## Manual Configuration Upgrade

To manually upgrade configuration of SETU VGRX, click the **Upgrade Configuration from Server** button.

## Backup Configuration

- To save the existing configuration files as backup, click the **Backup Configuration** button.

A **Opening config.zip** window will open.

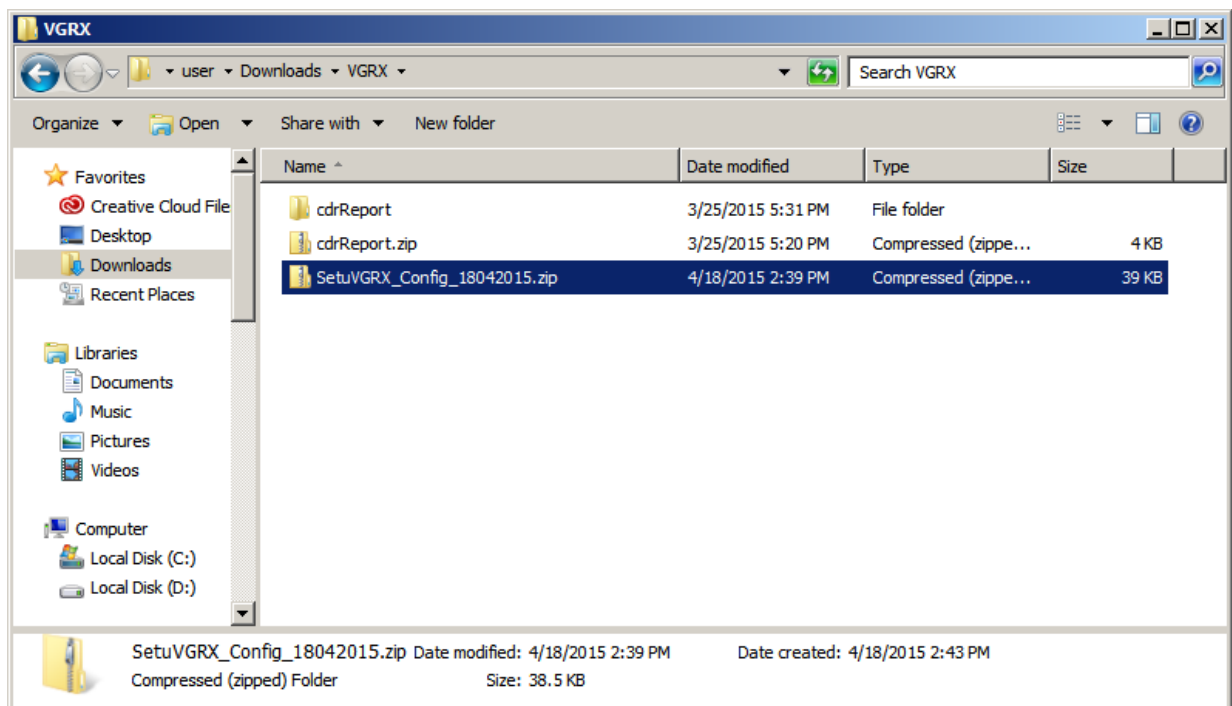


- You can either open the **config.zip** file or save the file to a location.



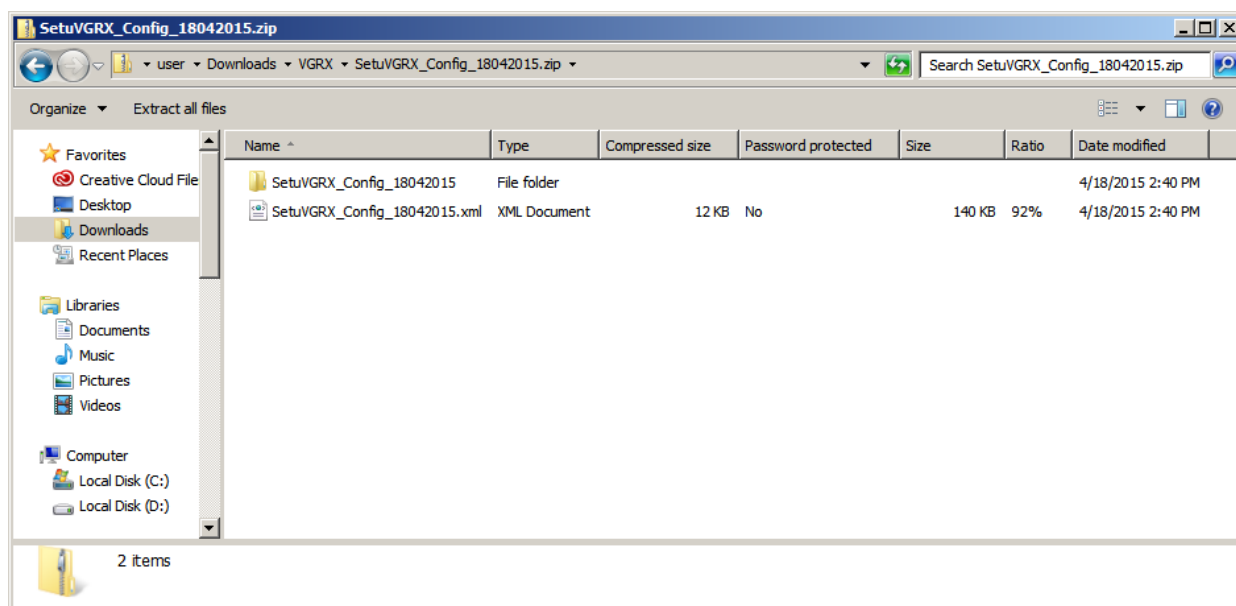
*If you are using Mozilla Firefox (version 3.5 recommended), before you save the configuration files, set the **Downloads** option of your browser as **Always ask me where to save the files**.*

- Save the file on the local disk.

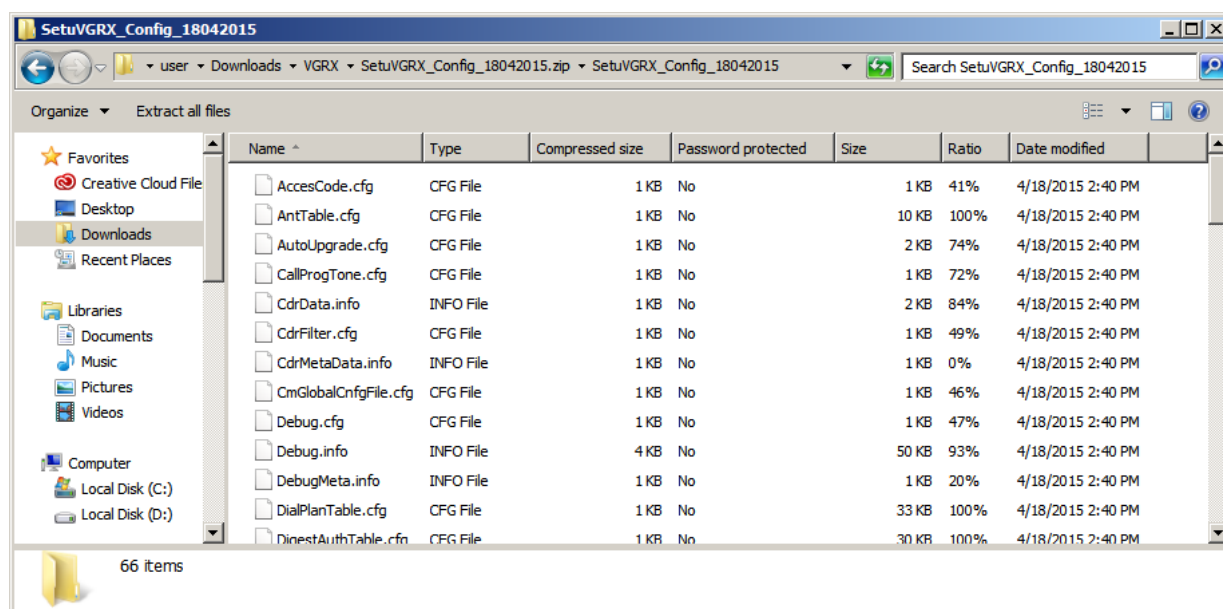


*Save the back up configuration files by tagging the file name with the Version-Revision of the Firmware and tag the name of the backup folder on your computer with the date. This will help you at the time of restoring the back up configuration files.*

- Open the configuration file (.zip) from the location you saved.



- The zip file contains all the system configuration files in .cfg format and xml format. You cannot edit the configuration files in .cfg format, however you can edit the configuration files in xml format and then upgrade the system with it.
- Open the SETU VGRX\_Config folder to view the configuration files.

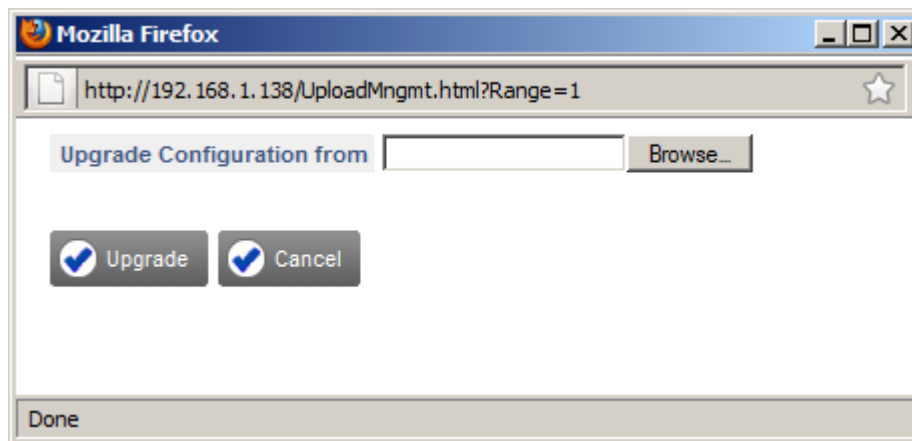


- Keep the Configuration folder as a backup. In case there is a problem with the system configuration files these backup files can be restored back in the system.

## Upgrading Configuration from a Personal Computer

You can upgrade configuration of SETU VGRX with the configuration files—.cfg format or xml format— stored on your computer. To do so,

- Click the **Upgrade Configuration from PC** button. A new window - **Upgrade Configuration From** opens.



- Click the **Browse** button to reach the location on the local disk on which the configuration file is stored.
- Select the required configuration files from the location on the local disk.
- Click the **Upgrade** button.



*At a time, you can upgrade configuration either manually or automatically from Auto Configuration Server or manually from a Personal Computer.*

# System Debug

---

Debugging is a method used for recording actions and events of the system. Debugs are the primary record keepers of the system and network activity. Debugging has several benefits which include troubleshooting, security and system administration.

SETU VGRX supports Syslog Client for sending debug messages to the remote syslog server on the IP network.

## Configuring System Debug

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **System Debug** link.



- Click the **Debug Settings** button and configure the following.

## Debug Settings

**Debug Settings**

Debug Enable	<input checked="" type="checkbox"/>
Save Debug In File	<input type="checkbox"/>
Syslog Server IP Address	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Server Port	<input type="text" value="514"/>
VoPP Packet Recording	<input type="checkbox"/>
VoPP Packet Recording IP Address	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
DSP Debug Parameter 1	<input type="text" value="0"/>
DSP Syslog Server Address	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
DSP Syslog Server Port	<input type="text" value="514"/>
PCM Capture Port	<input type="text" value="None"/>

- Select the **Debug Enable** check box to enable system debug. Default: Disabled.
- You will be able to configure the Debug Settings only after you enable this check box.
- Select the **Save Debug In File** check box, if you want to save the debug file in the system. Default: Disabled.
  - In **Syslog Server IP Address**, enter the remote Syslog Server IP Address. Default: Blank.
  - In the Syslog **Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.
  - Select the **VoPP Packet Recording** check box, if you want to enable VoPP Packet recording. Default: Disabled.
  - If you have enabled VoPP Packet Recording, configure the **VoPP Packet Recording IP Address**.
  - Enter the **DSP Debug Parameter 1** value for which you require the debug. Default 0
  - In **DSP Syslog Server Address**, enter the remote DSP Syslog Server Address. Default: Blank.
  - In **DSP Syslog Server Port**, enter the port number. The range of the server port is 514, 1024 to 65535. Default: 514.
  - In **PCM Capture Port**, select Radio and then the Receive or Transmit debug of the desired port. Default: None.
- To start PCM Capturing, click **Start**.
  - To stop PCM Capturing, click **Stop**.



There is an executable file which needs to be installed in the PC for PCM Capture. For the executable file, contact the Matrix Support Team.

## Miscellaneous

Miscellaneous	
Call	<input checked="" type="checkbox"/>
Config	<input checked="" type="checkbox"/>
Media Channel	<input checked="" type="checkbox"/>
Time	<input checked="" type="checkbox"/>
Webjeeves	<input checked="" type="checkbox"/>
SNMP	<input checked="" type="checkbox"/>
Network	<input checked="" type="checkbox"/>

- Select the check boxes of the events/processes you want to debug from the following.
  - Call
  - Config
  - Media Channel
  - Time
  - Webjeeves
  - SNMP
  - Network

By default, all debug levels, are enabled. To disable a debug level, clear the respective check box.

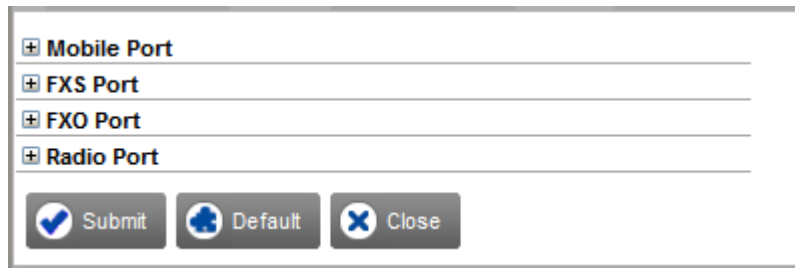
## SIP

SIP					
SIP	<input checked="" type="checkbox"/>	STUN	<input checked="" type="checkbox"/>	NAT	<input checked="" type="checkbox"/>
Call	<input checked="" type="checkbox"/>	Call Message	<input checked="" type="checkbox"/>	Stack Message	<input checked="" type="checkbox"/>
Register	<input checked="" type="checkbox"/>	OPTIONS	<input checked="" type="checkbox"/>	SUBSCRIBE	<input checked="" type="checkbox"/>

- For **SIP Port**, select the desired debug level:
  - SIP
  - STUN
  - NAT
  - Call
  - Call Message
  - Stack Message
  - Register
  - OPTIONS
  - SUBSCRIBE

By default, all debug levels, are enabled. To disable a debug level, clear the respective check box.

## Ports Debug



- To debug the Mobile Port 1 to Mobile Port 4, keep the respective check boxes enabled.
- To debug the **FXS Port 1** and **2**, keep the respective check boxes enabled.
- To debug the **FXO Port 1** and **2**, keep the respective check boxes enabled.
- To debug the **Radio Port 1** to **Radio Port 4**, keep the respective check boxes enabled.
- Click **Submit** to save changes.



*If debug is enabled, atleast one debug level should be selected. If no debug level is selected, SETU VGRX will prompt you to select a debug level.*

- The window closes, and you return to the System Debug page.
- All the Debug events appear on the screen.



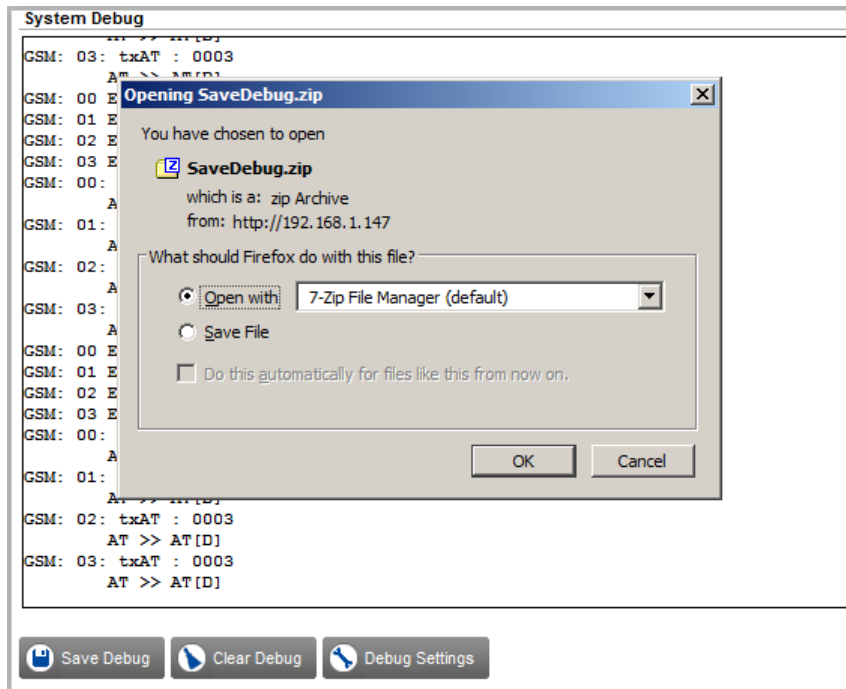
*Events will be displayed only if you enable Debug.*

- Whenever you want the system to fetch an updated debug report, click the **Reload** button on the System Debug page.



- If you want to delete all the events, click the **Clear Debug** button.

- If you want to save the debug events, click the **Save Debug** button.
- You will get a prompt with the option to open the **debug.zip** file or save the file to a location.



- Save the file on the local disk.
- Open the **debug.zip** file from the location you saved. The zip file contains the system debug file **debug.txt**.
- Once you have enabled Debug and set the filters, you can view the debug event log at any time on the **System Debug** page.
- You may log out of Jeeves.

# Simple Network Management Protocol (SNMP)

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Simple Network Management Protocol (SNMP) is an application-layer protocol used for exchanging management information between network devices. Using SNMP, you can manage and monitor network elements, audit network usage, detect network faults or inappropriate network access.

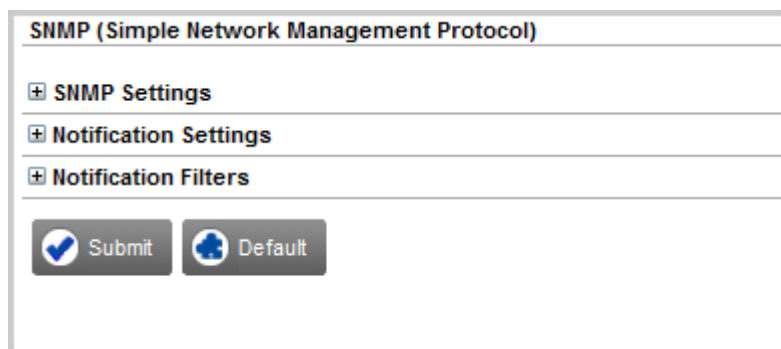
The SNMP architecture consists of:

- An **SNMP Agent** is a program that is bundled within the managed device. SNMP agent allows a managed device to collect the Management Information Base from the device and make it available to the SNMP Manager on request. It receives SNMP requests and generates SNMP responses or notifications (traps/informs). The SNMP Agents are SNMP Servers.
- **SNMP Manager**, usually the Network Management Station. The manager communicates with multiple SNMP Agents implemented in the network. It generates SNMP requests and receives SNMP responses and notifications (traps/informs). The SNMP Manager is an SNMP Client.
- **Managed device** or the network element is a part of the network that requires some form of monitoring and management. For example, switch, routers, servers.
- **Management Information Base** is the commonly shared database between the Agent and the Manager.

SNMP uses UDP (User Datagram Protocol) as the transport protocol for passing information between Managers and Agents. The Agent listens on UDP port 161 for requests from Manager and the Manager listens on UDP port 162 for notification from Agent.

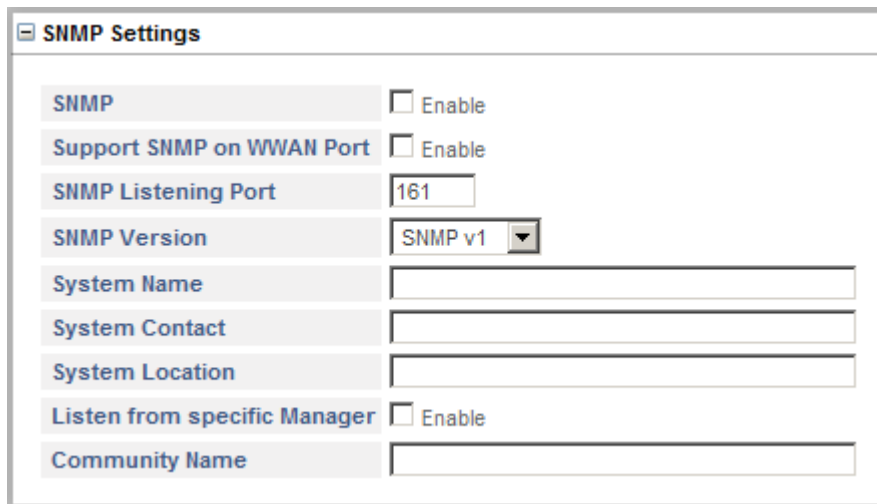
To configure SNMP parameters,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **SNMP** link.

A screenshot of a web-based configuration interface for SNMP. The title bar reads "SNMP (Simple Network Management Protocol)". Below the title bar, there are three expandable sections: "SNMP Settings", "Notification Settings", and "Notification Filters", each preceded by a plus icon. At the bottom of the interface, there are two buttons: "Submit" with a checkmark icon and "Default" with a reset icon.

## SNMP Settings

- Click **SNMP Settings** to expand.



SNMP	<input type="checkbox"/> Enable
Support SNMP on WWAN Port	<input type="checkbox"/> Enable
SNMP Listening Port	161
SNMP Version	SNMP v1 ▼
System Name	
System Contact	
System Location	
Listen from specific Manager	<input type="checkbox"/> Enable
Community Name	

- Select the **Enable SNMP?** check box. Default: Disabled.
- Select the **Support SNMP on WWAN Port** check box, if you want the system to allow incoming SNMP messages from WWAN port and also to send the Trap/Inform messages over WWAN Port. Default: Disabled.
- Configure the **SNMP Listening Port**. Valid Range:161, 1031-65535. Default: 161.
- Select the **SNMP Version** as supported by your SNMP Manager. You can select from:
  - SNMPv1
  - SNMPv2c
  - SNMPv3

For enhanced security, you must select SNMPv3.

- Configure the **System Name**. When there are multiple devices connected in the same network, the name configured helps to identify the SNMP Agent within the network. The System Name can be a maximum of 40 characters. Default: Blank.
- Configure the **System Contact**. It is the name and number of the person to be contacted, in case of notification. The System Contact can be of a maximum of 40 characters. Default: Blank.
- Configure the **System Location**. This is the physical location of SETU VGRX. This information is helpful to the administrator. The System Location may consist of a maximum of 40 characters. Default: Blank.
- Select the **Listen from Specific Manager** check box, if you want the system to listen to the incoming SNMP messages from a specific manager. Default: Disabled.
  - If you have enabled **Listen from Specific Manager** check box, you must configure the specific **Manager's Address**.

The Manager's Address can be a Domain Name or an IP Address. It can be a maximum of 64 characters. Default: Blank.

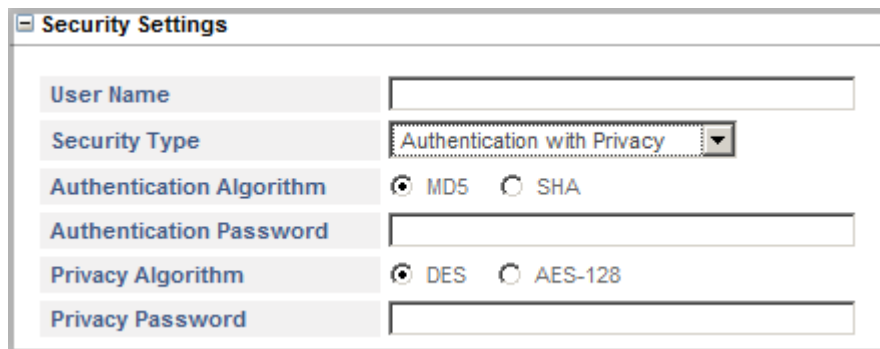
- If SNMP version is set as **SNMPv1** or **SNMPv2c**, configure **Community Name**.

Community Name identifies the SNMP community in which the sender and recipient of the message are located. It enables communication between SETU VGRX and the Manager. The Community Name can be a maximum of 40 characters. Default: Blank.

- If SNMP version is set as **SNMPv3**, the **System's Engine ID** is displayed in this field. This is a unique identification of the system. It is a hexadecimal field with length of 22 characters. The ID consists of:
  - Enterprise Number (800086df03 which is fixed)
  - MAC Address of the system (MAC address of Network port)

## Security Settings

- If SNMP version is set as **SNMPv3**, click **Security Settings** to expand and configure the following.



- Enter the **User Name**. The User Name can be a maximum of 40 characters. User Name will be used for authentication and privacy in SNMPV3.
- Select the appropriate **Security Type** as per your requirement. Security Type defines the level of security.
  - When Authentication and Privacy are not required, select **No Authentication-No Privacy**
  - When only Authentication is required, select **Authentication without Privacy**. Incoming SNMP Messages will require authentication.

If you select this method, select the **Authentication Algorithm** as **MD5** or **SHA**. Default: MD5.

In the **Authentication Password**, enter a password of your choice as Authentication Password for the User Name you have assigned. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

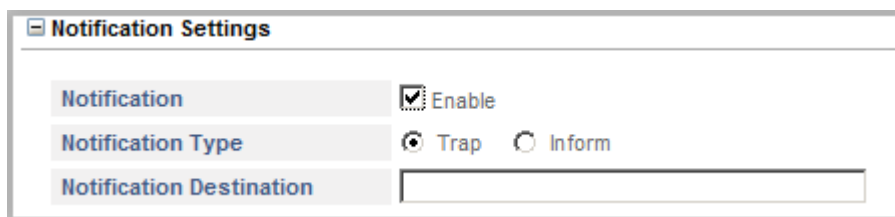
- When both Authentication and Privacy are required, select **Authentication with Privacy**. Incoming SNMP Message will require authentication and these messages will be encrypted, which will be decrypted at the receivers end only.

If you select this method,

- Select the **Authentication Algorithm** as **MD5** or **SHA**. Default: MD5.
- Enter **Authentication Password** for the User Name you have assigned. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.
- Select the **Privacy Algorithm** as **DES** or **AES-128**. Default: DES.
- Enter the **Privacy Password** of your choice. The Privacy Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

## Notification Settings

- Click **Notification Settings** to expand.



If SNMP version is set as **SNMPv1**, configure the following parameters.

- If you want SETU VGRX to generate Trap message for an error, select the **Enable Trap?** check box. Default: Disabled.
- You must configure the **Notification Destination**, if you have enabled **Trap**. SETU VGRX will send the notification (error message) to the destination configured.

The Notification Destination can be an IP Address or a Domain Name and the Port of the Manager or of any other device where you want to receive the trap messages. IP Address/Domain Name can be a maximum of 64 characters. Valid range of the port is 0-65535. Default port is 162.

- Click **Submit** button to save the settings.

If SNMP version is set as **SNMPv2c** or **SNMPv3**, configure the following parameters.

- Select **Notification Enable** check box, if you want SETU VGRX to generate Trap or Inform message for an error.
- Select the **Notification Type**. You may select **Trap** or **Inform**.

If you want the system to send notification message without acknowledgement, select **Trap**.

If you want the system to send notification message with acknowledgement, select **Inform**.

- If you select **Inform** as the *Notification Type*, you must configure Retry Attempts and Retry Interval.

If acknowledgement is not received from the Manager for the notification sent, the system will keep retransmitting the message for the number of attempts you have configured as the **Retry Attempts**. Default: 3.

The system will retransmit the messages at regular time intervals you have configured as **Retry Interval**. Default: 10 seconds.

- Configure the **Notification Destination**. SETU VGRX will send the notification (error message) to the destination configured.

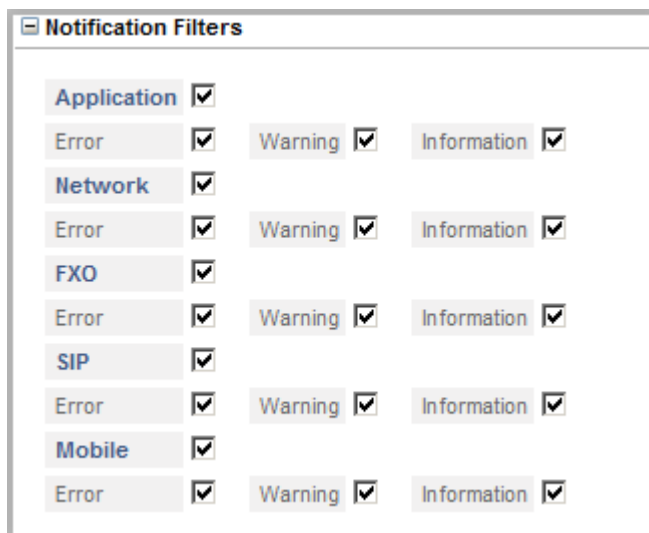
The Notification Destination can be an IP Address or a Domain Name and the Port of the Manager or of any other device where you want to receive the trap messages. IP Address/Domain Name can be a maximum of 64 characters. Valid range of the port is 0-65535. Default port is 162.

- Click **Submit** button to save the settings.

## Notification Filters

By default, you get error notifications, information and warnings for events related to the Application, Network and all Port Types. See table at the end of this topic for the event list. You can choose the type of notification you want by setting the notification filters.

To set filters, click **Notification Filters** link to expand.



To disable any filter, clear the respective check box.



*You must upload MIB file in your SNMP Manager to get the status and notifications for SNMP. Contact Matrix Support team for the MIB file.*

The List of Events for which you will receive notification is presented in the following table.

### Application

Error	Warning	Information
	System Reboot/Gateway Restarted	System boot/initialized

Error	Warning	Information
	Web Login - Authentication failure	Web JEEVES Login/Logout status
	CDR Buffer full	Password change
	SE Login blocked for IP = IP Address	System Config set to default
		Page config set to default

## Network

Error	Warning	Information
	LAN Link Down	LAN Link Up
	WAN Link Down	WAN Link UP
		IP Address of the Gateway
		New IP Address of Gateway
		MAC Address of Gateway
		DNS address of Gateway
		DynDNS status

## SIP

Error	Warning	Information
SIP Stack construction error	DHCP Error	SIP Trunk registering to registrar/ OB Proxy
VOPP Download failed	PPPoE Error	SIP Trunk gets active.
SIP Trunk Registration failed	STUN Error	Network Connection Disable
	SIP Trunk disabled	

## FXO

Error	Warning	Information
	FXO - Line Disconnected	FXO - Line Connected

## Mobile

Error	Warning	Information
	SIM PUK Required	SIM Absent
	SIM PIN Required	SIM Present
	SIM PIN Wrong	Network Absent

Error	Warning	Information
	Call Budget consumed	Network Present
		Current balance in the SIM/Mobile Port

# System Port Activity

You can view the state and activity of each Port of SETU VGRX.

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **System Port Activity** link.
- The port states and activity on each Port appear on this page.

MATRIX		SETU VGRX	
<b>Basic Settings</b>		<b>System Port Activity</b>	
<b>Advanced Settings</b>			
<b>Maintenance</b>			
→ Firmware			
→ Configuration			
→ System Debug			
→ SNMP			
→ <b>System Port Activity</b>			
→ PCAP Trace			
→ Manual Call Test			
→ AC Impedance Test (FXO)			
→ Default System			
→ Soft Restart			
<b>Status</b>			
Port	Name	Status	State
FXS Port 1		Idle	
FXS Port 2		Idle	
FXO Port 1		Inactive	
FXO Port 2		Inactive	
Mobile Port 1		Inactive	
Mobile Port 2		Inactive	
Mobile Port 3		Inactive	
Mobile Port 4		Inactive	
SIP Trunk 1		Idle	
SIP Trunk 2		Disable	
SIP Trunk 3		Disable	
SIP Trunk 4		Disable	
SIP Trunk 5		Disable	
SIP Trunk 6		Disable	
SIP Trunk 7		Disable	
SIP Trunk 8		Disable	
SIP Trunk 9		Disable	
Radio Port 1		Idle	
Radio Port 2		Idle	
Radio Port 3		Idle	
Radio Port 4		Idle	

- The **Port** column displays all the Ports present in the system.
- In the **Name** column, the names assigned to the ports on their respective Port Parameters page appear.
- In the **Status** column, the port status is displayed as:
  - **Disable**, when the port is disabled.
  - **Inactive**, when the port is enabled, but is unable to route calls or accept calls due to any reason.
  - **Idle**, when the port is enabled and is currently in use, but there is no call present currently on this port.

- **Active**, when the port is enabled, in use and a call is present on the port.
- In the **State** column, the state of **Active** ports is displayed as:
  - **Dial**, when the port is in Dial state, i.e. the call has been answered by the system or the FXS port is Off-hook but no called party number is received.
  - **Call in Progress**, when the destination Number is outdialed on the destination port (except FXS/Radio port)
  - **Speech**, when source port and destination port are in speech.
  - **Ringing**, when Ring event is detected on FXS/Radio Port.
  - **Incoming Call Proceeding**, when Ring event is detected on FXO, Mobile Port or SIP Trunk.
  - **Remote Held**, when Hold message is received on SIP Trunk or the FXS Port has put the call on Hold.
  - **Error**, when the other party disconnects the call.
  - **Sending SMS**, when the Mobile Port is sending an SMS.
  - **Processing Balance Inquiry**, when Mobile Port is processing a Balance Inquiry request.
  - **Processing Balance Recharge**, when Mobile Port is processing a Balance Recharge request.

As multiple calls are supported on SIP Trunks, the status and state of each call will appear.

- You may log out of Jeeves.

# PCAP Trace

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.

SETU VGRX supports PCAP Trace, which you can use to detect and diagnose network related problems; for example, when the SIP account is not getting registered, or a SIP related feature is not functioning.

Packets traveling over a network are captured and saved in the system. You can save these trace files (packets captured by the system) on a computer and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

A maximum of 2 MB of packets can be captured and stored in the system.

SETU VGRX also supports Filters and Promiscuous mode for capturing packets, which you can use to specify the types of data packets to be captured.

To use PCAP Trace,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **PCAP** link.

**PCAP**

Filter Setting

Enable Promiscuous mode ☐

**Last Status**

Packets captured

0

Total Bytes

0

Status

mpcap\_init : done : net = 192.168.2.0, mask = 255.255.255.0

Start

Stop

Save Trace File

**Note:** To see what is going on on the network level, you can generate PCAP files on this page. This file can be read with various network tools, for example Ethereal, Wireshark. To start recording, press the start button and to stop, press the stop button.

**Examples of Filter Setting**

Filter Type	Filter Setting	Comment
src port <i>port number</i>	src port 5060	Capture packets if the packet has a source port value of 5060.
dst port <i>port number</i>	dst port 80	Capture packets if the packet has a destination port value of 80.
port <i>port number</i>	port 5060	Capture packets if the packet has either source or destination port value of 5060
src host <i>ip address</i>	src host 192.168.1.176	Capture packets if the source field of packet is 192.168.1.176
dst host <i>ip address</i>	dst host 192.168.1.176	Capture packets if the destination field of packet is 192.168.1.176
host <i>ip address</i>	host 192.168.1.176	Capture packets if either source or destination field of packet is 192.168.1.176

- Decide the type of packets to be captured and set the **Filter Setting** accordingly. Filter Settings can be of maximum 60 characters. By default, it is blank. So, all packets will be captured.

You may view examples of Filter Settings on this page.



*It is not mandatory to set Filters. When the Filter Settings field is left blank, the system will capture all packets.*

- You may enable **Promiscuous Mode** by selecting the check box. Default: Disabled.

When you enable Promiscuous Mode, the SETU VGRX will capture all network traffic. However, this will work only in a non-switched environment.

When Promiscuous Mode is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through the SETU VGRX will be picked up by the PCAP Trace.



*'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power failure.*

- Click the **Start** button to begin the capturing of the packets.
- Click the **Stop** button to stop packet capture.

OR

Wait for the system to stop packet capturing. The system stops packet capturing once the maximum allotted memory of 2 MB (RAM) is utilized.

The Number of Packets and bytes captured as per the filter setting will be displayed in the fields **Packets Captured** and **Total Bytes** respectively.

The **Status** displays the current activity of packet capturing.



*Capturing of packets will not stop if you open any other page of Jeeves. So, you may continue using Jeeves for any other purpose while PCAP Trace is being used.*

- When the packet capturing is stopped (by you or the system), click the **Save Trace File** button to save the files on your computer or on another computer.

A dialog box opens. 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.*

- You may log out of Jeeves.
- Now, you can open the trace files using Wireshark/Ethereal or any other software which supports opening of trace files.

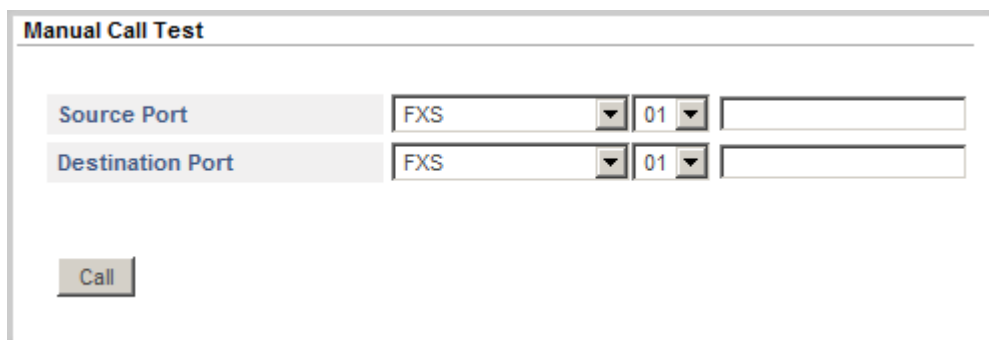
# Manual Call Test

---

Manual Call Test enables you to check the quality of Speech between two ports—Source Port and Destination Port—of SETU VGRX without altering the existing call routing configuration.

To conduct Manual Call Test,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **Manual Call Test** link.



The screenshot shows a web interface titled "Manual Call Test". It contains two rows of input fields. The first row is labeled "Source Port" and the second row is labeled "Destination Port". Each row has a dropdown menu for "Port Type" (currently showing "FXS"), a dropdown menu for "Port Number" (currently showing "01"), and a text input field for the phone number. At the bottom left of the form is a "Call" button.

In **Source Port**,

- Select the **Port Type** you want to test from the list.
- Select the **Port Number** you want to test from the list.
- Enter the **Phone Number** in the corresponding field. The phone number can be of maximum 16 characters. Valid characters are 0-9, \*, #, + and dot (.).

In **Destination Port**,

- Select the **Port Type** you want to test from the list.
  - Select the **Port Number** you want to test from the list.
  - Enter the **Phone Number** in the corresponding field. The phone number must be a valid number that the system can outdial. It can be of maximum 16 characters. Valid characters are 0-9, \*, #, + and dot (.).
- Click the **Call** button. SETU VGRX will out dial the phone number you entered to make a test call between the Source Port and the Destination Port.
  - As soon as the test call is made, the **System Port Activity** page will open. You can view the call states and status of the ports you are testing on this page.

For more information on Call States and Port Status, see ["System Port Activity"](#).

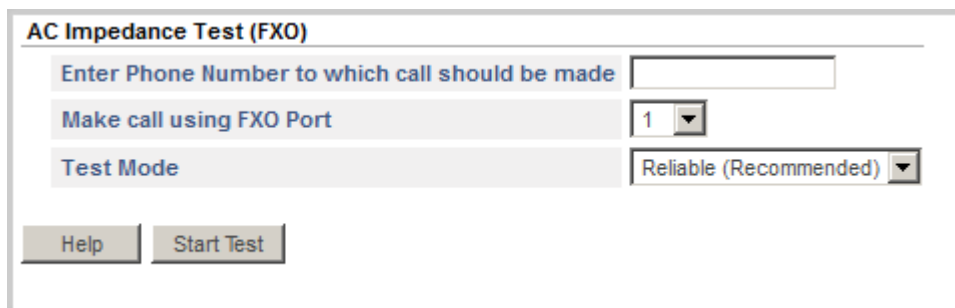
# AC Impedance Test (FXO)

---

SETU VGRX supports the AC Impedance Test for clear, audible and echo-free speech over FXO Ports. This test helps you to set the most appropriate values for the FXO Port Parameters —AC Termination Impedance, CO Termination and CO Line Type— to correct the line impedance mismatch between the AC Termination Impedance presented by the FXO Port of SETU VGRX to the line and the CO Termination Impedance presented by the Central Office to the line.

To conduct the AC Impedance Test,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **AC Impedance Test (FXO)** link.



- In **Enter Phone Number to which call should be made**, enter the phone number on which you want to make a test call. The number can be a landline or a mobile number. We recommend you to use a mobile number for the test call.



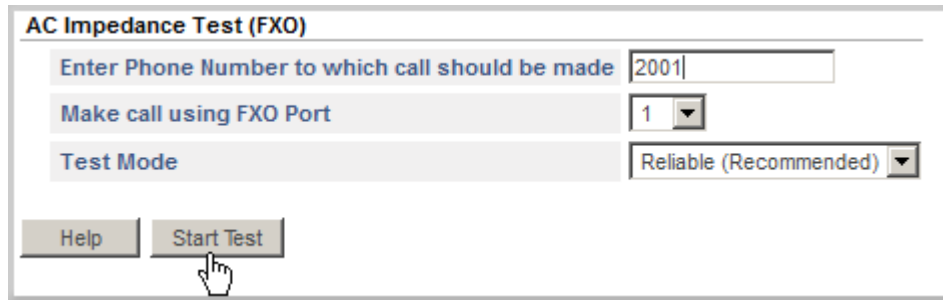
*If you are using a mobile phone number, make sure the handset of the configured mobile number supports the Mute function.*

- In **Make call using FXO Port**, select the FXO Port using which you want to make the test call. This must be the same FXO Port for which AC Impedance is to be set.
- Select the **Test Mode**. You may select **Reliable (Recommended)** or **Accurate**.

The **Reliable Test** mode suggests the AC Impedance settings on the basis of most commonly used AC Impedances, CO Terminations and CO Line Types across the globe. The test using Reliable Test mode takes approximately 5 minutes to complete.

The **Accurate Test** mode suggests the AC Impedance settings on the basis of all the possible AC Impedances, CO Terminations and CO Line Types across the globe. The test using the Accurate Test mode takes approximately 1 hour and 20 minutes to complete.

- Click the **Start Test** button. The system will call the phone number, you have configured. The message 'Starting....' appears on your screen.



The screenshot shows a web-based interface titled "AC Impedance Test (FXO)". It contains three input fields: "Enter Phone Number to which call should be made" with the value "2001", "Make call using FXO Port" with a dropdown menu showing "1", and "Test Mode" with a dropdown menu showing "Reliable (Recommended)". Below these fields are two buttons: "Help" and "Start Test". A mouse cursor is pointing at the "Start Test" button.



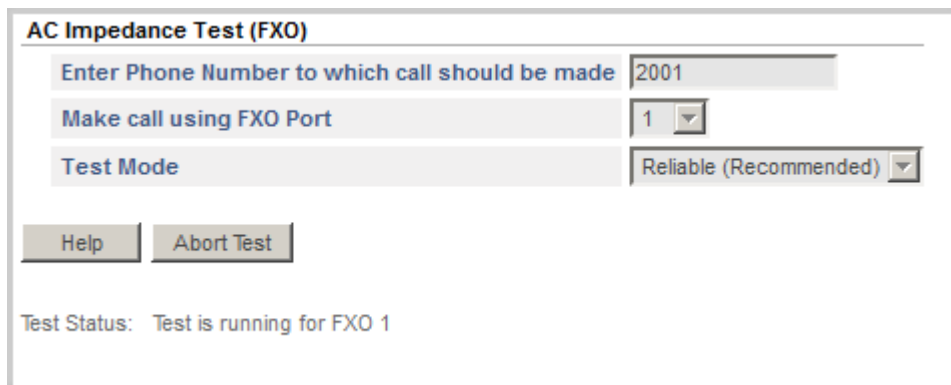
*While the test is being conducted, you will hear pulsating tone on all the ports of the system.*

- Answer the test call from the telephone, you have configured.

If you are using a Mobile phone, Mute the microphone of your mobile phone.

If you are making the test call on a landline number, mute the call using the Mute key of the phone. If your phone does not have a Mute key, unplug the handset cable from the phone body. This is to prevent test signals from reflecting back into the mic of the handset.

- After approximately 5 seconds, you will hear the test signals being transmitted by the system for the duration of the test. The message 'Test is running for FXO 1...' appears on your screen.



The screenshot shows the same web-based interface as before, but now the "Start Test" button has been replaced by an "Abort Test" button. The "Test Status" section at the bottom of the interface now displays the text "Test is running for FXO 1".

If you wish to abort the test midway, you may click the **Abort Test** button.

- On completion of the test, the system will automatically disconnect the call. The message 'Test Status: Successfully completed' appears on the screen.

- At the end of the test, the page displays the **Test Result**. Suggested Impedance Settings for the AC Termination Impedance, CO Termination and CO Line Type to be configured for the FXO Port you have tested, appears on the screen as shown below.

The screenshot shows the 'AC Impedance Test (FXO)' interface. At the top, there are three input fields: 'Enter Phone Number to which call should be made' with the value '2001', 'Make call using FXO Port' with a dropdown set to '1', and 'Test Mode' with a dropdown set to 'Reliable (Recommended)'. Below these are 'Help' and 'Start Test' buttons. The 'Test Status' is 'Successfully Completed'. The 'Test Result' section displays a table with three rows: 'AC Termination Impedance' (600 Ω), 'CO Termination' (600 Ω), and 'CO Line Type' (EIA-0). At the bottom are three buttons: 'Generate Test Report', 'Apply Test Result to FXO Port', and 'Apply Test Result to all FXO Ports'.

Test Result	
AC Termination Impedance	600 Ω
CO Termination	600 Ω
CO Line Type	EIA-0

- Click the **Apply Test Result to FXO Port** button, to apply the test result to the FXO Port you have tested.
- Click the **Apply Test Result to all FXO Ports** button, to apply the test result to all the FXO Ports of the system.
- Verify the settings by making a trial call. There should be no echo and speech should be audible and clear.

If there is no echo/mild echo, and the volume level is low/high, you may adjust the **Rx Gain** and **Tx Gain** of the FXO Port manually. See "[Hardware Settings](#)" under the "[FXO Port](#)" for details.

If you still hear echo during the trial call, you may re-run the test using the **Accurate Test** mode.



*It is possible that the AC Impedance Settings may differ for different CO Trunks subscribed from the same exchange. In such a case, you must run the test for each CO Trunk connected to the FXO Port separately and configure the settings accordingly.*

- To generate the detailed test report, click the **Generate Test Report** button.

The detailed test report appears in a new window.

Index	AC Termination Impedance	CO Termination	CO Line Type	Return Loss
1	600 $\Omega$	None	2000 ft. 22 awg	4.04dB
2	600 $\Omega$	150 $\Omega$ + 510 $\Omega$ + 47 nF	2000 ft. 22 awg	17.16dB
3	600 $\Omega$	220 $\Omega$ + 820 $\Omega$ + 150 nF	2000 ft. 22 awg	11.00dB
4	600 $\Omega$	600 $\Omega$	2000 ft. 22 awg	25.04dB
5	600 $\Omega$	600 $\Omega$ + 1.5 $\mu$ F	2000 ft. 22 awg	18.30dB
6	600 $\Omega$	900 $\Omega$ + 2.16 $\mu$ F	2000 ft. 22 awg	13.77dB
7	600 $\Omega$	1200 $\Omega$ + 376 $\Omega$ + 112 nF	2000 ft. 22 awg	8.57dB
8	270 $\Omega$ + (750 $\Omega$    150 nF) and 275 $\Omega$ + (780 $\Omega$    150 nF)	220 $\Omega$ + 120 $\Omega$ + 115 nF	2000 ft. 22 awg	15.82dB
9	220 $\Omega$ + (820 $\Omega$    120 nF) and 220 $\Omega$ + (820 $\Omega$    115 nF)	220 $\Omega$ + 820 $\Omega$ + 115 nF	2000 ft. 22 awg	10.03dB
10	370 $\Omega$ + (620 $\Omega$    310 nF)	220 $\Omega$ + 820 $\Omega$ + 120 nF	2000 ft. 22 awg	9.88dB
11	370 $\Omega$ + (620 $\Omega$    310 nF)	370 $\Omega$ + 620 $\Omega$ + 310 nF	2000 ft. 22 awg	12.08dB
12	320 $\Omega$ + (1050 $\Omega$    230 nF)	200 $\Omega$ + 560 $\Omega$ + 100 nF	2000 ft. 22 awg	12.03dB
13	320 $\Omega$ + (1050 $\Omega$    230 nF)	270 $\Omega$ + 750 $\Omega$ + 150 nF	2000 ft. 22 awg	10.32dB
14	320 $\Omega$ + (1050 $\Omega$    230 nF)	300 $\Omega$ + 1000 $\Omega$ + 220 nF	2000 ft. 22 awg	9.29dB
15	320 $\Omega$ + (1050 $\Omega$    230 nF)	370 $\Omega$ + 620 $\Omega$ + 310 nF	2000 ft. 22 awg	11.97dB
16	600 $\Omega$	None	2000 ft. 24 awg	5.04dB
17	600 $\Omega$	150 $\Omega$ + 510 $\Omega$ + 47 nF	2000 ft. 24 awg	16.49dB
18	600 $\Omega$	220 $\Omega$ + 820 $\Omega$ + 150 nF	2000 ft. 24 awg	11.26dB
19	600 $\Omega$	600 $\Omega$	2000 ft. 24 awg	22.22dB
20	600 $\Omega$	600 $\Omega$ + 1.5 $\mu$ F	2000 ft. 24 awg	17.38dB

Print Close

- You may print the report by clicking the **Print** button in the test report window.
- Select your Printer in the Printer options.

**Print**

Printer:

Name: \\192.168.101.61\Canon LBP3300(2) Properties...

Status: Adobe PDF

Type: Fax

Where: USB001

Comment: ☐ Print to file

Print range:

☒ All

☐ Pages from: 1 to: 1

☐ Selection

Copies:

Number of copies: 1

☒ Collate

Print Frames:

☐ As laid out on the screen

☐ The selected frame

☐ Each frame separately

OK Cancel

- You can also save the report in PDF format by selecting the **Adobe PDF** in the Printer options.

# Default System

---

You can restore the system configuration to default values:

- using the Web Jeeves.
- using the System Command.
- using the Reset button.

## Restoring Default Settings using Web Jeeves

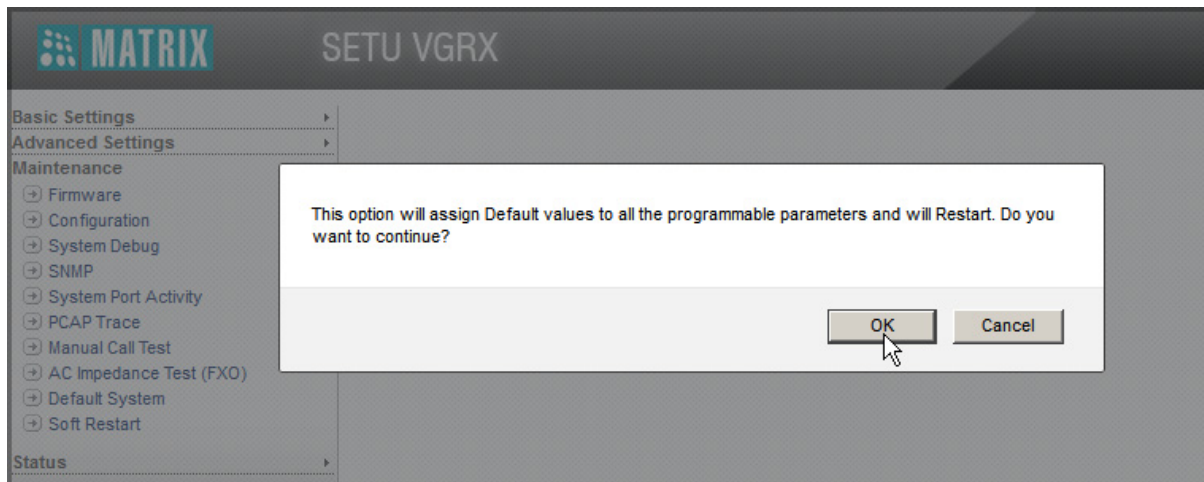
When you restore default settings using the Web Jeeves, all the parameters will be assigned default values **except** the following:

- Real Time Clock
- Call Detail Records
- Region
- Language
- Network
  - Connection Type
  - DNS Settings
  - DYN DNS
- System Parameters - NAT
  - Route Public IP Address
  - STUN Server Address
  - STUN Server Port
- System Parameters - Server Ports
  - HTTP Web Server Port
  - HTTPS Web Server Port
  - FTP Server Port
  - Telnet Server Port
- Mobile Port
  - SIM PIN
- Firmware Parameters
- Configuration Parameters
- Login Password (Jeeves and Command)

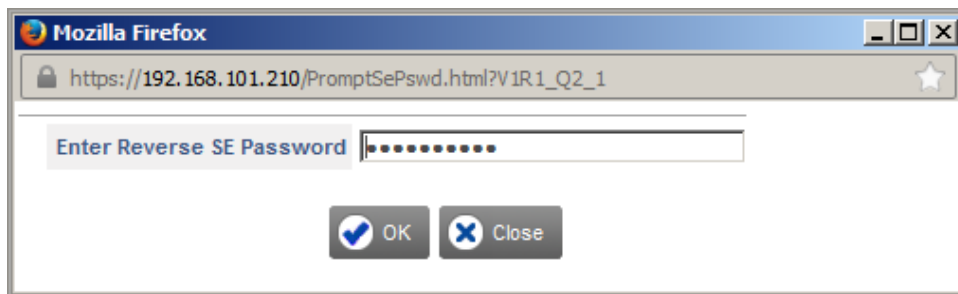
To restore the default settings using the Web Jeeves,

- Log into Jeeves.
- Click the **Maintenance** link.

- Click the **Default System** link.



- An alert message will appear, “**This option will assign default values to all the programmable parameters and will Restart. Do you want to continue?**”.
- Click **OK**.



- You will be prompted to enter the reverse SE password. Enter the current SE password backwards. For example, if your password is 5699, enter 9965. Click **OK**. The system will restart.

## Restoring Default Settings using System Command

When you restore default settings using the System Command, all the parameters will be assigned default values **except** the following:

- Real Time Clock
- Call Detail Records
- Region
- Language
- Network
  - Connection Type
  - DNS Settings
  - DYN DNS
- System Parameters - NAT
  - Route Public IP Address
  - STUN Server Address
  - STUN Server Port
- System Parameters - Server Ports
  - HTTP Web Server Port

- HTTPS Web Server Port
- FTP Server Port
- Telnet Server Port
- Mobile Port
  - SIM PIN
- Firmware Parameters
- Configuration Parameters
- Login Password (Jeeves and Command)

To restore the default settings by dialing the system command,

- Lift the handset of phone connected to the FXS Port.
- Dial **#19-Command Password** to enter the Programming Mode.
- You will get programming tone.
- Dial **51-Reverse Command Password-#\***

For example, if your password is 5699, enter 9965.

- Replace handset of the phone.
- The system will restart.

## Restoring Default Settings using the Reset button

Using Reset button, you can restore the following parameters to default values:

- SE Password
- LAN Port Parameters
  - IP Address
  - Subnet Mask
- WAN Port Parameters
  - Connection Type
  - Web Server Access from WAN
  - FTP Server Access from WAN
  - Telnet Server Access from WAN
  - Allow Server Access from specific IP Address
    - IP Address table for Server Access
- System Parameters - Server Ports
  - HTTP Web Server Port
  - HTTPS Web Server Port
  - FTP Server Port
  - Telnet Server Port

To restore the default settings using the Reset button,

- Press the Reset button for more than four seconds.
- Release the Reset button.



*If you press the Reset button for less than four seconds, SETU VGRX will restart.*

# Soft Restart

---

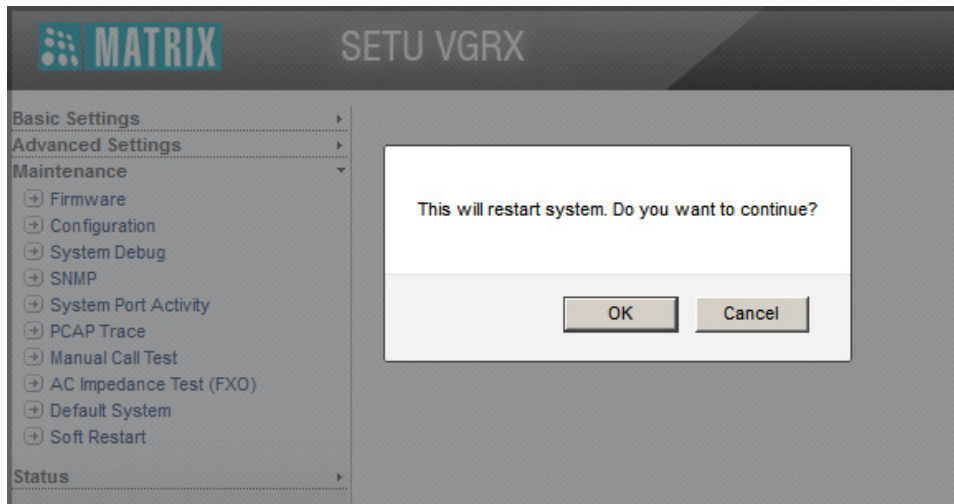
If you need to restart SETU VGRX, you may do it by

- pressing the Reset button.  
or
- use *Soft Restart* from Jeeves.

When you restart the system, all active calls will be disconnected and the ports in use will be released. The system configuration however, will remain unaffected.

To use Soft Restart,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **Soft Restart** link.



- An alert message will appear, "**This will Restart System. Do you want to continue?**".
- Click **OK** to restart the system.

To restart the system using the Reset button,

- Use a blunt pin to press and release the Reset button.
- Press the Reset button for less than 4 seconds, the system will restart.

You can view the System Details and the status of Auto-Firmware upgrade, Auto-Configuration upgrade, the LAN Port, the WAN (Ethernet) Port, the SIP Trunks, the Mobile Ports, the FXO Ports from Jeeves.

To view status,

- Log into Jeeves.
- Click the **Status** link.

## System Details

- Click the **System Detail** link.

The screenshot displays the MATRIX SETU VGRX web interface. On the left is a navigation menu with categories: Basic Settings, Advanced Settings, Maintenance, and Status. Under the Status category, 'System Detail' is selected and highlighted. The main content area is titled 'System Detail' and contains a table of system parameters.

Product Name	SETU VGRX
WAN Port	1
LAN Port	1
FXS Port	2
FXO Port	2
Mobile Port	4
Radio Port	4
VoIP DSP Module	1
Software Version-Revision	V1R2.1.1
Kernel Date	#1 Thu Jul 16 12:25:36 IST 2015
Stack Status	Constructed
CPLD Version-Revision	V1R1
WAN Port MAC Address	00:1b:09:02:f3:a5
LAN Port MAC Address	00:1b:09:02:f3:a6

The following System Details will be displayed on this page.

- **Product Name:** This field displays the name of the product.
- **WAN Port:** This field displays the number of WAN Ports supported in the system.
- **LAN Port:** This field displays the number of LAN Ports supported in the system.
- **FXS Port:** This field displays the number of FXS Ports supported in the system.
- **FXO Port:** This field displays the number of FXO Ports supported in the system.
- **Mobile Port:** This field displays the number of Mobile Ports supported in the system.
- **Radio Port:** This field displays the number of Radio Ports supported in the system.
- **VoIP DSP Module:** This field displays the number of VoIP DSP Modules present in the system.
- **Software Version-Revision:** This field displays the current version and revision of the firmware of SETU VGRX.
- **Kernel Date:** This field displays the Kernel compilation date.
- **Stack Status:** This field displays the SIP Stack Status.
- **CPLD Version Revision:** This field displays the CPLD version revision.
- **WAN Port MAC Address:** This field displays the factory set MAC Address of the WAN (Ethernet) Port.



*If you have cloned the MAC Address of the WAN (Ethernet) Port, you can view it in Network Status.*

- **LAN Port MAC Address:** This field displays the factory set MAC Address of the LAN Port.
- **Serial Number of the Product:** This field displays the Serial Number of the product.
- **Hardware Design of Main Board:** This field displays the Hardware Design of the Main Board.
- **Hardware Design of GSM Module:** This field displays the Hardware Design of the GSM Module.
- **Hardware Design of DSP Module:** This field displays the Hardware Design of the DSP Module.
- **VoIP DSP Module (AudioCodes AC490 - 12 Channel):** This field displays whether the VoIP DSP Module is present in the system.

# Firmware

- Click the **Firmware** link.

Firmware Status	
Last Upgraded On	
Next Upgrade On	Schedule Not Available
Last time when Synchronized with Server	
Status of Last Synchronization	Disable

The following information related to Auto-Firmware upgrade will appear on your screen.

- Last Upgraded On:** This field displays the firmware with which SETU VGRX last upgraded itself through the provisioning server, along with the date (DD:MM:YYYY) and time (HH:MM) of the upgradation.
- Next Upgrade On:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VGRX will again check for new firmware updates on the server.
- Last time when Synchronized with Server:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VGRX last synchronized with the server for new firmware updates.
- Status of Last Synchronization:** This field displays the status of last synchronization. The possible status messages that may appear are listed in the table below.

Possible Responses	Event
Invalid Parameters	When parameters are not valid.
Local Failure	When internal error occurs, like Thread Creation failed.
Resolving Server Address	When IP Address is not found using DNS query.
Server Not Found	When server is not connected after the expiry of Retry Timer and Retry Counter.
Send Request Failed	When there is Curl Internal Error
Connecting to Server	When system is establishing TCP connection with server until the expiry of Retry Timer and Retry Counter.
TCP Connection Failed	When no response is received for TCP connection until expiry of Retry Timer and Retry Counter.
Connection Failed	When no response is received for TCP connection after expiry of Retry Timer and Retry Counter.
	When there is an open SSL error.
	When the maximum file size is exceeded.
	When there are too many Redirect or illegal operation from curl response.

Possible Responses	Event
Permission Denied	<p>When access is denied.</p> <p>When there is permission problem on the server.</p> <p>When login fails.</p>
Downloading Firmware Index File	When the system is retrieving Firmware Index file.
Downloading Firmware	When the system is retrieving Firmware zip file.
File Not Found	When the remote file is not found.
Waiting for Firmware File Name	When <i>Check Firmware Available on Server</i> button is clicked manually and the list of available firmware is presented.
No File Found for Up-gradation	<p>When selected firmware benchmark is not found.</p> <p>When user does not select the firmware name manually.</p> <p>When matrix_firmware.html file is received but current product name is not found from this file.</p> <p>Single firmware name is received in matrix_firmware.html but this benchmark file does not match with current firmware benchmark.</p> <p>Multiple firmware names are received but all files are below the current firmware.</p>
Firmware Version Below	When the received firmware version is below the current firmware version.
Firmware Version Same	When the received firmware version is same as the current firmware version.
Firmware Decryption Failed	<p>When the firmware zip file decryption has failed.</p> <p>When the firmware file name does not match or benchmark is less than the current firmware version-revision in the text file.</p>
Auto Upgrade stop due to parameter change	When Auto Upgrade is in process and the Firmware parameters are changed.
Auto Upgrade stop by system	When Auto Upgrade process is stopped due to network restart.
Auto Upgrade Stop on User request	When Firmware upgrade process is started manually but user clicks Cancel button after display of list of firmware files.
Successfully Updated	When firmware is updated successfully.

# Configuration

- Click the **Configuration** link.

Configuration Status	
Last Upgraded On	
Next Upgrade On	Schedule Not Available
Last time when Synchronized with Server	
Status of Last Synchronization	Disable

The following information related to Auto-Configuration upgrade will appear on your screen.

- **Last Upgraded On:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VGRX last upgraded its configuration through the server.
- **Next Upgrade On:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VGRX will again check for new configuration on the server.
- **Last time when Synchronized with Server:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VGRX last resynchronized with the server for new configuration.
- **Status of Last Synchronization:** This field displays the status of last synchronization. The possible status messages that may appear are listed in the table below.

Possible Responses	Event
Invalid Parameters	When parameters are not valid.
Local Failure	When internal error occurs, like Thread Creation failed.
Resolving Server Address	When IP Address is not found using DNS query.
Server Not Found	When server is not connected after the expiry of Retry Timer and Retry Counter.
Send Request Failed	When there is Curl Internal Error
Connecting to Server	When system is establishing TCP connection with server until the expiry of Retry Timer and Retry Counter.
TCP/TFTP Connection Failed	When no response is received for TCP/TFTP connection until expiry of Retry Timer and Retry Counter.
Connection Failed	When no response is received for TCP connection after expiry of Retry Timer and Retry Counter.  When there is an open SSL error.  When the maximum file size is exceeded.  When there are too many Redirect or illegal operation from curl response.

Possible Responses	Event
Permission Denied	When access is denied.  When there is permission problem on the server.  When login fails.
Downloading Config File	When the system is retrieving config file.
File Not Found	When the remote file is not found.
Config Decryption Failed	When the config decryption has failed.
Config Parsing Failed	When the file parsing has failed.  When the root tag is not found.
Successfully Updated	When configuration is updated successfully.

## Network

- Click the **Network** link.

**LAN Port**

IP Address	192.168.2.100
Subnet Mask	255.255.255.0
MAC Address	00:1b:09:01:8f:f3

**Ethernet Port**

Status	Using WAN Port
IP Address	192.168.1.100
Subnet Mask	255.255.255.0
Gateway IP Address	192.168.1.254
DNS Address	
System MAC Address	00:1b:09:01:8f:f4
Dynamic DNS Status	Dynamic DNS update is disabled
Stack Status	Constructed

**NAT**

NAT Type	Unknown - STUN server address is not programmed
Router's Public IP Address	
IP Address fetched using STUN	
SIP Port fetched using STUN	

The current values of the following parameters will appear on your screen:

## LAN Port

- **IP Address:** This field displays the current IP address assigned to the LAN Port of SETU VGRX.
- **Subnet Mask:** This field displays current Subnet Mask assigned to the LAN Port of SETU VGRX.
- **MAC Address:** This field displays the MAC Address assigned to the LAN Port of SETU VGRX.

## WAN (Ethernet) Port

- **Status:** This field displays the status of the Ethernet Port of SETU VGRX.
- **IP Address:** This field displays the IP address assigned to the Ethernet Port of SETU VGRX.
- **Subnet Mask:** This field displays the Subnet Mask assigned to the Ethernet Port of SETU VGRX.
- **Gateway IP Address:** This field displays the Gateway Address assigned to the Ethernet Port of SETU VGRX.
- **DNS Address:** This field displays the DNS address.
- **System MAC Address:** This field displays the MAC Address assigned to the Ethernet Port of SETU VGRX.



*If you have cloned the MAC Address, this field will display the cloned MAC Address. You can view the factory set MAC Address in System Detail.*

- **Dynamic DNS Status:** This field displays the response received from DDNS server while sending the IP Address update request to the server. Any of the following responses can appear in this field:

Possible Responses	Event
Please Wait.....!!	When system is waiting for error/ successful response from DDNS server
Updated Successfully - IP Address	IP Address updated successfully in DDNS server
Host has been blocked	When 'abuse' is received
Authentication Fail	When authentication check is failed either problem in user id or password
No such host in the system	When 'no host' is received
Invalid hostname format	When 'notfqdn' is received
Host not in this account	When '!Yours' is received
DNS error encountered	When 'dnserr' is received
Server goes under schedule maintenance	When '911' is received
No Response	No response is received from DDNS server due to any reason
DDNS Failed	For all remaining cases
In all remaining cases, the default messages supported by DDNS client will appear in this field.	

- **Stack Status:** This field displays the SIP Stack Status.

## NAT

- **NAT Type:** This field displays the NAT Type, if STUN is enabled in SETU VGRX. The commonly used NAT types are:
  - Unknown
  - Open
  - Conenat
  - Restrictednat
  - Portrestrictednat
  - Symmetricnat
  - Symmetricfirewall
  - Blocked
- **Router's Public IP Address:** This field displays the Router's Public IP address programmed in the System Parameters.
- **IP Address fetched using STUN:** This field displays the IP address fetched using STUN, if STUN server address is programmed in the system.
- **SIP Port fetched using STUN:** This field displays the SIP Port fetched using STUN, if STUN server address is programmed in the system.

## FXO Port

- Click the **FXO Port** link.

FXO Port Status	
FXO Port Number	Status
1	Line not Connected
2	Line not Connected

- The status of the FXO Ports appears on this page, as 'Line not Connected' or 'Line Connected'.

# Mobile Port

- Click the **Mobile Port** link.

The screenshot shows the MATRIX SETU VGRX web interface. On the left is a navigation menu with categories: Basic Settings, Advanced Settings, Maintenance, and Status. Under Status, there are links for System Detail, Firmware, Configuration, Network, FXO Port, Mobile Port (which is highlighted), and SIP Trunk. The main content area is titled 'Mobile Port' and has tabs for Mobile 1, Mobile 2, Mobile 3, and Mobile 4. Below the tabs, there is a form with the following fields: Activity Status (Network Present), Module Firmware (T7.53.1.A3.201510141142.SL3010T Rev 1.0 1854468 10), IMEI (80C5A825), IMSI (empty), Network Operator Code (4040000), Network Operator Name (empty), Registered with Network (CDMA), SMS Service Center Number (empty), Signal Strength (-79 dbm with a blue bar graph and 'High' label), Allowed Call Minutes (empty), and Consumed Minutes (empty). There is a 'Reset Consumed Minutes' button next to the Consumed Minutes field.

The following parameters will be displayed for Mobile Ports.

- **Activity Status:** This field displays port activity status listed below:
  - Module Initialization
  - SIM PUK Required
  - SIM PIN Wrong
  - SIM Absent
  - SIM Present
  - Network Absent
  - Network Present
- **Module Firmware:** This field displays the current version-revision of the engine's firmware.
- **IMEI:** This field displays the International Mobile Equipment Identity (IMEI) Number, the unique identity number of the GSM engine.
- **IMSI:** This field displays the International Mobile Subscriber Identity (IMSI), the unique identity number of the SIM Card present in the Mobile Port.
- **Network Operator Code:** This field displays the code of the network with which the Mobile Port is registered.
- **Network Operator Name:** This field displays the name of the network with which the Mobile Port is registered.
- **Registered with Network:** This field displays the type of network with which the SIM is registered, that is GSM, UMTS, 3G, 2G. If the SIM is not registered this field displays the status as Not Registered.

- **SMS Service Center Number:** This field displays the Number of the SMS Service Center of the Service Provider.



**SMS Service Center Number** is not applicable, if a CDMA module is installed in your SETU VGRX.

- **Signal Strength:** This field displays the current signal strength.
- **Allowed Call Minutes:** This field displays the Call Minutes allowed to the Mobile Port. To know more about this feature, see [“Call Minutes”](#).
- **Consumed Minutes:** This field displays the Call Minutes used up by the Mobile Port. You can reset the consumed minutes by clicking the **Reset Consumed Minutes** button. To know more, see [“Call Minutes”](#).

## SIP Trunk

- Click the **SIP Trunk** link.

SIP Trunk Status				
SIP Trunk Number	Status	Registration Time	Registration Retry Count	Failed Reason
1	Active	0	0	
2	Disabled	0	0	
3	Disabled	0	0	
4	Disabled	0	0	
5	Disabled	0	0	
6	Disabled	0	0	
7	Disabled	0	0	
8	Disabled	0	0	
9	Disabled	0	0	

The following status indications will appear for the SIP Trunks.

- **SIP Trunk Number:** The number of the SIP Trunk.
- **Status:** The possible status indications that will be displayed in this column for the respective SIP Trunk numbers are described in the table below.

Status Message	Meaning
Disable	The SIP Trunk is disabled.
Registering	The SIP Trunk is enabled and is waiting for response from the SIP server.
Active	The SIP Trunk is registered with the SIP server.

Status Message	Meaning
Failed	Some error has occurred in the SIP Trunk and no calls can be made using the SIP Trunk (applicable only if the SIP Trunk mode is configured as 'Proxy').
Network Connection Disable	The SIP Trunk is enabled but the active <i>Network Connection</i> does not match the option selected for <i>Use SIP Trunk for Network Connection</i> parameter.
Inactive	The Proxy Server is unavailable (no response is received from the server).

- **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 the SIP Trunk.
- **Failed 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:

Failure Message	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 constraints, or resource limitation or 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- 4xx to 6xx	This is the error response code.
No contact header in 2xx	This reason is displayed when no contact address is received in the 2xx response from the SIP server.
Authentication Failed	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's 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.



If for a SIP Trunk, you have enabled **Fallback Server** and **Registration Behavior** is set to **Register with all Servers**, the SIP Trunk Status page will display status of all the servers for that SIP Trunk as shown below.

SIP Trunk Status				
SIP Trunk Number	Status	Registration Time	Registration Retry Count	Failed Reason
1	Registering	0	1	Local Failure
	Registering	0	0	
	Registering	0	0	
2	Disabled	0	0	
3	Disabled	0	0	
4	Disabled	0	0	
5	Disabled	0	0	
6	Disabled	0	0	
7	Disabled	0	0	
8	Disabled	0	0	
9	Disabled	0	0	

## MWI Status

If you have subscribed for Message Wait Indication service from your ITSP, you can view the status of the messages received on the SIP Trunk.

MWI						
Subscription Status	New Messages	Old Messages	Urgent New Messages	Urgent Old Messages	Message Notificaiton on	Failed Reason
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	
Disabled	0	0	0	0	No	

The following status indications will appear for the MWI subscription on SIP Trunks.

- **SIP Trunk Number:** The number of the SIP Trunk on which MWI is subscribed.

- **Subscription Status:** It displays the MWI Subscription Status. The possible status indications that will be displayed in this column for the respective SIP Trunk numbers are described in the table below.

Status Message	Description
Active	When 200 OK with Event as message-summary and Subscription-State as active is received against SUBSCRIBE for MWI sent from SIP Trunk.
Active	When the NOTIFY with the Event header field as message-summary, Subscription-State as active and the Message Body containing current status of the pending messages, is received on SIP Trunk.
Corresponding 4xx/5xx/6xx response along with the text of the error message as it is received with 4xx/5xx/6xx response.	When any 4xx/5xx/6xx response is received against the SUBSCRIBE for MWI sent from SIP Trunk.
Corresponding internal error message (same error messages for relative condition, displayed in case of REGISTER failure)	When any internal error occurs.
Disable	When “Subscribe for MWI” Flag is disabled for SIP Trunk.

- **New Messages:** It displays the number of unread new messages for the SIP Trunk.
- **Old Messages:** It displays the number of old messages for the SIP Trunk.
- **Urgent New Messages:** It displays the urgent unread new messages for the SIP Trunk.
- **Urgent Old Messages:** It displays the urgent old messages for the SIP Trunk.
- **Message Notification on:** It displays the FXS Port number on which the Message Wait Indication Notification is to be sent for the new messages received on the SIP Trunk.



# *Appendix*

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## Acronyms

<b>ACS</b>	Auto Configuration Server
<b>ASCII</b>	American Standard Code for Information Technology
<b>ANT</b>	Automatic Number Translation
<b>CA</b>	Certificate Authority
<b>CDR</b>	Call Detail Record
<b>CLI</b>	Caller Line Identification
<b>CLIR</b>	Calling Line Identification Restriction
<b>CO</b>	Central Office
<b>COS</b>	Class of Service
<b>CPT</b>	Call Progress Tone
<b>DDI</b>	Direct Dialing In
<b>DHCP</b>	Dynamic Host Control Protocol
<b>DND</b>	Do Not Disturb
<b>DNS</b>	Domain Name Service
<b>DTMF</b>	Dual Tone Multi-Frequency
<b>FIFO</b>	First In First Out
<b>FoIP</b>	Fax over IP
<b>FSK</b>	Frequency Shift Keying
<b>FTP</b>	File Transfer Protocol
<b>FXO</b>	Foreign Exchange Office
<b>FXS</b>	Foreign Exchange Subscriber
<b>GMT</b>	Greenwich Mean Time
<b>GSM</b>	Global Systems for Mobile Communications
<b>HF</b>	High Frequency
<b>HTTP</b>	Hypertext Transfer Protocol
<b>ICMP</b>	Internet Control Message Protocol
<b>IMEI</b>	International Mobile Equipment Identity
<b>IP</b>	Internet Protocol
<b>ITSP</b>	Internet Telephony Service Provider
<b>LAN</b>	Local Area Network
<b>LED</b>	Light Emitting Diode
<b>MAC</b>	Media Access Control
<b>MWI</b>	Message Wait Indication
<b>NAT</b>	Network Address Translation

<b>PBX</b>	Private Branch Exchange
<b>PIN</b>	Personal Identification Number
<b>PPPoE</b>	Point-to-Point Protocol over Ethernet
<b>PSTN</b>	Public Switched Telephone Network
<b>PTT</b>	Press to Talk
<b>PUK</b>	Personal Unlock Key
<b>PWR</b>	Power
<b>QoS</b>	Quality of Service
<b>RCOC</b>	Returned Call to Original Caller
<b>RTC</b>	Real Time Clock
<b>RTP</b>	Real Time Protocol
<b>RUIM</b>	Re-usable Identification Module
<b>SE</b>	System Engineer
<b>SIM</b>	Subscriber Identification Module
<b>SIP</b>	Session Initiation Protocol
<b>SMS</b>	Short Message Service
<b>SMSC</b>	SMS Service Center Number
<b>SNMP</b>	Simple Network Management Protocol
<b>SNTP</b>	Simple Network Time Protocol
<b>STUN</b>	Simple Traversal of UDP over NAT
<b>TLS</b>	Transport Layer Security
<b>UDP</b>	User Datagram Protocol
<b>UHF</b>	Ultra High Frequency
<b>URI</b>	Uniform Resource Identifier
<b>URL</b>	Universal Reference/Resource Locator
<b>VHF</b>	Very High Frequency
<b>VoIP</b>	Voice over IP
<b>WAN</b>	Wide Area Network
<b>WWAN</b>	Wireless WAN

# Default Region Table

The country-specific default settings of various parameters that will be loaded on changing the **Region** are presented in the table below.

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type	Country Code	Companding Type	FXS Port - CLI Type	FXO Port - CLI Type
1	Afghanistan	English	GMT+04:30				93			
2	Algeria	English	GMT+01:00				213	A-law		
3	Antigua and Barbuda	English	GMT-04:00				1 268			
4	Argentina	Spanish	GMT-03:00		4		54	A-law		
5	Australia (Perth)	English	GMT+08:00	2	5	8	61			
6	Australia (Adelaide)	English	GMT+09:30	2	5	8	61			
7	Australia (Brisbane, Canberra, Melbourne, Sydney)	English	GMT+10:00		5	8	61			
8	Austria	German	GMT+01:00	1			43			
9	Bahamas	English	GMT-05:00				1 242			
10	Bahrain	English	GMT+03:00	3			973			
11	Bangladesh	English	GMT+06:00				880			
12	Belarus	English	GMT+02:00				375			
13	Belgium	French	GMT+01:00	2	39	11	32	A-law		
14	Bhutan	English	GMT+06:00				975			
15	Bolivia	Spanish	GMT-04:00				591			
16	Bosnia and Herzegovina	English	GMT+01:00				387			
17	Botswana	English	GMT+02:00				267			
18	Brunei	English	GMT+08:00				673			
19	Brazil (Fernando De Noronha)	Portuguese	GMT-02:00		6	6	55	A-law		
20	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	Portuguese	GMT-03:00	4	6	6	55	A-law		
21	Brazil (Manaus)	Portuguese	GMT-04:00		6	6	55	A-law		
22	Brazil (Acre)	Portuguese	GMT-05:00		6	6	55	A-law		
23	Bulgaria	English	GMT+02:00				359			
24	Cambodia	English	GMT+07:00				855			
25	Cameroon	English	GMT+01:00				237			
26	Canada (St. John's)	English	GMT-03:30	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore
27	Canada (Halifax)	English	GMT-04:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore
28	Canada (Montreal, Ottawa, Toronto)	English	GMT-05:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore
29	Canada (Winnipeg)	English	GMT-06:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore
30	Canada (Calgary)	English	GMT-07:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore
31	Canada (Vancouver)	English	GMT-08:00	5	7	7	1	U-law	FSK Bellcore	FSK Bellcore
32	Chile	Spanish	GMT-04:00	6			56			
33	China	English	GMT+08:00		8	11	86	A-law		
34	Colombia	Spanish	GMT-05:00				57			
35	Costa Rica	Spanish	GMT-06:00				506			
36	Croatia	English	GMT+01:00				385			
37	Cuba	Spanish	GMT-05:00	18			53	A-law		
38	Cyprus	English	GMT+02:00				357			
39	Czech Republic	English	GMT+01:00				420			
40	Denmark	English	GMT+01:00	7			45	A-law		
41	Egypt	English	GMT+02:00	11	9	7	20	A-law		
42	Fiji	English	GMT+12:00				679			
43	Finland	English	GMT+02:00	8			358	A-law		

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type	Country Code	Companding Type	FXS Port - CLI Type	FXO Port - CLI Type
44	France	French	GMT+01:00	2	10	14	33	A-law		
45	Germany	German	GMT+01:00	2	11	6	49	A-law		
46	Greece	English	GMT+02:00	2	12	6	30			
47	Guyana	English	GMT-04:00				592			
48	Hong Kong	English	GMT+08:00				852			
49	Hungary	English	GMT+02:00	2			36			
50	India	English	GMT+05:30		13	8	91	A-law		
51	Indonesia	English	GMT+07:00		14		62			
52	Iran	English	GMT+03:30		15		98			
53	Iraq	English	GMT+03:00	9	16		964			
54	Ireland	English	GMT	7			353			
55	Israel	English	GMT+02:00		17	15	972			
56	Italy	Italian	GMT+01:00	2	18	6	39			
57	Japan	English	GMT+09:00		19	10	81	U-law		
58	Jordan	English	GMT+02:00				962	A-law		
59	Kazakhstan	English	GMT+06:00				7			
60	Kenya	English	GMT+03:00		20		254			
61	Korea – North	English	GMT+09:00		21	11	850			
62	Korea – South	English	GMT+09:00		21	11				
63	Kuwait	English	GMT+03:00				965			
64	Kyrgyzstan	English	GMT+06:00	10			996			
65	Lebanon	English	GMT+02:00	12			961			
66	Libya	English	GMT+02:00				218			
67	Malaysia	English	GMT+08:00		22	15	60			
68	Maldives	English	GMT+05:00				960			
69	Mauritius	English	GMT+04:00				230			
70	Mexico (Mexico City)	Spanish	GMT-06:00	3	23		52	A-law		
71	Mexico (Chihuahua)	Spanish	GMT-07:00	3	23		52	A-law		
72	Mexico (Tijuana)	Spanish	GMT-08:00	3	23		52	A-law		
73	Mongolia	English	GMT+08:00				976			
74	Mozambique	Portuguese	GMT+02:00				258			
75	Myanmar	English	GMT+06:30				95			
76	Namibia	English	GMT+01:00	13			264			
77	Nepal	English	GMT+05:45				977			
78	Netherlands	English	GMT+01:00				31	A-law		
79	New Zealand	English	GMT+12:00	14	24	15	64			
80	Nigeria	English	GMT+01:00				234			
81	Norway	English	GMT+01:00	15			47	A-law		
82	Oman	English	GMT+04:00				968			
83	Pakistan	English	GMT+05:00				92			
84	Paraguay	Spanish	GMT-04:00	16			595			
85	Peru	Spanish	GMT-05:00				51			
86	Philippines	English	GMT+08:00		25		63	A-law		
87	Poland	English	GMT+01:00	1	26	15	48			
88	Portugal	Portuguese	GMT	7	27	12	351			
89	Qatar	English	GMT+03:00				974			
90	Romania	English	GMT+02:00				40			
91	Russia (Moscow, St. Petersburg)	English	GMT+03:00	1	28	11	7			
92	Russia (Novosibirsk)	English	GMT+06:00	1	28	11	7			
93	Russia (Vladivostok)	English	GMT+10:00	1	28	11	7			

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type	Country Code	Companding Type	FXS Port - CLI Type	FXO Port - CLI Type
94	Singapore	English	GMT+08:00		30	8	65	A-law		
95	Slovakia	English	GMT+01:00				421			
96	South Africa	English	GMT+02:00		31	8	27			
97	Spain	Spanish	GMT+01:00	1	32	13	34	A-law		
98	Sri Lanka	English	GMT+05:30				94			
99	Sudan	English	GMT+03:00				249			
100	Sweden	English	GMT+01:00	2			46	A-law		
101	Switzerland	German	GMT+01:00	2			41			
102	Syria	English	GMT+02:00	17			963			
103	Taiwan	English	GMT+08:00				886			
104	Tajikistan	English	GMT+05:00				992			
105	Thailand	English	GMT+07:00		33	15	66	A-law		
106	Turkey	English	GMT+02:00		34		90			
107	Uganda	English	GMT+03:00				256			
108	Ukraine	English	GMT+02:00				380			
109	United Arab Emirates	English	GMT+04:00		35	15	971	A-law		
110	United Kingdom	English	GMT	7	36	8	44	A-law		
111	United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indianapolis, Miami, NY, Philadelphia, Washington)	English	GMT-05:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore
112	United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	English	GMT-06:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore
113	United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	English	GMT-07:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore
114	United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	English	GMT-08:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore
115	United States (Juneau)	English	GMT-09:00	3	37	7	1	U-law	FSK Bellcore	FSK Bellcore
116	United States (Hawaii)	English	GMT-10:00		37	7	1	U-law	FSK Bellcore	FSK Bellcore
117	Uzbekistan	English	GMT+05:00				998			
118	Venezuela	Spanish	GMT-04:30				58			
119	Vietnam	English	GMT+07:00				84			
120	Yemen	English	GMT+03:00				967			
121	Yugoslavia	English	GMT+02:00				381			
122	Zambia	English	GMT+02:00				260			
123	Zimbabwe	English	GMT+02:00				263			

# Call Progress Tones

Call Progress Tones (CPT) are audible tones sent by switching systems such as PSTN or PBX, to calling parties to show the status of the phone call.

Each CPT has a distinctive tone frequency and cadence assigned to it, for which some standards have been established by the ETSI.

On the basis of specific frequency, modulating frequency and cadence, the CPTs generated by SETU VGRX are categorized as:

- Dial Tone
- Ring Back Tone
- Busy Tone
- Error Tone 1
- Confirmation Tone
- Feature Tone/ Programming Tone
- Intrusion Tone
- Error Tone 2
- Routing Tone
- Stuttered Dial Tone

CPT standards are applied differently in different situations and in different countries. You can match call progress tones of SETU VGRX to that of the country standard where it is installed.

See the table for the **CPTG Type** (frequency and cadence of the different tones) supported by SETU VGRX. The table shows the CPTG Types supported for different countries.

When you select 'Region', the Call Progress Tones matching the country standards of the selected Region/Country will be automatically loaded. However, you may select a different CPTG Type, if required. You can also customize the frequency and cadence. For instructions, see ["Region"](#) under *Basic Settings*.



*Remote Hold Tone is fixed for all the countries; it is non-programmable.*

## CPTG Types (as per ETSI standard) supported by SETU VGRX

CPTG Type	Country	Dial tone		Ring Back Tone		Busy Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
1	Type1	440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	440	0.75on 0.75off
2	Type2	400	Continuous	400	0.6on 0.2off 0.2on 2.0off	400	0.5on 0.5off
3	Type3	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
4	Argentina	425	Continuous	425	1.0on 4.0 off	425	0.3on 0.2off
5	Australia	425*25	Continuous	400*25	.4on .2off .4on 2.0off	425	0.375on 0.375off
6	Brazil	425	Continuous	425	1.0on 4.0 off	425	0.25on 0.25off

CPTG Type	Country	Dial tone		Ring Back Tone		Busy Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
7	Canada	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
8	China	450	Continuous	450	1.0on 4.0off	450	0.35 on 0.36off
9	Egypt	425*50	Continuous	425*50	2.0on 1.0off	425*50	1.0on 4.0off
10	France	440	Continuous	440	1.5on 3.5off	440	0.5on 0.5off
11	Germany	425	Continuous	425	1.0on 4.0off	425	0.48on 0.48off
12	Greece	425	0.2on 0.3off 0.7on 0.8off	425	1.0on 4.0off	425	0.3on 0.3off
13	India	400*25	Continuous	400*25	.4on .2off .4on 2.0off	400	0.75on 0.75off
14	Indonesia	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off
15	Iran	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off
16	Iraq	400	0.4on 0.2off 0.4on 1.5off	400	Continuous	400	1.0on 1.0off
17	Israel	400	Continuous	400	1.0on 3.0off	400	0.5on 0.5off
18	Italy	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off
19	Japan	400	Continuous	400*25	1.0on 2.0off	400	.5on .5off
20	Kenya	425	Continuous	425	0.67on 3.0off 1.5on 5.0off	425	0.2on 0.6off 0.2on 0.6off
21	Korea	350+440	Continuous	440+480	1.0on 2.0off	480+620	0.5on 0.5off
22	Malaysia	425	Continuous	425	0.4on 0.2off 0.4on 2.0off	425	0.5on 0.5off
23	Mexico	425	Continuous	425	1.0on 4.0off	425	0.25on 0.25off
24	New Zealand	400	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off
25	Phillippines	425	Continuous	425+480	1.0on 4.0off	480+620	0.5on 0.5off
26	Poland	425	Continuous	425	1.0on 4.0off	425	0.5on 0.5off
27	Portugal	425	Continuous	425	1.0on 5.0off	425	0.5on 0.5off
28	Russia	425	Continuous	425	0.8on 3.2off	425	0.4on 0.4off
29	Saudi Arabia	425	Continuous	425	1.2on 4.6off	425	0.5on 0.5off
30	Singapore	425	Continuous	425*24	0.4on 0.2off 0.4on 2.0off	425	.75on .75off
31	South Africa	400*33	Continuous	400*33	0.4on 0.2off 0.4on 2.0off	400	.5on .5off

CPTG Type	Country	Dial tone		Ring Back Tone		Busy Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
32	Spain	425	Continuous	425	1.5on 3.0off	425	0.2on 0.2off
33	Thailand	400*50	Continuous	400	1.0on 4.0off	400	0.5on 0.5off
34	Turkey	450	Continuous	450	2.0on 4.0off	450	0.5on 0.5off
35	UAE	350+440	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off
36	UK	350+440	Continuous	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off
37	USA	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
38	Type4	400	Continuous	400	1.0on 2.0off	400	0.5on 0.5off
39	Belgium	425	Continuous	425	1.0on 3.0off	425	0.5on 0.5off
40	Type5	350+440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	400	0.75on 0.75off

CPTG Type	Country	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
1	Type1	440	0.25on 0.25 off	440	1on 1off	350+440	0.1on 0.1off
2	Type2	400	0.25on 0.25 off	400	1on 1off	400	0.1on 0.1off
3	Type3	440	0.25on 0.25 off	440	1on 1off	350+440	0.1on 0.1off
4	Argentina	425	0.3on 0.4off	425	1on 1off	425	0.1on 0.1off
5	Australia	425	0.375on 0.375off	425	1on 1off	425*25	0.1on 0.1off
6	Brazil	425	0.25on 0.25 off	425	1on 1off	425	0.1on 0.1off
7	Canada	480+620	0.25on 0.25off	480+620	1on 1off	350+440	0.1on 0.1off
8	China	450	0.7on 0.7off	450	1on 1off	450	0.1on 0.1off
9	Egypt	450	0.5on 0.5off	450	1on 1off	425*50	0.1on 0.1off
10	France	440	0.25on 0.25off	440	1on 1off	440	0.1on 0.1off
11	Germany	440	0.20on 0.48off	425	1on 1off	425	0.1on 0.1off
12	Greece	425	0.15on 0.15off	425	1on 1off	425	0.1on 0.1off
13	India	400	0.25on 0.25off	400	1on 1off	400	0.1on 0.1off

CPTG Type	Country	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
14	Indonesia	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
15	Iran	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
16	Iraq	400	0.25on 0.25off	400	1on 1off	400	0.1on 0.1off
17	Israel	400	0.25on 0.25off	400	1on 1off	400	0.1on 0.1off
18	Italy	425	0.2on 0.2off	425	1on 1off	425	0.1on 0.1off
19	Japan	400	0.25on 0.25off	400	1on 1off	400	0.1on 0.1off
20	Kenya	425	0.2on 0.6off	425	1on 1off	425	0.1on 0.1off
21	Korea	480+620	0.3on 0.2off	480+620	1on 1off	350+440	0.1on 0.1off
22	Malaysia	425	2.5on 0.5off	425	1on 1off	425	0.1on 0.1off
23	Mexico	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
24	New Zealand	400	0.25on 0.25off	400	1on 1off	400	0.1on 0.1off
25	Phillippines	480+620	0.25on 0.25off	480+620	1on 1off	425	0.1on 0.1off
26	Poland	425	0.5on 0.5off	425	1on 1off	425	0.1on 0.1off
27	Portugal	450	0.33on 1.0off	450	1on 1off	425	0.1on 0.1off
28	Russia	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
29	Saudi Arabia	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
30	Singapore	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
31	South Africa	400	0.25on 0.25off	400	1on 1off	400*33	0.1on 0.1off
32	Spain	425	0.25on 0.25off	425	1on 1off	425	0.1on 0.1off
33	Thailand	400	0.3on 0.3off	400	1on 1off	400*50	0.1on 0.1off
34	Turkey	450	0.2on 0.2off .6on .2off	450	1on 1off	450	0.1on 0.1off
35	UAE	400	0.4on 0.35off 0.225on 0.525off	400	1on 1off	350+440	0.1on 0.1off

CPTG Type	Country	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
36	UK	400	0.4on 0.35off 0.225on 0.525off	400	1on 1off	350+440	0.1on 0.1off
37	USA	480+620	0.25on 0.25off	480+620	1on 1off	350+440	0.1on 0.1off
38	Type4	400	0.25on 0.25 off	400	1on 1off	400	0.1on 0.1off
39	Belgium	425	0.167on 0.167 off	425	1on 1off	425	0.1on 0.1off
40	Type5	400	0.25on 0.25 off	400	1on 1off	350+440	0.1on 0.1off

CPTG Type	Country	Feature / Programming / Prompt Tone		Routing Tone		IntrusionTone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
1	Type1	350+440	0.1on 0.9off	350+440	0.1on 1.9off	440	0.1on 2.9off
2	Type2	400	1.5on 0.1off	400	0.1on 1.9off	400	0.2on 4.8off
3	Type3	350+440	0.1on 0.9off	350+440	0.1on 1.9off	440	0.1on 2.9off
4	Argentina	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
5	Australia	425*25	0.1on 0.9off	425*25	0.1on 1.9off	425	Continuous
6	Brazil	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
7	Canada	350+440	0.1on 0.9off	350+440	0.1on 1.9off	480+620	0.5on 0.5off
8	China	450	0.1on 0.9off	450	0.1on 1.9off	450	0.2on 0.2off 0.2on 0.6off
9	Egypt	425*50	0.1on 0.9off	425*50	0.1on 1.9off	450	0.5on 0.5off
10	France	440	0.1on 0.9off	440	0.1on 1.9off	440	0.1on 2.9off
11	Germany	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
12	Greece	425	0.1on 0.9off	425	0.1on 1.9off	425	0.15on 0.25off 0.15on 1.45off
13	India	400*25	0.1on 0.9off	400*25	0.1on 1.9off	400	0.15on 4.85off
14	Indonesia	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
15	Iran	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
16	Iraq	400	0.1on 0.9off	400	0.1on 1.9off	400	0.1on 2.9off
17	Israel	400	0.1on 0.9off	400	0.1on 1.9off	400	0.1on 2.9off

CPTG Type	Country	Feature / Programming / Prompt Tone		Routing Tone		IntrusionTone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
18	Italy	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
19	Japan	400	0.1on 0.9off	400	0.1on 1.9off	400*25	0.1on 2.9off
20	Kenya	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
21	Korea	350+440	0.1on 0.9off	350+440	0.1on 1.9off	350+440	0.1on 2.9off
22	Malaysia	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
23	Mexico	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
24	New Zealand	400	0.1on 0.9off	400	0.1on 1.9off	425	0.1on 2.9off
25	Phillippines	425	0.1on 0.9off	425	0.1on 1.9off	440	0.1on 2.9off
26	Poland	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
27	Portugal	425	0.1on 0.9off	425	0.1on 1.9off	425	0.2on 1.4off
28	Russia	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
29	Saudi Arabia	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
30	Singapore	425	0.1on 0.9off	425	0.1on 1.9off	425	0.25on 2.0off
31	South Africa	400*33	0.1on 0.9off	400*33	0.1on 1.9off	400	0.15on 0.25off 0.15on 1.45off
32	Spain	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
33	Thailand	400*50	0.1on 0.9off	400*50	0.1on 1.9off	400	0.1on 2.9off
34	Turkey	450	0.1on 0.9off	450	0.1on 1.9off	450	0.1on 2.9off
35	UAE	350+440	0.1on 0.9off	350+440	0.1on 1.9off	350+440	0.1on 2.9off
36	UK	350+440	0.1on 0.9off	350+440	0.1on 1.9off	400	0.2on 4.8off
37	USA	350+440	0.1on 0.9off	350+440	0.1on 1.9off	480+620	0.5on 0.5off
38	Type4	400	1.75on 0.1off	400	0.1on 1.9off	400	0.2on 0.2off 0.2on 2.5off
39	Belgium	425	0.1on 0.9off	425	0.1on 1.9off	440	0.1on 2.9off
40	Type5	350+440	0.1on 0.9off	350+440	0.1on 1.9off	350+440	0.5on 0.5off 1.0on 5.0off

#### Stuttered Dial Tone

- **Frequency:** 425 Hz (applicable for all Regions)
- **Cadence (msec):** 100-100-100-100-1000-1000 (applicable for all Regions)

# Ring Type

Default values for all ring types for all the countries as per ETSI standard is programmed in the system. Ring Type table is shown below:

Ring Type	Ring Cadence						Supported Country
	ON Time 1 (msec)	OFF Time 1 (msec)	ON Time 2 (msec)	OFF Time 2 (msec)	ON Time 3 (msec)	OFF Time 3 (msec)	
1	Infinite						
2	750	750	0	0	0	0	
3	500	1500	0	0	0	0	
4	750	2250	0	0	0	0	
5	1500	500	0	0	0	0	
6	1000	4000	0	0	0	0	Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
7	2000	4000	0	0	0	0	Egypt, USA, Canada, Namibia
8	400	200	400	2000	0	0	Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
9	400	200	400	200	400	2000	
10	1000	2000	0	0	0	0	Japan
11	1000	3000	0	0	0	0	China, Korea, Russia, Belgium, Taiwan
12	1000	5000	0	0	0	0	Portugal, Sweden
13	1500	3000	0	0	0	0	Spain
14	1500	3500	0	0	0	0	France
15	2000	3000	0	0	0	0	Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3500	5500	790	1100	0	0	



*The Ring Type is not set to default, if the system is set to default.*

# Product Specifications

## Port Description

Port Name	Application
FXS Port	To connect analog phone or fax machine
FXO Port	To connect PSTN or FXS Ports of PBX
Radio Port	To connect Radio devices such as Radio Handsets, Radio Base Stations.
Mobile Port	To connect GSM/CDMA Network
SIP Trunks	To connect Internet Network for VoIP
LAN Port	To connect PC/Computers
WAN Port	To connect Internet

## Different Configurations supported in the SETU VGRX

Sr. No.	Configuration	VoIP Channels	WAN Port	LAN Port	FXS Ports	FXO Ports	Radio Ports	Mobile Ports
1	SETU VGRX (GSM)	8	1	1	2	2	4	4
2	SETU VGRX (CDMA)	8	1	1	2	2	4	4

## Radio Parameters

Connector	RJ45
Line Input	Balanced, Transformer Isolated 1V rms, 600 ohm
Line Output	Balanced, Transformer Isolated 1V rms, 600 ohm
PTT Output	Opto Isolated Normally Open, 1A max

## GSM Parameters

GSM Frequency Band	Quad-band: GSM 850, EGSM 900, DCS 1800, PCS 1900
Compliant	ETSI GSM Phase 2/2+
SIM Card	One SIM per GSM Port
SIM Interface	1.8V, 3V
Transmission Power	Class 4 (2W) at EGSM900 and GSM850 MHz band Class 1 (2W) at DCS1800 and PCS1900 MHz band
RF Sensitivity	Better than -106dBm for GSM850, EGSM900, DCS1800, PCS1900

## CDMA Parameters

CDMA Frequency Band	800MHz CDMA cellular; 1900MHz PCS
Compliant	CDMA 1x; IS-95A
RUIM Card	One RUIM per CDMA Port

RUIM Interface	1.8V, 3V
Max. Output Power	+25dBm
RF Sensitivity	Typical -107dBm for 800MHz cellular and 1900MHz PCS

## Antenna

	2G	3G
Type of Antenna	One Antenna per Mobile Port, Fixed Omni Directional Swivel Antenna	One Antenna per Mobile Port, Fixed Omni Directional Swivel Antenna
Antenna Gain	1.8dBi	3.0dBi
Antenna Connector	SMA (Male), 50Ω Impedance	SMA (Male), 50Ω Impedance

## FXS Port Parameters

Signaling	Loop Start
Connector	RJ11
Off-hook Line Impedance	600Ω/900Ω/Complex
Number of Long Loop Extension	4
Loop Limit	1800Ω (Max) Excluding Telephone Set
On-hook Voltage (Tip/Ring)	-48V
Off-hook Current	25mA - 40mA (Max)
Ringing Voltage	Trapezoidal 60 Vrms at 25Hz and Sinusoidal 52Vrms at 25Hz
REN	3
DTMF Detection	ITU-T Q.23 and Q.24
CLI Presentation	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage and Over Current Secondary Protection
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -12dB to +6dB; Rx Gain: -12dB to +6dB
Answer Signaling on FXS	Battery Reversal
Disconnect Signaling on FXS	Battery Reversal and Open Loop Disconnect

## FXO Port Parameters

Signaling	Loop Start
Connector	RJ11
Off-hook Line Impedance	600Ω/900Ω/Complex
Loop Limit	1200Ω

Pulse Dialing	10PPS and 20PPS
DTMF Dialing and Reception	ITU-T Q.23 and Q.24
CLI Reception	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage and Over Current Secondary Protection
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -15dB to +10dB; Rx Gain: -15dB to +10dB
Answer Supervision on FXO	Battery Reversal and Delay
Disconnect Supervision on FXO	Battery Reversal and Open Loop Disconnect

## VoIP Parameters

Connector	RJ45
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833), SRTP
Network Protocols	IPv4, TCP, UDP, DHCP, PPPoE, SNTP, NAT, STUN, HTTP, TLS, DynDNS
SIP	9 SIP Trunks with Out Bound Proxy Support
NAT	STUN and NAT Keep Alive
Voice Codec's	G.711 (a-Law and mu-Law), G.729, G.723, GSM FR
Line Echo Cancellation	G.168 with 128ms Tail Length
Call Progress Tones	Dial Tone, Ring Back Tone, Busy Tone, Error Tone
Voice	Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Fax	T.38(UDPTL), T.38(RTP) and Pass Through
Quality of Service	Layer 3 Diffserv and TOS
Data Network	2 Ports Auto MDIX 10/100 Base-T (RJ-45)
Security	Password Protected Administration

## Telephony features

- Voice Calls using SIP proxy
- Voice calls without using SIP proxy (Peer-to-Peer Calling)
- Battery Reversal- Useful when some billing machine is connected to SETU VGRX
- Black Listed Callers
- Call Disconnect Tone
- Call Forwarding
- Call Hold
- Call Pick-up
- Call Toggle- Used to toggle between active and held call
- Call Transfer
- Call Waiting
- Conference
- Disconnect Call using Access Code
- Do Not Disturb (DND)- Incoming calls can be rejected

- Hotline
- Making New Call
- Making Second Call
- Message Wait Indication

## Time Settings

Synchronizing with specific Time Server

## Provisioning, Administration and Maintenance

- Auto Firmware and Configuration upgrade
- Programmable using Web Jeeves

## LED Indication (Total 15 LEDs)

- Power = 1
- Status = 1
- Port = 12
- Reverse Battery = 1

## Packing

- **Dimension (W x H x D):** 278x44.2x169.5 (mm)
- **Unit Weight:** 1.1 Kg
- **Shipping Weight:** 2.750 Kg
- **Mounting:** Wall Mounting; Table-Top and 19" Rack Mount

## Power Supply

### Adapter

- **External Adapter:** 12V DC @ 2A
- **Power Consumption:** 15W Typical
- **Connector:** DC Power Jack

### Battery Terminal

- **Battery Supply Supported:** 12/24/48 VDC
- **Power Consumption:** 15W Typical
- **Connector:** 2-Pin DC Terminal Connector

## Environmental

- **Operating Temperature:** 0°C to 45°C
- **Storage Temperature:** -20°C to 70°C
- **Operating Humidity:** 5-95% RH (Non-Condensing)
- **Storage Humidity:** Max. 0-95% RH (Non-Condensing)

## Features at Glance

Feature Description	Feature Code
To Enter Programming Mode	<b>#19-Command Password (Default=1234)</b>
To Exit Programming Mode	<b>00#*</b>
To Set Hotline	<b>#151-1</b>
To Cancel Hotline	<b>#151-0</b>
To Enable Call Waiting	<b>#16-1</b>
To Disable Call Waiting	<b>#16-0</b>
To Set DND	<b>#18-1</b>
To Cancel DND	<b>#18-0</b>
To Set Call Forward Unconditional	<b>#131-1</b>
To Cancel Call Forward Unconditional	<b>#131-0</b>
To Set Call Forward Busy	<b>#132-1</b>
To Cancel Call Forward Busy	<b>#132-0</b>
To Set Call Forward No Reply	<b>#133-1</b>
To Cancel Call Forward No Reply	<b>#133-0</b>
To Program Hotline Number	<b>#152-Destination Number-End-of-Dialing<sup>a</sup></b>
To Program Hotline Timer	<b>#153-X (X is the timer value)</b>
To Program Call Forward Unconditional Number	<b>#135-Destination number-End-of-Dialing<sup>a</sup></b>
To Program Call Forward Busy Number	<b>#136-Destination Number-End-of-Dialing<sup>a</sup></b>
To Program Call Forward No Reply Number	<b>#137-Destination Number-End-of-Dialing<sup>a</sup></b>
To Program No-Reply Timer	<b>#139-XX (XX is time in seconds)</b>
For Call Pick-up	<b>#5</b>
For Call Hold	<b>Flash</b>
To Retrieve Held Call	<b>Flash</b>
For Call Toggle (Call Split)	<b>#2</b>
To Reject the Waiting Call and Speech with Current Call	<b>#31</b>
To Ignore the Waiting Call and Speech with Current Call	<b>#32</b>
To Accept the Waiting Call and Hold Current Call	<b>#33</b>
To Accept the Waiting Call and Release Current Call	<b>#34</b>
For Blind Transfer	<b>#6</b>
For Conference	<b>#8</b>
For Using Supplementary Services of Service Provider	<b>#4</b>
For Using Voice Mail of the Service Provider	<b>#7</b>

Feature Description	Feature Code
For Attended Transfer	^
For Making a New Call	<b>#91</b>
To Disconnect Call	<b>#92</b>
To release a Radio Port forcibly	<b>#93</b>

a. *Dial # as end of dialing, if it has been configured by you or the system will wait till the expiry of the inter digit wait timer.*

# System Commands

---

Certain parameters of SETU VGRX can be configured by dialing System Commands from a telephone connected to the FXS Port. You can configure certain Network Parameters, like IP Address, Subnet Mask, Connection Type, set the system to default and also view current IP Address, Subnet Mask, Connection Type, DNS and Gateway Address by dialing System Commands.

To be able to view these details, the telephone connected to the FXS Port must have an LCD display.

To dial System Commands,

- Pick up the handset of the telephone connected to the FXS Port of SETU VGRX.
- Dial **#19-Command Password** to enter programming mode.  
You will get programming tone.
- To change WAN Port IP Address, dial **11-IP Address-#\***  
For example, to change IP address to 192.168.1.120, dial **11-192168001120-#\***  
Default: 192.168.1.100
- To change WAN Port Subnet Mask, dial **12-Subnet Mask-#\***  
Where,  
Valid range for Subnet Mask is 0, 128, 192, 224, 240, 248, 252, 254 and 255.  
For example, to change Subnet Mask to 255.255.254.0, dial **12-255255254000-#\***  
Default: 255.255.255.000
- To configure WAN Port Gateway IP Address, dial **13-Gateway Address-#\***  
Where,  
Gateway IP Address is of 12 digits.  
For example, to change Gateway Address to 192.168.9.10, dial **13-192168009010-#\***  
Default: Blank  
  
To change Gateway IP Address to Blank, dial **13-#\***
- To change LAN Port IP Address, dial **41-IP Address-#\***  
For example, to change IP address to 192.168.1.140, dial **41-192168001140-#\***  
Default: 192.168.2.100
- To change LAN Port Subnet Mask, dial **42-Subnet Mask-#\***  
Where,  
Valid range for Subnet Mask is 0, 128, 192, 224, 240, 248, 252, 254 and 255.  
For example, to change Subnet Mask to 255.255.253.0, dial **42-255255253000-#\***  
Default: 255.255.255.000
- To select the Connection type, dial **10-Code-#\***  
Where,  
Code is 1 for Static, 2 for DHCP, 3 for PPPoE  
Default: Static
- To enable/disable VLAN Tag, dial **31-Code-#\***  
Where,  
Code is 1 for Enable, 0 for Disable  
Default: Disable

- To restore factory defaults, dial **51-Reverse Command Password-#\***  
Replace the handset. The system will restart.
- To view Connection Type, dial **20-#\*** and go On-hook
- To view the Network IP Address, dial **21-#\*** and go On-hook
- To view the Subnet Mask, dial **22-#\*** and go On-hook
- To view the Gateway Address, dial **23-#\*** and go On-hook
- To view the DNS Address, dial **24-#\*** and go On-hook
- To display the LAN Port IP address on Phone, dial **46-#\*** and go On-hook.
- To display the LAN Port Subnet Mask on Phone, dial **47-#\*** and go On-hook.
- To view the Status of SIP Trunks, dial **27-SIP Trunk number-#\*** and go On-hook. SIP Trunk Number is from 1 to 9.

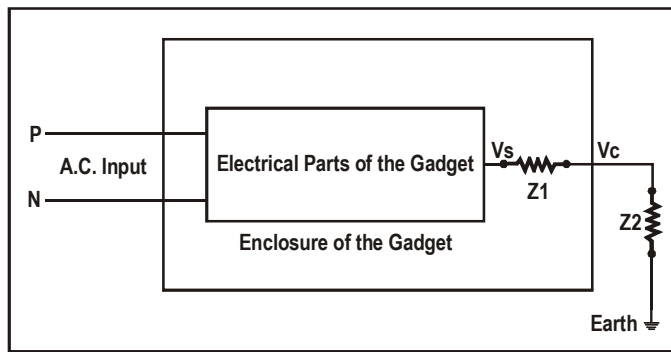
The value of the parameter will be displayed on the LCD of the telephone instrument.

- To exit programming mode, dial **00-#\***

# How to Make the Telecom Earth

The Earth (Ground) is the most important safety procedure to prevent electrical shocks and fires. It protects from lightning strikes, electrical transients, static discharges, electromagnetic interference and electrical hazards.

A proper earth must be in place to protect people and the system. The following explanation shows how a perfect electrical earth can save lives.



In the above diagram,  $V_c = V_s * Z_2 / Z_1 + Z_2$

Where,

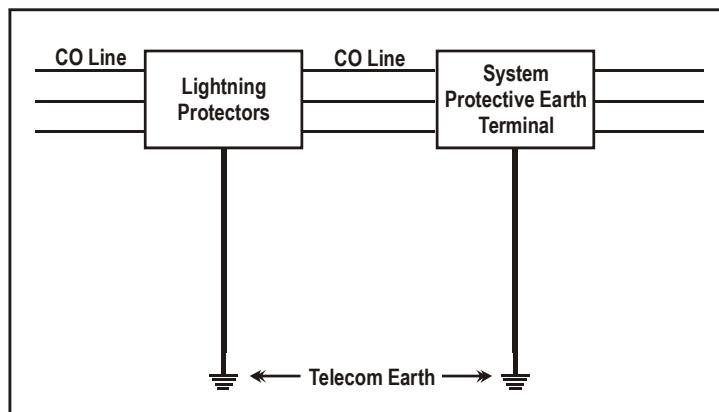
- $Z_1$  is the stray impedance between the electrical parts of the Gadget and the Chassis.
- $Z_2$  is the stray impedance between the Chassis and the Earth.
- If  $Z_2 = 0$  then  $V_c = 0$

This formula implies that if the impedance between the Chassis and the Earth is reduced to Zero, then the Voltage on the Chassis, that is,  $V_c$ , would be Zero and hence any person touching the enclosure will not get an electric shock. Hence  $Z_2$  should be made Zero.

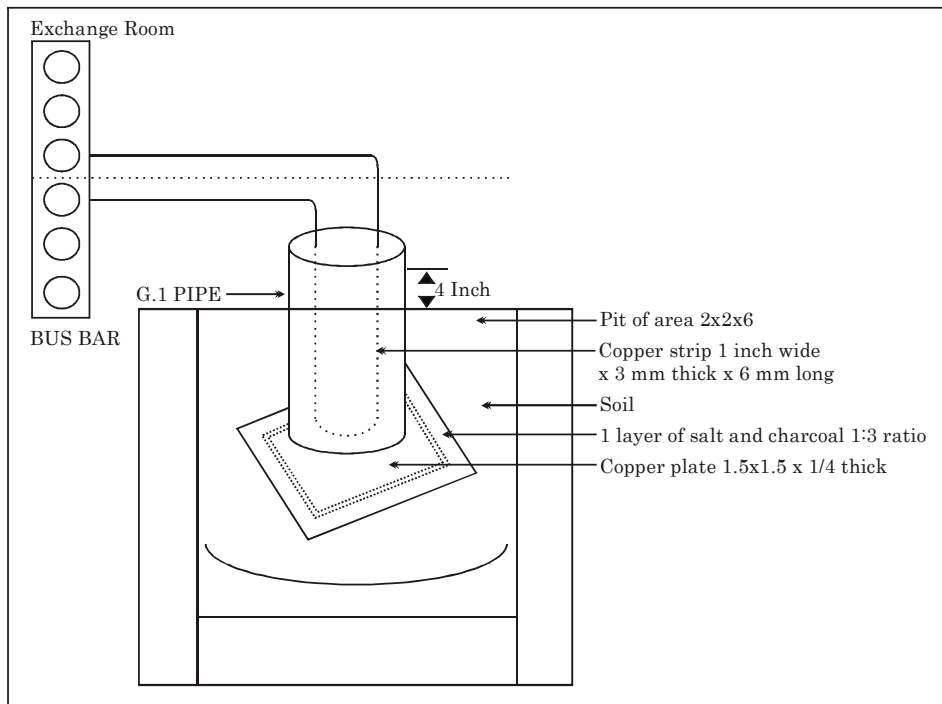
This can be done by providing a perfect earth to the electrical equipment.

It is recommended that you provide a dedicated earth for the system/any other telecom equipment. This dedicated earth is called the Telecom Earth (Ground).

Providing a separate Telecom Earth to the telecom equipment eliminates the possibility of any back-voltage on the earth.



## How to make a perfect Earth



- Dig a pit of area 2 feet x 2 feet x 6 feet (L x B x D).
- Get a copper plate of size 1.5 feet x 1.5 feet x 0.25 feet.
- Connect a copper strip of size 1-inch wide, 3 mm thick and 6 feet length at the center of the copper plate by welding or nuts and bolts.
- Insert a G.I. pipe onto the copper strip till it reaches the copper plate.
- Place this set up into the pit. Make sure that at least 4 inches of the G.I. pipe is above the ground level.
- Fill the bottom of the pit with a 1-inch layer of charcoal and salt in the proportion of 3:1 (3 parts charcoal, 1 part salt) and then cover with the soil.
- Connect a bare 14 SWG copper wire (double) on the top of the copper strip and run it to the exchange room and connect it on the bus bar.
- The Bus bar is a copper strip, 4 inches long with 6 screws and nuts mounted on it. It has to be fixed on the wall in the exchange room.
- The earth wire of the Primary Protection Modules (PPM)/system should be connected to this Bus bar.
- Water the earth at regular intervals.



- *Make sure you comply with all applicable laws, regulations and guidelines.*
- *Proper earthing is very important to protect the system from external noise and to reduce the risk of electrocution in the event of a lightning strike.*

# Warranty Statement

---

Matrix warrants that its products will be free from defects in material and workmanship, under normal use and service for a period of twelve (12) months from the date of installation.

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5. If the product is operated outside the product's specifications or used without designated protections.
6. If the completely filled warranty cards have not been received by Matrix within 15 days of the installation.

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- **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"**

# Regulatory Information

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## MILITARY Compliance

SETU VGRX complies with the following MIL-STD-461F Standards:

- MIL-STD-461F\_CS101
- MIL-STD-461F\_CS114
- MIL-STD-461F\_CS115
- MIL-STD-461F\_CS116
- MIL-STD-461F\_CE102
- MIL-STD-461F\_RE102



*Please note that all the above mentioned tests are performed under specific lab conditions.*

## ROHS Certificate



### Declaration of Conformity

Manufacturer's Name : Matrix Comsec Pvt. Ltd.

Manufacturer's Address : 15 & 19-GIDC, Waghodia,  
Dist: Vadodara 391760  
Gujarat, India

Declares that the product/s  
Product : Radio-over IP Gateway with Integrated GSM/ CDMA Connectivity

Model/ Type : SETU VGRX

Trade Name : MATRIX

The designated products are in conformity with the below mentioned European RoHS (Restriction of use of certain Hazardous Substances) directive ;

**RoHS Recast (RoHS 2) : Directive 2011/65/EU (as per standard EN 50581:2012)**



**Mr. Ganesh Jivani**  
Director  
Date: 20.01.2016  
Vadodara

**MATRIX COMSEC PVT. LTD.**

Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: [inquiry@MatrixComSec.com](mailto:inquiry@MatrixComSec.com) • [www.MatrixComSec.com](http://www.MatrixComSec.com)

Factory: 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73

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```
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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.
```

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```
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`Gnomovision' (which makes passes at compilers) written by James Hacker.
```

```
<signature of Ty Coon>, 1 April 1989
Ty Coon, President of Vice
```

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