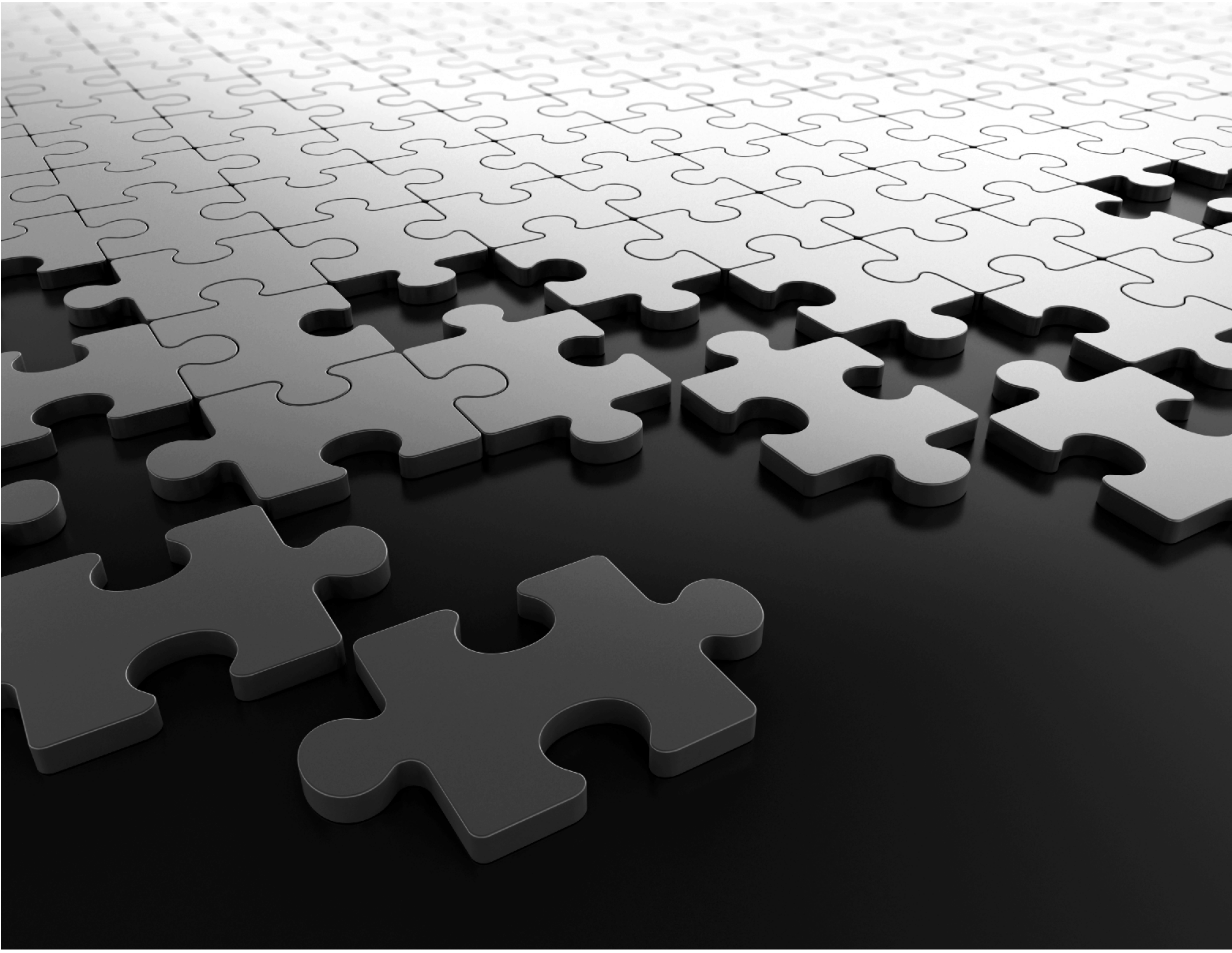
SETU VFX
System Manual



TELECOM SOLUTIONS

SETU VFX

Low-Density Multi-Port VoIP-FXO-FXS Gateway

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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Contents

CHAPTER 1

Introduction

Welcome

Thank you for choosing SETU VFX! We hope you will make optimum use of this intelligent, intuitive, feature-packed Multi-Port VoIP to FXO-FXS Gateway. Please read this document carefully before installing your SETU VFX.

About this System Manual

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

You may also refer to the *SETU VFX Quick Start* for quick installation. For instructions on using the features, you may refer the *SETU VFX User Card*. To view or download these documents, scan the QR Code printed on the Product Label/Packaging Label.

The documentation can be found at https://www.matrixtelesol.com/product-manuals.html

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

This system manual is written with reference to two variants of SETU VFX: SETU VFX880 (VoIP-FXO Gateway) and SETU VFX808 (VoIP-FXS Gateway).

Intended Audience

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

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-FXO-FXS Gateways.

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

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 VFX: Provides an overview of SETU VFX.

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

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

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

Features: Describes the telephony features of SETU VFX and provides instructions on using these features.

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

Status: Describes the indicators of the System, Network, SIP Trunks 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 these 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 to remind you of something you need to do when you are using the SETU VFX.*

Tip: *It indicates a helpful hint giving you an alternative way to operate the SETU VFX 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 VFX or your property.*

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

Terminology used in this System Manual

In this system manual, words **SETU VFX**, **System** and **Gateway** are used synonymously. Some of the terms specific to this document are defined below:

Term	Usage in the document
System Engineer (SE)	The person who installs, configures and maintains SETU VFX.
User	The person who uses SETU VFX.

Term	Usage in the document
Caller / Calling party	The person who make calls to SETU VFX.
Callee / Called party	The person to whom calls are made using SETU VFX.
Source / Originating Port	A port from which a call originates.
Destination / Terminating Port	A port on which a call terminates.

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

CHAPTER 2

Know Your SETU VFX

Overview of SETU VFX

SETU VFX is a SIP based gateway that provides voice services over IP network. It is an effective solution for accessing internet-based telephone services in the existing established LAN.

It is an innovative enterprise gateway that offers excellent functionality and sound quality. It supports following voice ports: FXS, FXO and SIP.

It is a feature packed gateway to fulfil the requirements of SOHO (Small Office-Home Office) users as well as Medium scale Enterprises.

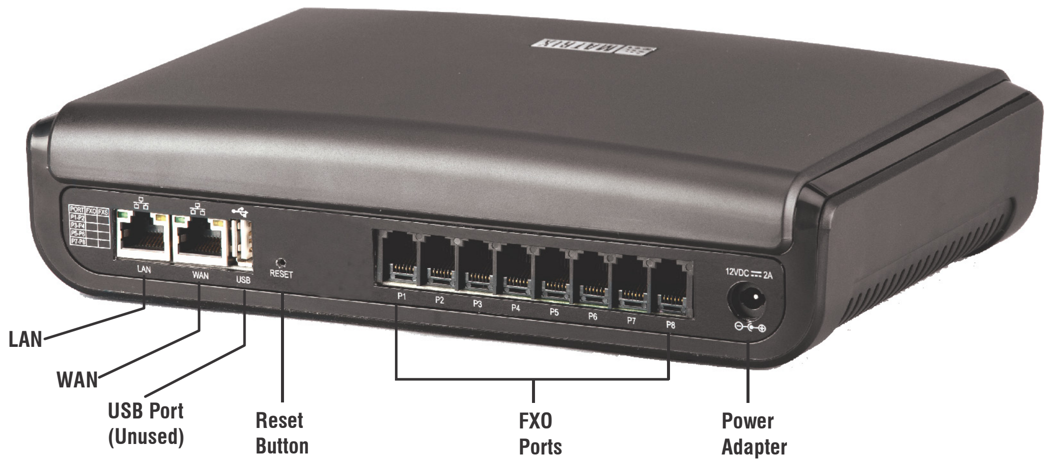
This product is available in following configurations:

Sr. No.	Configuration	Voice Channels	FXO Ports	FXS Ports	WAN Port	LAN Port	Lifeline Port
1	SETU VFX880	8	8	0	1	1	0
2	SETU VFX808	8	0	8	1	1	1
4	SETU VFX440	4	4	0	1	1	0
3	SETU VFX404	4	0	4	1	1	1

This is the common document for all the above mentioned configurations. For a complete list of Hardware and Software features, refer “Product Specifications” in the Appendix.

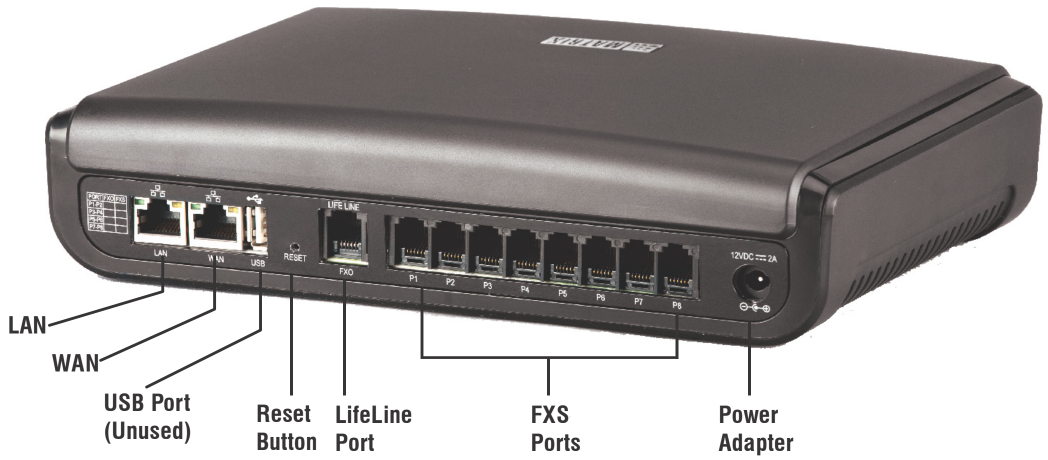
SETU VFX880 (VoIP-FXO Gateway)

SETU VFX880 has a WAN Port, a LAN Port, a USB Port, a Reset button, 8 FXO Ports, 9 SIP Trunks, a Power Socket.



SETU VFX808 (VoIP-FXS Gateway)

SETU VFX808 a WAN Port, a LAN Port, a USB Port, a Reset button, a Lifeline Port, 8 FXS Ports, 9 SIP Trunks, a Power Socket.



Ports and Connectors

Port	Connector	Description
12VDC-2A	DC Jack	To connect 12VDC, 2A Power Adapter.
P1 to P8 (FXO[a])	RJ11	To connect PSTN lines or a PBX.
P1 to P8 (FXS[b])	RJ11	To connect standard Telephone Instruments, or a Fax Machine or a PBX.
LifeLine Port[c]	RJ11	To access PSTN line during power failure/emergency conditions.
Reset Button	---	To restart the system or to restore the default LAN IP Address.

Port	Connector	Description
USB Port	---	Unused
WAN Port	RJ45	To connect to the IP network over a DSL Modem or Router or a LAN Switch.
LAN Port	RJ45	To connect a computer or a LAN Switch.

a. FXO Port is available only in VoIP-FXO Gateway.
b. FXS Port is available only in VoIP-FXS Gateway.
c. Life Line Port is available only in VoIP-FXS Gateway.

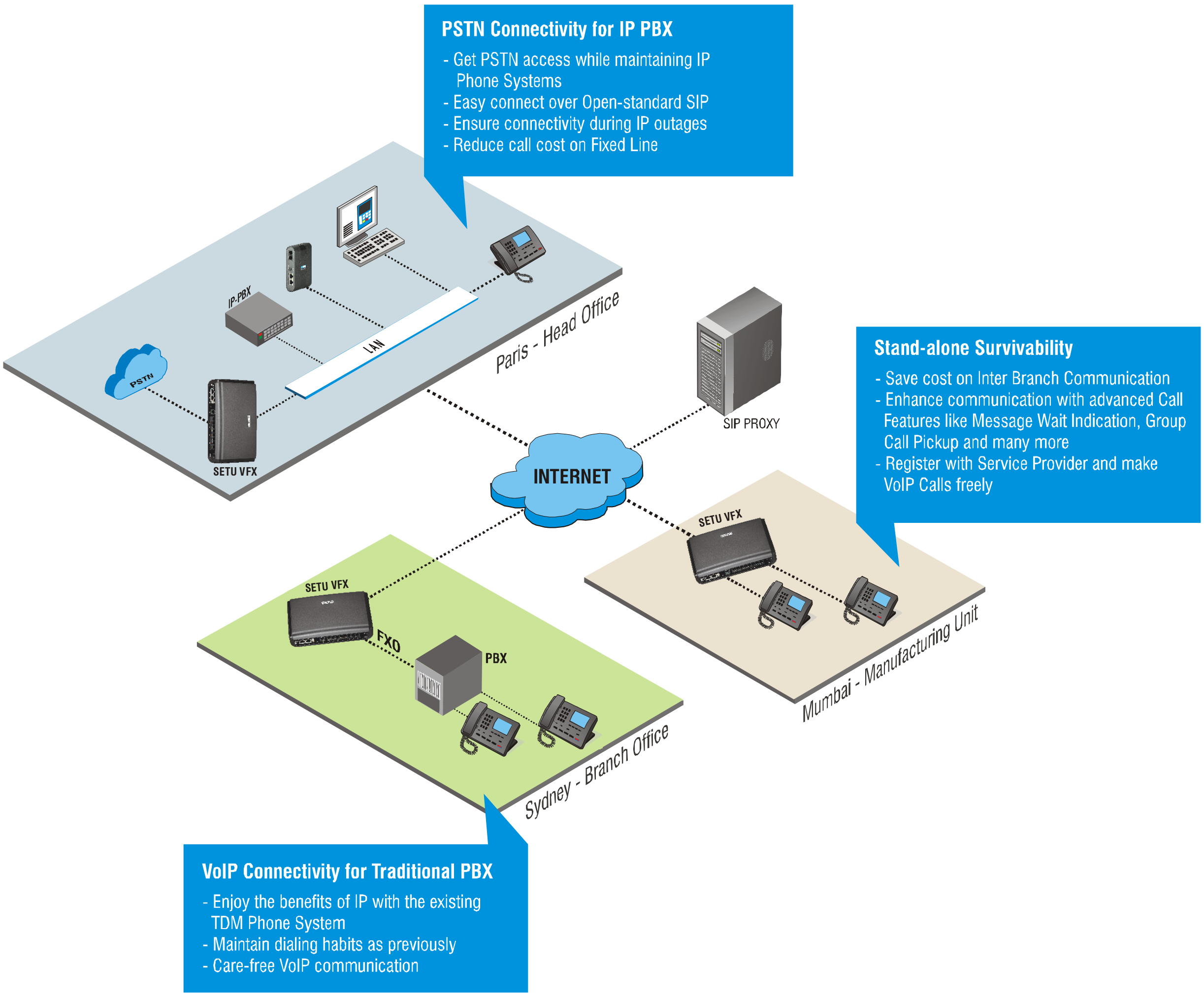
LEDs

SETU VFX has total 10 LEDs. One for Power (PWR), one for Status (STS) and eight Port LEDs (P1 to P8). The LEDs indicate the status of the ports and various events occurring on the ports, including errors.

To each LED, except Power and Status, you can assign a port of your choice. So, if required, you can assign some or all of these LEDs to SIP Trunks. To know more, see “Port LED”.

SETU VFX 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 VFX



CHAPTER 3

Installing SETU VFX

Before You Start

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

- A suitable location to install SETU VFX.
- 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 VFX.
- Appropriate cables and connectors to set up and test the WAN interface of SETU VFX and the LAN connection.
- An Analog Trunk Line from the CO/PSTN to connect to the FXO Port, if applicable.
- At least one standard telephone instrument to connect to the FXS Port, if applicable. 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.

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

Protect SETU VFX 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.

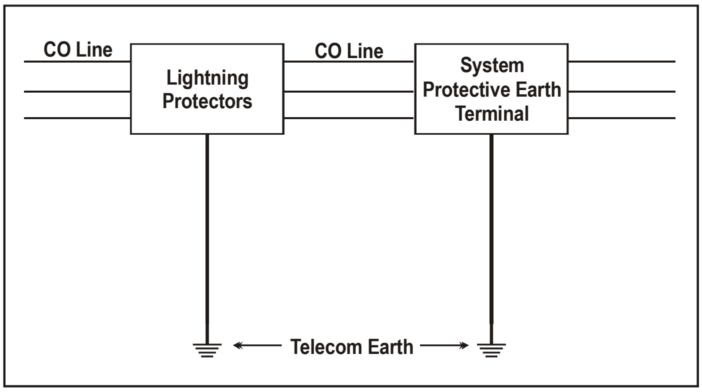
- Always wear an electrostatic discharge preventive wrist strap or belt and use a grounding mat when handling the system.
- Unplug the system from the power outlet before cleaning. Do not use liquid cleaners, use only a dry and soft cloth.
- Do not turn on the power supply until the installation is complete.
- Never open SETU VFX 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.

Protecting the system from Lightning[1]

- You must protect the system from high current surges by installing a primary protection device. A lightning protector is used to prevent a dangerous surge from entering the building and damaging the system. It is necessary to protect the system with primary protectors such as PPM, when it is installed at a location which is frequently exposed to lightning. Problems due to lightning surges have increased with the increase in the use of electronic equipments. A dangerous surge can occur if a telephone line comes in contact with a power line. You must install a lightning protector on an outside (CO) line to prevent a dangerous surge from entering the building and damaging the system.

1. *Applicable only for VoIP-FXO Gateway.*

- 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. In addition, grounding (connection to earth ground) is very important to the system.

Battery

SETU VFX 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. See "Disposal of Products/Components after End-Of-Life".

Getting Started

- Select an appropriate site to install the SETU VFX, considering the safety precautions listed earlier in this chapter.
- Unpack SETU VFX and verify the package contents:
  - SETU VFX Unit
  - Power Adapter - 12VDC, 2A (Country Specific)
  - Ethernet Cable (RJ45)
  - Line Cord (RJ11)[2]
  - Two Screws M7/30 with grips
  - Mounting Template
  - Warranty Card Set

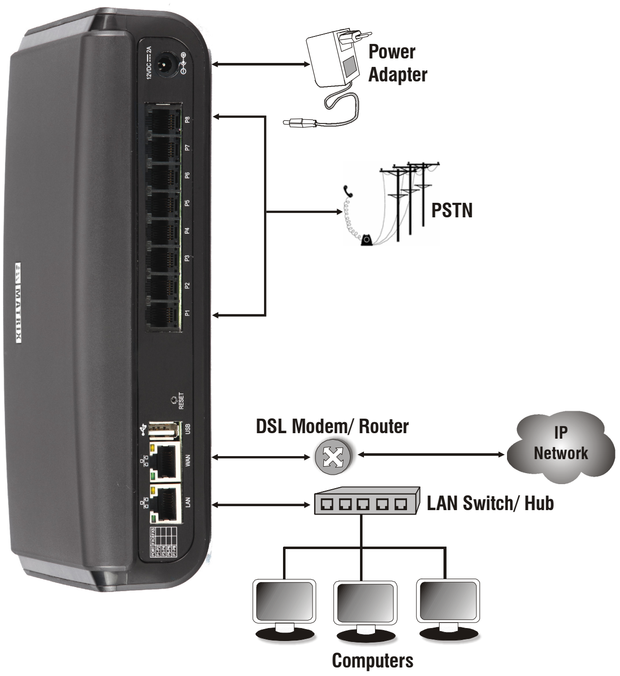
2. *Applicable only for VoIP-FXS Gateway.*

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

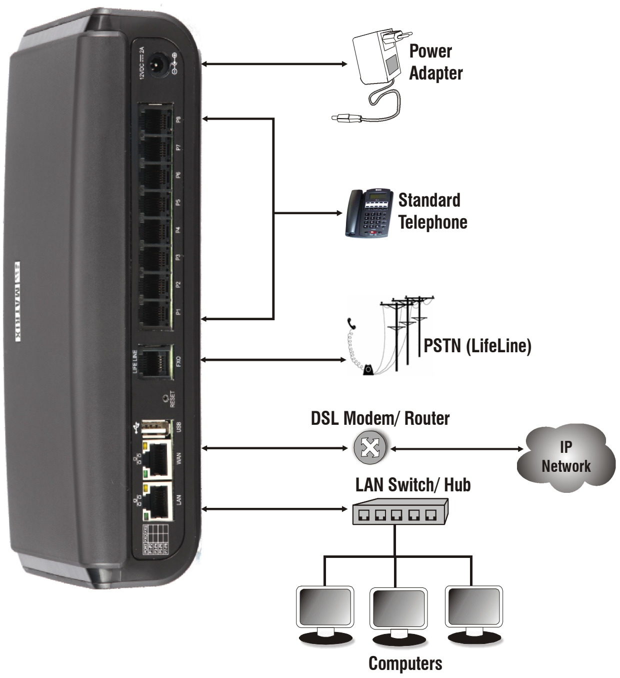
- Place the system at the selected location.
- You may mount the system on a wall. Refer the mounting template for dimensions and accordingly drill the holes on the wall.

Connecting SETU VFX

SETU VFX880 (VoIP-FXO Gateway)



SETU VFX808 (VoIP-FXS Gateway)

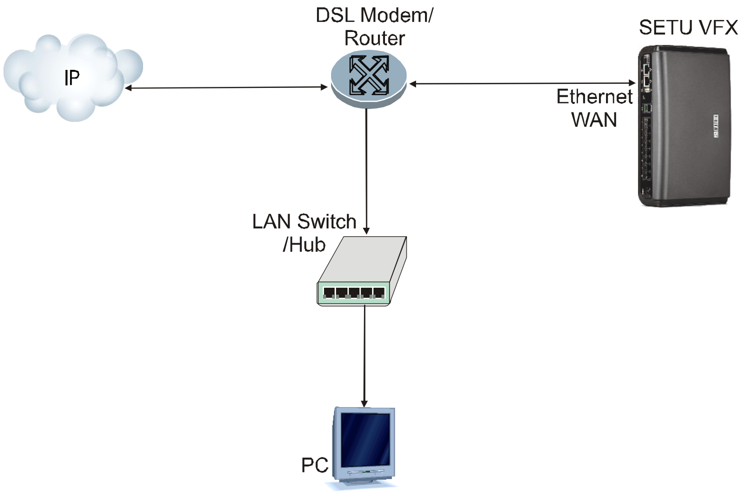


Connecting to the IP Network

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

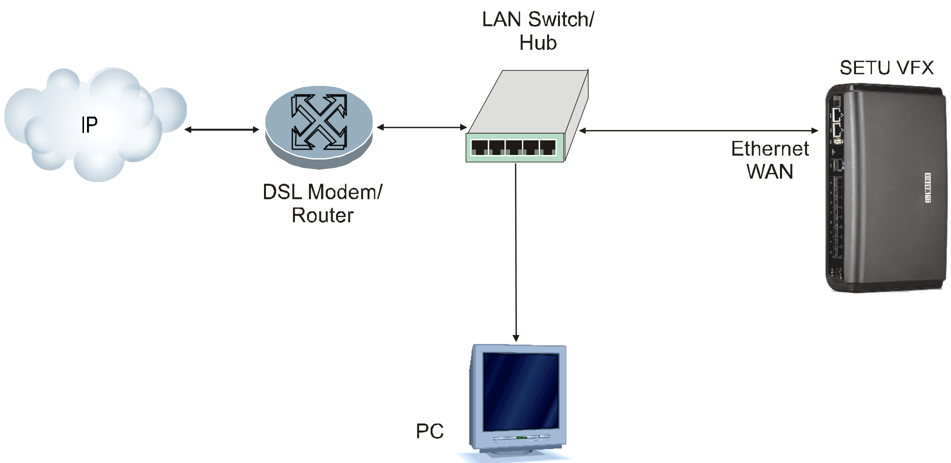
If connecting to the Public IP Network,

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



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

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



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 CO Network[3]

- 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[4]

- 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.

Power ON SETU VFX

- Connect the **Power Adapter** into the power jack, and plug it into a power outlet.

- Switch ON 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	Color	STS	P1	P2	P3	P4	P5	P6	P7	P8	Time in MS
Power ON - UBOOT	Red	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
Kernel UP	Red	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
Application and LED Driver Loaded	Green	OFF	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	
Few Initialisation (i.e SysConfig, Resolver, SysLog etc.)	Green	OFF	ON	ON	OFF	OFF	OFF	OFF	OFF	OFF	
Few Initialisation (i.e WebJvs, CallManager, PortCnfg etc.)	Green	OFF	ON	ON	ON	OFF	OFF	OFF	OFF	OFF	
VOPP Program Download Success	Green	OFF	ON	ON	ON	ON	OFF	OFF	OFF	OFF	

3. *Applicable only for VoIP-FXO Gateway.*
4. *Applicable only for VoIP-FXS Gateway.*

System Status	Color	STS	P1	P2	P3	P4	P5	P6	P7	P8	Time in MS
All Init Done, System goes Live	Green	ON	ON	ON	ON	ON	ON	ON	ON	ON	1000 ms
	Red	ON	ON	ON	ON	ON	ON	ON	ON	ON	1000 ms
		OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	1000 ms
	Green	ON	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	1000 ms
		OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	1000 ms
	(Continuous Green blinking as given in Last 2 steps)										

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.
1sec On - 1sec Off	Green	SETU VFX 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.
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 down. 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.

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

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

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

You can assign any of the port status LEDs to SIP Trunks. To know more, see "Port LED".

When an LED is assigned to a SIP Trunk, it will display the following error/event/status:

LED Status	Color	Event/State/Status
Continuous Off	-	SIP Disable
Continuous On	Green	SIP Registered
Continuous On	Red	SIP Registration Failed
200ms On - 200ms Off - 200ms On - 3400 Off (2 Blinks)	Red	SIP Authentication Failed

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

Configuring SETU VFX

SETU VFX 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 VFX.

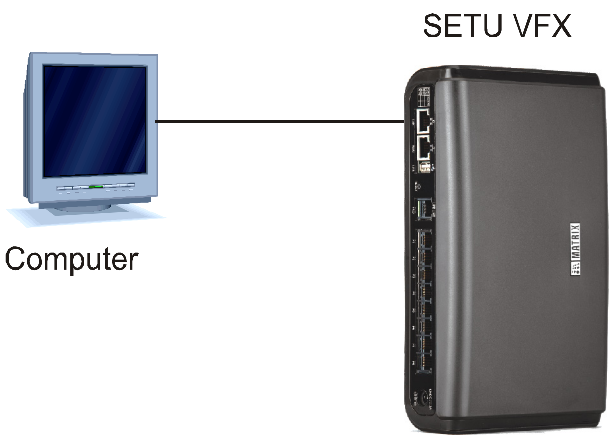
Connecting a Computer

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

- *Connect a standalone computer to SETU VFX, 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 VFX 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 VFX. 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 VFX do not conflict, and that both are in the same Subnet.

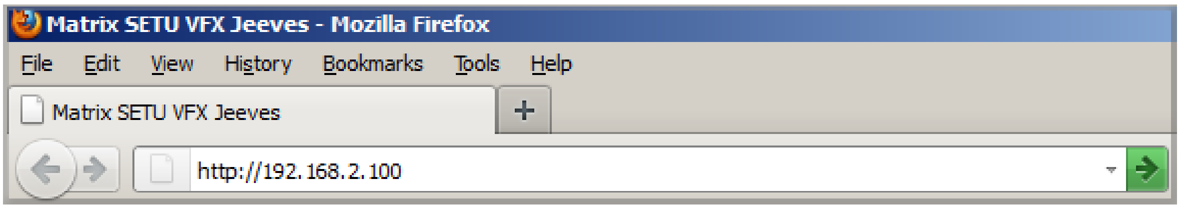
  The default IP Address of the LAN Port of SETU VFX is: **192.168.2.100**

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

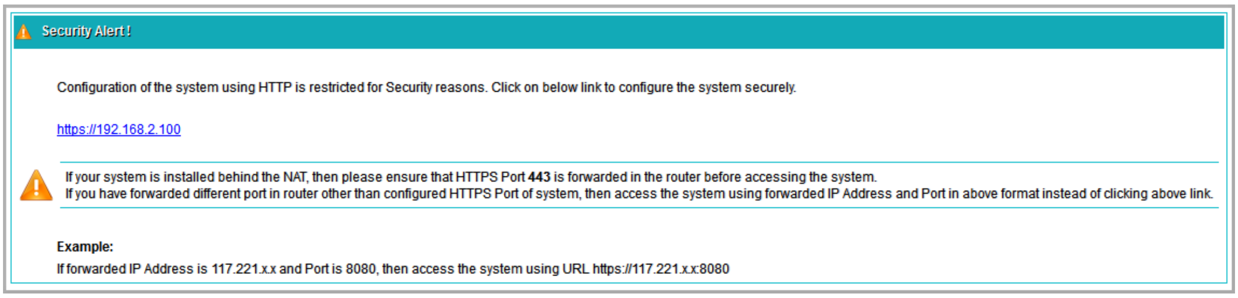
  Change the Subnet of the computer, if necessary.

- 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**



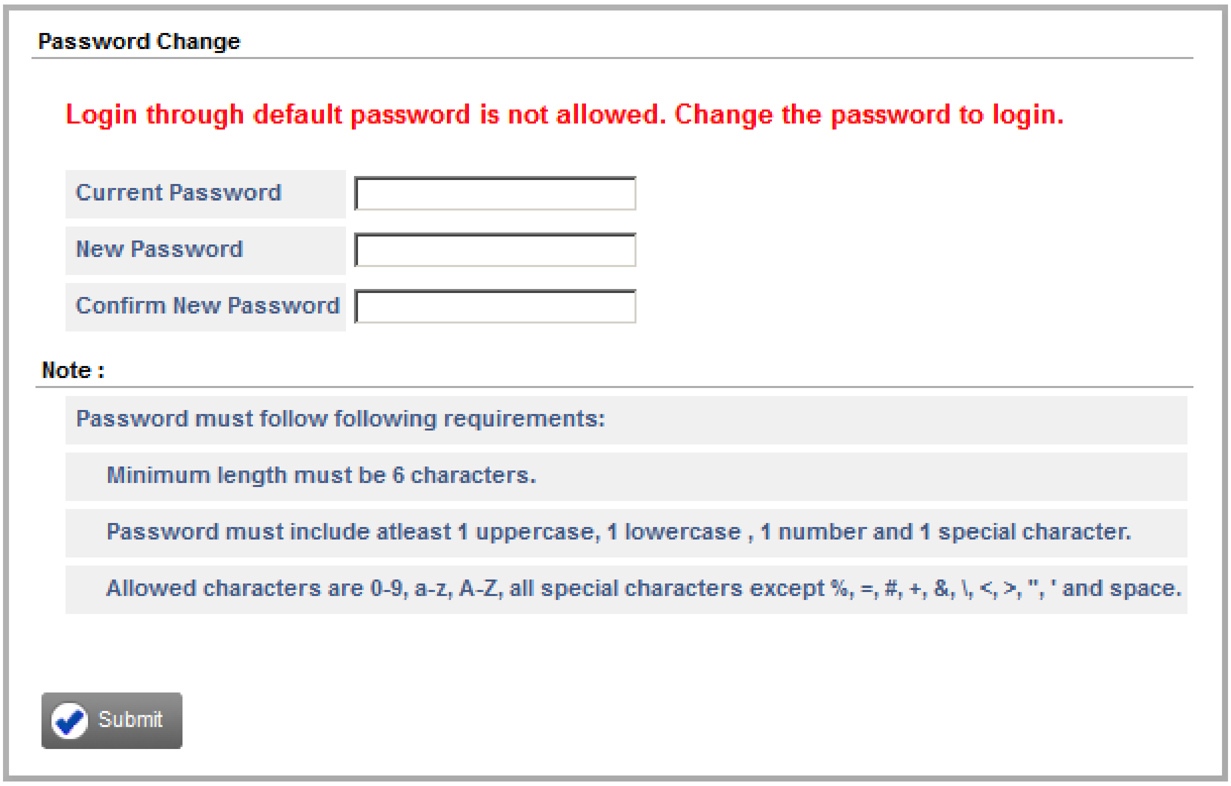
- You will be redirected to the HTTPS protocol for security reasons.



- 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.

Before you start configuring the system, if you wish to view or download the SETU VFX Quick Start/ User Card, you can scan the QR Code present on the login page of Jeeves.

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



- 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 VFX.

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 status of the ports - FXO Ports (if applicable), Network and SIP Trunks.

You may now configure the Basic Settings of SETU VFX.

If you need to change the IP Address and the Subnet Mask of the LAN Port and the WAN Port of SETU VFX, 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*.

CHAPTER 4

Basic Settings

The Basic Settings enable you to configure SETU VFX 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.

There are two ways to configure Basic Settings:

- Using the **Wizard**. The Wizard will guide you step-by-step through the configuration.

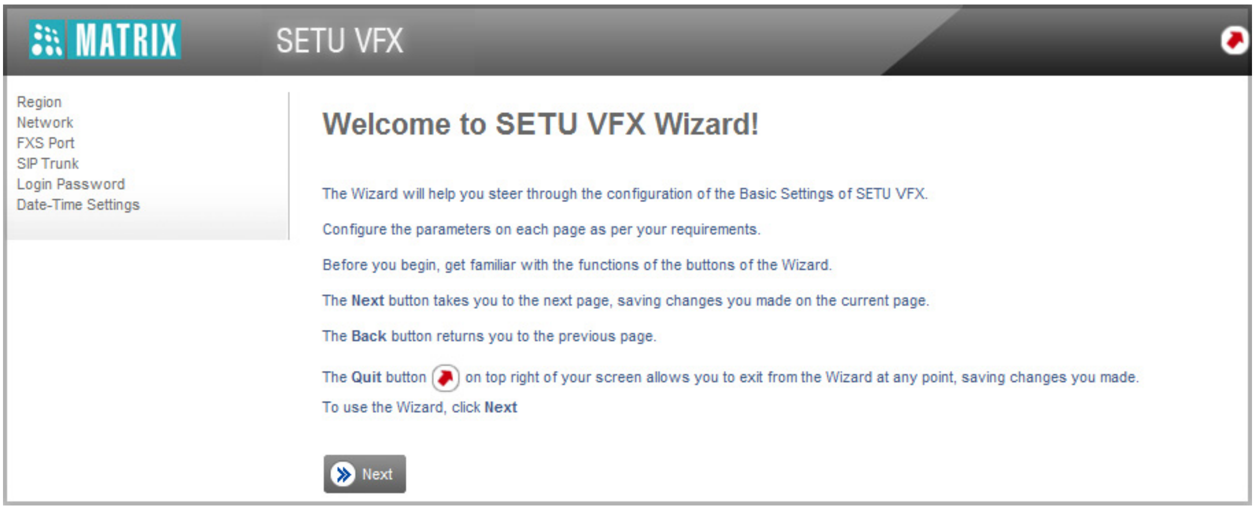
OR

- Selectively configuring the Basic Settings parameters, by clicking the links on the left navigation bar.

To use the **Wizard**,

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

- The welcome page of the Wizard opens



- Get familiar with the functions of the following buttons of the Wizard before you begin to use them.
  - **Next:** Takes you to the next page, saving the changes you made on the current page.
  - **Back:** Returns you to the previous page.
  - **Expand** ⊞ **:** Expands a parameter link to display all parameters under the link.
  - **Collapse** ⊟ **:** Collapses a link; hides all parameters under the link.
  - **Settings** **:** Enables you to configure / edit the settings of a parameter further or to edit an entry or a record.
  - **Default:** Assigns factory set values to all the parameters on the page.

- **Add:** Enables you to add a new record.
- **Delete:** Enables you to delete a record.
- **Close:** Enables you to exit a window.
- **Copy:** Enables you to copy the parameters of a port to another port.
- **Quit** **:** Enables you to exit the Wizard at any stage, saving changes you made before exiting.

To use **Selective Configuration**,

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

- Click the link of the required parameter: **Region**, **Network, FXS Port (if applicable)**, **FXO Port (if applicable)**, **SIP Trunk**, **Login Password**, **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.

You may use the Wizard or selectively configure the Basic Settings pages, whichever works best for you.

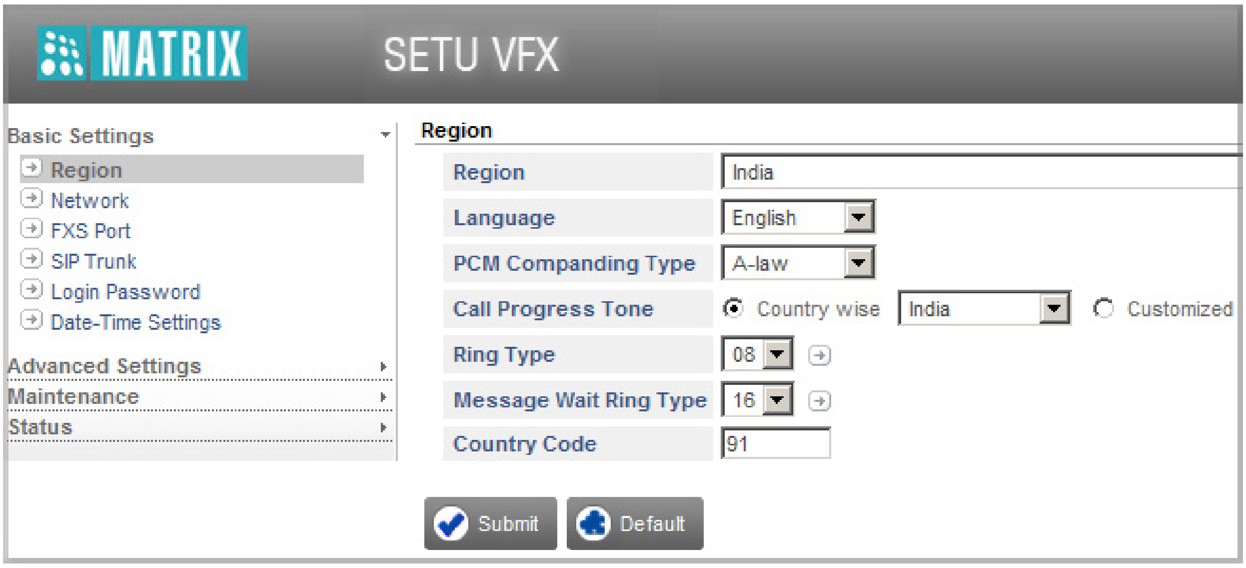
The instructions provided in this chapter describe *selective* configuration of the Basic Settings pages.

- Click **Basic Settings** to expand.

- The following links appear under Basic Settings:
  - Region
  - Network
  - FXS Port[5]
  - FXO Port[6]
  - SIP Trunk
  - Login Password
  - Date-Time Settings

Each of these is explained in detail in the following.

Region



To configure Region and other region specific parameters,

- Click the **Region** link.

5. *Applicable only for VoIP-FXS Gateway.*
6. *Applicable only for VoIP-FXO Gateway.*

Region

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

  When you change the Region, an alert message will appear on your screen **"Changing Region shall assign default values to all the 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 the **Language** list, click the language in which you want the pages of the GUI, Jeeves, to be presented.

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

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

  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.

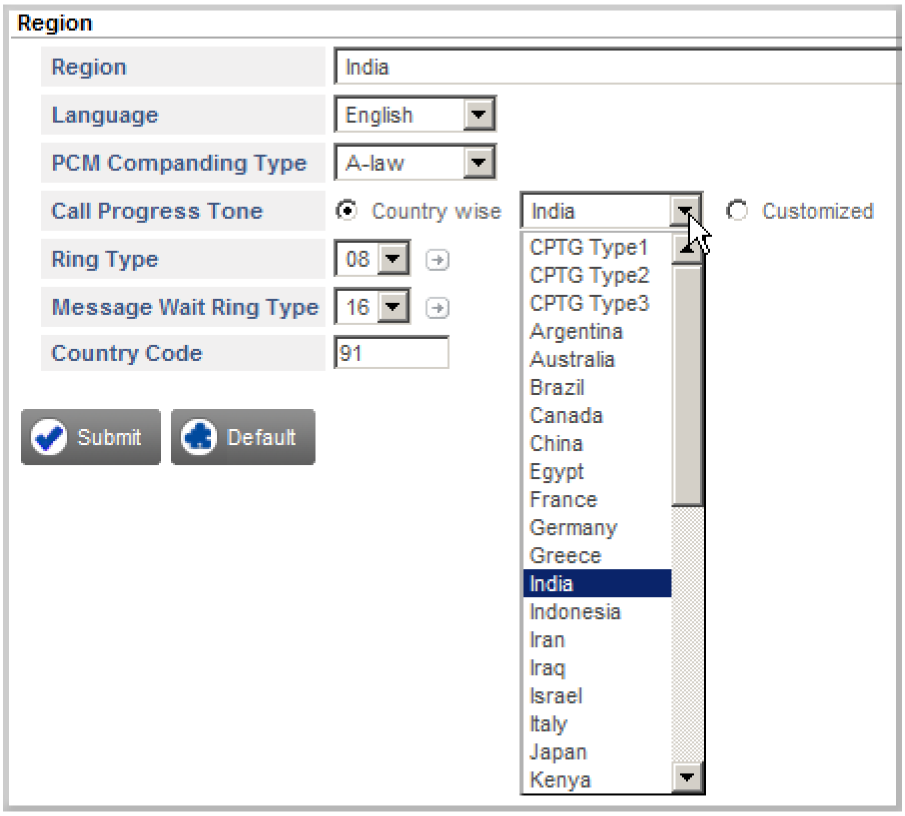
PCM Companding Type

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

Call Progress Tones

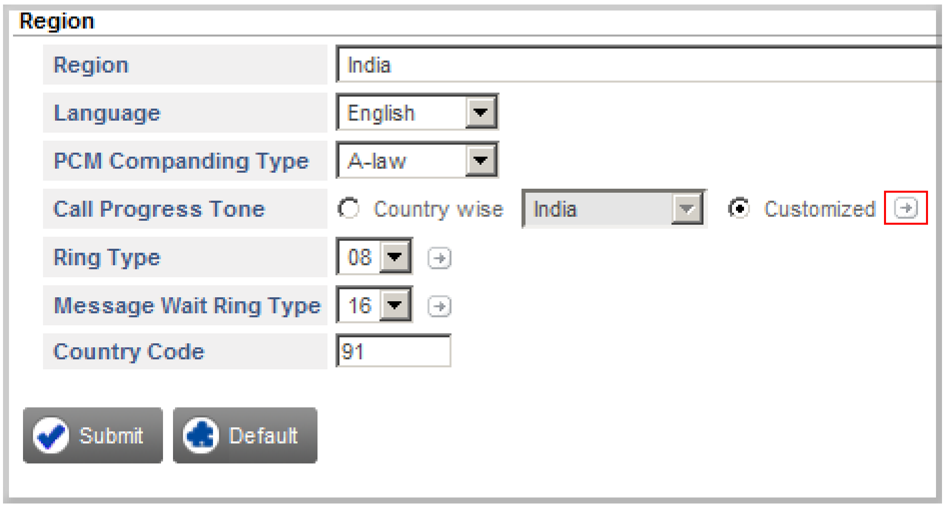
- Select the **Call Progress Tone**. SETU VFX supports country specific Call Progress Tone Generation (CPTG) to simulate the same tones of the local PSTN to which it is connected. The Call Progress Tones supported by SETU VFX for different countries is presented in the Appendix.

- To match the call progress tone of the country where SETU VFX is installed, select the **Countrywise** option and select the country from the list box. Default: India.



- 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** .



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

Call Progress Tone Cadence Table

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 in this field. The range of frequency1 is 300-1400 Hz for all tones.

- **Frequency2 (Hz):** Configure frequency2 in this field. The range of frequency2 is 20-1400 Hz for all tones.

- **Operator:** Operator parameter has three options:

  1) No: If *No* is programmed, Frequency2 will not be applicable.

  2) * (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.

- *The Call Progress Tone will not be set to default when the system is set to default.*

- Click **Submit**.

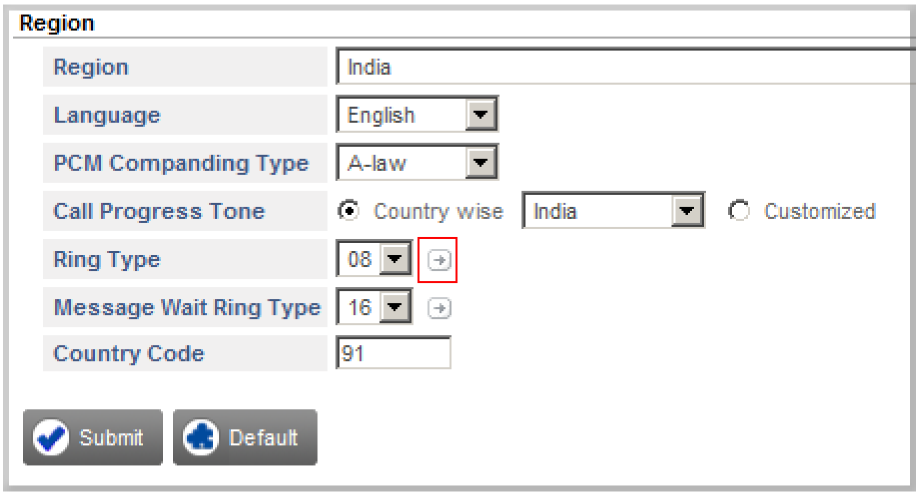
- Close the window to return to the **Region** page.

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 VFX will restart and your changes will be saved.

Ring Type[7]

- If required, you may change the **Ring Type** set automatically by SETU VFX according to the Region you select. Default: 08 (India).

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



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

7. *Applicable only for VoIP-FXS Gateway.*

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.

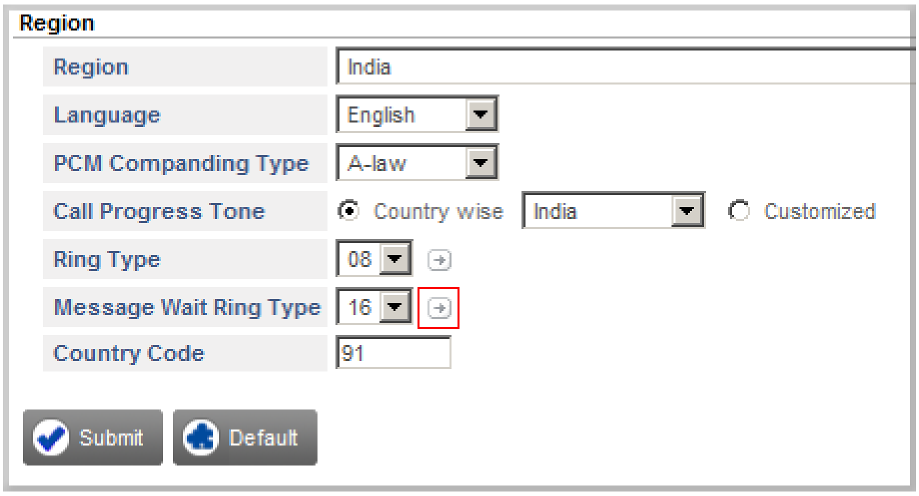
- Note the Ring Type number that you want to assign.
- Close the window to return to the **Region** page.
- In the **Ring Type** list, select the Ring Type number of your choice.

Message Wait Ring Type[8]

- 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** .



8. *Applicable only for VoIP-FXS Gateway.*

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

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

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

Country Code

- If required, you may change the **Country Code**, set automatically by SETU VFX 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

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

When SETU VFX is installed in a Public IP Network,

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

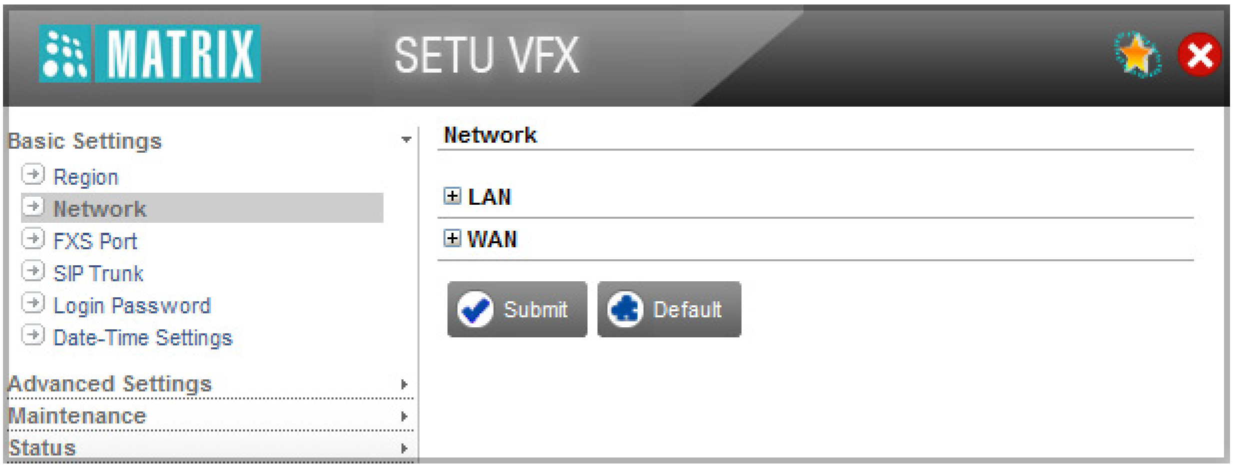
When SETU VFX is installed in a Private Network, behind a NAT Router,

- the WAN Port of SETU VFX 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.

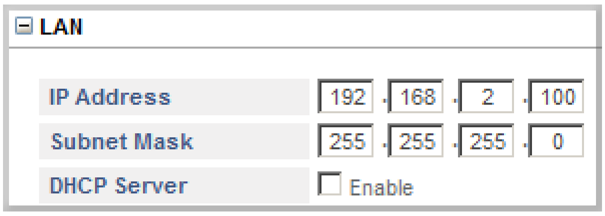
To configure Network Parameters,

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



LAN

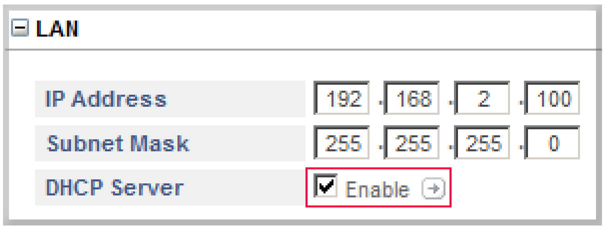
- Click **LAN** to expand.



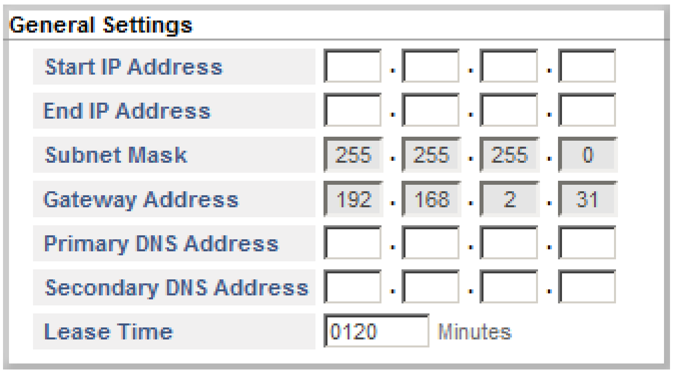
- In **IP Address**, the current IP Address of the LAN Port is displayed. Default: 192.168.2.100
- In **Subnet Mask**, the current Subnet Mask of the LAN Port is displayed. Default: 255.255.255.0

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

- Select the **DHCP Server Enable** check box, if you want SETU VFX to dynamically assign IP Addresses to the LAN devices on request. Default: Disabled.



- Click **Settings** to configure the DHCP Server parameters.

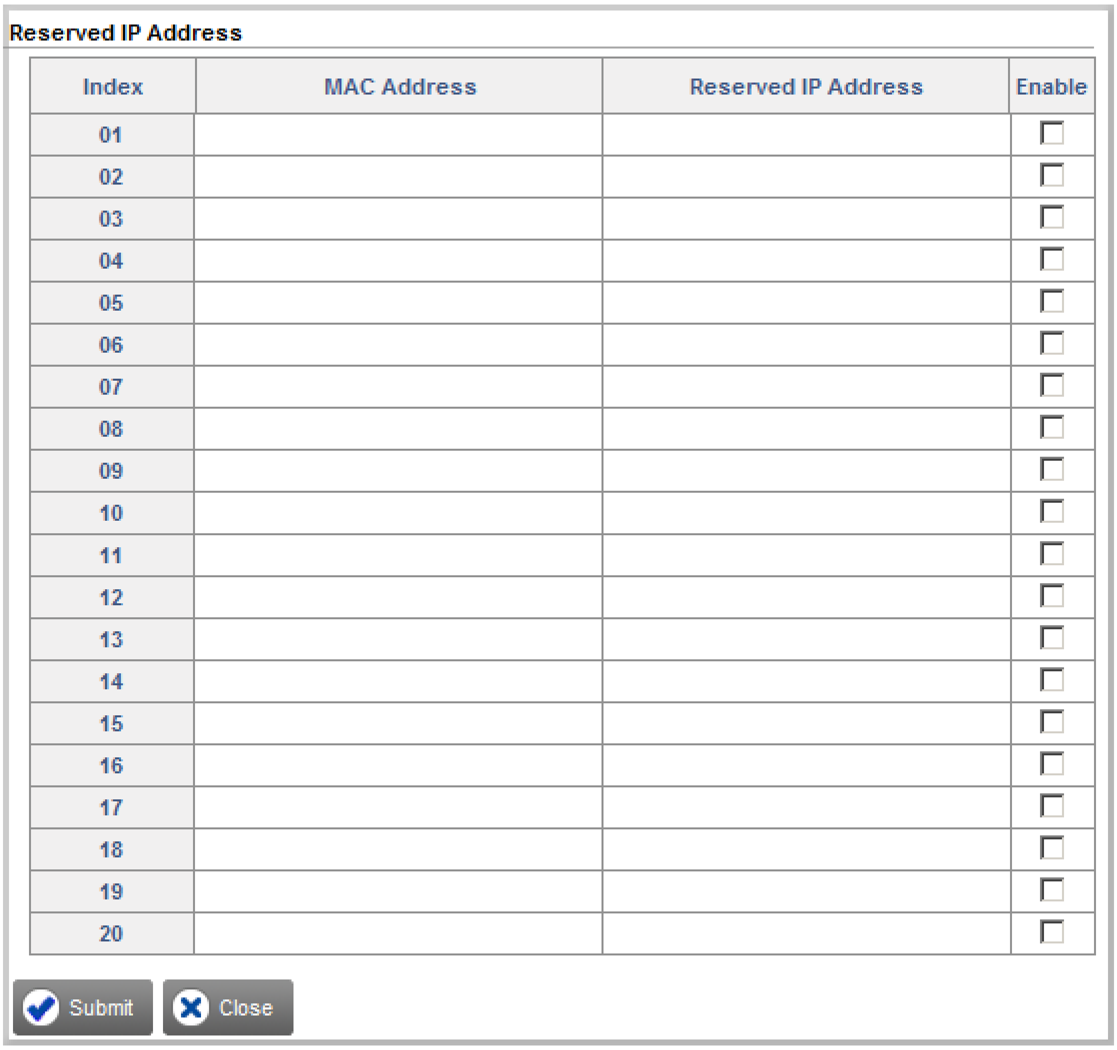


- In **General Settings**, configure the following parameters.
  - **Start IP Address:** Enter the Start IP Address. It is the starting IP Address to be assigned to the DHCP client (LAN device). SETU VFX will start assigning IP Addresses to the DHCP clients from this IP Address onwards. Default: Blank.
  - **End IP Address:** Enter the End IP Address. It is the last IP Address to be assigned to the DHCP client (LAN device). Default: Blank.
  - **Subnet Mask:** This field displays the Subnet Mask assigned to the DHCP client (LAN device).
  - **Gateway Address:** This field displays the Gateway Address assigned to the DHCP client (LAN device).
  - **Primary DNS Address:** Enter the Primary DNS Address to be assigned to the DHCP client.
  - **Secondary DNS Address:** Enter the Secondary DNS Address to be assigned to the DHCP client. Secondary DNS Address is considered, when the request to the Primary DNS Address fails.
  - **Lease Time:** Enter the Lease time (Minutes) for which IP Address is allocated to the DHCP client. The Lease Time ranges from 1 to 2880 minutes. Default: 120 minutes.

    DHCP client shall renew its lease within the Lease Timer. If the client does not renew its lease before the expiry of the configured Lease Time, SETU VFX will free the IP Address and this IP Address may be assigned to another DHCP client in LAN.

If you want the DHCP Server to assign fixed IP Address to certain DHCP clients, you must configure the Reserved IP Address table.

In the IP Address Reservation table, you can configure upto 20 entries. Each entry is stored against an Index number. Configure the following parameters for each entry.



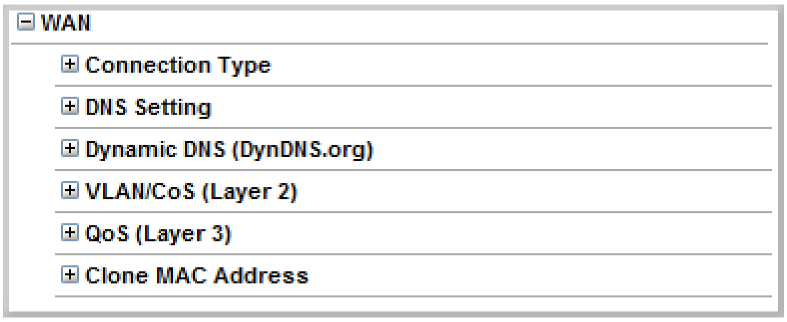
- **MAC Address**: Enter the MAC address of the LAN device to which you want to assign a fixed IP Address. Default: Blank.
- **Reserved IP Address**: Configure the fixed IP Address that you want to assign to the LAN device. This IP Address will be reserved for the MAC Address you configured. Default: Blank.
- **Enable**: Select this check box for each entry you configure. Default: Disabled.

You can view the status of the LAN Devices to which the IP Addresses have been leased by SETU VFX. See "DHCP Clients" in the *Status*.

- *When the system is set to default, all the DHCP Server parameters will be restored to default values.*
- *When your SETU VFX is installed in a Private Network, make sure the LAN Port and the WAN Port are connected in different subnets.*

WAN

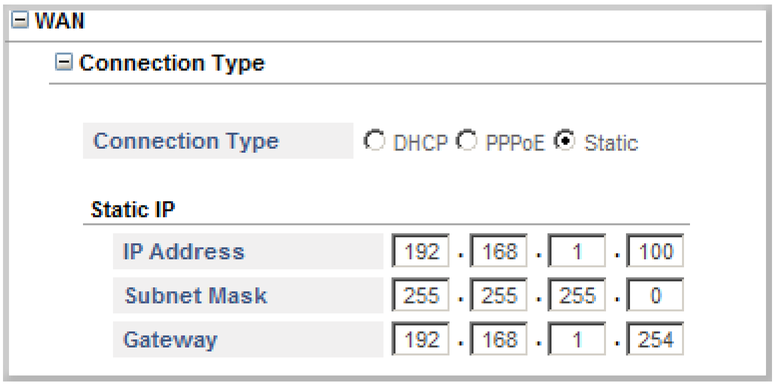
- Click **WAN** to expand.



Configure the following parameters.

Connection Type

- Click **Connection Type**.



- Select 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.
    - **IP Address:** Enter the IP Address you obtained from your Network Administrator for the WAN Port of SETU VFX in this field. Make sure that the IP Address does not conflict with that of any other device on the LAN. Default: 192.168.1.100
    - **Subnet Mask:** Enter the Subnet Mask you obtained from your Network Administrator for the WAN Port in this field. Default: 255.255.255.0
    - **Default Gateway:** Enter the IP Address of the Router's LAN Interface as the Default Gateway IP Address.
  - **DHCP:** Whenever SETU VFX 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 must configure the Domain Name Server (DNS) Address, 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 VFX. You need to configure the following parameters provided by your Internet Service Provider:

    - **PPPoE User ID:** Enter the User Name provided by the Internet Service Provider. The User ID may be a maximum of 64 characters.

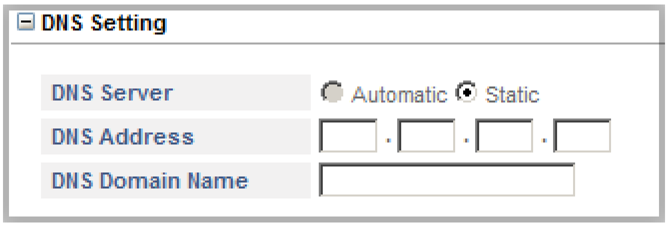
    - **PPPoE Password:** Enter the User Password provided by the Internet Service Provider. The password may have a maximum of 64 characters.

    - **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 Server

Configure the Domain Name Server (DNS) settings as provided by your Internet Service Provider. You may consult your LAN 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** and enter the DNS Server Address in the **DNS Address** field. 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)

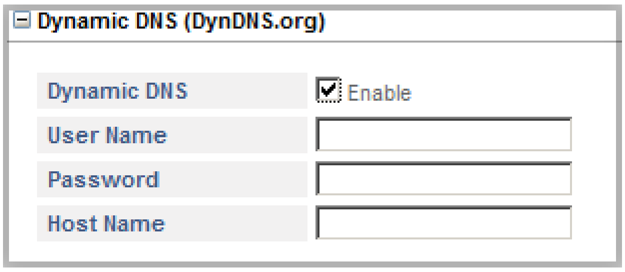
Dynamic DNS (DDNS) is a service that maps internet domain names to IP addresses. DDNS Service Provider provides the host name/domain name to the internet devices and also embed DDNS client in the internet device. By doing so, whenever a new IP Address is assigned to internet host, the DDNS client running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

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

SETU VFX supports Dynamic DNS Server client of the Service Provider Dynamic DNS.org. If you want to use the DNS Service of DynDNS.org, configure these parameters:

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

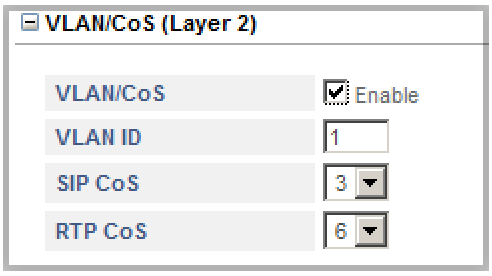


- Select the **Dynamic DNS Enable** check box.
- Enter the **User Name** you created on DynDNS.org. The name can be of maximum 40 characters.
- Enter the **Password** you created for the User Name on DynDNS.org. The password can be of maximum 24 characters.
- Enter the **Host Name** you created on the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

VLAN/CoS

If SETU VFX is connected in VLAN network, configure the **VLAN/CoS**. This parameter enables the SETU VFX 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[9].

- 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.

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

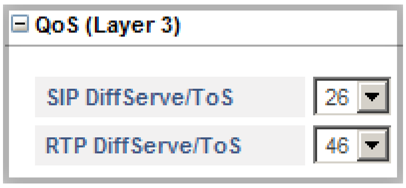
- Enter the **VLAN ID** that you have assigned to the VLAN in which the SETU VFX is connected. The valid range for this is 0-4094. Default: 1.

- 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)**.

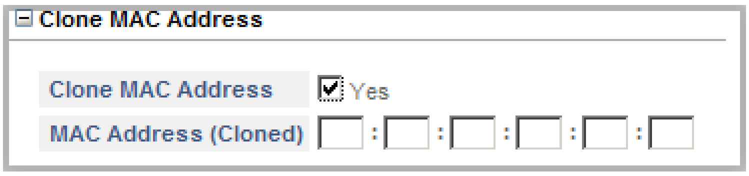


- Select the **SIP DiffServe/ ToS** as per your requirement, SETU VFX will send all SIP messages using SIP QoS setting. Valid range is from 00-63, Default: 26

- Select the **RTP DiffServe/ ToS** as per your requirement, SETU VFX will send all the RTP packets with RTP QoS setting. 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 the **MAC Address (Cloned)** field, enter the desired MAC address you want to clone in hexadecimal format, for example, 00:50:c2:55:b0:10.

After you have set all the 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 in two ways:

1. Using Reset Button.
2. By changing the Jumper position on the PCB.

Using Reset Button

To restore the Default LAN IP Address using Reset button,

- 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 VFX will restart.*
- *Along with the WAN IP Address, a few other parameters will also be set to default. See “Restoring Default Settings using the Reset button” for details.*

By changing the Jumper position on the PCB.

To restore the Default LAN IP Address by changing the Jumper position,

- Be sure to take all electrostatic discharge preventive measures.
- Switch off the power supply.
- Remove the top cover of the enclosure.
- Locate and change the position of the Jumper **J3** from ***BC*** to ***AB***.
- Replace the cover of the enclosure.
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system and remove the enclosure cover.
- Change the Jumper position from ***AB*** to the original position ***BC***.
- Replace the enclosure cover.
- Switch ON the system.

The LAN IP Address will be restored to the default value: **192.168.2.100**

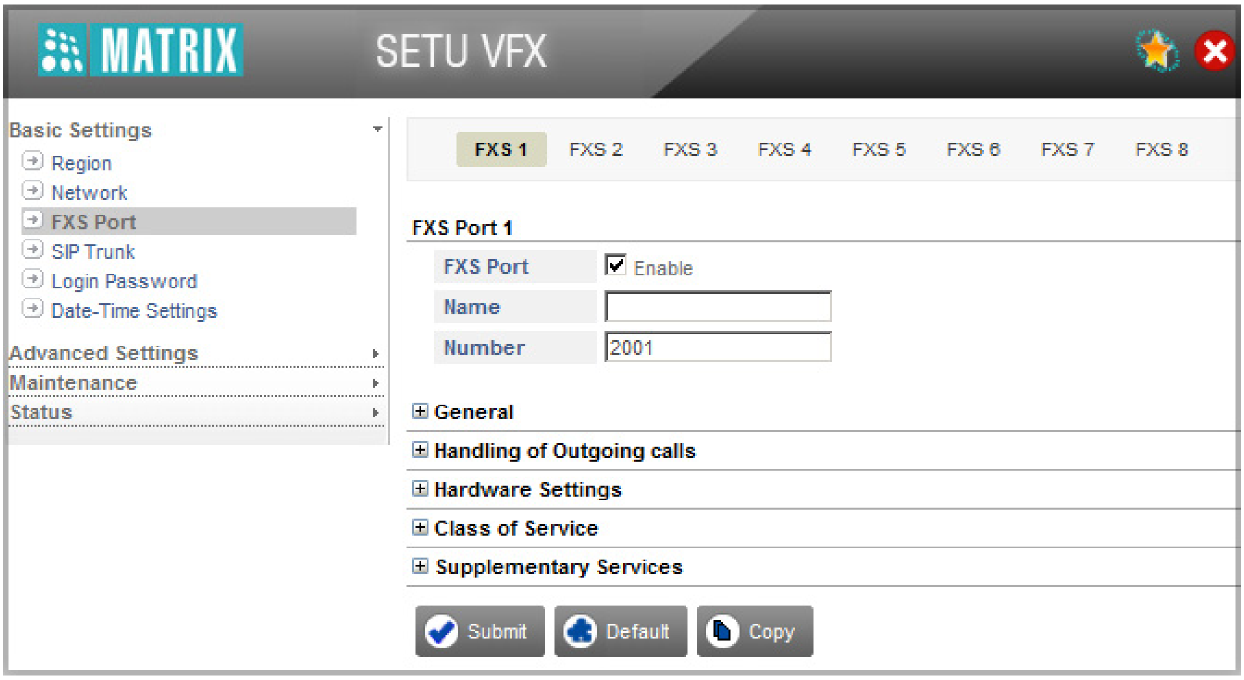
*When you restore the default IP Address (**192.168.2.100**) by changing the Jumper position, a few other parameters will also be set to default. See “Restoring Default Settings by changing the Jumper Position” for details.*

FXS Ports

SETU VFX808 has eight 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.
- Click the FXS number tab, **FXS1** to **FXS8**, you want to configure. The FXS Port page opens.



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

  Clear this 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 the phone supports name display.

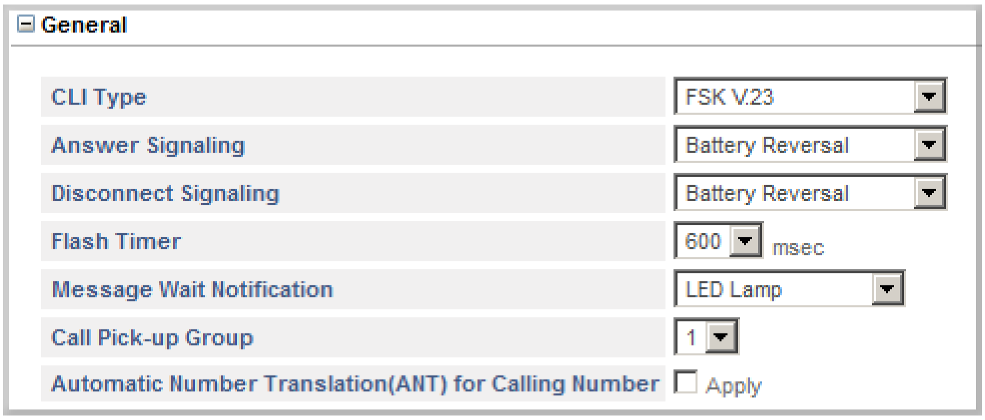
  The name you assign may consist of a maximum of 12 characters. Default: Blank.

- 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 the phone supports number 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**.



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

  SETU VFX supports three signaling protocols for CLI on the FXS Port:
  - DTMF
  - FSK BellCore
  - FSK V.23

  Default: FSK V.23

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

  Answer Signaling is a signal generated on the FXS Port to indicate that the called party has answered the call (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.

  - 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.

    - **Open Loop Disconnect Timer:** Enter the desired value for Open Loop Disconnect Timer. Valid range: 001 to 999 milliseconds. Default: 500 milliseconds.

Default: Battery Reversal.

- Configure the **Flash Timer**, as per your requirement. The Flash timer signifies the time period for which the loop current breaks. SETU VFX uses this event to activate various features such as Call Hold, Call Transfer. Default: 600 milliseconds.

- 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[10], you may select the desired **Message Wait Notification** type from the following options.

  - **Stuttered Dial Tone:** Select this option, if you want new message indication in the form of a stuttered dial tone, when the user picks up the phone connected to the FXS Port.

    You can change the frequency and cadence of the Stuttered Dial tone, if required. For instructions, see “Call Progress Tones”, under “Region” in the *Basic Settings* chapter.

  - **LED Lamp:** Select this option, 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.

  - **Ring:** Select this option, 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 (see “Message Wait Ring Type”), you can 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, in the *Advanced Settings* chapter, under “System Parameters”, see “Message Wait”.

  Default: LED Lamp

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

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

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

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

  You can assign the Port to any group from 1 to 4. Default: 1

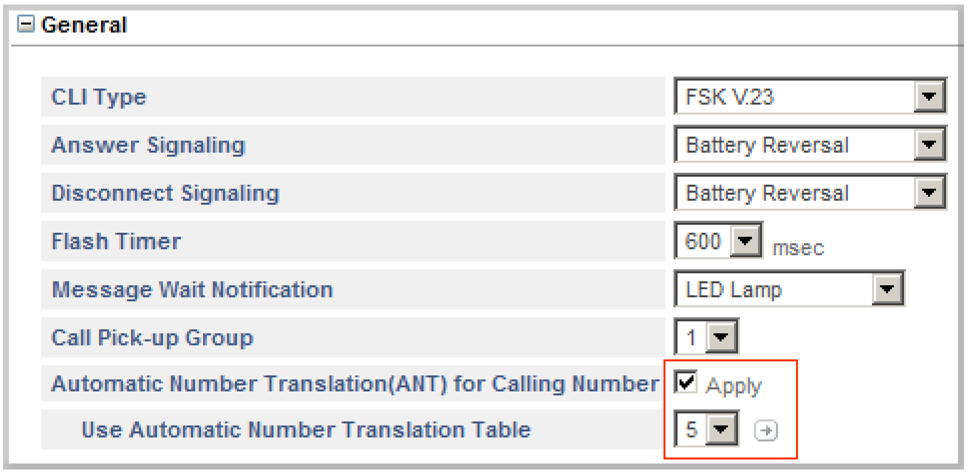
  Make sure that the Call Pick-up feature is included in the *Class of Service* of all FXS Ports in the group. See “Class of Service” later in this topic.

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

- You can apply Automatic Number Translation (ANT) logic on the outgoing calls made from the FXS Port.

10. *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.*

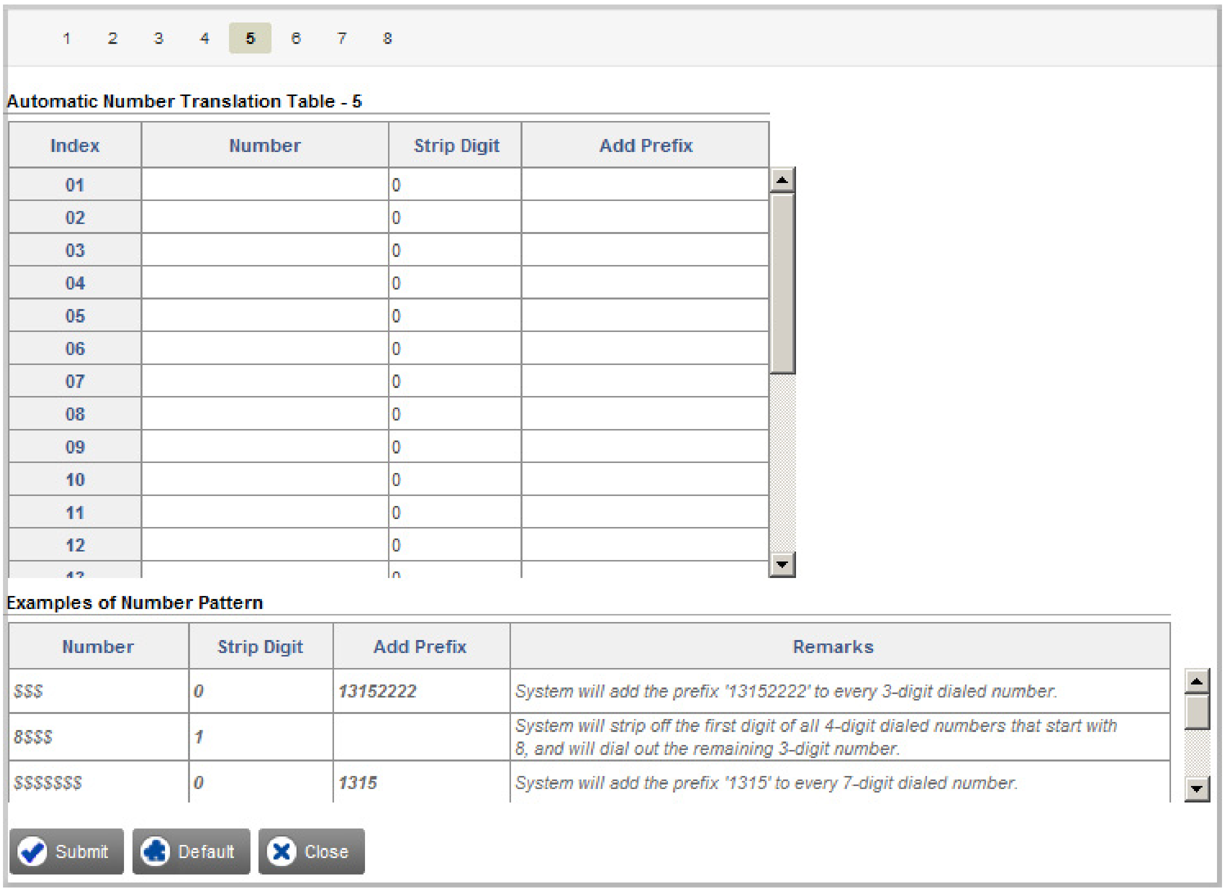
- To apply ANT logic on the Calling Numbers, click the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.



- 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 page will open in a new window.

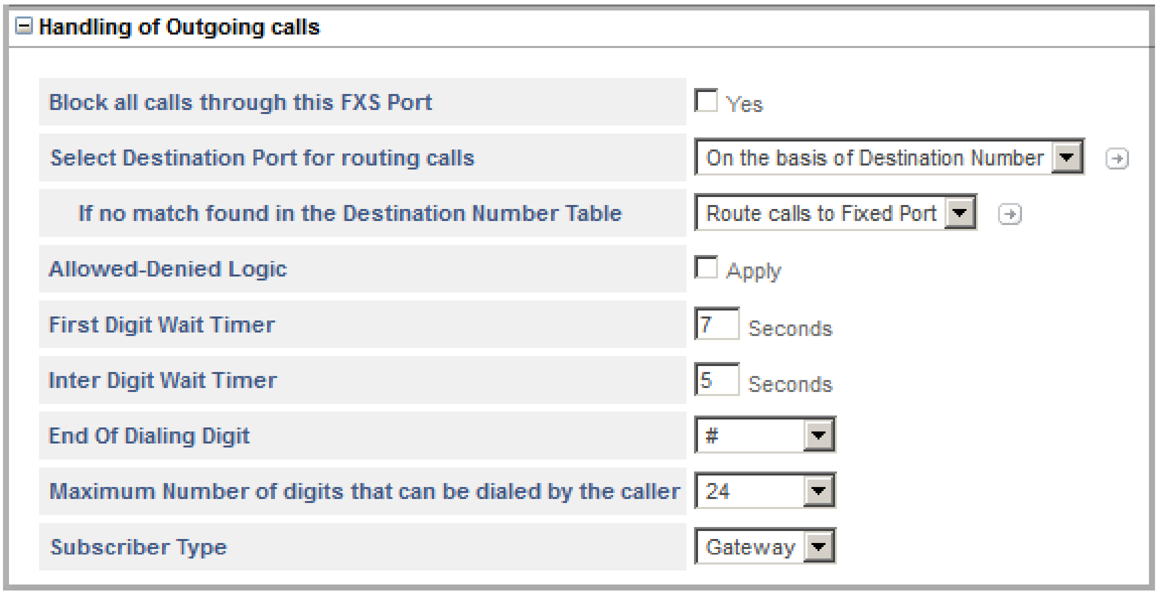


  - You may configure the default Automatic Number Translation Table 5 or any other Table. See “Automatic Number Translation (ANT)” to configure the ANT Table.

- 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.

Handling of Outgoing Calls

- Click **Handling of Outgoing Calls**.



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

Destination Port Determination

- In the **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.

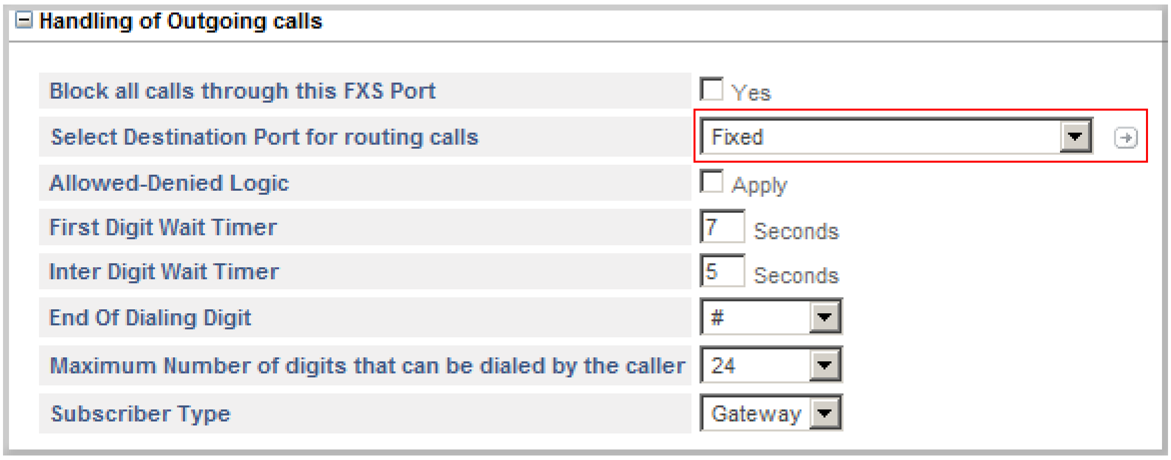
If the destination number to be dialed out is an IP Address, SETU VFX will not check the Destination Port Determination Method. Instead, it will route the call using the SIP Trunk / Group programmed for IP Dialing. (To know more, see the feature description "IP Dialing").

Fixed

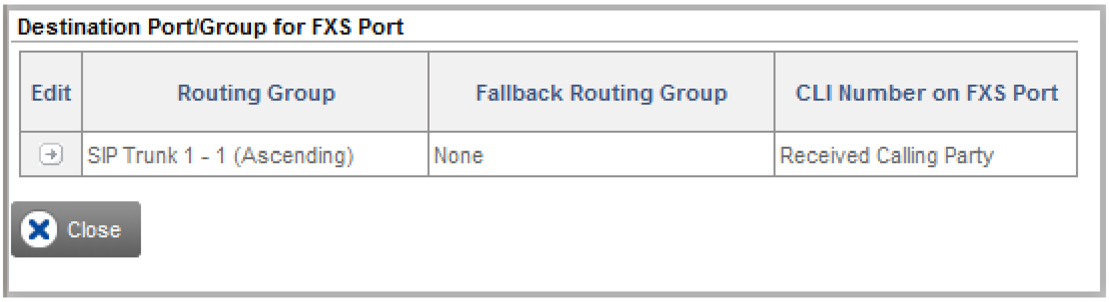
In this method, outgoing calls made from the FXS Port are routed to a Fixed Destination Port, irrespective of the number dialed from the FXS Port.

To apply this method, do the following:

- In the **Select Destination Port for routing calls** box, click **Fixed**.
- Click **Settings** .

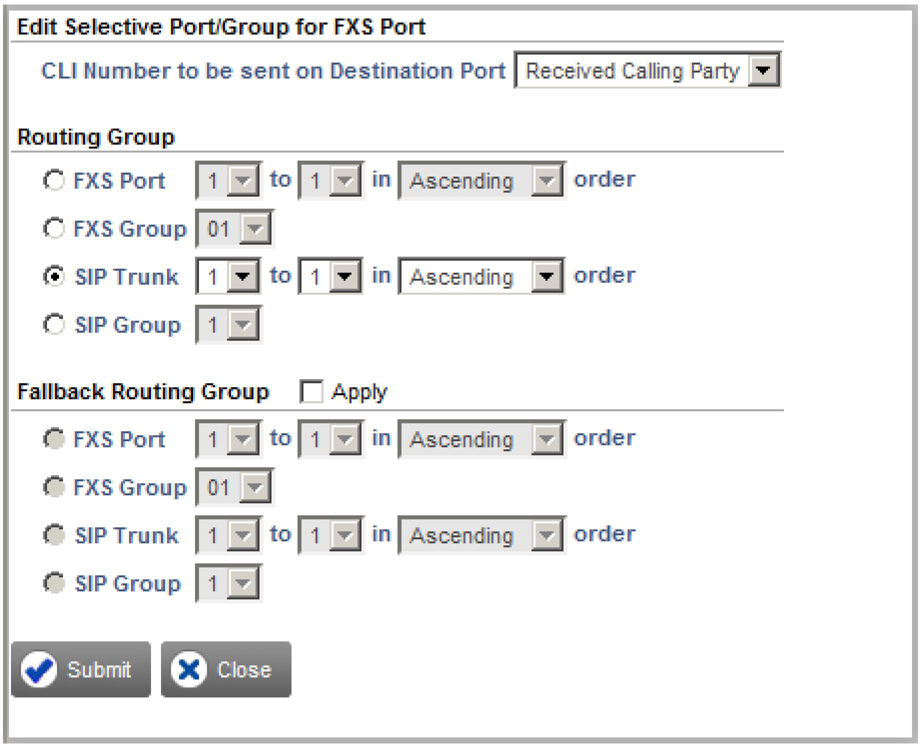


The **Destination Port/Group for FXS Port** window opens.



- To change the default Routing Group and to create the Fallback Routing Group, under **Edit**, click **Settings** .

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



- Select the **CLI Number to be sent on Destination Port** from these options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

CLI Number to be sent on Destination Port *is applicable only when* ***FXS Port*** *or* ***FXS Group*** *is selected as the Destination Port (Routing Group).*

- 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.

      To start hunting from the first to the last member FXS Port, select **Ascending**.

      To start hunting from the last to the first member FXS Port, select **Descending**.

    Default: Ascending.
  - 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 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 *sequential* and *not-sequential* 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 and SIP Trunks.

- 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 FXS Port page.

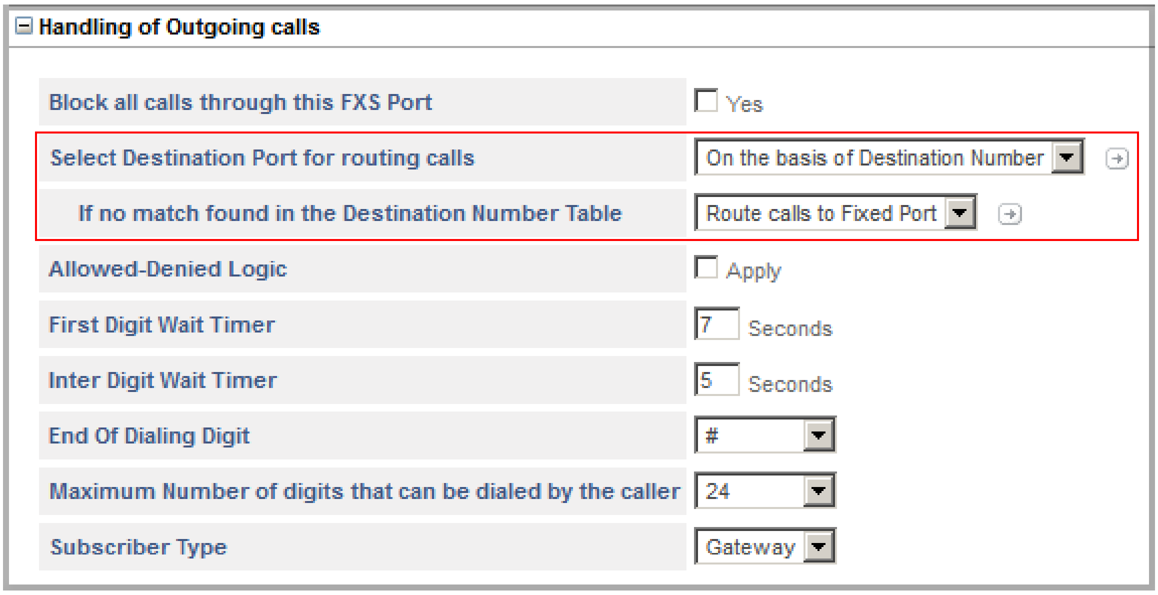
On the basis of Destination Number

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.

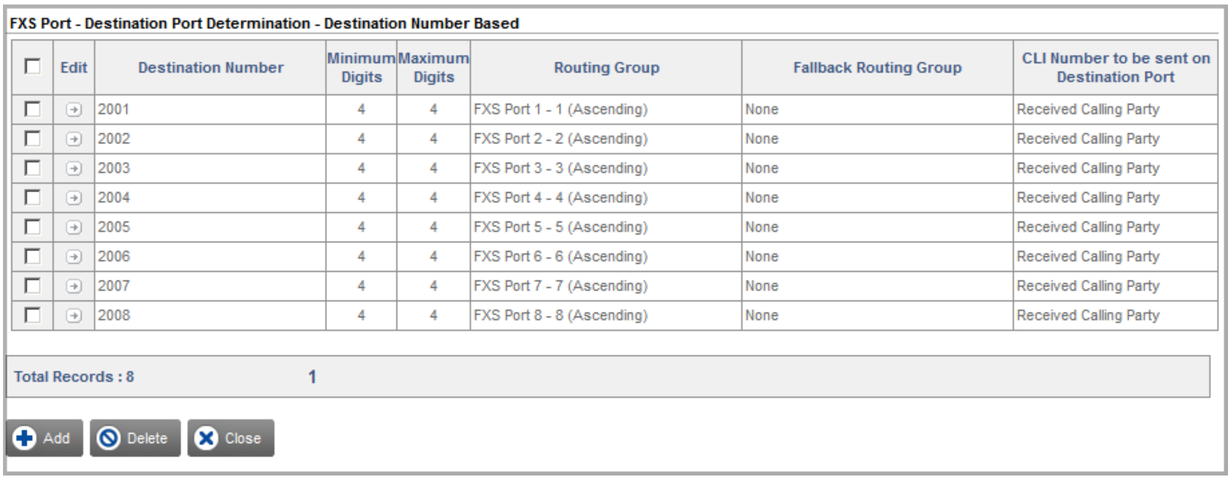
To apply this method, do the following:

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

- Click **Settings** .



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



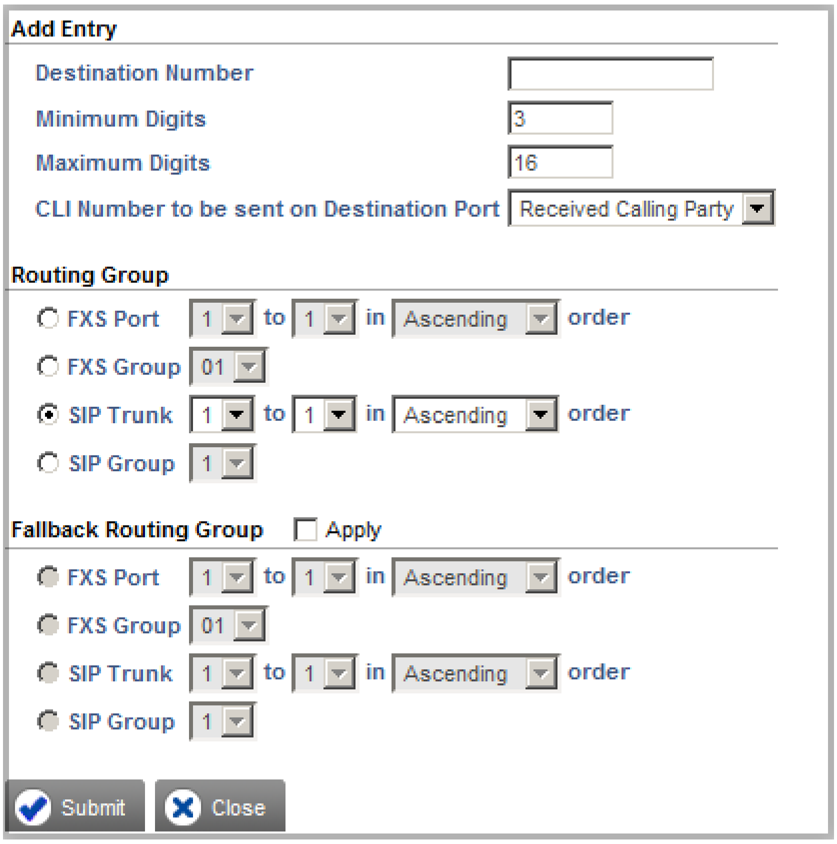
FXS Port - Destination Port Determination - Destination Number Based

☐	Edit	Destination Number	Minimum Digits	Maximum Digits	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
☐		2001	4	4	FXS Port 1 - 1 (Ascending)	None	Received Calling Party
☐		2002	4	4	FXS Port 2 - 2 (Ascending)	None	Received Calling Party
☐		2003	4	4	FXS Port 3 - 3 (Ascending)	None	Received Calling Party
☐		2004	4	4	FXS Port 4 - 4 (Ascending)	None	Received Calling Party
☐		2005	4	4	FXS Port 5 - 5 (Ascending)	None	Received Calling Party
☐		2006	4	4	FXS Port 6 - 6 (Ascending)	None	Received Calling Party
☐		2007	4	4	FXS Port 7 - 7 (Ascending)	None	Received Calling Party
☐		2008	4	4	FXS Port 8 - 8 (Ascending)	None	Received Calling Party

Total Records : 8 1

Add Delete Close

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



- In the **Destination Number** field, enter the number (24 characters, maximum) that you expect callers to dial. Valid digits: 0–9, *, #, (dot). Default: Blank.

- In the **Minimum Digits** field, enter the minimum number of digits of the destination number that the caller must dial for the system to route the call. Valid range: 01–24. Default: 3.

  If the dialed number string is less than the configured minimum length, the call will be rejected.

- In the **Maximum Digits** field, enter the maximum number of digits to be dialed by the caller for the system to consider it as end-of-dialing for routing the call. Valid range: 01–24. Default:16.

  If the number string dialed by the caller exceeds the maximum length configured, the system will strip off the extra digits, and route the call.

- Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

***CLI Number to be sent on Destination Port** is applicable only when **FXS Port** or **FXS Group** is selected as the Destination Port (Routing Group).*

- 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.

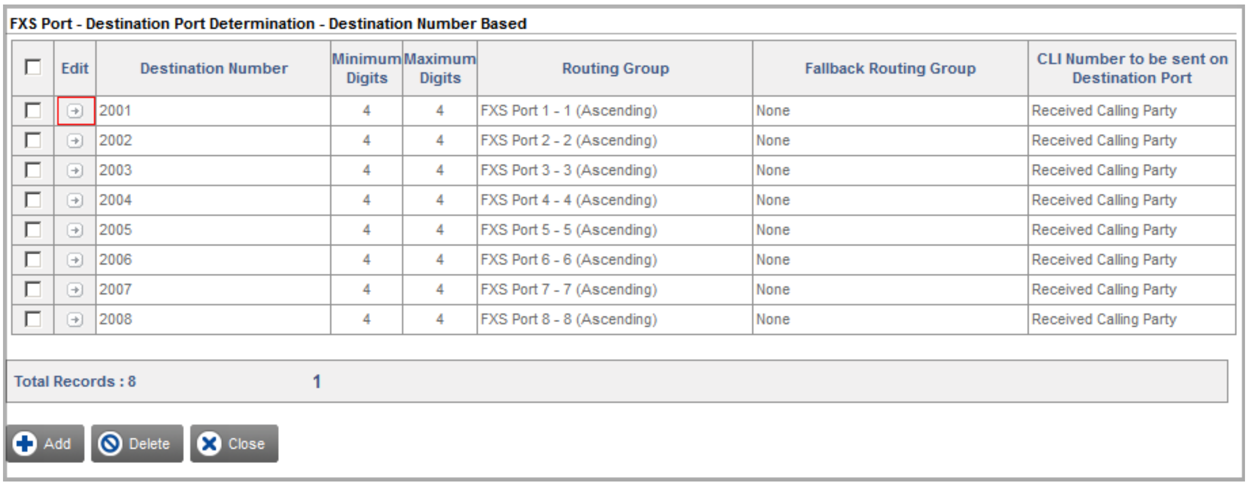
    To start hunting from the first to the last member FXS Port, select **Ascending**.

    To start hunting from the last to the first member FXS Port, select **Descending**.

    Default: Ascending.
  - To create a group of *not-sequential* **FXS Ports** as members,
    - Select an **FXS Group**.
    - Select **FXS Group** number. Default:1.
    - Click **Settings** . The **FXS Groups** window opens. Create the FXS Group. For detailed instructions, see “Group”.
  - Similarly, you can create group of *sequential* and *not-sequential* SIP Trunks as members.
- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier to create *sequential* and *not-sequential* groups of FXS Ports and SIP Trunks.
  - Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **FXS Port - Destination Port Determination - Destination Number Based** window.

To edit the Routing Group and/or the Fallback Routing Group, do the following:

- Under **Edit**, click **Settings** .



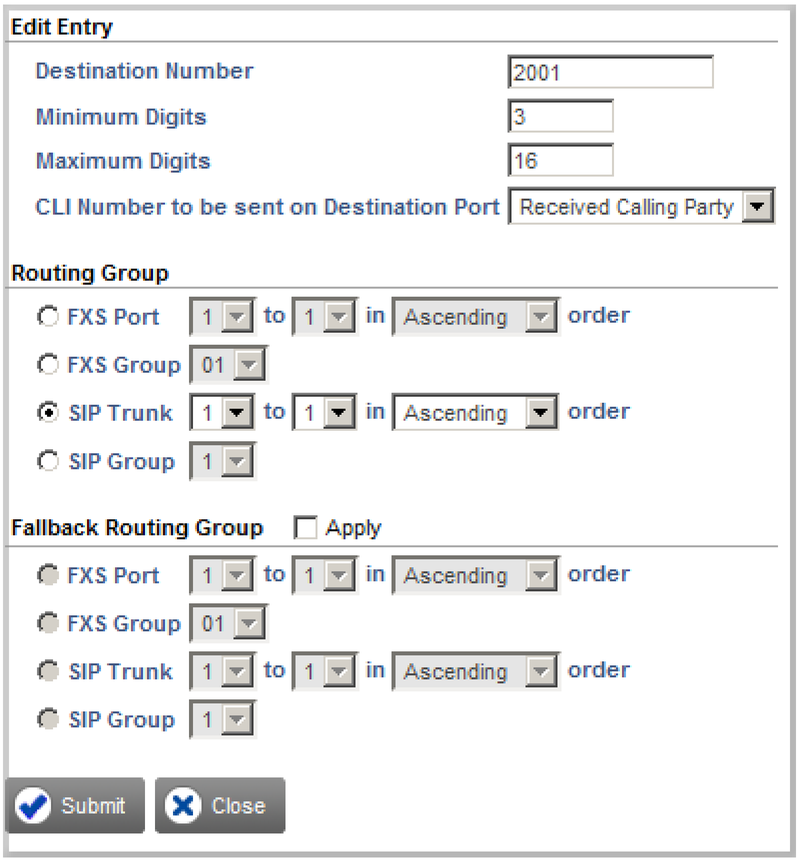
FXS Port - Destination Port Determination - Destination Number Based

☐	Edit	Destination Number	Minimum Digits	Maximum Digits	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
☐		2001	4	4	FXS Port 1 - 1 (Ascending)	None	Received Calling Party
☐		2002	4	4	FXS Port 2 - 2 (Ascending)	None	Received Calling Party
☐		2003	4	4	FXS Port 3 - 3 (Ascending)	None	Received Calling Party
☐		2004	4	4	FXS Port 4 - 4 (Ascending)	None	Received Calling Party
☐		2005	4	4	FXS Port 5 - 5 (Ascending)	None	Received Calling Party
☐		2006	4	4	FXS Port 6 - 6 (Ascending)	None	Received Calling Party
☐		2007	4	4	FXS Port 7 - 7 (Ascending)	None	Received Calling Party
☐		2008	4	4	FXS Port 8 - 8 (Ascending)	None	Received Calling Party

Total Records : 8 1

Add Delete Close

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as described earlier.

- Click **Submit** and close the window.

- Follow the same steps as above to add another entry to this table.

- To delete an entry, select the check box of the entry and click the **Delete** button.

- Close the window, if you have finished adding/editing entries.

  You can also configure the **Destination Number Based** Table from *Advanced Settings*. 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 the **If no match found in the Destination Number Table** box, click the desired option for routing the call. You may select from the following options.
  - Route calls to Fixed Port
  - 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 (Toll-Control)

With the Allowed-Denied Numbers feature you can permit and restrict the dialing of particular numbers from the FXS Port.

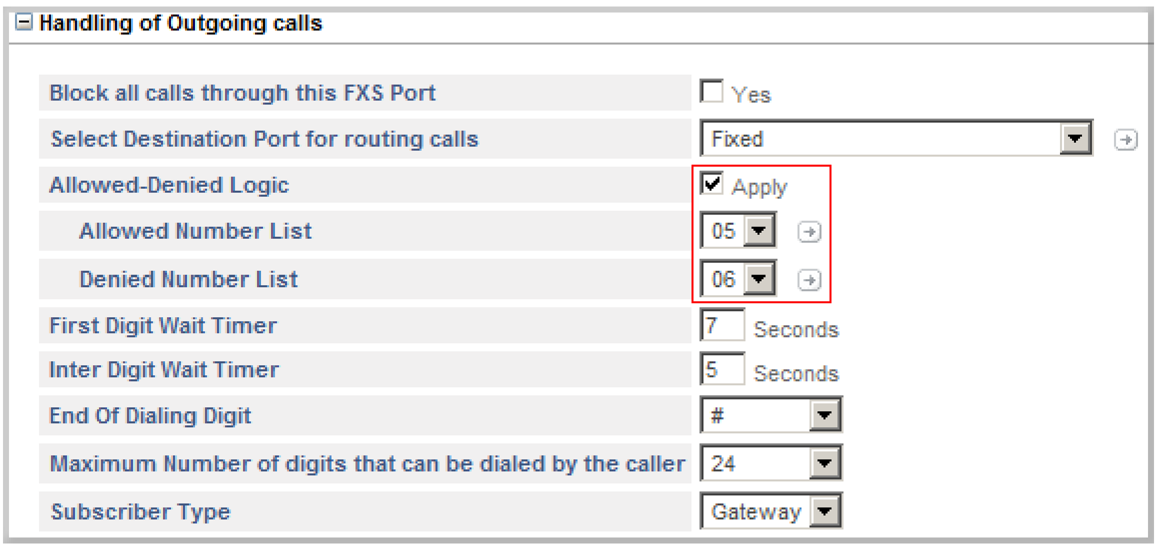
The Allowed Denied Number Logic makes use of two Number lists:

- **Allowed Numbers List:** This is the list of numbers that can be dialed out from the FXS Port. By default, List Number 5 is assigned to the FXS Port.

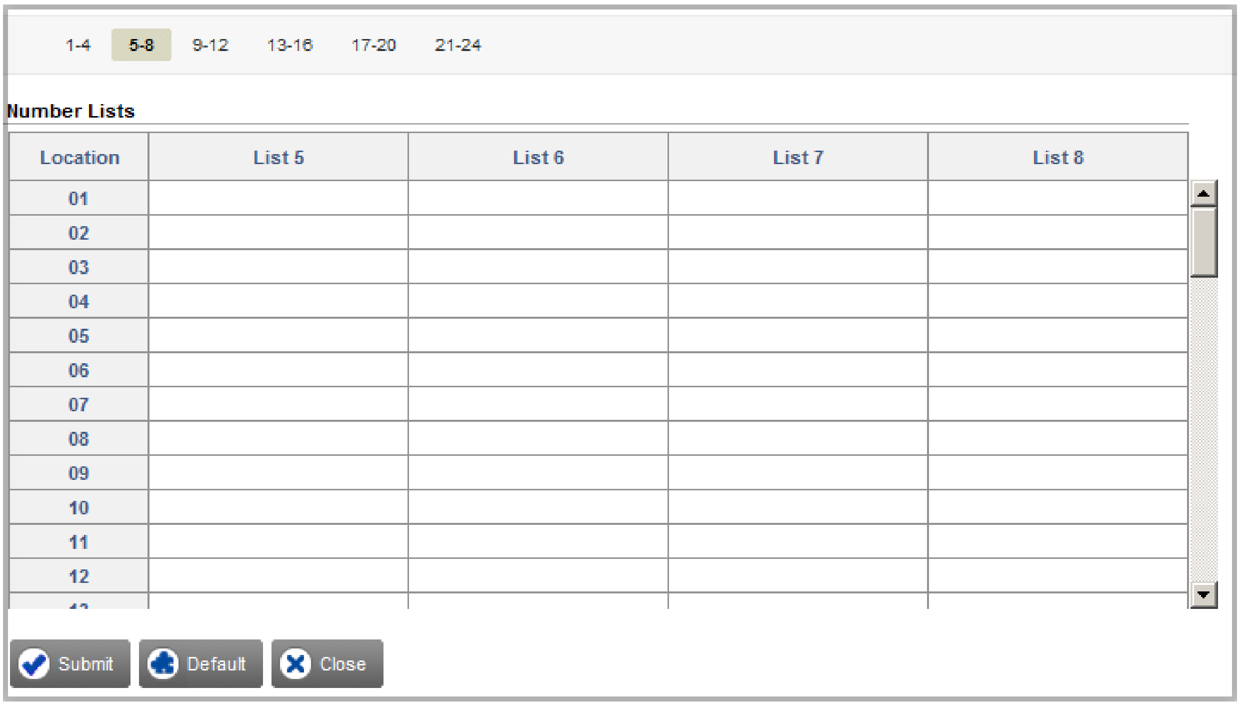
- **Denied Numbers List:** This list contains the numbers that are to be restricted from being dialed out from the FXS Port. By default, List Number 6 is assigned to the FXS Port.

To apply Allowed - Denied Logic on the FXS Port, do the following:

- Click the **Allowed - Denied Logic** check box. Default: Disabled.



- To configure the **Allowed Number List**, click **Settings** .

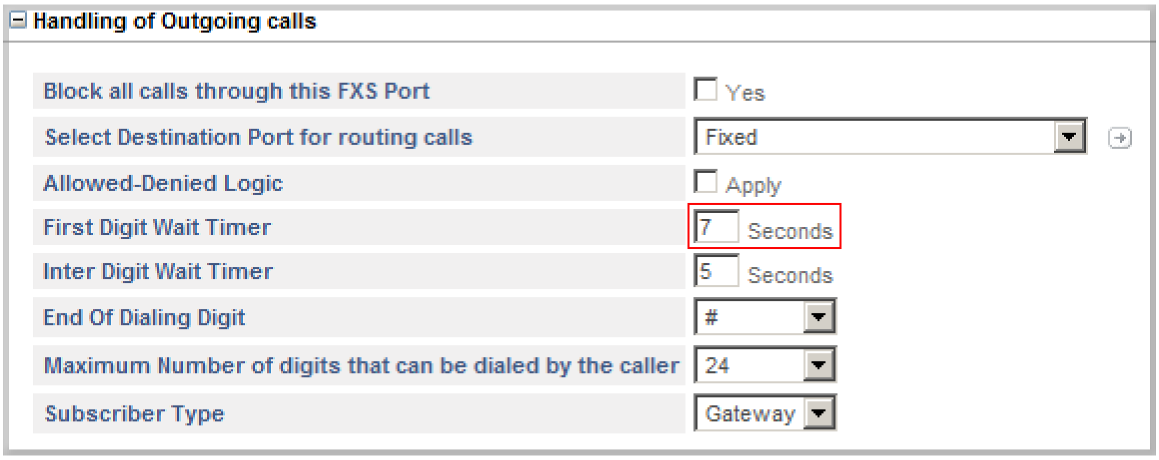


  - The Number List page opens in a new window.
  - By default, Number List 5 is assigned as Allowed Number List.
  - Enter the numbers you want the system to allow to be dialed in this list.
  - Click **Submit** to save the entries and close the window.

- To configure the **Denied Number List**, click **Settings** .
  - The Number List page opens in a new window.
  - By default, Number List 6 is assigned as Denied Number List.
  - Enter the numbers you want the system to restrict from being dialed out in this list.
  - Click **Submit** to save the entries and close the window.

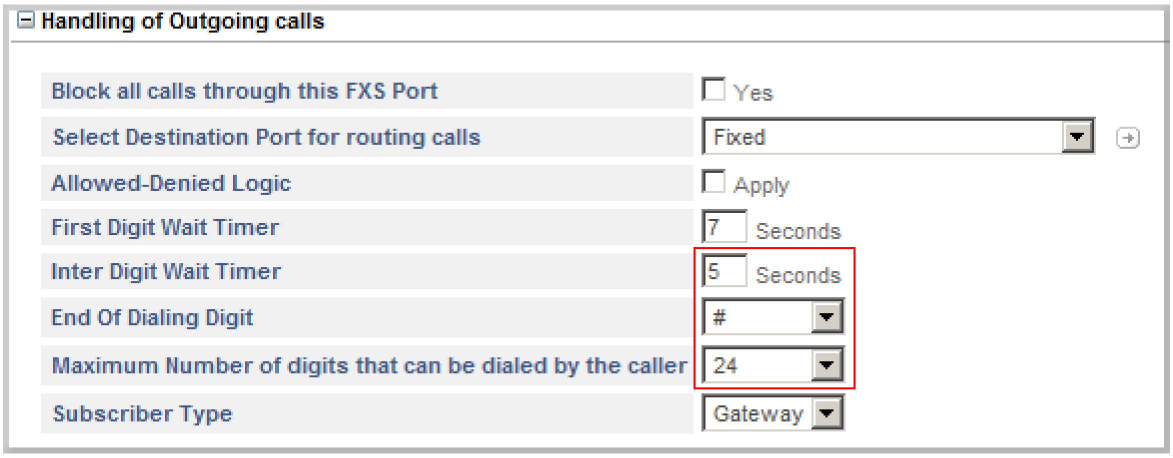
You may configure a different Number List as Allowed and Denied List. To know more about this feature, see "Allowed-Denied Logic" under "Number Lists".

First Digit Wait Timer



- If required, you may change the duration of the **First Digit Wait Timer (FDWT)**. This is the time in seconds for the which the system will wait for the user to dial the destination number. Valid range: 01–99 seconds. Default: 7 seconds.

End-of-Dialing



- In the **Inter Digit Wait Timer** field, define the number of seconds the system should 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: 01–99 seconds. Default: 5 seconds.

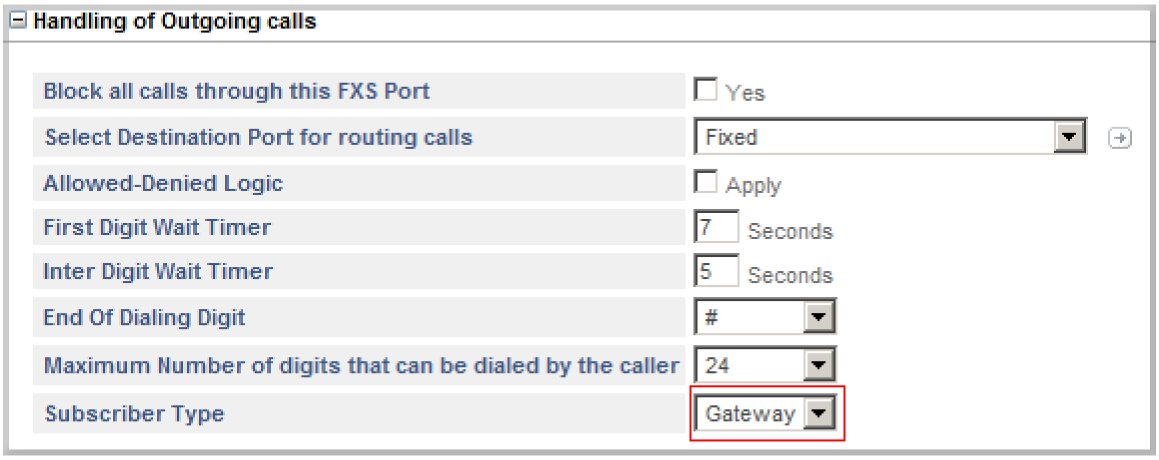
- In the **End of Dialing Digit** box, select **#** or ***** as the termination digit that the system should consider to detect end of dialing. Default: #

- In the **Maximum number of dialed digits that can be dialed by the caller** field, select the maximum number of digits to be dialed by the user for the system to consider it as end-of-dialing**.** Valid range: 01–24 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 VFX.



When SETU VFX is interfaced with a service provider server—ITSP, the Matrix ETERNITY IP-PBX, or any other PBX—that supports supplementary services that require dialing of Flash, like Call Hold, Call Transfer, Call Waiting, you must select the Subscriber Type for SETU VFX according to the extent of feature access you want on the FXS Port connected to the system.

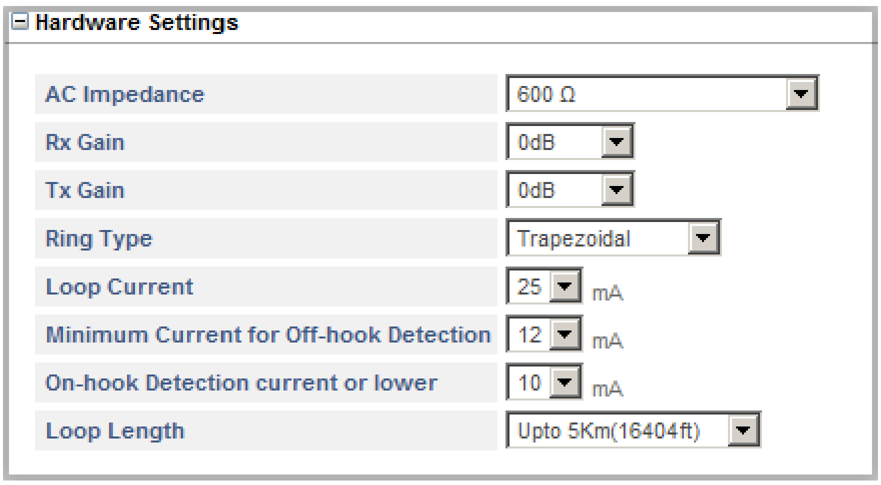
- Select **Network,** if you want to use the supplementary services supported by the PBX. When you set SETU VFX in the Network mode, you can access the service provider features by dialing Flash. You will not be able to access the local features of SETU VFX.

- Select **Gateway,** if you want to use primarily the supplementary features of SETU VFX. 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.**



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

  You may select from the following impedance options:
  - 600 Ω
  - 900 Ω
  - 350 Ω + (1000 Ω || 0.21 μF)
  - 220 Ω + (820 Ω || 120 nF)
  - 270 Ω + (750 Ω || 150 nF)

  Default = 600 Ω.

- You may adjust the **Rx Gain** (Receive) on FXS Port to increase or decrease the volume of the remote party's voice being transmitted to you. Select the required Rx Gain level. Default: 0 db.

- If required, you may adjust the **Tx Gain** (Transmit) on the FXS Port to increase or decrease the volume of your voice being transmitted to the remote party. Select the appropriate Tx Gain. Default: 0 db.

- Select the **Ring Type** to be generated on the FXS Port. SETU VFX supports the following Ring Types:
  - Low Sinusoidal
  - Low Trapezoidal
  - Sinusoidal
  - Trapezoidal

  Default: Trapezoidal.

- Select the **Loop Current** as per the Loop Length. SETU VFX supports the following options:
  - 25 mA
  - 30 mA
  - 35 mA
  - 40 mA

  Default: 25 mA.

- Select the **Minimum Current for Off-hook Detection (mA)** on the FXS Port. You can select from the following options:
  - 10mA
  - 12mA
  - 14mA
  - 16mA

  Default: 12 mA.

- Select the **On-hook Detection current or lower** on the FXS Port. You can select from the following options:
  - 10mA
  - 12mA
  - 14mA
  - 16mA

  Default: 10 mA.

- Select the **Loop Length** as **Upto 5 Km (16404 ft)** or **Above 5 Km (16404 ft)** depending on your installation scenario. The Loop Length is the distance between the Central Office and the telephone instrument connected to the FXS Port. Default: Upto 5 Km (16404 ft).

Class of Service

- Click **Class of Service**.

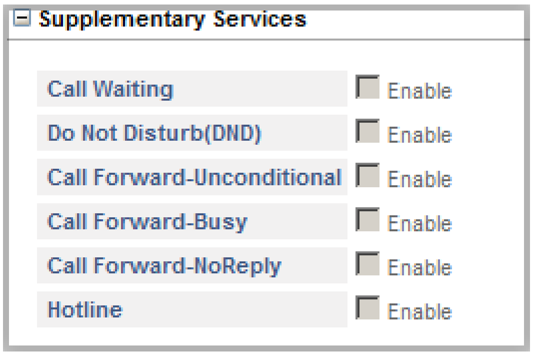
- Select the features of SETU VFX that you want to allow in **Class of Service**[11] (CoS) of the FXS Port.

  By default all the features are denied.

  To allow a feature to the FXS Port, select the respective check box.

  To deny a feature, clear the respective check box.

Supplementary Services



SETU VFX offers the following set of supplementary features, which you can set or cancel on this page, if the same have been enabled in the CoS:

- Call Waiting
- Do Not Disturb (DND)
- Call Forward - Unconditional
- Call Forward - Busy
- Call Forward - No Reply
- Hotline

- To set any of these features, select the respective **Enable** check box.

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

- To cancel the feature, clear the check box of the feature.

- By default all the features are disabled.

To use the Supplementary feature, make sure it is allowed to you in the Class of Service.

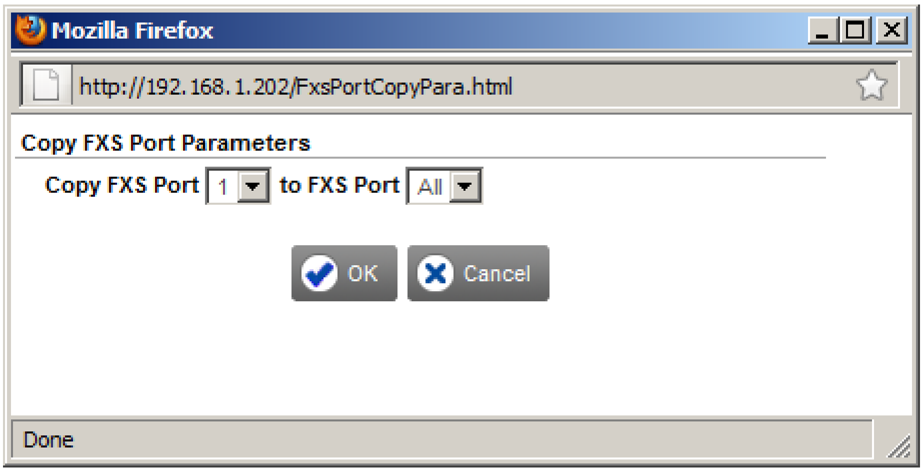
- If you have completed the configuration of FXS 1, click **Submit** to save settings.

- To configure the next FXS Port, click the FXS Port number tab and follow the same instructions as given earlier.

Copy Port Settings

- You can also copy the settings of an 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 the **Copy FXS Port** box, select the number of the port you want to copy settings *From*. In the **to FXS Port** box, select the number of the port you want to copy the settings *To*.

  - Click **OK** button and close the window.

- Once you have copied the settings, you can again edit the specific parameters of the FXS Port you copied the settings to.

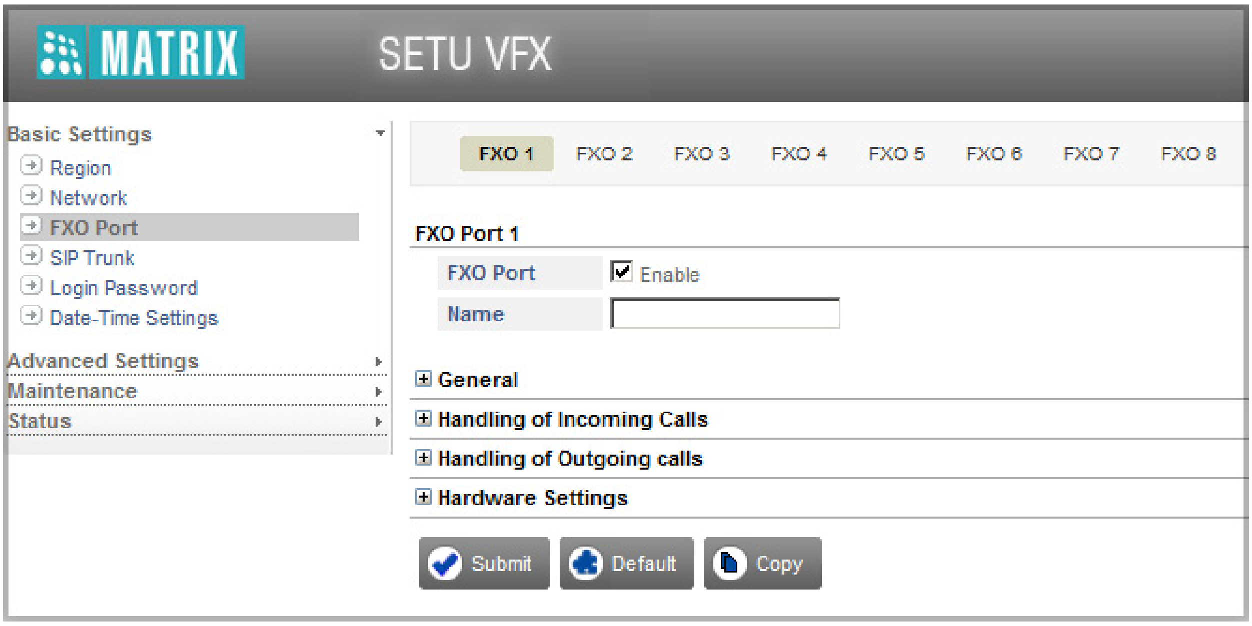
FXO Port

SETU VFX880 supports eight FXO Ports. The FXO port can be,

- interfaced with the analog trunk from CO and used to route incoming calls to the 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.



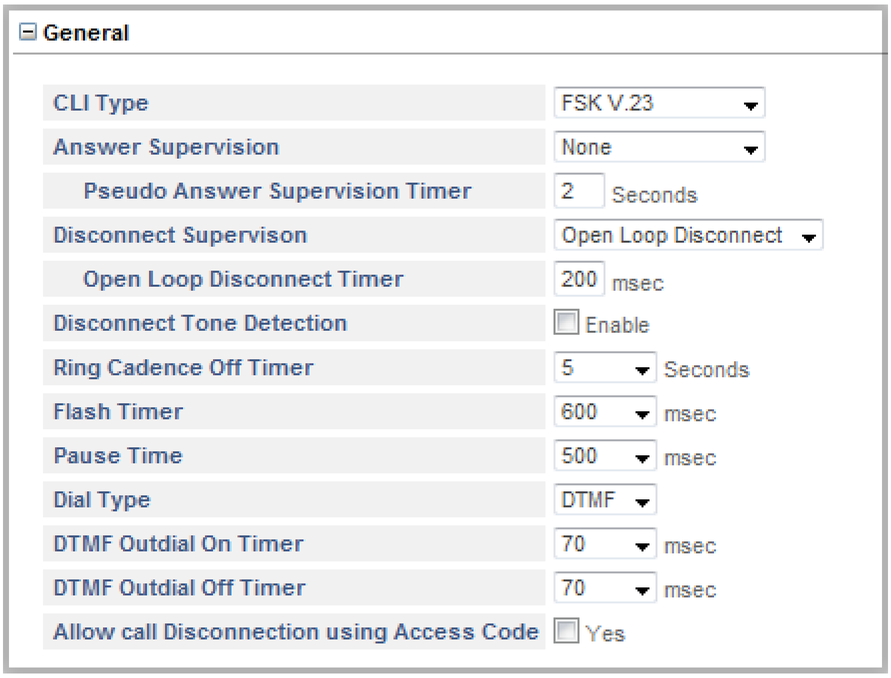
- Click the required FXO Number tab, **FXO 1** to **FXO 8**, you want to configure.
- Keep the **FXO Port** check box enabled.

  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 name display. Default: Blank

  The name you assign may consist of a maximum of 12 characters.

General

- Click **General**.



- Select the **CLI Type** from the options according to the CLI type supported by your CO network:
  - DTMF-India
  - DTMF-ETSI
  - DTMF-Denmark
  - DTMF-Brazil
  - DTMF-Any
  - FSK V.23
  - FSK Bellcore

  Default: FSK V.23

- SETU VFX 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** Type from the options according to the type supported by your CO network:

  - **None:** This option is to be selected when no signaling is available from the CO. If this option is selected, the call will be considered as matured on the expiry of the Pseudo Answer Supervision Timer (programmable; default: 2 seconds), irrespective of whether the call gets matured or not. After this timer, the system will start detection of Disconnect signal or Tone programmed on the FXO Port.

    - When the Answer Supervision type is selected as **None**, configure the **Pseudo Answer Supervision Timer.** The range of this timer is from 00 to 99 seconds. Default: 2 seconds.

  - **Battery Reversal:** If this option is selected, SETU VFX will consider the call as matured only if polarity reversal 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 to detect far-end disconnection (when the called party has gone On-hook).

  Select the appropriate **Disconnect Supervision** type from the options according to the type supported by your CO network:

  - **None:** Select this when no signaling is available from the CO.
  - **Battery Reversal:** Select this when call disconnection is signaled in the form of Battery Reversal. When the call is disconnected by the remote user, the polarity of line gets reversed, the FXO Port is released (free) and caller gets error tone.
  - **Open Loop Disconnect:** Select this option when call disconnection is signaled in the form of Open Loop Disconnect signal. The system will check Open Loop Disconnect signal for the time configured as the Open Loop Disconnect Timer.

    If Open Loop signal is detected for the time less than the Open Loop Disconnect Timer configured, it will not be considered as a valid Open Loop signal for releasing the port. If Open Loop is detected continuously at least for the time set in the Open Loop Disconnect Timer, it is considered as a valid Disconnect signal. After this signal is detected, the call will be released and caller will get error tone.

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

    - **Open Loop Disconnect Timer:** If you selected Open Loop Disconnect, you may set the value of this timer, as required. Default: 200 milliseconds. Valid range: 001 to 999 milliseconds.

    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 made on/ from the FXO Port.

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

  - If you enable Disconnect Tone Detection, you must select the **Disconnect Tone Type** you want the system to considered as Call Disconnect Tone on the FXO Port. You can select from:
    - Disconnect Tone 1
    - Disconnect Tone 2
    - Disconnect Tone 3
    - Disconnect Tone 4
    - Ring Back Tone
    - Error Tone 1
    - Error Tone 2
    - Busy Tone
    - Confirmation Tone
    - Feature Tone
    - Routing Tone
    - Intrusion Tone

Default: Disconnect Tone 1

If required, you can customise the frequencies and cadences of the Disconnect Tone as per your requirement. See “Disconnect Tone” for more instructions.

- You may configure the **Ring Cadence-OFF Timer** for the FXO Port to set OFF time for Ring cadence.

  During an incoming call on the FXO Port, if the CO gives ring with a long Ring OFF period, the system will consider that the ring has stopped, and it will stop ringing the FXS Port, even though the incoming call is still present.

  To get correct indication, the system supports the Ring Cadence OFF timer on the FXO Port so that ring can continue, even for incoming calls with long Ring OFF period.

  The range of the Ring Cadence OFF timer is from 1 to 9 seconds. Default: 5 seconds.

- Configure the **Flash Timer**, as per your requirement. The Flash timer signifies the time period for which the loop current breaks. SETU VFX uses this event to activate various features such as Call Hold, Call Transfer. Default: 600 milliseconds.

- Configure the **Pause Timer** to add delay, while a call is being made from the FXO Port. After the FXO Port goes Off-hook, SETU VFX 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: 500 milliseconds.

  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 appropriate **Dial Type** option as supported by your CO Network. You can select either Pulse or DTMF. Default: DTMF.

  - If you have selected the **Dial Type** as **Pulse**, you must set the **Pulse Dial Ratio** as per your country. You can select the required pulse dial ratio from the following options: 40:60, 50:50 and 33:67. Default: 33:67.

- In the **DTMF Dialout On Time** field, set the time as per the CO network. The DTMF On Time is the time for which the DTMF digit which is to be out dialed remains ON. Valid range: 50 to 200 milliseconds. Default: 70 milliseconds.

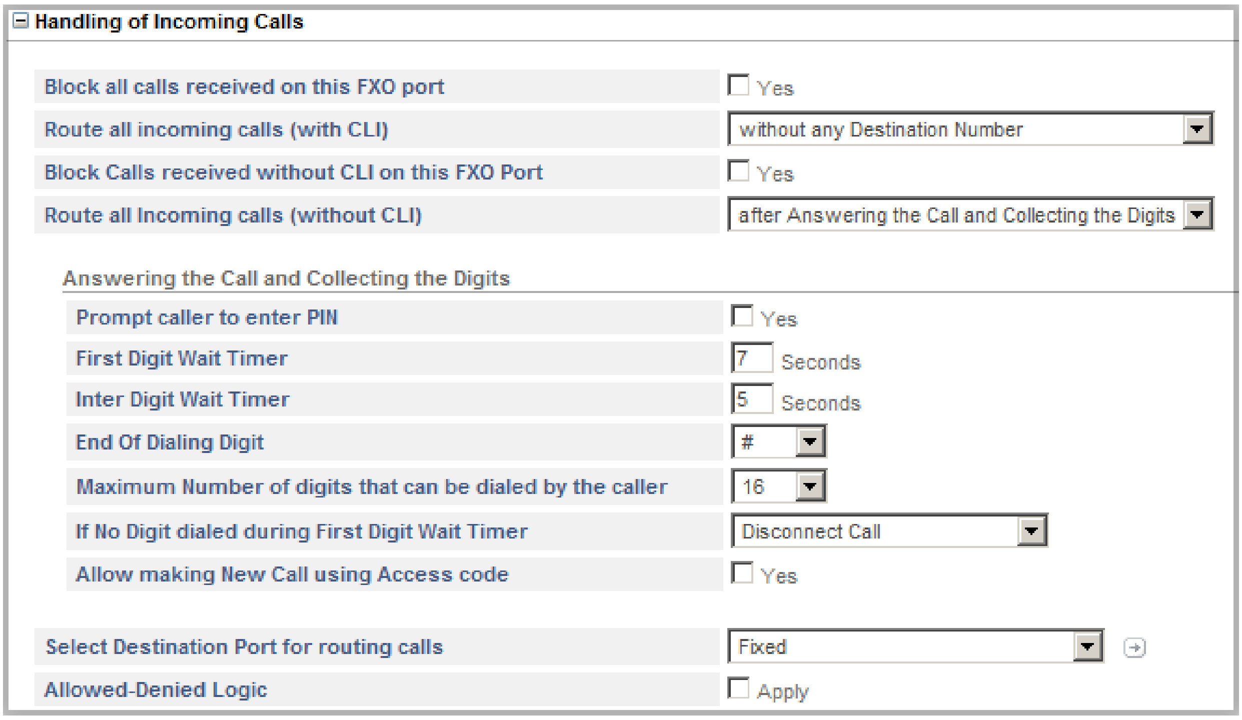
- In the **DTMF Dialout Off Time** field, set the time for which system should wait before dialing the successive DTMF digits so that the CO network can detect the dialed digits. Valid range: 50 milliseconds to 200 milliseconds. Default: 70 milliseconds.

- 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 VFX 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" to know more).

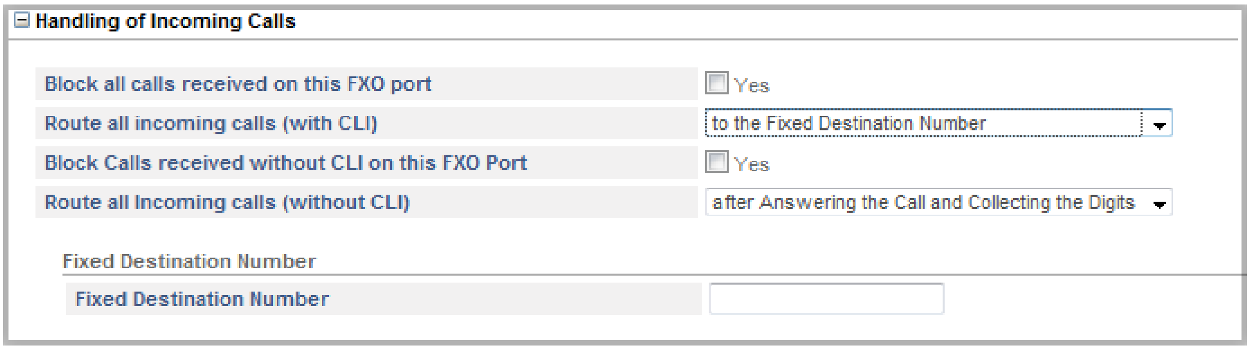
Route Calls without any Destination Number

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

- To apply this method, in the **Route all incoming calls (with CLI)** list, click **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.



To apply this method, do the following:

- In the **Route all incoming calls (with CLI)** list, click **to the Fixed Destination Number**.
- In the **Fixed Destination Number** field that appears, 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.

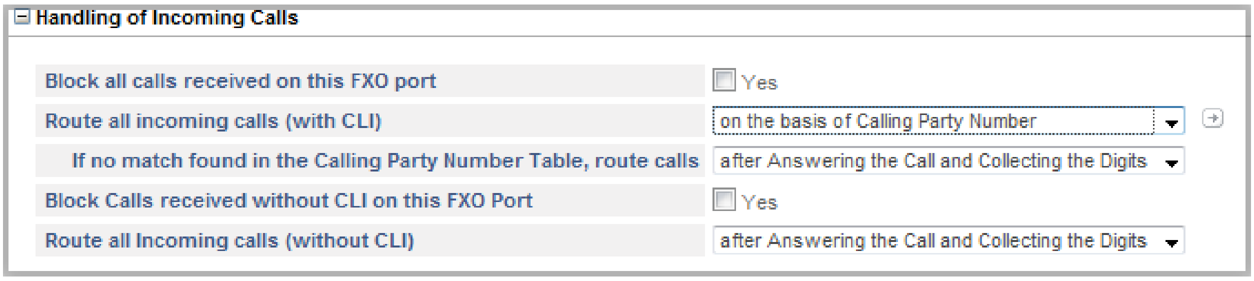
Route on the basis of Calling Party Number

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.

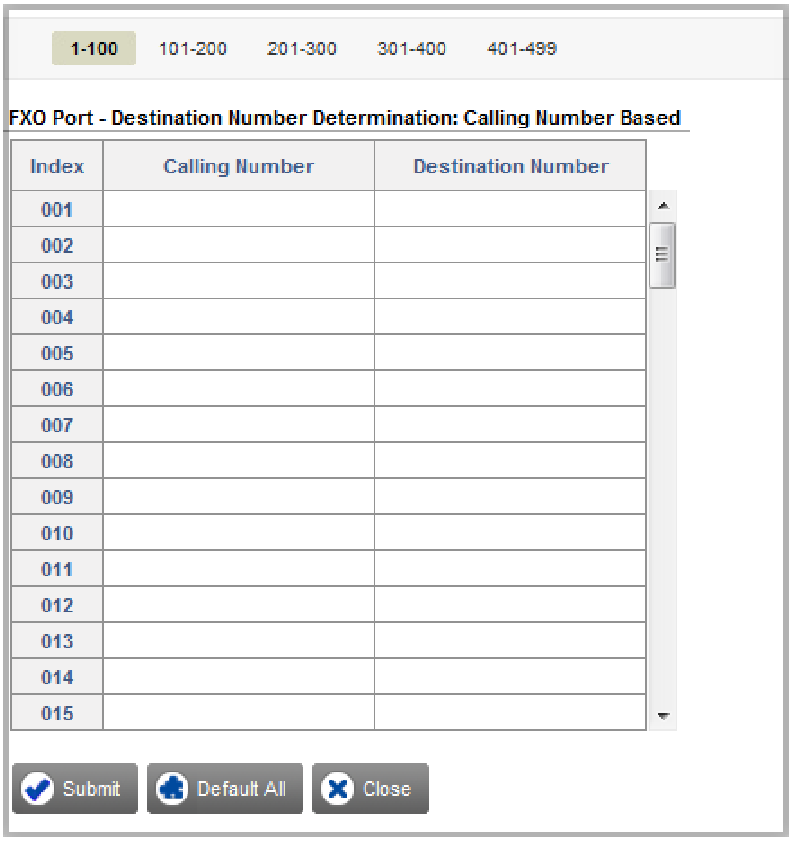
Whenever there is an incoming call on the FXO Port, SETU VFX will match the Calling Party Number with the entries of the Calling Party Number Based Table. If a match is found, the call will be routed to the destination number configured for that Calling Party Number.

To apply this method, do the following:

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



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



Configure the **Calling Number Based** table for the FXO Port. You can enter up to 500 Calling Party Numbers and their corresponding Destination Numbers in this table.

- In the **Calling Number** column, enter the calling party numbers. The Calling numbers may consist of a maximum of 24 characters. Default: Blank.
- For each calling party number, enter a corresponding destination number in the **Destination Number** column. 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 page.

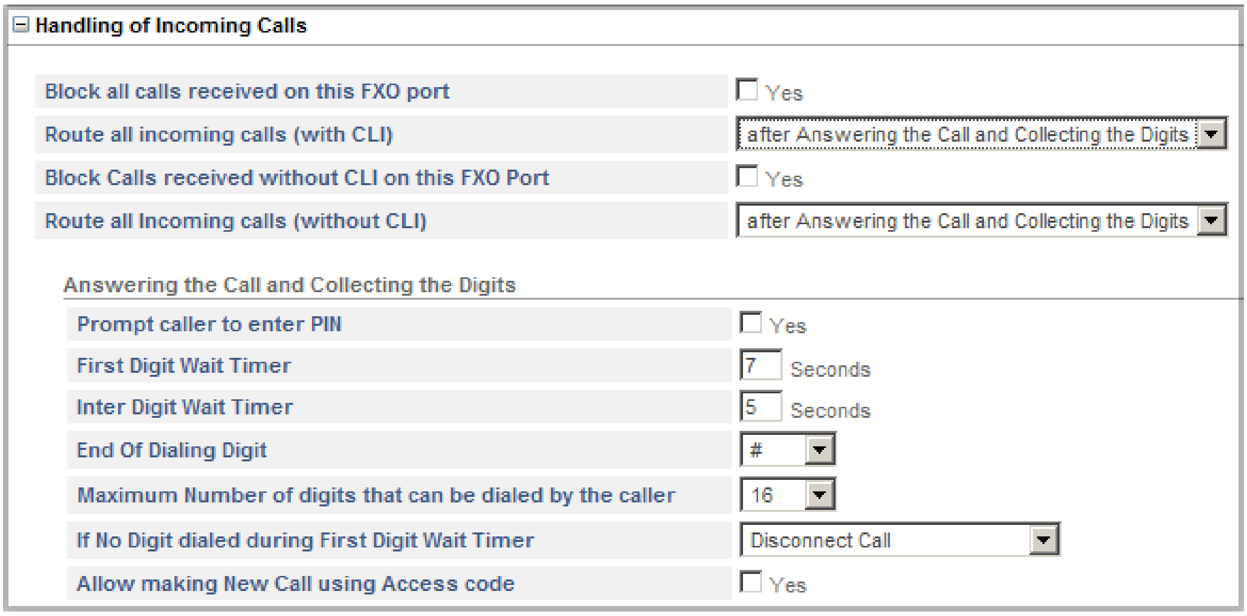
  You can also configure the **Calling Number Based** Table from *Advanced Settings* links. For instructions, see “Destination Number Determination” under *Advanced Settings*.

- 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 the **If no match found in the Calling Party Number Table, route calls** box, you may select the option **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.



To apply this method, do the following:

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

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

- Set the following **Answering the Call and Collecting the Digits** parameters to the desired values.

  - To enable PIN Authentication for incoming calls on this 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*.

  - You may change the duration of the **First Digit Wait Timer (FDWT)**, if required. FDWT is the time for which the system waits for the caller to dial the destination number after it has played dial tone to the caller. The valid range of this timer is 01 to 99 seconds. Default: 7 seconds.

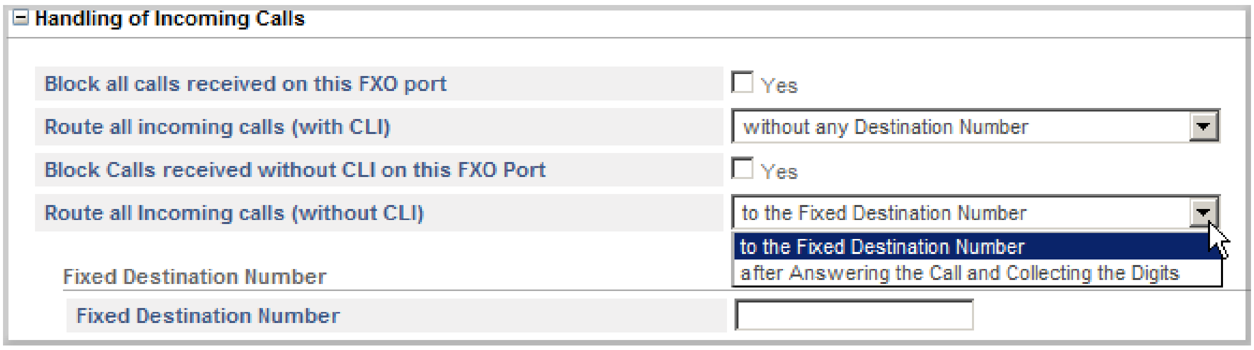
- You may configure the following options as end-of-dialing indication:
  - In the **Inter Digit Wait Timer** field, define the number of seconds the system should wait while receiving the dialing digits, to consider it as end-of-dialing. You may change this timer, if required.The valid range is 01 to 99 seconds. Default: 05 seconds.
  - In the **End of Dialing Digit** box**,** select # or * as termination digit the system should consider to detect end of dialing. Default: #
  - In the **Maximum number of digits that can be dialed by the Caller** box, 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: 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 the **If No Digit dialed during First Digit Wait Timer (FDWT)** box, select the desired option: **Disconnect the Call** or **Use Fixed Destination Number**. Default: Disconnect Call.

- If you selected **Use Fixed Destination Number**, enter the desired destination number in the **Fixed Destination Number** field**.** The Destination number may consist of a maximum of 24 digits. Valid digits are 0 to 9, *, # and (.) dot. Default: Blank.
- 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 received without CLI through this port, select the **Block Calls received without CLI on this FXO Port** check box.



- 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 for Routing Calls

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 VFX 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" to know more).

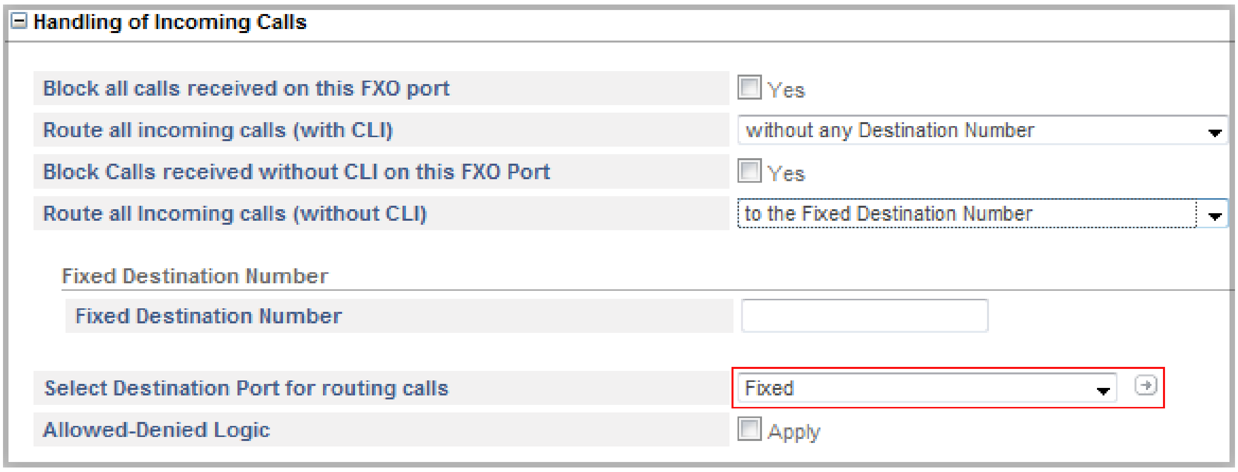
Fixed

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.

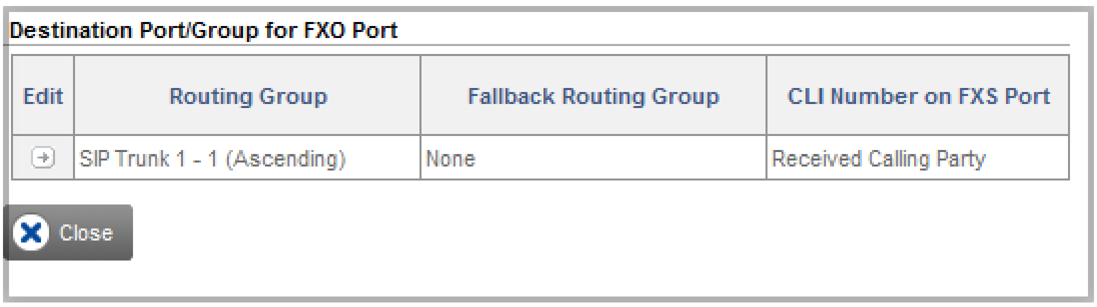
To apply this method, do the following:

- In **Select Destination Port of routing calls**, click **Fixed**.

- Click **Settings** .

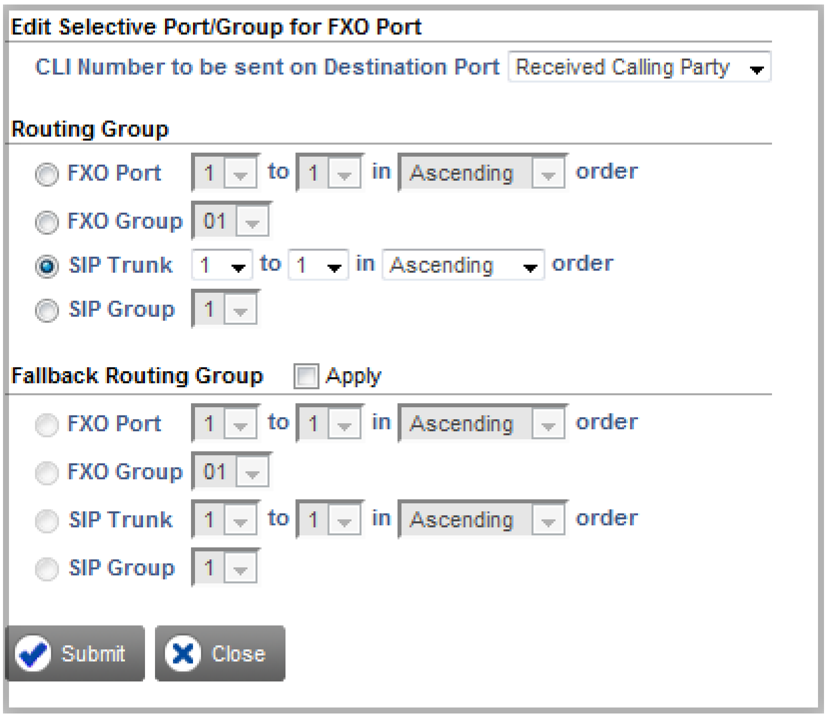


The **Destination Port/Group for FXO Port** window opens.



- To change the default Routing Group and create the Fallback Routing Group, under **Edit**, click **Settings** .

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



- Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
    - Received Calling Party
    - Received Called Party

  Default: Received Calling Party

CLI Number 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* **FXO Ports** as members,
        - Select the desired **FXO Port** numbers as members. Default: 1.
        - In the **in - order** field, 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 FXS Group. See “Group” for detailed instructions.

- Similarly, you can create group of *sequential* and *not-sequential* SIP Trunks as members.

- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions for creating *sequential* and *not-sequential* groups for FXO Ports and SIP Trunks.

- Click **Submit** to save changes. The **Edit** window closes.

- The edited entry appears in the **Destination Port/Group for FXO Port** window. Close this window to return to the main page.

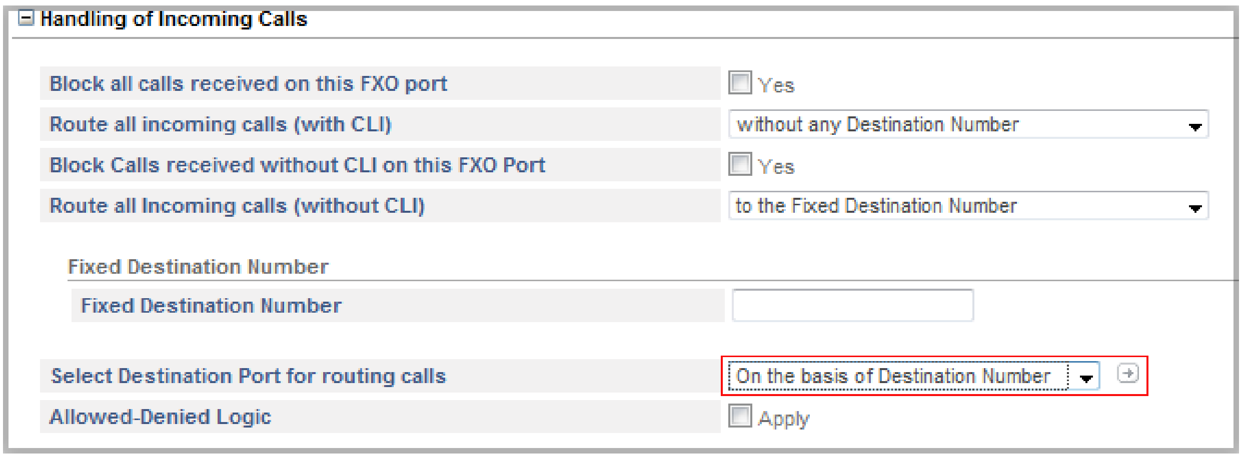
On the basis of Destination Number

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.

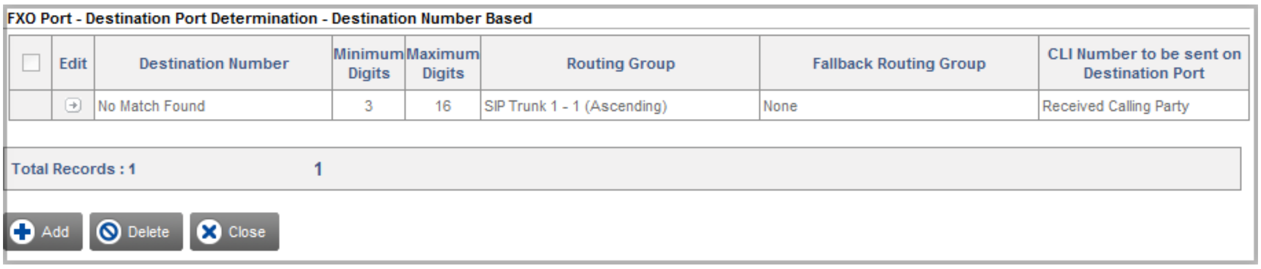
To apply this method, do the following:

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

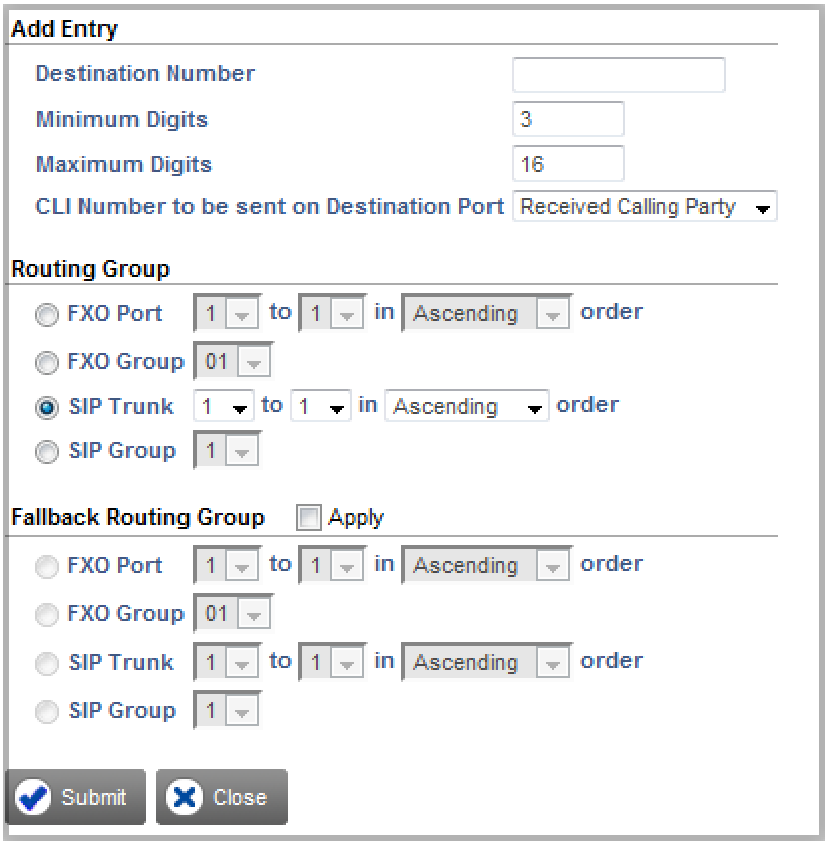
- Click **Settings** .



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



- Click **Add** to add a new entry. The **Add Entry** window opens. You can add upto 100 entries.



- In the **Destination Number** field, enter the number (max. 24 characters) you expect callers to dial. Valid digits: 0 to 9, *, #, (dot). Default: Blank.
- In the **Minimum Digits** field, enter the minimum number of digits of the destination number that the caller must dial for the system to route the call. Range: 01 to 24. Default: 03.

  If the dialed number string is less than the configured minimum length, the call will be rejected.
- In the **Maximum Digits** field, enter the maximum number of digits of the destination number the caller must dial for the system to route the call. Maximum length range: 01 to 24. Default:16.

  If the number string dialed by the caller exceeds the maximum length configured, the system will strip off the extra digits, and routes the call.

- Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

CLI Number to be sent on Destination Port *is applicable only when* ***FXS Port*** *or* ***FXS Group*** *is selected as the Destination Port (Routing Group).*

- 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 the **in - order** field, select the order in which the system should hunt for a free member FXO Port to route the call.

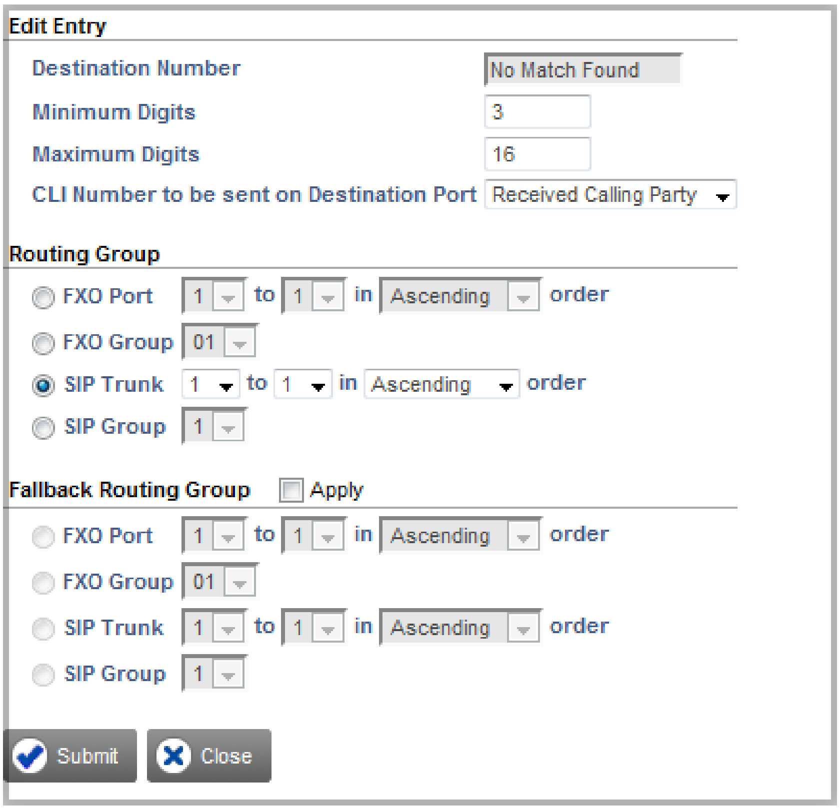
      To start hunting from the first to the last member FXO Port, select **Ascending**.

      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 group of *sequential* and *not-sequential* SIP Trunks as members.
- You may create the **Fallback Routing Group**.
  - Select the Fallback Routing Group **Apply** check box.
  - Follow the same instructions for creating *sequential* and *not-sequential* groups, for FXO Ports and SIP Trunks.
  - 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** window.
- 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.



- Create the **Routing Group** and **Fallback Routing Group**.
- Click **Submit** and close the window.

- 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.
- Close the window if you have finished adding/editing entries.

You can also configure the **Destination Number Based** Table from *Advanced Settings*. See "Destination Port Determination" under Advanced Settings.

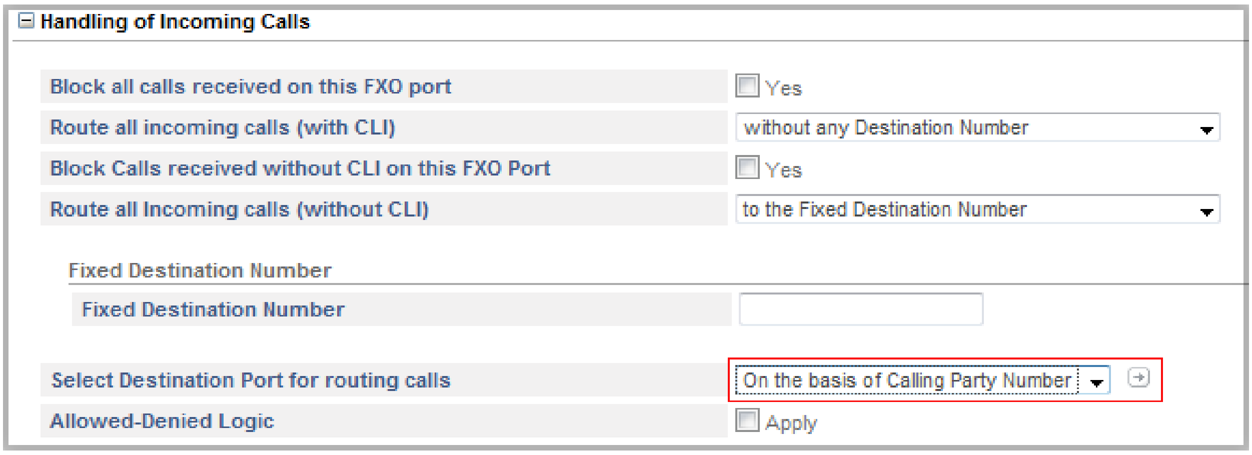
On the basis of Calling Party Number

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

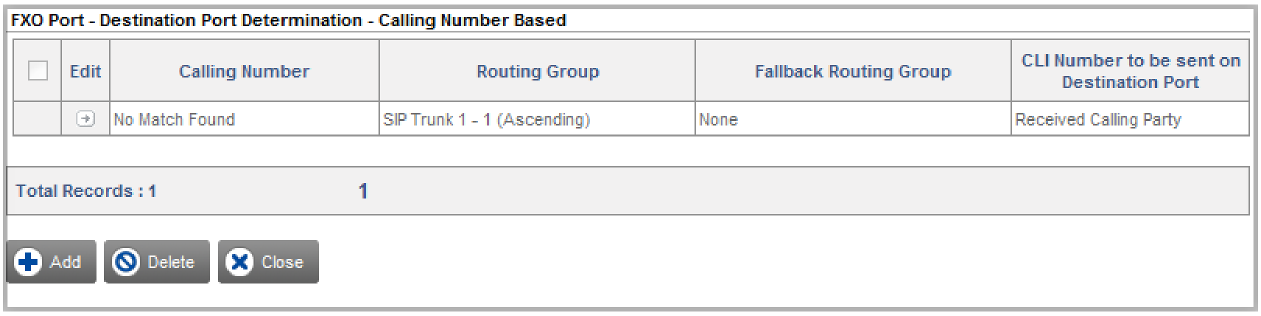
To apply this method, do the following:

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

- Click **Settings**.



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

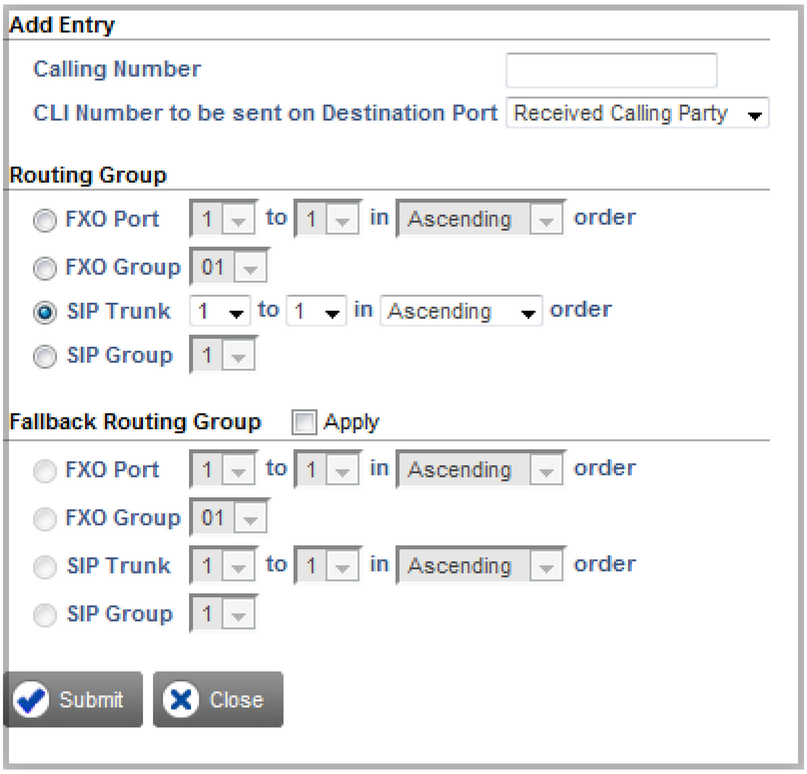

FXO Port - Destination Port Determination - Calling Number Based

	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

Add Delete Close

- Click **Add** to add a new entry. The **Add Entry** window opens. You can add upto 500 entries.



- In the **Calling Number** field, enter numbers (max. 24 characters) from which you expect calls to be received. Valid digits: 0 to 9, *, #, (dot). Default: Blank.

- Select the **CLI Number to be send on Destination Port**. You may select from the following options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

***CLI Number to be sent on Destination Port** is applicable only when **FXS Port** or **FXS Group** is selected as the Destination Port (Routing Group).*

- 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 the **in - order** field, 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.

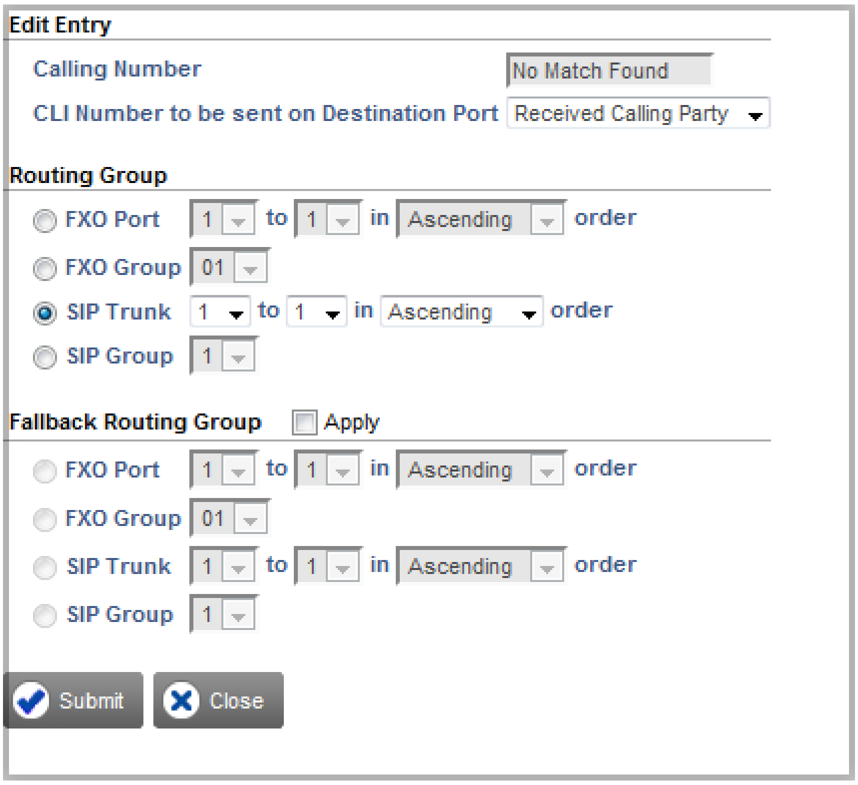
- To create a group of *not-sequential* **FXO Ports** as members,
  - Select a **FXO Group**.
  - Select **FXO Group** number. Default:1.
  - Click **Settings** . The **FXO Groups** window opens. Create the FXO Group. See “Group” for detailed instructions.
- Similarly, you can create group of *sequential* and *not-sequential* FXO Ports and SIP Trunks as members.

- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions for creating *sequential* and *not-sequential* groups, for FXO Ports and SIP Trunks.
  - 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** window.
- 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 No Match Found numbers entry,
  - For the **No Match Found** entry in the table, under **Edit,** click **Settings**

.

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group**.
- Click **Submit** and close the window.

- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click **Delete** button.
- Close the window if you have finished adding/editing entries.

You can also configure the **Calling Number Based** Table from *Advanced Settings*. See "Destination Port Determination" under Advanced Settings.

Allowed - Denied Logic (Toll-Control)

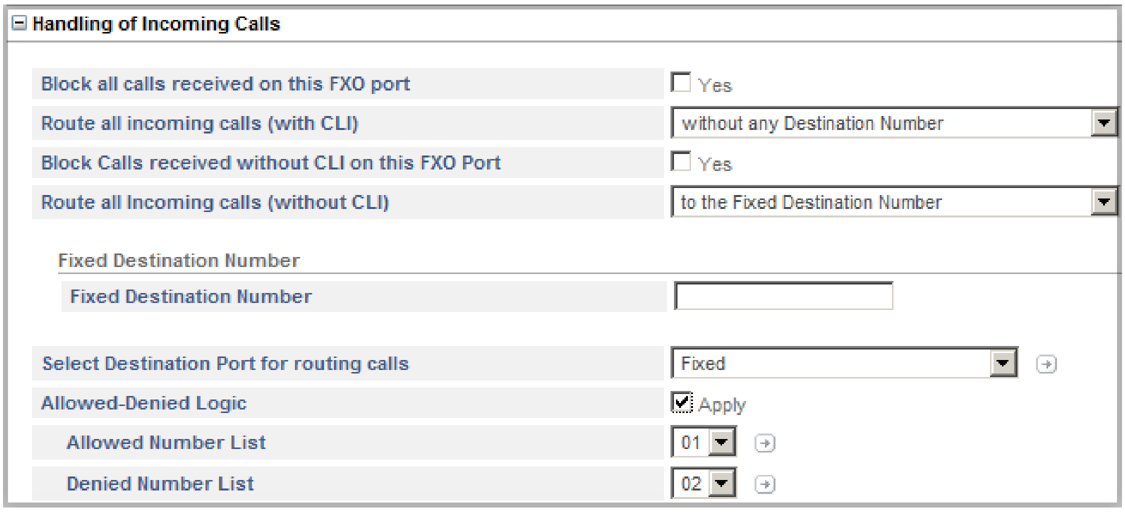
With the Allowed-Denied Numbers feature you can permit and restrict the dialing of particular numbers from the FXO Port.

Allowed Denied Number Logic makes use of two Number Lists:

- **Allowed Numbers List:** This is the list of numbers that are allowed to be dialed out by the caller on the FXO Port. By default, List Number 1 is assigned to the FXO Port.
- **Denied Numbers List:** This list contains the numbers that are denied to be dialed out by the caller on the FXO Port. By default, List Number 2 is assigned to the FXO Port.

To apply Allowed - Denied Logic on the FXO Port,

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



- To configure the **Allowed Number List**, click **Settings** .

  - The Number List page opens in a new window.
  - By default, Number List 1 is assigned as Allowed Number List.
  - Enter the numbers you want the system to allow to be dialed in this list.
  - Click **Submit** to save the entries and close the window.

- To configure the **Denied Number List**, click **Settings** .

  - The Number List page opens in a new window.

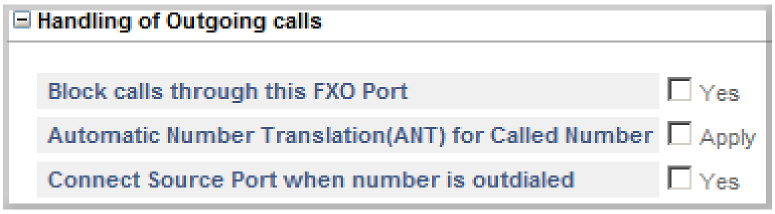
- By default, Number List 2 is assigned as Denied Number LIst.
- Enter the numbers you want the system to restrict from being dialed out in this list.
- Click **Submit** to save the entries and close the window.

You may configure a different Number List as Allowed and Denied List. See "Allowed-Denied Logic" under "Number Lists".

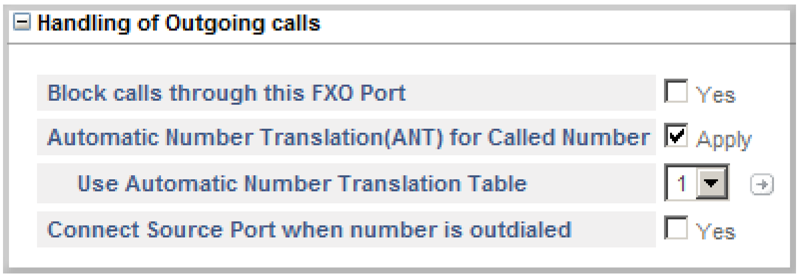
Handling of Outgoing Calls

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

- Click **Handling of Outgoing Calls**.



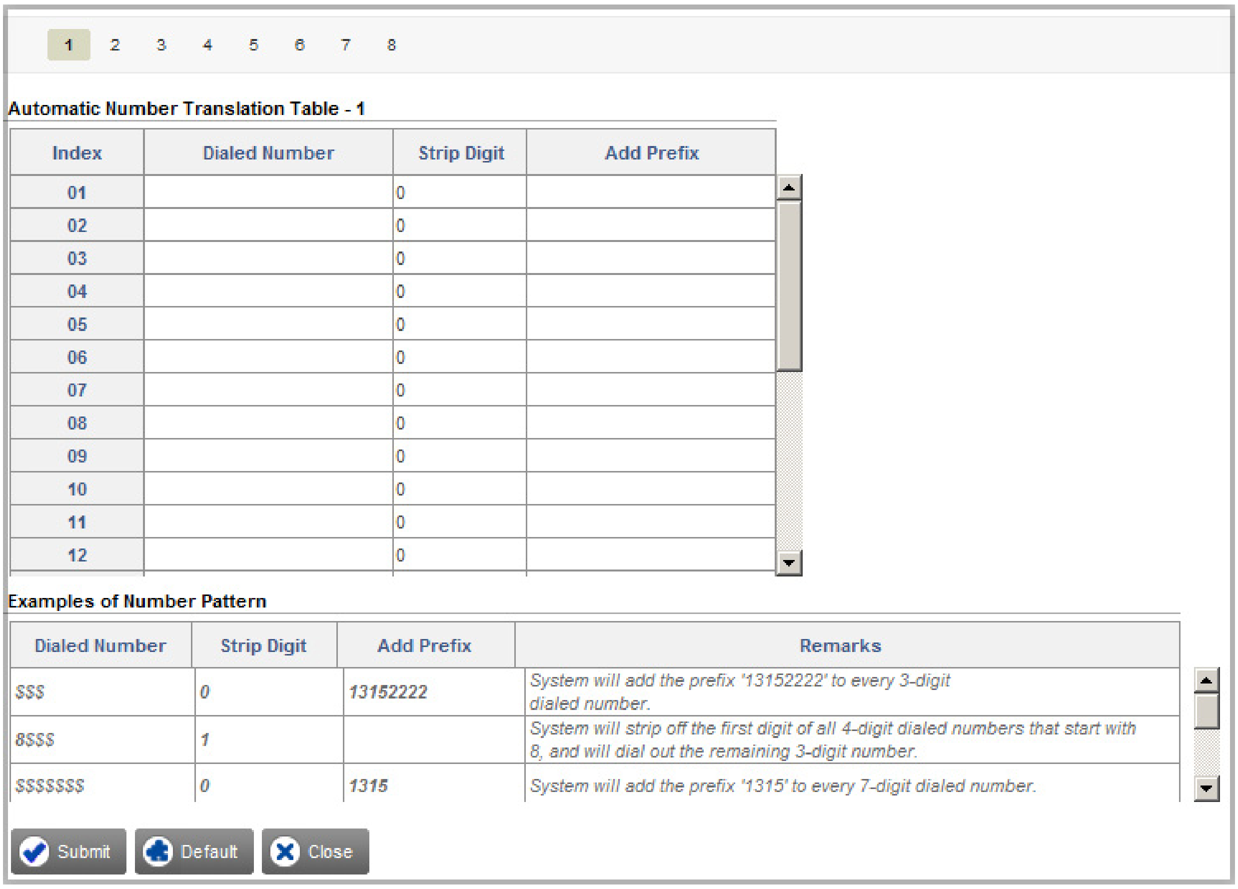
- If you do not want to route outgoing calls through this port, select the **Block calls through this FXO Port Yes** check box.
- You can apply **Automatic Number Translation logic** on outgoing calls made from the FXO Port.
    - To apply ANT logic on the Called Numbers, click the **Automatic Number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



- In the **Use Automatic Number Translation Table**, select the ANT Table number you have configured for the FXO Port. By default, Table 1 is assigned to the FXO Port.

  If you have not configured the Automatic Number Translation Table,

- Click **Settings** . The Automatic Number Translation Table page will open in a new window.



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 1 or any other Table number (2 to 8) for the FXO Port. 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.

- 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, i.e. the called party has answered the call. Until the call matures, the caller hears only Ring Back Tone played by the network.

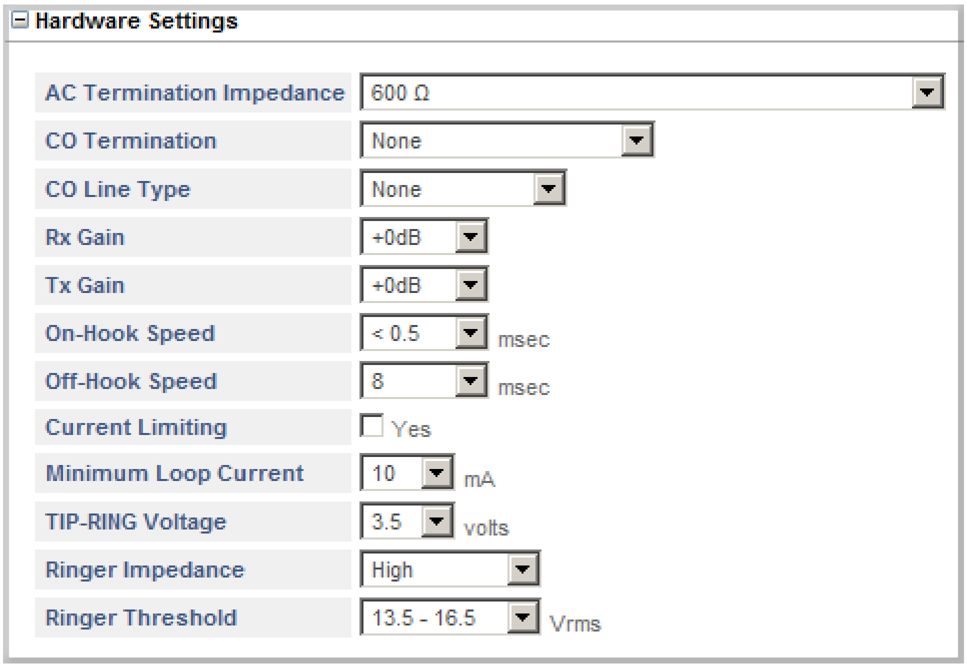
  By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

- 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.*

Hardware Settings

- Click **Hardware Settings**.



- In the **AC Termination Impedance** field, select the appropriate Impedance of the FXO Port as per the AC Termination Impedance supported by your CO network. Default: 600 Ω.

- In the **CO Termination** field, select the appropriate line impedance match. This would depend on the region where SETU VFX is deployed. Default: None.

  This parameter allows you to increase near-end echo cancellation on the FXO Port. Near-end echo is primarily caused by the mismatch between AC Termination Impedance (presented by FXO Port of SETU VFX to the line) and CO Termination (Impedance presented by the Central Office to the line), and to some extent by 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 and CO Termination, and selecting a Line Type that most closely models the line that connects the FXO Port of SETU VFX to the Central Office.

  Now, select the line model to be used from the CO Line Type list.

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

  This parameter allows you to select the Line model for the AC Impedance-CO Termination you selected.

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)”.

- If required, you may adjust the **Rx Gain** (Receiving Volume) on the FXO Port to increase or decrease the volume of the calling party's voice being transmitted to you. Select the required Rx Gain. Default: +0 db.

- If required, you may also adjust the **Tx Gain** (Transmit Volume) on the FXO Port to increase or decrease the volume of your voice being transmitted to the remote party. Select the desired Tx Gain. Default: +0 db.

- In the **On-Hook Speed** field, set the amount of time required for the line-side device (DAA) to go On-hook.

  The On-hook speed specified is measured from the time the On-hook bit is cleared until loop current equals zero. Select the desired On-hook Speed from the following options:
  - <0.5ms
  - 3 ms
  - 26 ms

  Default: <0.5 milliseconds.

- In the **OFF- Hook Speed** field, define the time required by the line transients to settle, after which transmission or reception can occur. Select the desired Off-hook Speed from the following options:
  - 512 ms
  - 128 ms
  - 64 ms
  - 8 ms

  Default: 8 milliseconds.

- 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.

- For the line-side device (DAA) of the FXO Port to operate, set the **Minimum Loop Current**. You can select the minimum operational loop current from the following options as per your requirement:
  - 10
  - 12
  - 14
  - 16

  Default: 10 mA.

- To adjust the TIP/Ring Voltage on the line side, set the **Tip Ring Voltage (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 from the following options:
  - 3.1
  - 3.2
  - 3.35
  - 3.5

  Default:3.5 volts.

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

  High signifies 20Mohm 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 validates the Ring signal. Set Ringer Threshold from the following options:
  - 13.5 - 16.5
  - 19.35 - 23.65
  - 40.5 - 49.5

  Default: 13.5 - 16.5 Vrms.

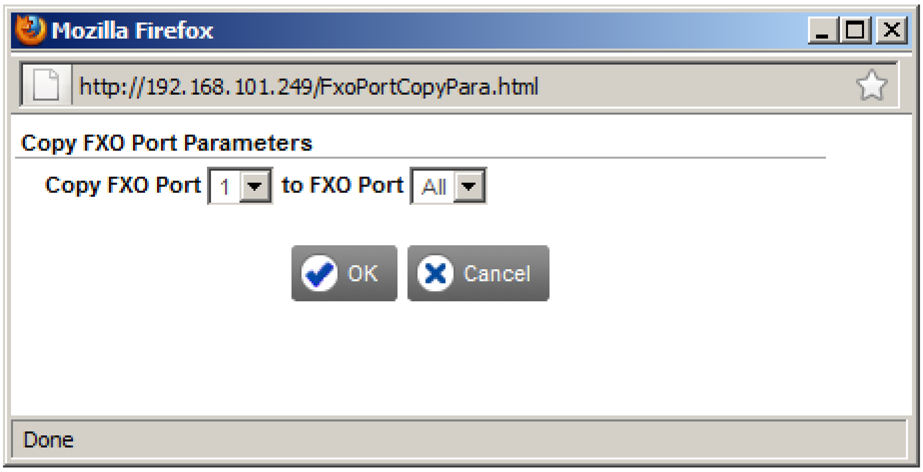
- Click **Submit** to save.

- To configure the next FXO Port, click the FXO Port number tab and follow the same instructions as given earlier.

Copy Port Settings

You can also copy the settings of one FXO Port to another FXO Port using the **Copy** button. To do this,

- Click the **Copy** button. The **Copy FXO Port Parameters** window opens.



- In the **Copy FXO Port** box, select the number of the port you want to copy settings *From*. In the **to FXO Port** box, select the number of the port you want to copy the settings *To*.

- Click the **OK** button and close the window.

- Once you have copied the settings, you can again edit the specific parameters of the FXO Port you copied the settings to.

Life Line Port[12]

You must connect a PSTN line to the Lifeline Port of SETU VFX. The Lifeline Port is used to make calls during emergency conditions listed below.

- When the Ethernet link is down.
- When the SIP Trunk fails to register due to any reason.
- When an emergency number is to be dialed.
- When you are unable to achieve toll quality speech through SIP due to some problem in the IP network.

The Lifeline Port is not a true FXO Port as you cannot receive incoming calls on this port. You can only make outgoing calls during an emergency, by dialing the Lifeline Port access code from the FXS Port 1.

However, during Power failure, FXS Port 1 will be automatically connected to the Lifeline Port. You will be able to receive and make calls through this port without dialing the Lifeline Port access code.

By default, the access code of the Lifeline Port is **#***. You may change this access code, if required. For instructions on changing the access code, see “Access Codes”.

- *Lifeline Port can be accessed through the FXS Port 1 only.*
- *Lifeline Port access code cannot be dialed, if any held call or waiting call is present on the FXS Port 1.*

How to use Lifeline Port?

- Go OFF-Hook from the telephone connected to the FXS Port 1.
- You will get dial tone.
- Dial the access code (**#***) for accessing the Lifeline Port.
- You will get the PSTN dial tone.
- Dial the desired number.
- Talk when the called party answers the call.
- Go ON-Hook to disconnect.

12. *Applicable only for VoIP-FXS Gateway.*

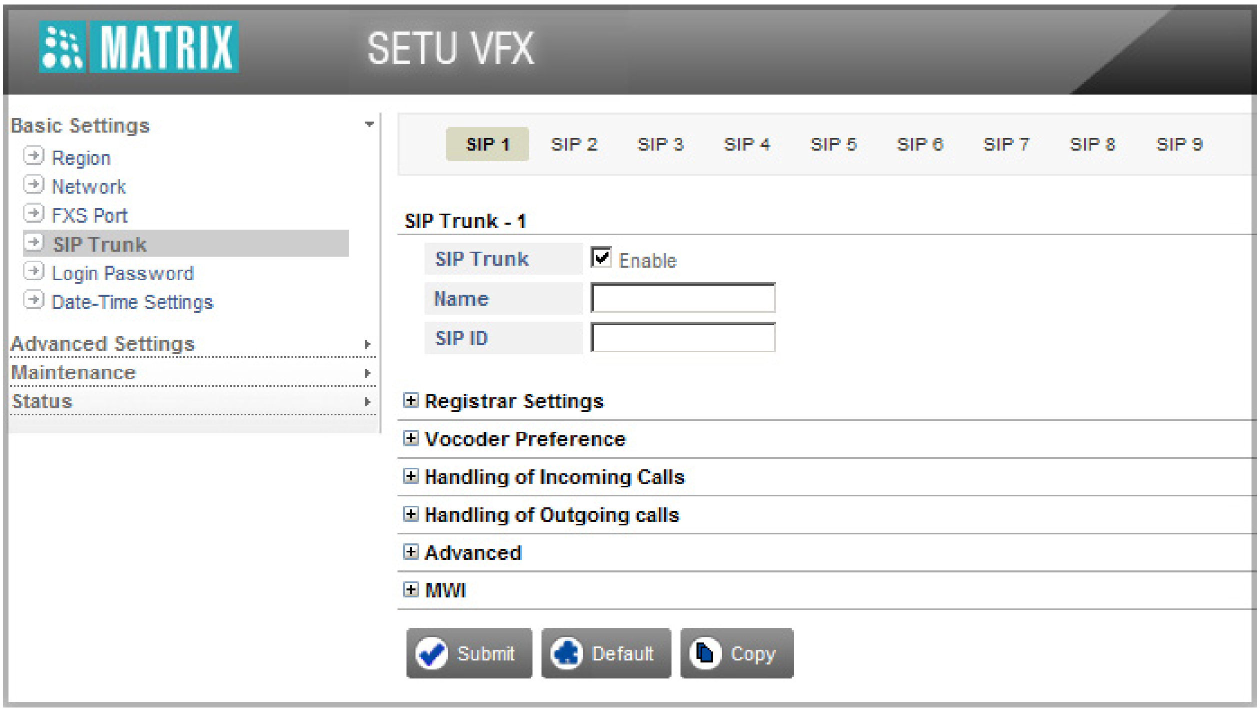
SIP Trunks

SETU VFX supports nine 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 Non-Proxy, that is, Peer-to-Peer.

To configure a SIP Trunk,

- Click the **SIP Trunk** link.
- Click the desired SIP Trunk Number tab, **SIP 1** to **SIP 9**, you want to configure.

  The SIP Trunk page opens.

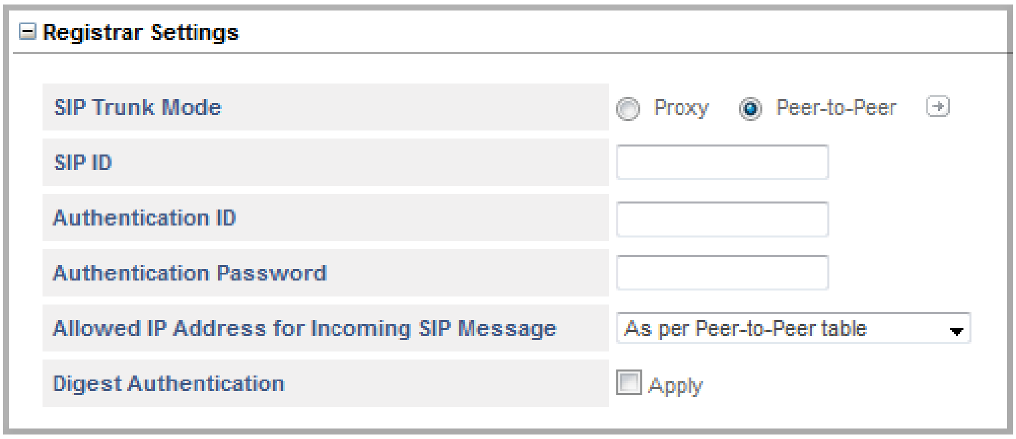


- Select the **SIP Trunk Enable** check box to use the SIP Trunk. Keep it disabled, if you do not want to route calls through this Trunk. Default: Disabled.
- You can assign a **Name** to the SIP Trunk for identification of this Trunk. Default: Blank.

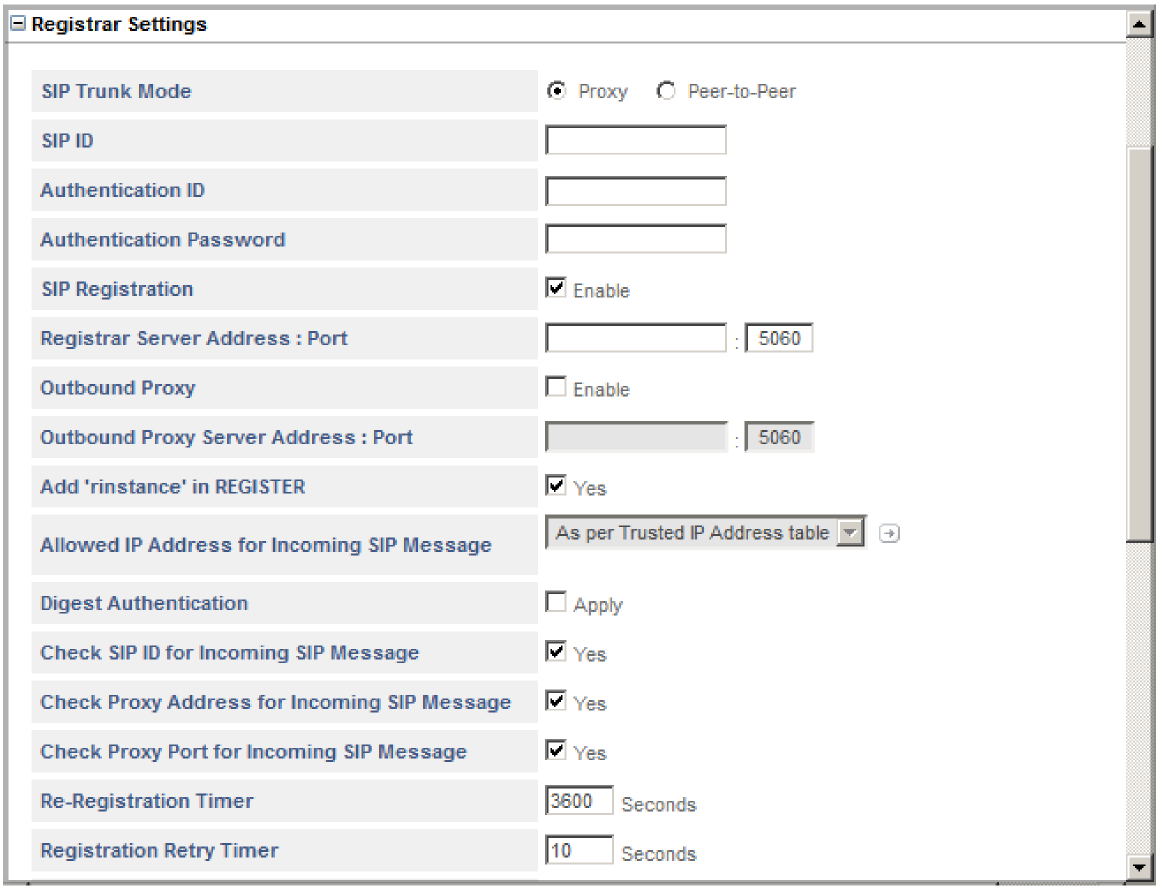
  The name you assign to the SIP Trunk will appear on the display of the remote party's phone when a call is made through this Trunk.
- In the **SIP ID** field, the SIP ID that you assign under *Registrar Settings* will appear.

Registrar Settings

- Click **Registrar Settings**.



- Select the **SIP Trunk Mode** according to your installation. Default: Peer-to-Peer.
  - Select **Proxy**, if you want to register this SIP Trunk with an ITSP or a Registrar Server.
  - Select **Peer-to-Peer**, if you want to use the Trunk for Peer-to-Peer (non-proxy) calls.
- To configure **SIP Trunk Mode** as **Proxy**, do the following:



  - Select **Proxy**. You will be presented with the related registrar settings.

- In the **SIP ID** field, enter the SIP ID provided by your ITSP. For example, if the SIP URI provided by the ITSP is 12345@abc.com, enter 12345 in this field. 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.

- Keep the **SIP Registration** check box enabled.

  SETU VFX will send the REGISTER message to Registrar proxy or Outbound proxy as applicable.

  Clear the check box only if you want to disable registration.

- In the **Registrar Server Address: Port** field, 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 port range: 1025–65534. Default Port: 5060.

- If your Service Provider uses outbound proxy for handling voice calls, select the **Outbound Proxy** check box. Default: Disabled.

- In the **Outbound Proxy Server Address: Port** field, 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 port range:1025–65534. Default Port: 5060.

- To add 'rinstance' in REGISTER message, keep the **Add 'rinstance' in REGISTER** check box selected.

  'rinstance' is any random value which can be used by the SETU VFX to fetch its own contact binding, that is, to know the Registration Expiry Timer assigned by the server.

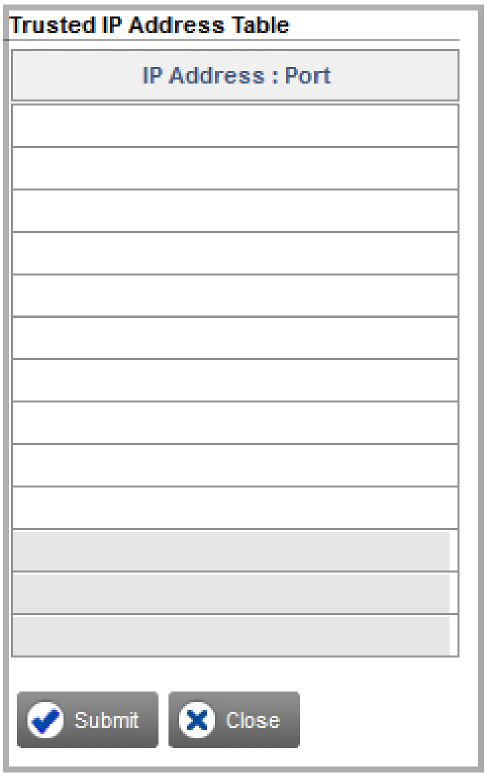
  When you enable 'rinstance' in Register, SETU VFX will generate any random value of 'rinstance' and include it in the REGISTER message. The system will use the registration expiry timer of that contact binding.

- 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.



  - 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 feature on the SIP Trunk, you must 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 VFX 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 VFX 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 VFX to validate the Proxy Port during an incoming call. Default: Enabled.

- Select **Send OPTIONS message as Heartbeat** check box, if you want SETU VFX to send OPTIONS messages periodically to the Proxy Server to monitor its availability. Default: Disabled.

Calls can be made and received only if the Proxy Server is alive. If the Proxy Server is unavailable (no response is received from the server), the status of the SIP Trunk will display *"Inactive"* along with the Reason for Failure.

- *To view status of the Proxy Server, go to "SIP Trunk" Status.*
- *The **Send OPTIONS message as Heartbeat** will work only if,*
  - ***SIP Trunk Mode** is configured as Proxy.*
  - ***SIP Registration** is disabled.*

If you enable *Send OPTIONS message as Heartbeat*, you must configure the **Heartbeat Interval**.

- Set the **Heartbeat Interval** (Seconds). It is the time period after which SETU VFX will send the OPTIONS message to the Proxy Server to check its availability. Valid range of Heartbeat Interval is from 10 to 999 seconds. Default: 60 seconds.

- Set the **Re-registration Timer**. This is the time period after which the SETU VFX will send registration request to maintain registration binding with the Registrar server. Valid range: 00001–65535 seconds. Default: 3600 seconds.

*The **Re-registration Timer** will be applicable only if, **SIP Registration** is enabled.*

- Define the **Registration Retry Timer**. When a registration attempt fails, SETU VFX will resend registration request to the Registrar Server after the expiry of the Re-registration Timer. Valid range: 00001–65535. Default: 10 seconds.

*The **Registration Retry Timer** will be applicable only if, **SIP Registration** is 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.

- 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** field, enter addresses of the alternate Registrar Servers and their respective listening ports. The Fallback Registrar Server Address can be of maximum 64 characters. Valid port range: 1025–65534. Default Port: 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** field, 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 port range:1025–65534. Default Port: 5060.

- In the **Fallback Event** list, select the event on occurrence of which SETU VFX should fallback to an alternate Registrar/Outbound Proxy Server, if available.
  - No Response

- 503 or No Response
- 5xx or No Response

Default: 503 or No Response

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

If the SIP General Request Timer configured in the System Parameters is less than the No Response Timer, then SETU VFX 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 VFX 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 VFX 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.*

- Keep the **Switch Registration to Alternate Server on Fallback** check box enabled. SETU VFX 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** 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.

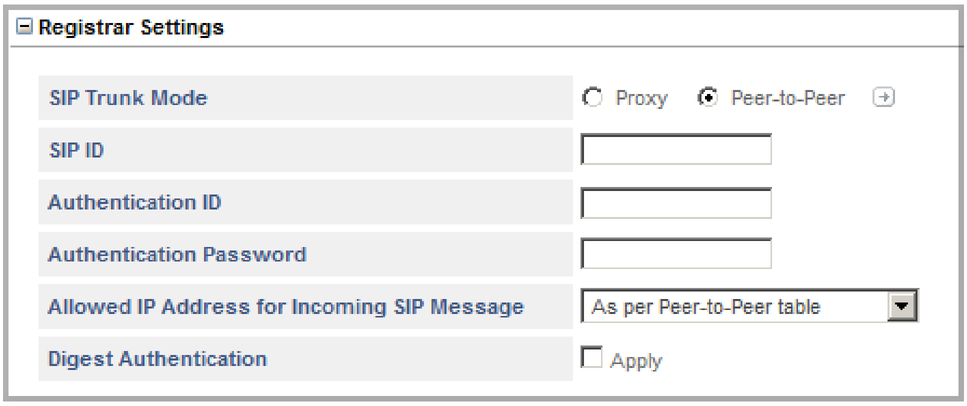
  - **Cyclic**: Each new call will be processed through the next active Registrar/Outbound Proxy Server.

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.

Default: Last Call Active.

*If you have disabled **SIP Registration**, it is recommended that you enable **Send OPTIONS message as Heartbeat** to use Fallback facility efficiently.*

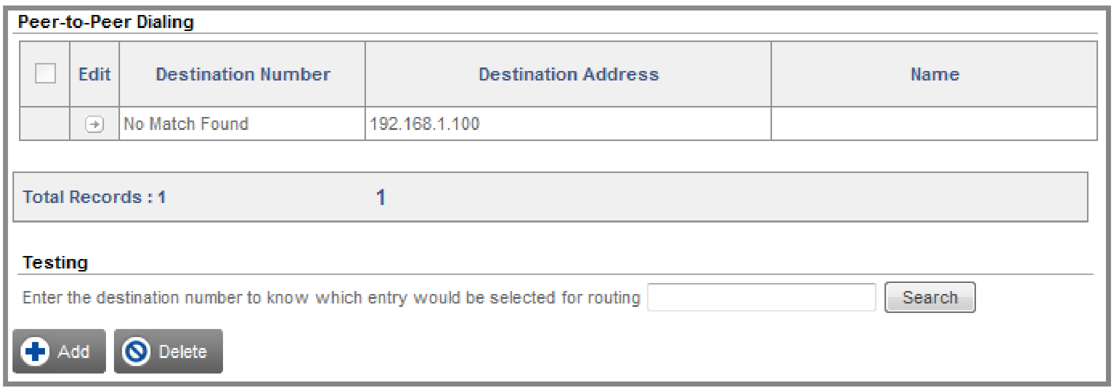
- To configure **SIP Trunk Mode** as **Peer-to-Peer**, do the following:



- Select the **Peer-to-Peer** option. You will be presented with the related peer-to-peer SIP Trunk parameters.
- In the **SIP ID** field, enter the desired SIP ID which the remote parties will use to call this SIP Trunk. Default: Blank.

  The SIP ID may be a number or text consisting of a maximum of 40 characters.
- In the **Authentication ID** field, enter the ID of your preference as Authentication ID. Default: Blank.
- In the **Authentication Password**, enter a password of your choice as Authentication Password for the Authentication ID you have assigned. Default: Blank.
- To configure the Peer-to-Peer Number strings, click **Settings** .

  The **Peer-to-Peer Dialing** table window opens.

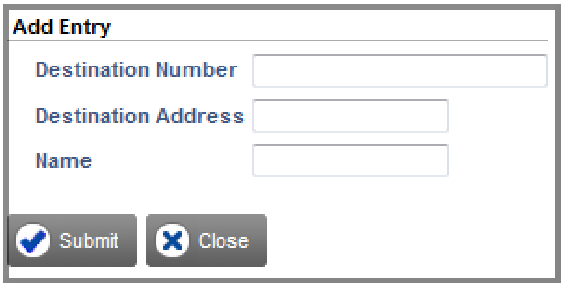


You can add as many as 500 number strings to this table. Each entry in the table consists of the Destination Number, Minimum and Maximum Digits, Destination Address and Name.

The first entry in the table is reserved for numbers that do no match with any of the entries in the table, the **No Match Found** entry. When the number dialed by users does not match with any of the entries in the table, the system uses the **Destination Address** assigned to the **No Match Found** entry in the table to route the call.

- To enter a new record in this table, click the **Add** button.

  The **Add Entry** window opens.



- In the **Destination Number** field, enter the peer-to-peer number string—prefix or entire number—that will be dialed. The number string must not exceed 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, 123@abc.com, you must enter 123 in this field.

Wildcard Characters

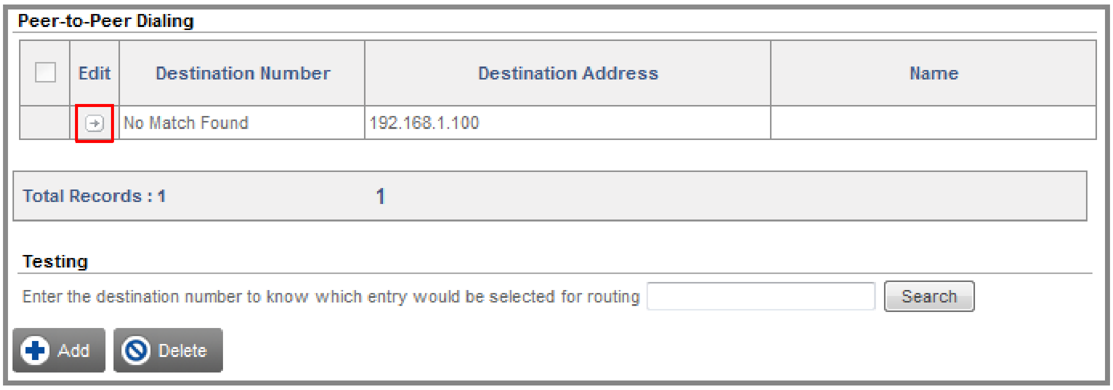
SETU VFX supports following characters.

Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.
+	+ (plus) can be configured as a first character of the Destination Number string in the *SIP Trunk-Destination Port Determination-Destination Number Based* 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.

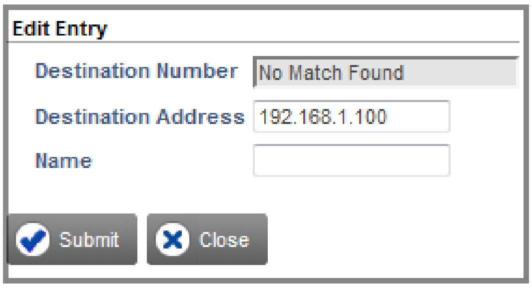
- In the **Destination Address** field, enter the domain name or IP Address where the dialed peer-to-peer number string is to be sent. The Destination Address may have up to 40 characters. Default: Blank.

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

- Enter a name in the **Name** field to identify the number string you configured. It may be the name of your contact or any name you want to assign to the number string. The name may consist of 12 characters (maximum). Default: Blank.
- Click **Submit** to save entries. The window closes.
- The records appear in the Peer-to-Peer Dialing table.
- You may also change the default values of the **No Match Found** entry in the table.
  - To change the default values of the No Match Found entry, under **Edit**, click **Settings** of the **No Match Found** entry.



  - The **Edit Entry** window opens.



  - Change the default values of the **Destination Address** and **Name** as per your requirement.
  - Click **Submit** and close the window.
- To delete an entry in the Peer-to-Peer Dialing Table, select the check box of the entry, and then click **Delete**.
- Close the window to return to the SIP Trunk page.

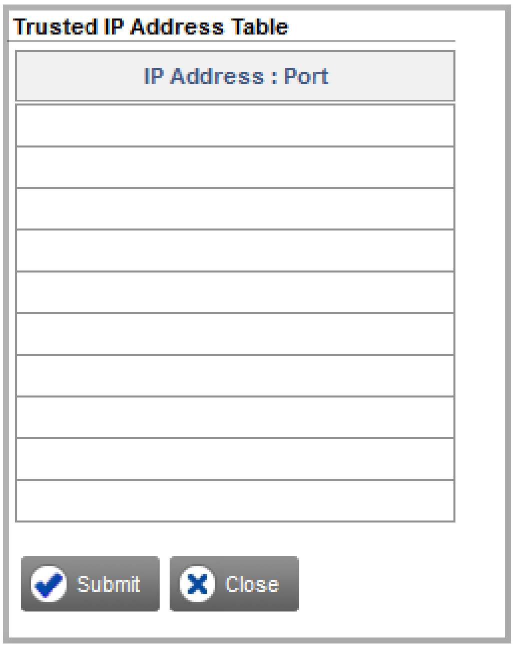
To know more about the Peer-to-Peer application and for detailed instructions, *see* “Peer to Peer Dialing” under *Advanced Settings*.

- Select desired option in the **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 .

    The **Trusted IP Address Table** opens in a new window.



    - 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 authenticates 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.

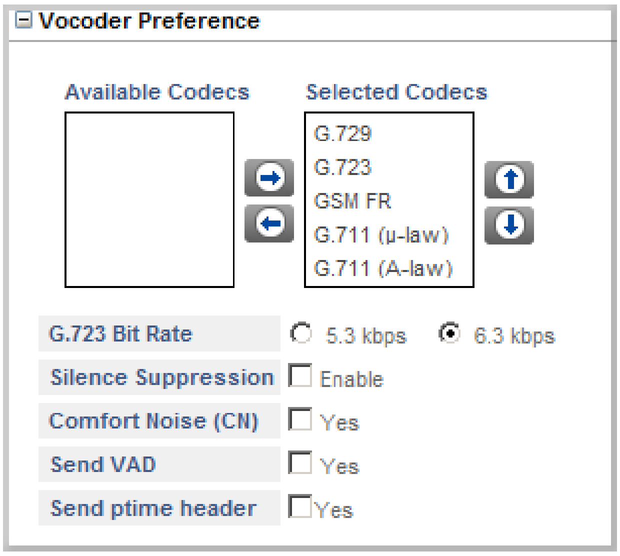
Default: **As per Peer to Peer table**.

- If you set *Allowed IP Address for Incoming SIP Message* to *As per Trusted IP Address table* or *As per Peer to Peer table*, you may also enable the **Digest Authentication**. Incoming calls on this SIP Trunk will be allowed only after the callers authenticate themselves with their User ID and Password. Default: Disabled.

  If you enable Digest Authentication feature on the SIP Trunk, you must configure the Digest Authentication Table. See "Digest Authentication" for more details.

Vocoder Preference

- Click **Vocoder Preference**.



Vocoders are voice codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality.

The Vocoders supported by SETU VGFX appear in the **Selected Codecs** list in the following order of preference:

1. G. 729
2. G.723
3. GSM FR
4. G.711 (µ – Law)
5. G.711 (A - Law)

- You can change the order of preference by moving the desired Vocoders up or down the list. To move a Vocoder up or down the list, do the following:

  - In the **Selected Codecs** list, click the Vocoder you want to move.

  - Click the UP/DOWN ARROW to move the Vocoder to the desired position in the list.

- To remove a Vocoder from the **Selected Codecs** list, click the Vocoder you want to remove, and then click the LEFT ARROW. The Vocoder is moved to the **Available Codecs** list.

  - To move a Vocoder from the **Available Codecs** list to the **Selected Codecs** list, click the Vocoder you want to move, and then click the RIGHT ARROW.

- If you have **G.723** as a Preferred Vocoder, select the **G.723 Bit Rate** as **5.3 Kbps** or **6.3 Kbps**. Default: 6.3kpbs.

  When G.723 is negotiated, the selected Bit Rate will be applied only when sending the RTP packets. When receiving RTP packets from the remote end, both the Bit Rates of G.723 will be accepted.

- Select the **Silence Suppression Enable** check box, if you want SETU VGFX to suppress the *Silence* packets and allow only the *Voice* packets to pass through. Default: Disabled.

Silence Suppression is applicable only when you have selected G.729 as Preferred Vocoder.

- Select the **Comfort Noise (CN)** check box, if you want SETU VGFX to negotiate the Comfort Noise received in the SDP body with the remote peer. Default: Disabled.

- Select the **Send VAD** check box, if you want SETU VGFX to send *VAD = no* in the SDP for any offer and answer. Default: Disabled.

  SETU VGFX will send *VAD = no* only if G.711 (μ – Law) or G.711 (A - Law) codec is present in the SDP offer or answer.

- Select the **Send ptime header** check box, if you want SETU VGFX to add ptime header in the SDP offer and answer. Default: Disabled.

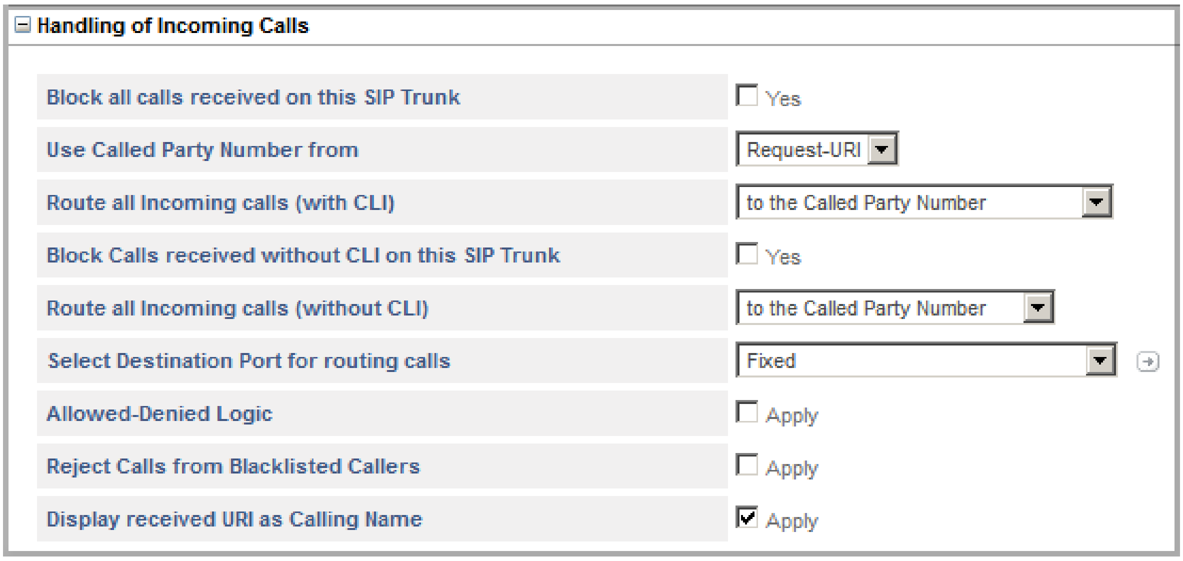
- You must select the **ptime value**, if you have enabled *Send ptime header* check box and have selected codec as—G. 729 and/or G.711 (μ – Law) and/or G.711 (A - Law). You can select from the following:
  - 10 msec
  - 20 msec
  - 30 msec
  - 40 msec

  Default: 20 msec

For Passthrough FAX, SETU VGFX will use the default ptime value (20 msec) only.

Handling of Incoming Calls

- Click **Handling of Incoming Calls**.



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

- By default, SETU VFX 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.

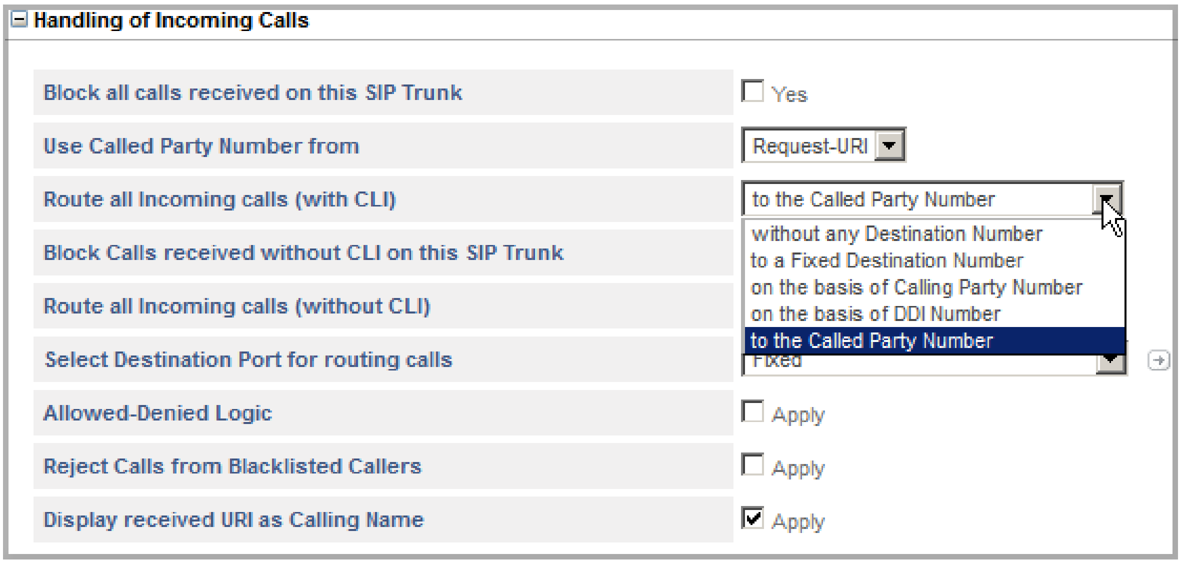
  If you want the system to identify the Called Party Number from the 'To Field' of the INVITE message, in the **Use Called Party Number From** parameter, select the **To Field** option.

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 the Fixed Destination Number
  - on the basis of Calling Party Number
  - on the basis of DDI Number
  - to the Called Party Number

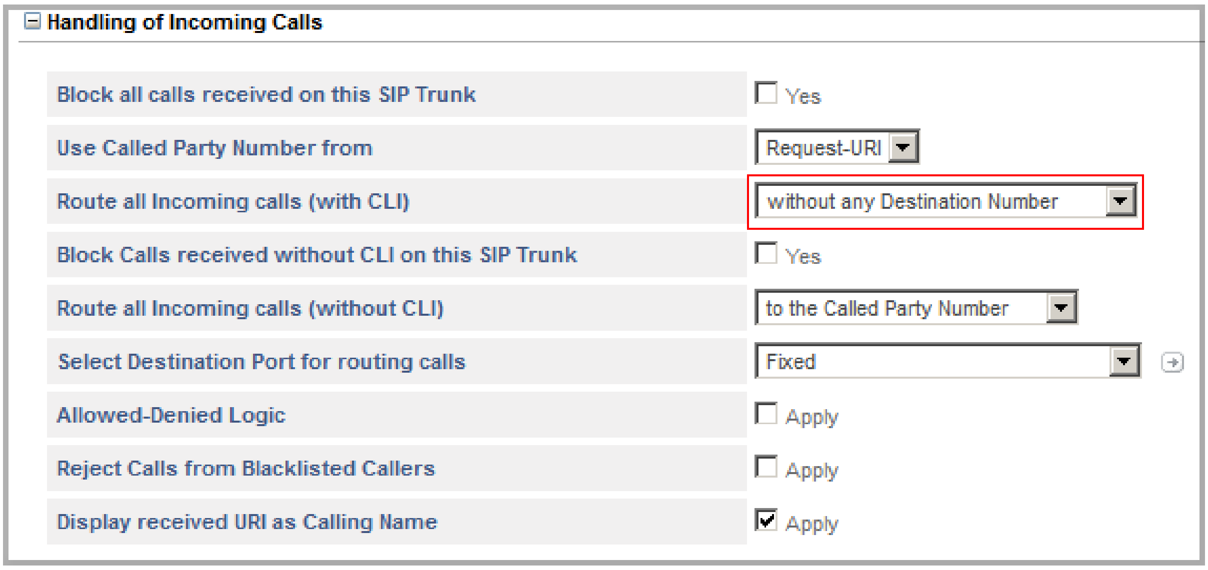
Default: to the Called Party Number.



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

Route Calls without any Destination Number

In this method, all calls received on the SIP Trunk are directly routed to a fixed destination port configured for the SIP Trunk, regardless of the Destination Number.



- To apply this method, in the **Route all incoming calls (with CLI)** list, click **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 configured for the SIP Trunk.

To apply this method, do the following:

- In the **Route all Incoming calls (with CLI)** list, click **to a Fixed Destination Number**.
- In the **Fixed Destination Number** box that appears, 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

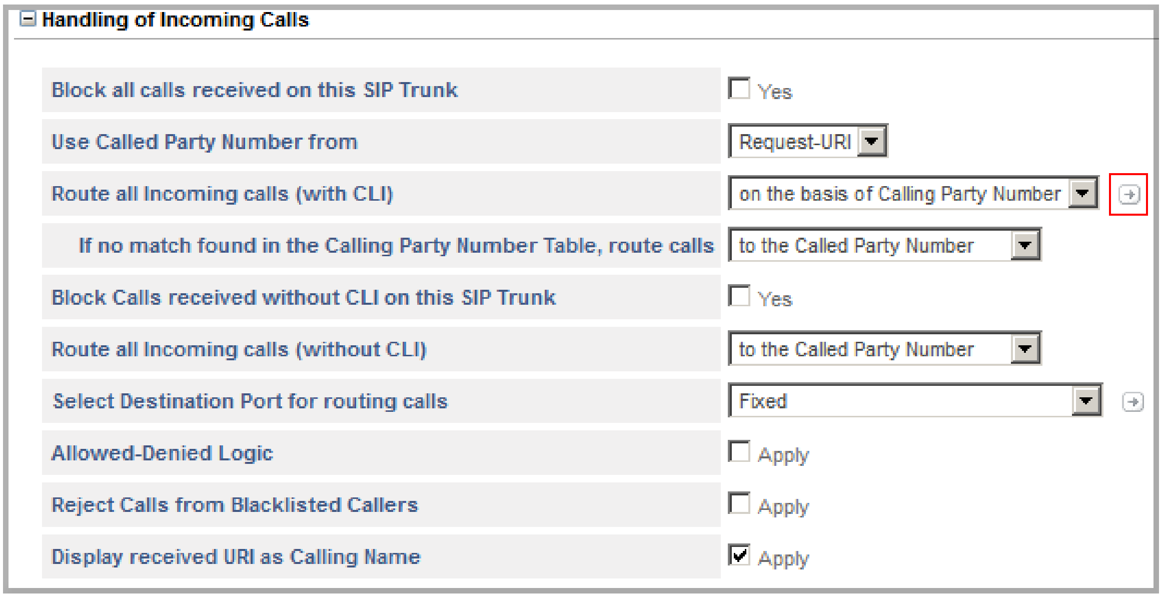
In this method, a call received on the SIP Trunk is routed to a specific number, as per the calling party's number.

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 VFX will match the CLI of the number received with the entries of this table. If a match is found for the number in the table, the call is routed to the destination port.

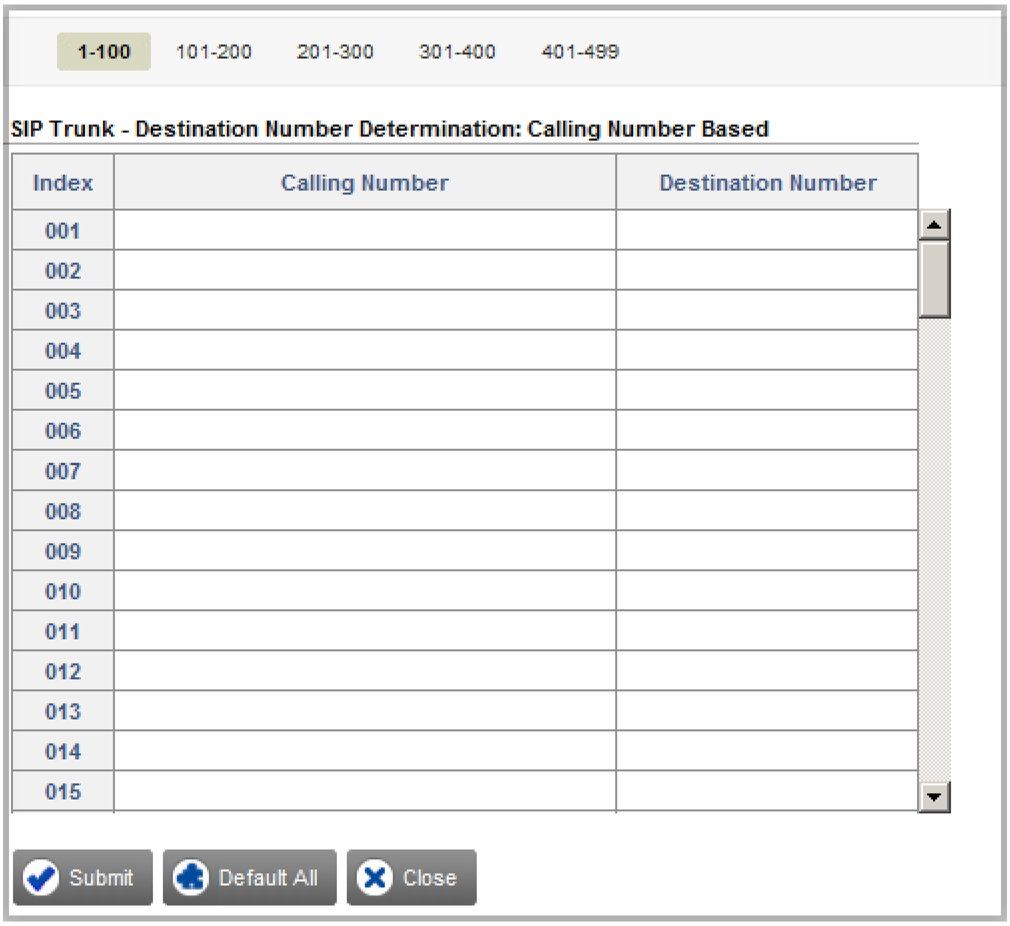
To apply this method, do the following:

- In the **Route all Incoming calls (with CLI)** list, click **on the Basis of Calling Party Number**.

- Click **Settings**



The **Calling Number Based Table** page opens. You can store up to 500 numbers in this table.



- In the **Calling Number** column, enter the calling party numbers. The calling party numbers may consist of a maximum of 24 characters. All ASCII characters are allowed. Default: Blank.

- For each calling party number, enter a corresponding destination number in the **Destination Number** column. 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 SIP Trunk 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 the **If no match found in the Calling Party Number Table, route calls** box, click 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

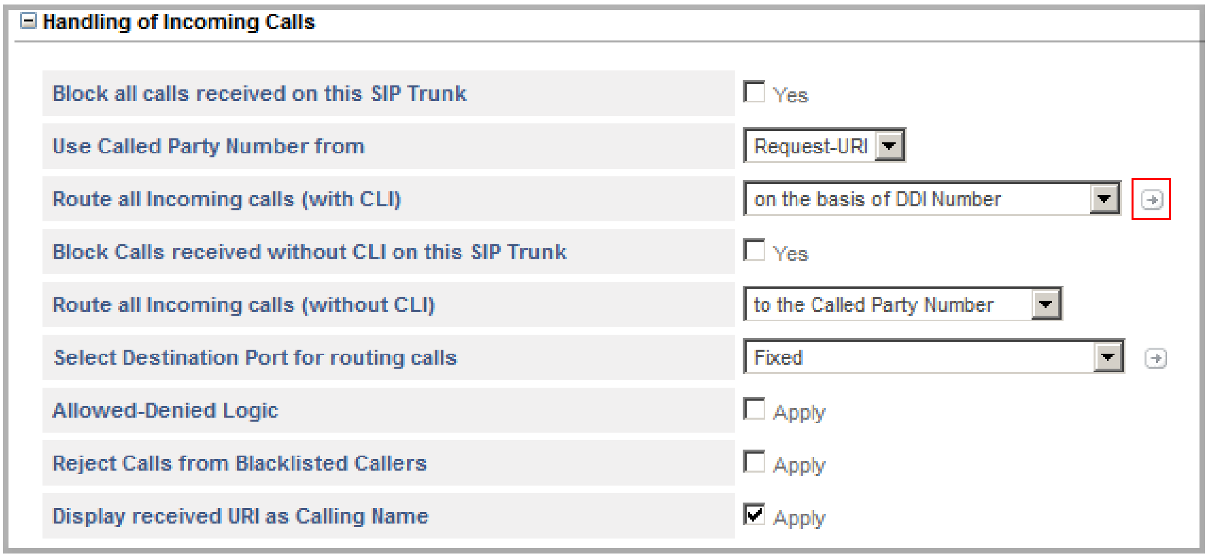
In this method, a call received on the SIP Trunk is routed to a specific number as per the DDI number received in the SIP INVITE message.

You must configure the DDI numbers in the DDI Number Based Table. When there is an incoming call on the SIP Trunk, SETU VFX will match the CLI of the number received with the entries of this table. If a match is found for the number in the table, the call is routed to the destination port.

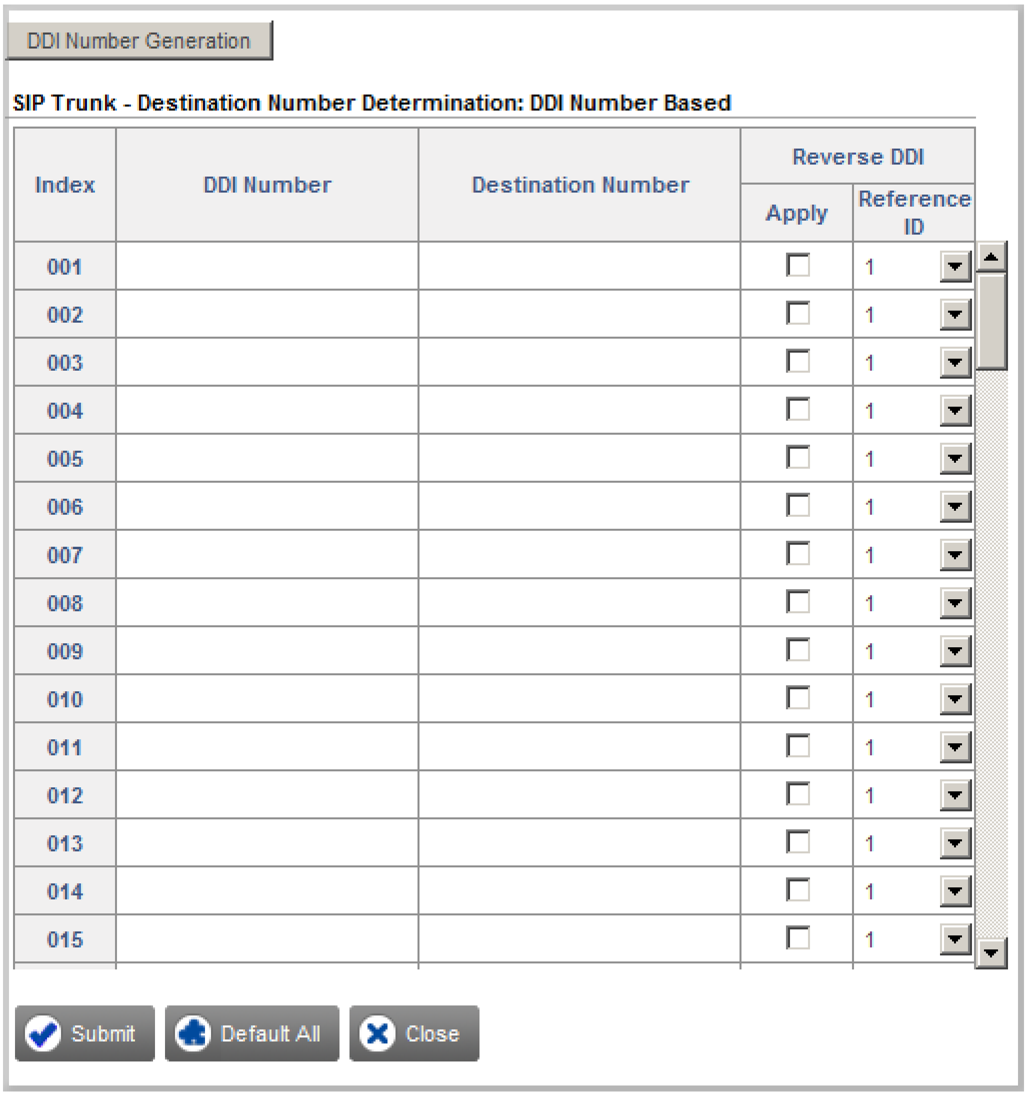
To apply this method, do the following:

- In the **Route all Incoming calls with CLI** list, click **on the Basis of DDI Number**.

- Click **Settings** .



The **DDI Number Based Table** page opens. You can store up to 100 numbers in this table.



DDI Number Generation

SIP Trunk - Destination Number Determination: DDI Number Based

Index	DDI Number	Destination Number	Reverse DDI	
			Apply	Reference ID
001			☐	1
002			☐	1
003			☐	1
004			☐	1
005			☐	1
006			☐	1
007			☐	1
008			☐	1
009			☐	1
010			☐	1
011			☐	1
012			☐	1
013			☐	1
014			☐	1
015			☐	1

Submit Default All Close

- There are two ways to generate the DDI Numbers:
  - Using the **DDI Number Generation** Button to automatically generate the DDI Number Table. See “Configuring SIP-DDI Number Based Table” in *Destination Number Determination* topic for instructions.

    OR

  - Entering each DDI Number manually.
    - In the **DDI Number** column**,** enter the DDI numbers allotted by your service provider. The DDI numbers may consist of a maximum of 24 characters. Characters 0-9, +, * and # are allowed. Default: Blank.
    - In the **Destination Number** column, enter a corresponding destination number for each DDI number. Destination numbers may consist of a maximum of 24 characters. Characters 0 to 9, * and # are allowed. Default: Blank.

- To apply **Reverse DDI** for each number, select the **Reverse DDI Apply** check box and select the **Reference ID** for the number. Default: Apply Reverse DDI is disabled and Reference ID is 1.

- Click **Submit** to save your entries. Close the window to return to the SIP Trunk page.

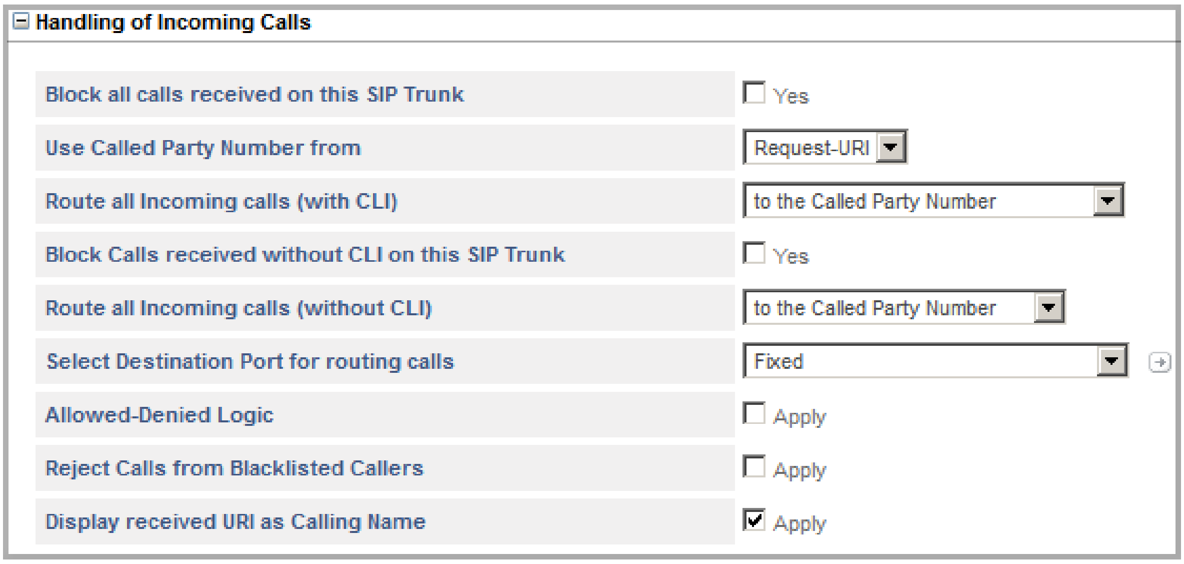
  You can also configure the **DDI Number Based** Table from *Advanced Settings.* See “Destination Number Determination” under *Advanced Settings* for instructions.

Route to the Called Party Number

In this method, a call is routed to a specific number depending upon the called party number received in the SIP ID of the Request URI of the INVITE message.

To apply this method,

- In the **Route all Incoming calls (with CLI)** list, click **to the Called Party Number**.



- If you do not want to route calls without CLI through this SIP Trunk, select the **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”.
  - to the Called Party Number, see “Route to the Called Party Number”.
  - on the basis of DDI Number, see “Route on the basis of DDI Number”.

  Default: to the Called Party Number.

Destination Port Determination

- In **Select the Destination Port for routing calls**, 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 VFX 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" to know more)*

- *SETU VFX does not support SIP to SIP calls.*

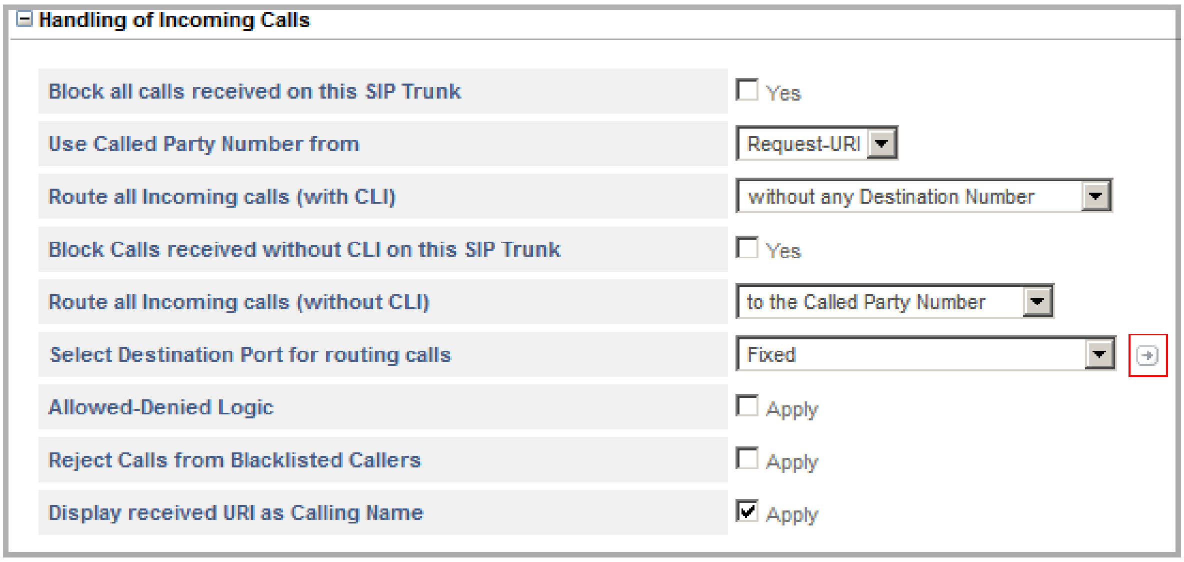
Fixed

In this method, calls received on the SIP Trunk are routed to a fixed destination port, irrespective of the number dialed on the source port.

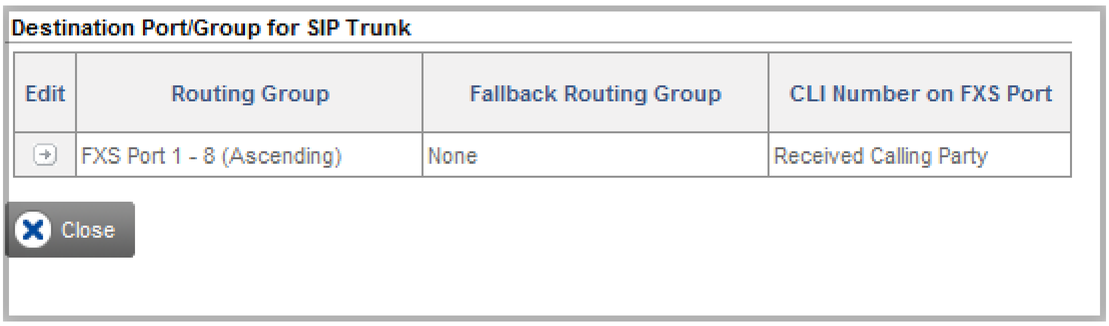
To apply this method, do the following:

- In the **Select Destination Port for routing calls** list, click **Fixed.**

- Click **Settings** .

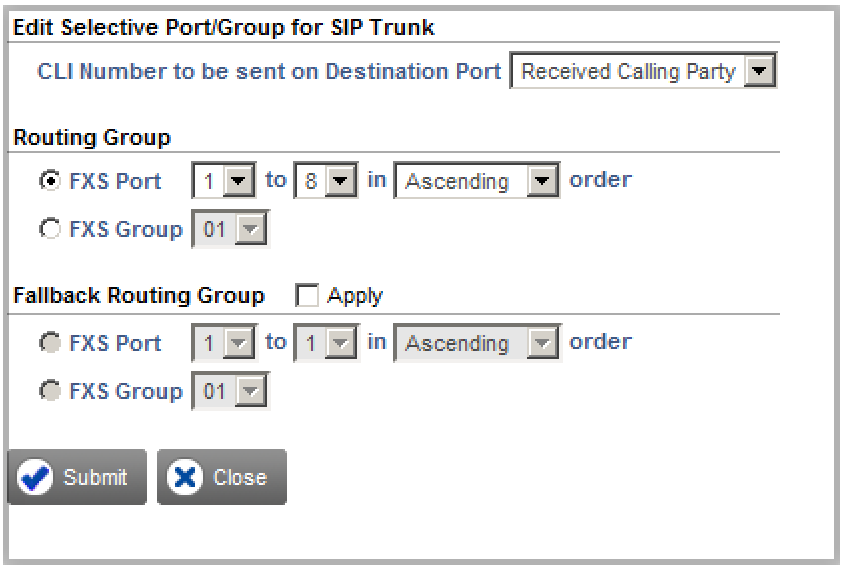


The **Destination Port/ Group for SIP Trunk** window opens.



- To change the default Routing Group and create the Fallback Routing Group, under **Edit**, click **Settings** .

The **Edit Selective Port/ Group for SIP Trunk** window opens.



- Select the **CLI Number to be sent on Destination Port.** You may select from the following options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

CLI Number 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** field, select the order in which the system should hunt for a free member FXS Port to route the call.

      To start hunting from the first to the last member FXS Port, select **Ascending**.

      To start hunting from the last to the first member FXS Port, select **Descending**.

      Default: Ascending.
  - To create a group of *not-sequential* **FXS Ports** as members,
    - Select an **FXS Group**.
    - Select **FXS Group** number. Default:1.
    - Click **Settings** . The **FXS Groups** window opens. Create the FXS Group. See "Group" for detailed instructions.

- Similarly, you can create group of *sequential* and *not-sequential* FXO Ports as members.

- You may create the **Fallback Routing Group**.

  - Select the Fallback Routing Group **Apply** check box.

  - Follow the same instructions for creating *sequential* and *not-sequential* groups for FXS Ports and FXO Ports.

- Click **Submit** to save changes. The **Edit** window closes.

- The changes you made appear in the **Destination Port/Group for SIP Trunk** window. Close this window to return to the main page.

On the basis of Destination Number

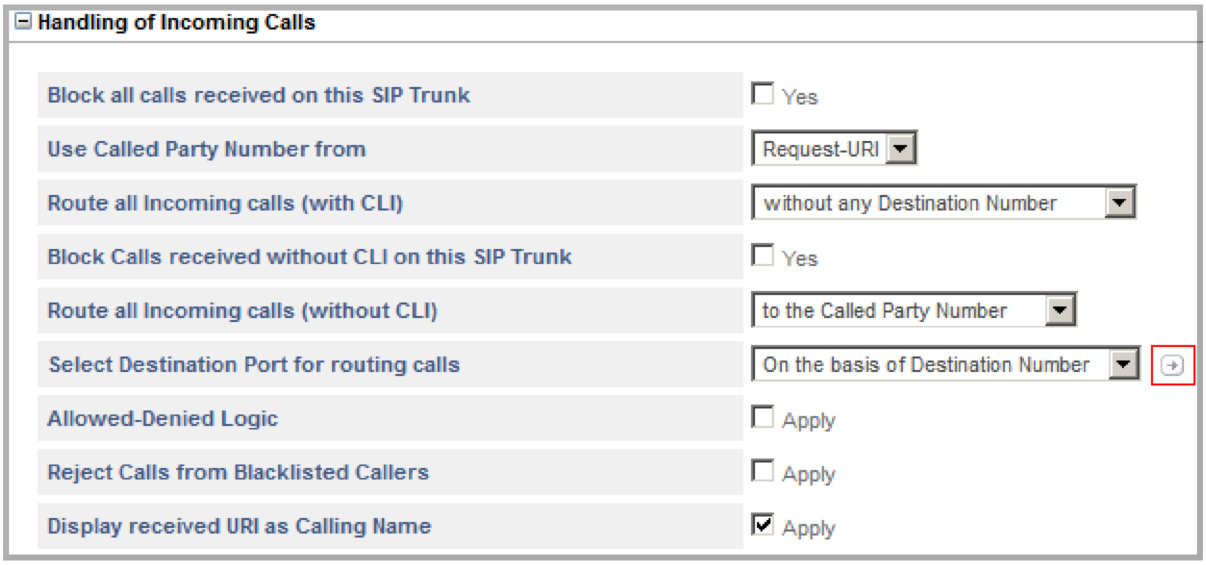
In this method, incoming calls on the source port are routed to the destination port on the basis of the destination number (called party number) dialed by the caller.

You must configure the called party numbers in the **Destination Number Based** Table. SETU VFX 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 port.

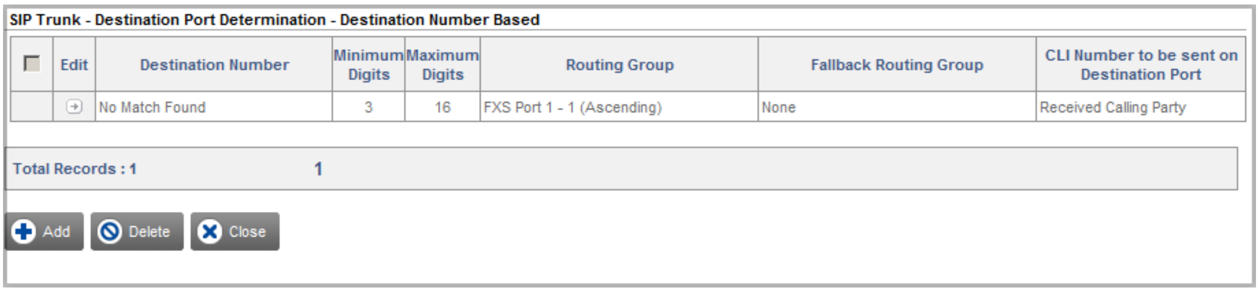
To apply this method, do the following:

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

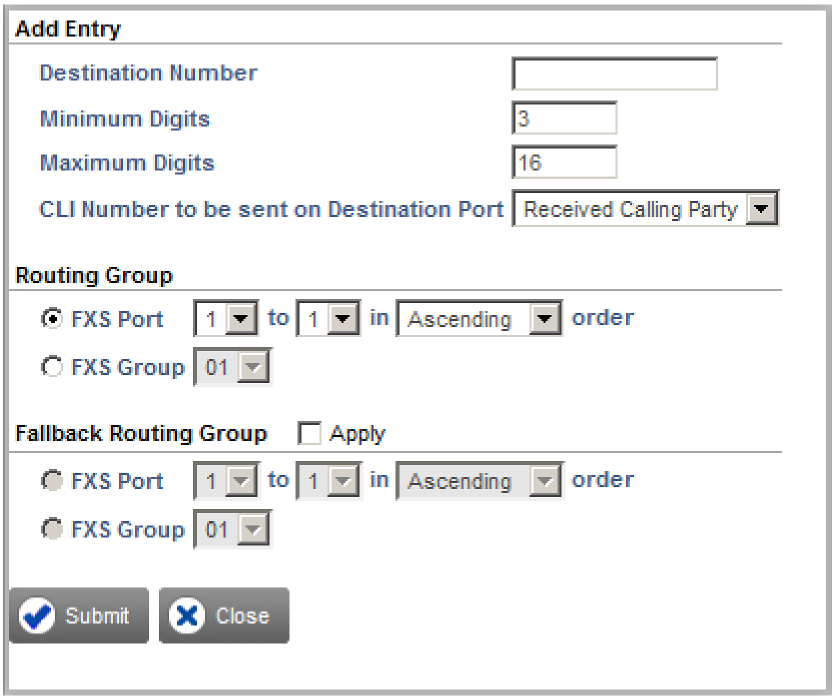
- Click **Settings** .



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



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



- In the **Destination Number** field, enter the number (max. 24 characters) you expect callers to dial. Valid characters: 0 to 9, +, * and #. Default: Blank.

- In the **Minimum Digits** field, enter the minimum number of digits of the destination number that the caller must dial for the system to route the call. Range: 01 to 24. Default: 03.

  If the dialed number string is less than the configured minimum length, the call will be rejected.

- In the **Maximum Digits** field, define the maximum number of digits to be dialed by the user for the system to consider it as End of Dialing for routing the call. The Maximum digit length range is 01 to 24. Default: 16.

  If the number string dialed by the caller exceeds the maximum digit length configured, the system will strip off the extra digits and route the call.

- Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
  - Received Calling Party
  - Received Called Party

Default: Received Calling Party

***CLI Number to be sent on Destination Port** is applicable only when **FXS Port** or **FXS Group** is selected as the Destination Port (Routing Group).*

- 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** field, select the order in which the system should check for a free member FXS Port to route the call.

          To start hunting from the first to the last member FXS Port, select **Ascending**.

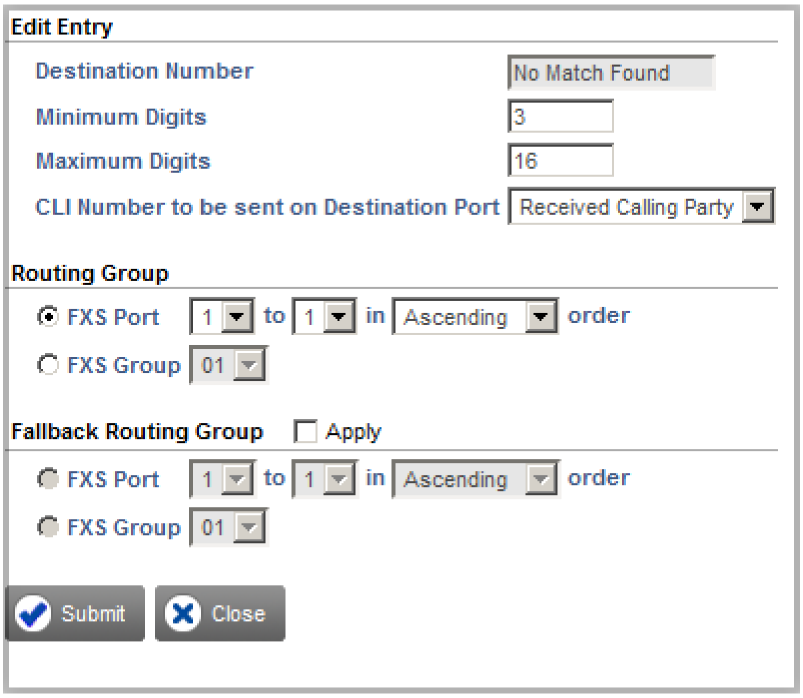
          To start hunting from the last to the first member FXS Port, select **Descending**.

          Default: Ascending.
    - 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 Groups** window opens. Create the FXS Group. See "Group" for detailed instructions.
    - Similarly, you can create group of *sequential* and *not-sequential* FXO Ports as members.
- You may create the **Fallback Routing Group**. To do this,
    - Select the **Apply** check box.
    - Follow the same instructions for creating *sequential* and *not-sequential* groups, for FXS Ports and FXO 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** window.
- 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, that is, the **No Match Found** entry.

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

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

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group**.

- Click **Submit** and close the window.

- Follow the same steps as above to add another entry to this table.

- To delete an entry, select the check box and click **Delete** button.

- Close the window if you have finished adding/editing entries.

  You can also configure the **Destination Number Based** Table from *Advanced Settings*. See “Destination Port Determination” under Advanced Settings.

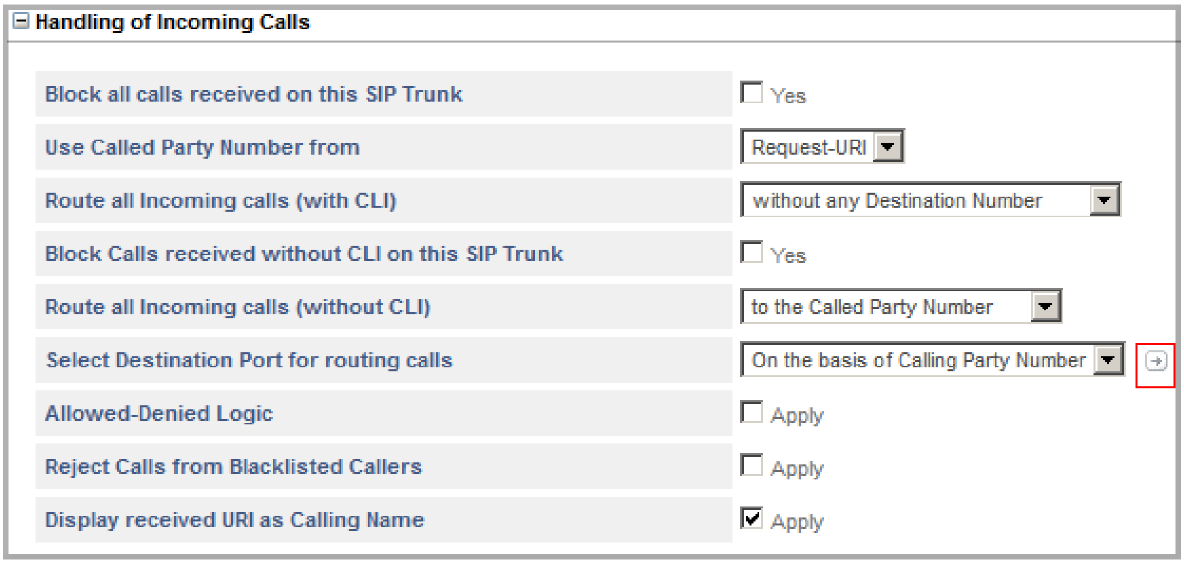
On the basis of Calling Party Number

In this method, incoming calls on the SIP Trunk will be routed to a specific port as per the calling party's number.

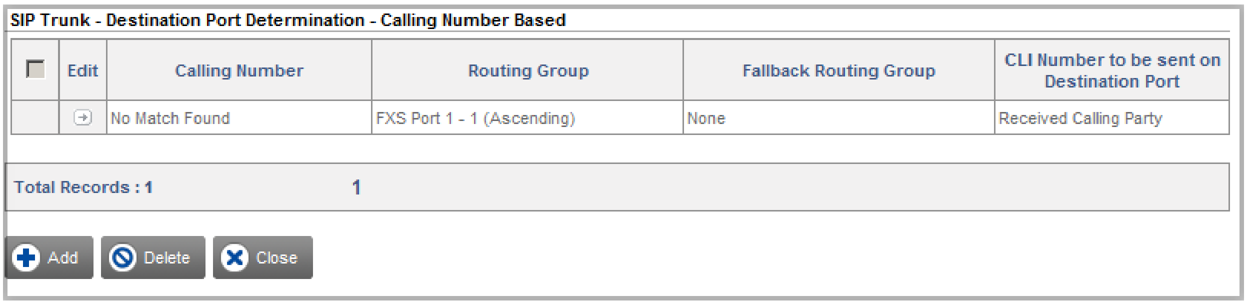
To apply this method, do the following:

- In the **Select Destination Port for routing calls** list, click **on the basis of Calling Party Number**.

- Click **Settings** .



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

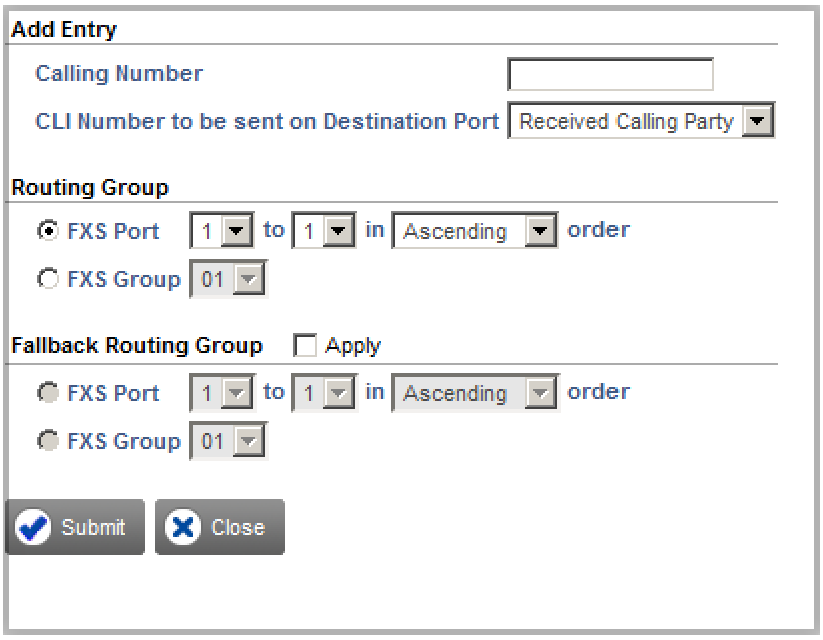

SIP Trunk - Destination Port Determination - Calling Number Based

	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 500 entries



- In the **Calling Number** field**,** enter numbers (max. 24 characters) from which you expect calls to be received. All ASCII characters are allowed. Default: Blank.

- Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

***CLI Number to be sent on Destination Port** is applicable only when **FXS Port** or **FXS Group** is selected as the Destination Port (Routing Group).*

- 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** field, select the order in which the system should hunt for a free member FXS Port to route the call.

      To start hunting from the first to the last member FXS Port, select **Ascending**.

      To start hunting from the last to the first member FXS Port, select **Descending**.

      Default: Ascending.
  - To create a group of *not-sequential* **FXS Ports** as members,
    - Select **FXS Group**.

        - Select **FXS Group** number. Default:1.

        - Click **Settings** . The **FXS Groups** window opens. Create the FXS Group. See “Group” for detailed instructions.

    - Similarly, you can create group of *sequential* and *not-sequential* FXO Ports as members.

- You may create the **Fallback Routing Group**. To do this,

    - Select the **Apply** check box.

    - Follow the same instructions for creating *sequential* and *not-sequential* groups, for FXS Ports and FXO Ports.

    - Click **Submit** to save changes. The **Add Entry** window closes.

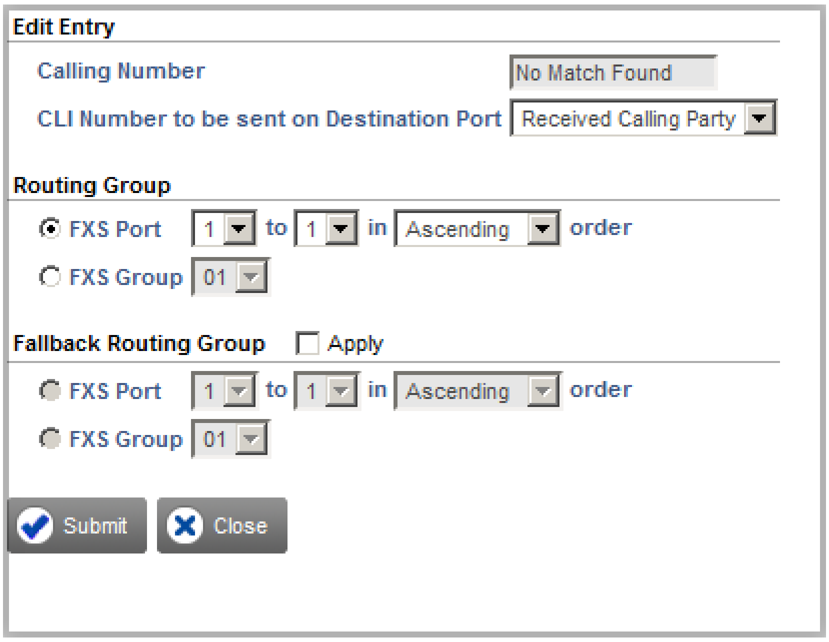
- The entry you added appears in the **SIP Trunk- Destination Port Determination - Calling Number Based** window.

- 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 in the Calling Number Based Table, that is, the **No Match Found** entry.

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

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

    - The **Edit Entry** window opens.



    - Create the **Routing Group** and **Fallback Routing Group**.

- Click **Submit** and close the window.

- Follow the same steps as above to add another entry to this table.
- To delete an entry, select the check box and click **Delete** button.
- Close the window if you have finished adding/editing entries.

  You can also configure the **Calling Number Based** Table from *Advanced Settings*. See "Destination Port Determination" under Advanced Settings.

Allowed - Denied Logic (Toll Control)

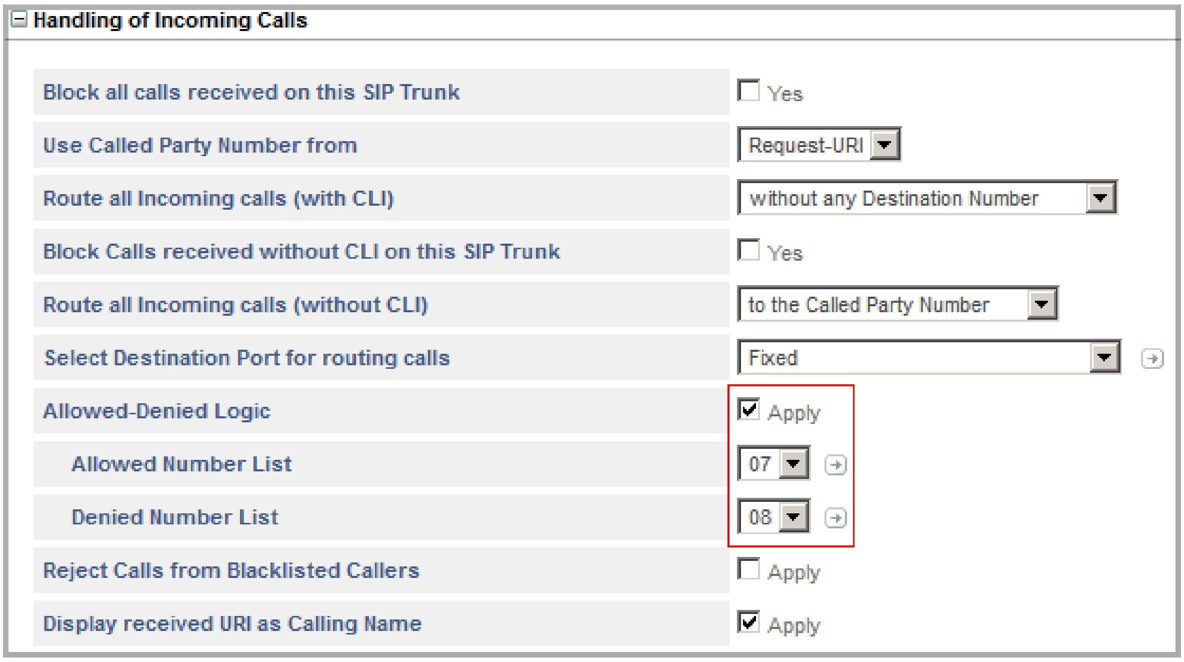
You can apply the Allowed-Denied logic on the SIP Trunk (source port) if you want to allow or restrict the dialing of particular numbers. You can use this feature for Toll Control.

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

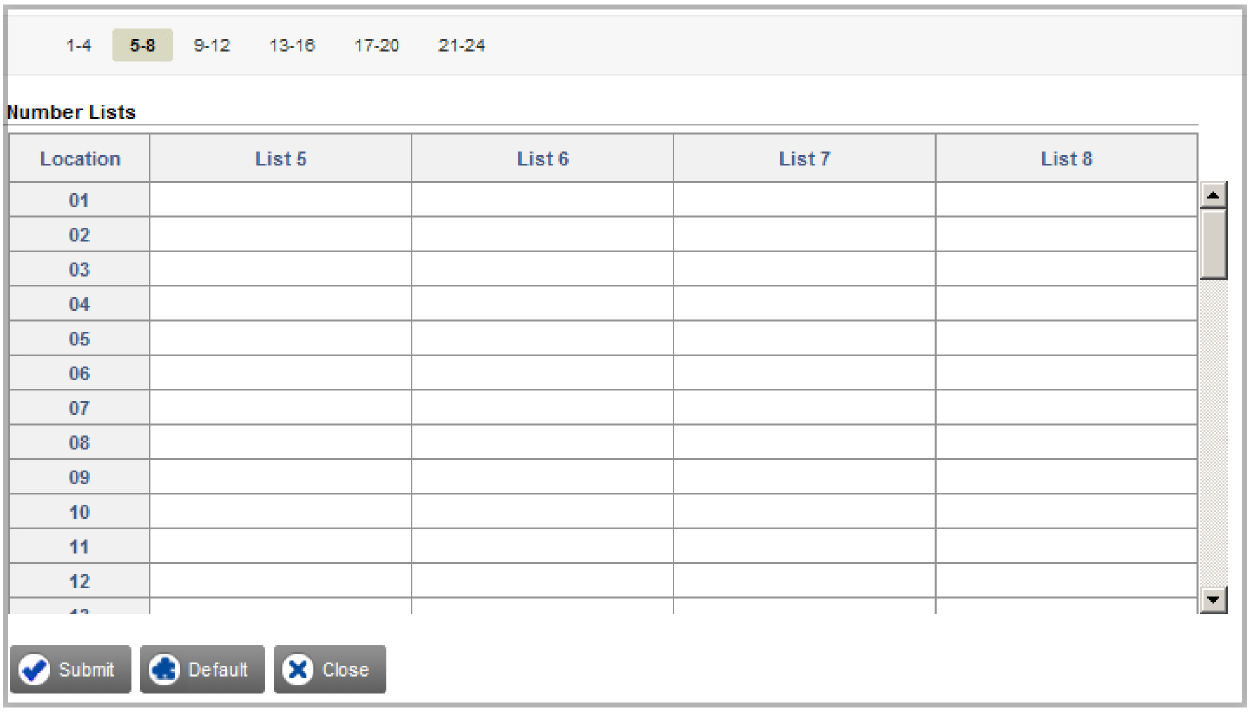
- **Allowed Numbers List:** This is the list of numbers that are allowed to be dialed out by the caller on the SIP Trunk.
- **Denied Numbers List:** This list contains the numbers that are denied to be dialed out by the caller on the SIP Trunk.

To apply Allowed - Denied Logic on the SIP Trunk,

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



- To configure the **Allowed Number List**, click **Settings** .



  - The Number List page opens in a new window.
  - By default, Number List 7 is assigned as Allowed Number List.
  - Enter the numbers you want the system to allow to be dialed in this list.
  - Click **Submit** to save the entries and close the window.

- To configure the **Denied Number List**, click **Settings** .

  - The Number List page opens in a new window.
  - By default, Number List 8 is assigned as Denied Number LIst.
  - Enter the numbers you want the system to restrict from being dialed out in this list.
  - Click **Submit** to save the entries and close the window.

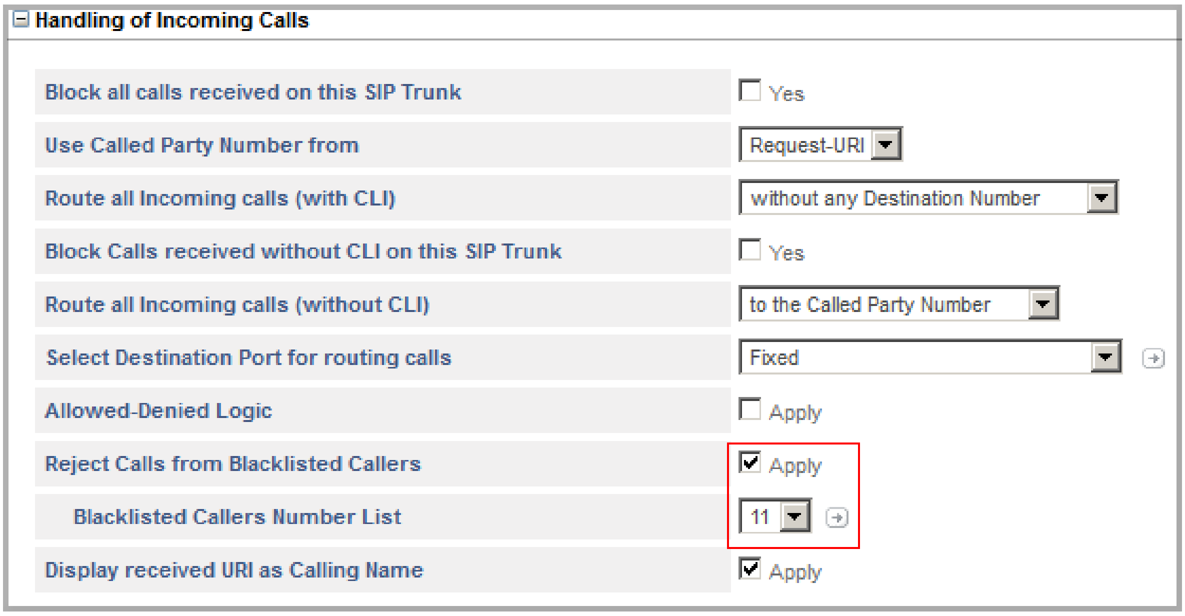
- You may configure a different Number List as Allowed and Denied List. See “Allowed-Denied Logic” under “Number Lists”.

Black Listed Caller

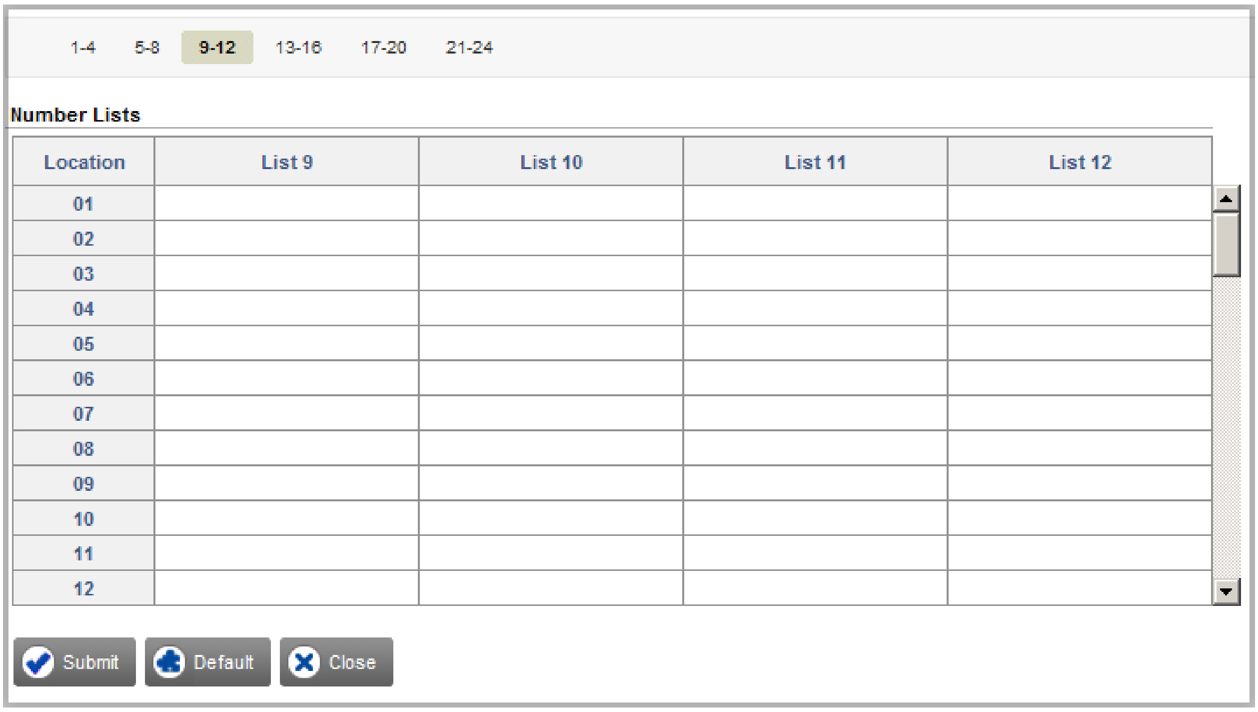
With the Black Listed Callers feature you can block incoming calls from specific addresses/numbers on SIP Trunks. Thus all incoming calls from the numbers you have 'blacklisted' will be automatically rejected by SETU VFX.

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** .
  - The Number List page opens in a new window.



  - By default, Number List 11 is assigned as Black Listed Callers List.
  - Enter the numbers of unwanted callers in this list.

Make sure you have configured the full SIP URI (for example: 12345@abc.com) of the unwanted callers in the Blacklisted Callers Number List.

Display Received URI as Calling Name

- Keep the **Display Received URI as Calling Name** check box enabled.

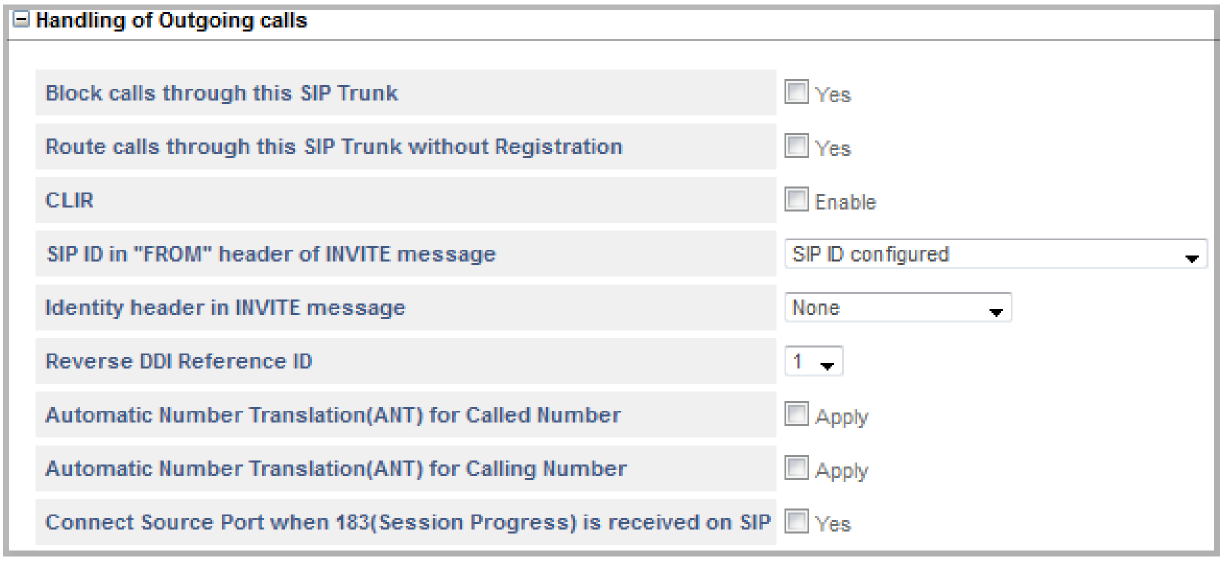
  When Name is received in the "FROM" header for incoming call on the SIP Trunk, SETU VFX will display name and received URI as calling name. When Name is not received, SETU VFX will display only the received URI as calling name.

  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** to save the entries and close the window to return to the main page.

Handling of Outgoing Calls

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



- 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.
- Select an option you want to send as **SIP ID in "FROM" header of INVITE message**. You may select:

- SIP ID configured
- Caller ID received on Source Port
- Caller ID after applying Reverse DDI logic

Default: SIP ID configured

- SETU VFX also offers flexible options for sending Identity header in INVITE message during an outgoing call. Select the desired option according to the Identity header supported by your service provider. You may select:
  - None
  - P-Preferred Identity
  - P-Asserted Identity

  Default: None.

*If you have enabled **CLIR** and **Identity header in INVITE message** is configured other than None, then SETU VFX will add **Privacy = ID** header in the INVITE message during an outgoing call from the SIP Trunk.*

  - If you select *P-Preferred Identity* or *P-Asserted Identity* in **Identity header in INVITE message**, then select the desired option for sending **SIP ID in Identity header of INVITE message** during an outgoing call. You may select:
    - Send SIP ID configured,
    - Send Caller ID received on Source Port,
    - Send Caller ID after applying Reverse DDI logic,
    - Send Fixed Number.

    Default: Send SIP ID configured.

    - If you select *Caller ID after applying Reverse DDI logic*, SETU VFX allows you to configure the desired option for **If no match found using Reverse DDI logic**. You may select:
      - SIP ID configured
      - Caller ID received on Source Port
      - Fixed Number.

      Default: SIP ID configured

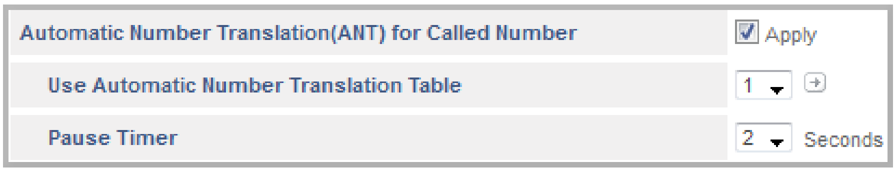
    - If you select *Send Fixed Number* option for **SIP ID in "FROM" header of INVITE message** or **If no match found using Reverse DDI logic**, you must configure the Fixed Number. The Fixed Number can be a maximum of 24 characters. Characters 0-9, +, * # and dot(.) are allowed. Default: Blank.

- Select the **Reverse DDI Reference ID**, if you have selected *Caller ID after applying Reverse DDI logic* as **SIP ID in "FROM" header of INVITE message** and/or *Send Caller ID after applying Reverse DDI logic* as **SIP ID in Identity header of INVITE message**.

  SETU VFX 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 VFX 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.

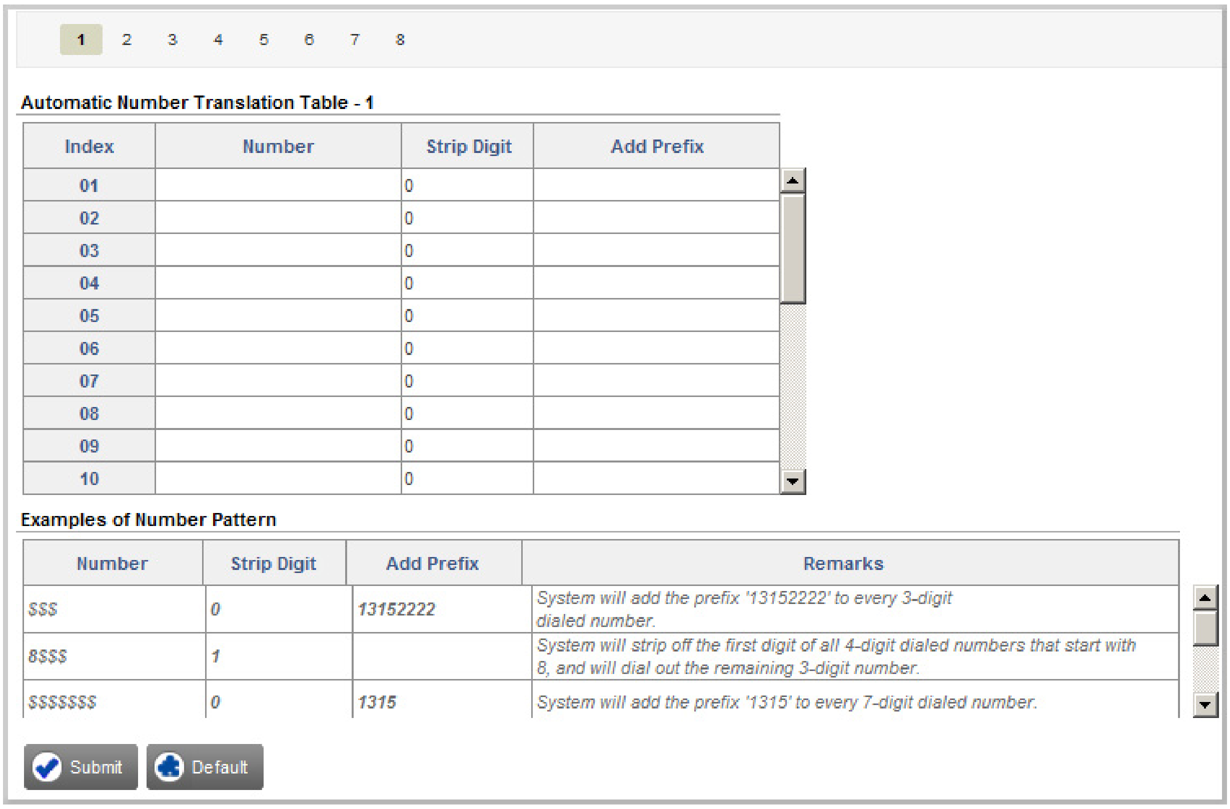
- To apply ANT logic on the Called Numbers, click the **Automatic Number Translation (ANT) for Called Number** check box. Default: Disabled.



- In the **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 page will open in a new window.



  - You may configure the default Automatic Number Translation Table 1 or any other Table (2 to 8). See "Automatic Number Translation (ANT)" to configure the ANT Table.

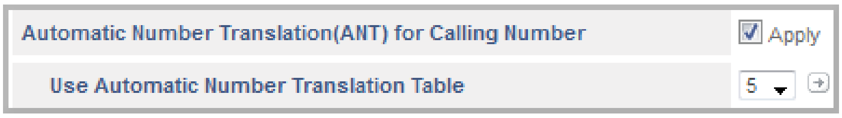
  - Click **Submit** to save the ANT Table and close the window.

  - Return to ANT parameter and assign the ANT Table you configured.

  - Click **Submit** to apply List.

- Configure the **Pause Timer**, if you have configured ^ (Pause) in the Add Prefix column of the ANT Table. The valid range of the Pause Timer is 1 to 9 seconds. Default: 2 seconds.

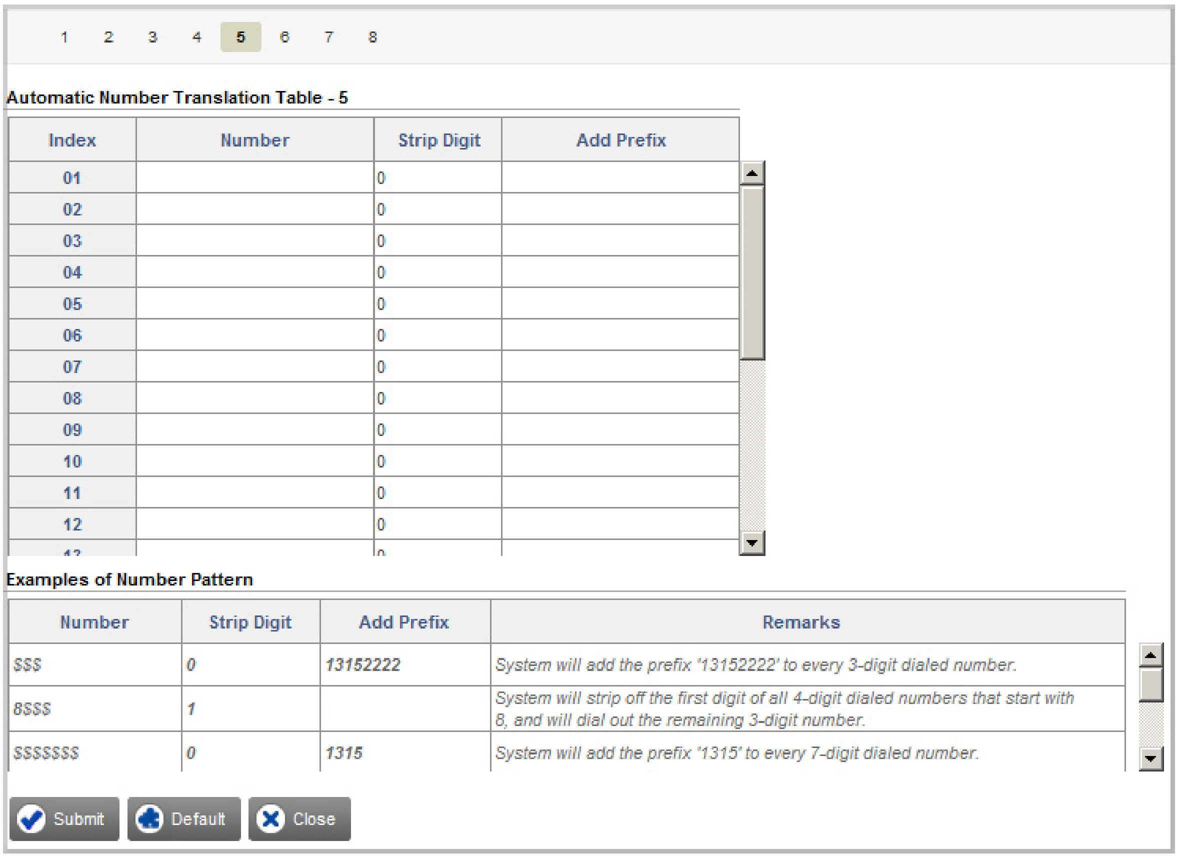
- To apply ANT logic on the Calling Numbers, click the **Automatic Number Translation (ANT) for Calling Number** check box. Default: Disabled.



- 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 page will open in a new window.



  - 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.

- 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 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.

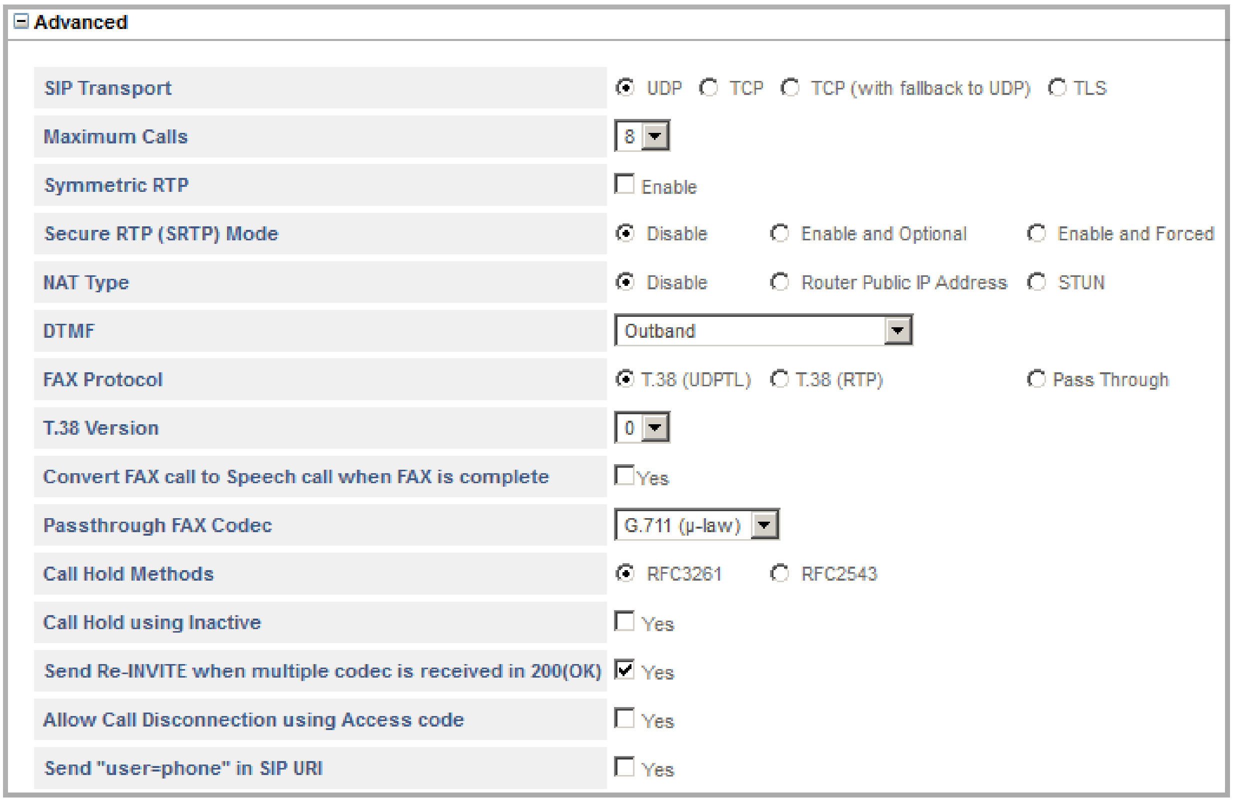
By connecting the Source Port with the Destination Port immediately after the number is dialed, the caller can know the state of the call; if the called party is busy, not responding, not reachable or is rejecting the call.

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.

Advanced

- Click **Advanced** and configure the following parameters:



- 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):** TCP is used for outgoing messages. However, if the TCP connection fails, the system will attempt to send the 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** "System Parameters" in the page.*
- *To use TLS, you must enable **SIP over TLS** in "System Parameters".*

- In the **Maximum Calls** field, configure the number of simultaneous calls you want to allow on this SIP Trunk. Default: 8.

  This depends on the number of simultaneous calls allowed by the ITSP with whom you have subscribed this SIP Trunk.

  The SETU VFX supports 8 simultaneous calls. Ask your ITSP about the number of simultaneous SIP calls supported on this SIP Trunk.

- If you want the system to send RTP packets to the original IP and Port from where RTP packets are received, by ignoring the contact information received in SDP, enable the **Symmetric RTP** check box. Default: Disabled.

- Select the appropriate **Secure RTP (SRTP)** mode from the following options:

  - **Disable:** SRTP will not be used.
  - **Enable and optional:** Either RTP or SRTP will be used. If you select this mode, you must select the SRTP Media Type. You can select AVP or SAVP. Default: AVP.
  - **Enable and forced:** Only SRTP will be used.

  Default: Disabled.

- When the system is installed behind a NAT Router, select the specific NAT traversal mechanism to be used as **NAT Type**. Default: Disabled.

  - Select **Router's IP Address,** if your SETU VFX is located behind the NAT router (any type).

    Make sure you disable Outbound Proxy on SIP Trunk and configure 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 configure the STUN Server Address and port in "System Parameters".

- Select the appropriate **DTMF** sending/receiving mechanism that is compatible with the DTMF sending/receiving mechanism of your ITSP or remote peer. SETU VFX 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 is 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 is negotiated, SIP INFO will have priority over Outband.

  Default: Outband

- To send and receive the Fax over IP, select the desired **Fax Protocol**:

  - **T.38(UDPTL):** If you select this option, the device you are sending the fax to must also support this protocol.

  - **T.38(RTP):** 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 (UDPTL).

- If you have selected *T.38(UDPTL)* or *T.38(RTP)* as Fax Protocol, you must select an appropriate **T.38 Version** that is compatible with your ITSP proxy server/remote peer. You may select:
  - 0
  - 1
  - 2

  Default: 0

- Select the **Convert FAX call to Speech call when FAX is complete** check box, if you want SETU VFX to convert the fax call to a speech (voice) call after the fax complete event is received. Default: Disabled.

- If you have selected *Pass Through* as Fax Protocol, you must select an appropriate **Passthrough FAX Codec** that is compatible with your ITSP proxy server/remote peer. You may select the Codec as:
  - G.711(μ-law)
  - G.711 (A-law)

  Default: G.711 (μ-law)

- Select an appropriate **Call Hold Method** that is compatible with your ITSP proxy server/remote peer. You may select:
  - RFC 2543
  - RFC 3261

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.

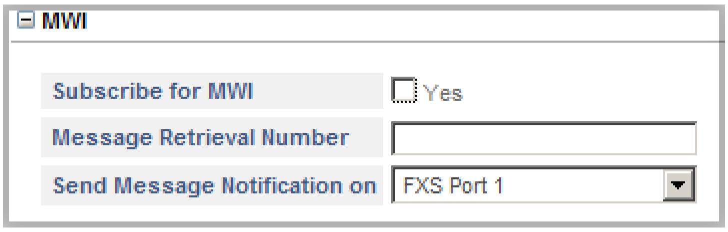
- Clear the **Send Re-INVITE when multiple codec is received in 200(OK)** check box, if you do not want SETU VFX to send Re-INVITE message and use only the first codec from the multiple codec received in 200(OK). Default: Enabled.

- Select the **Allow Call Disconnection using Access code,** if you want to enable the feature Disconnect Call using Access Code on the SIP Trunk. See "Disconnecting a Call using Access Code".

- Select **Send "user=phone" in SIP URI** check box, if you want SETU VFX to add user=phone in the Request URI/From/To header of the INVITE message. Default: Disabled.

  SETU VFX will send user=phone in SIP URI, only if the SIP ID is numeric.

Message Wait Indication (MWI)[13]



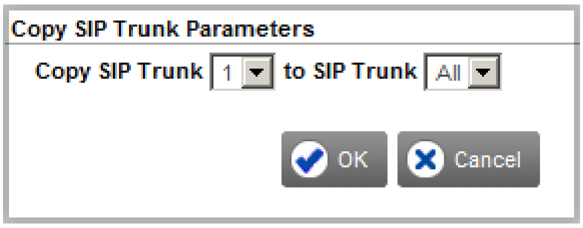
Message Wait Indication parameters are applicable only when SIP Trunk is configured as Proxy or Gateway.

- 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. 0-9, * and # are allowed. 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.
  - If you have completed the configuration of SIP Trunk 1, click **Submit** to save settings.
- To configure the next SIP Trunk, click the SIP Trunk number tab and follow the same instructions as given earlier.

13. *Applicable only for VoIP-FXS Gateway.*

Copy SIP Trunk Settings

- 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.



  - In the **Copy SIP Trunk** box, select the number of the port you want to copy settings *From*. In the **to SIP Trunk** box, select the number of the port you want to copy the settings *To*.
  - Click the **OK** button and close the window.
- 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 VFX 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.

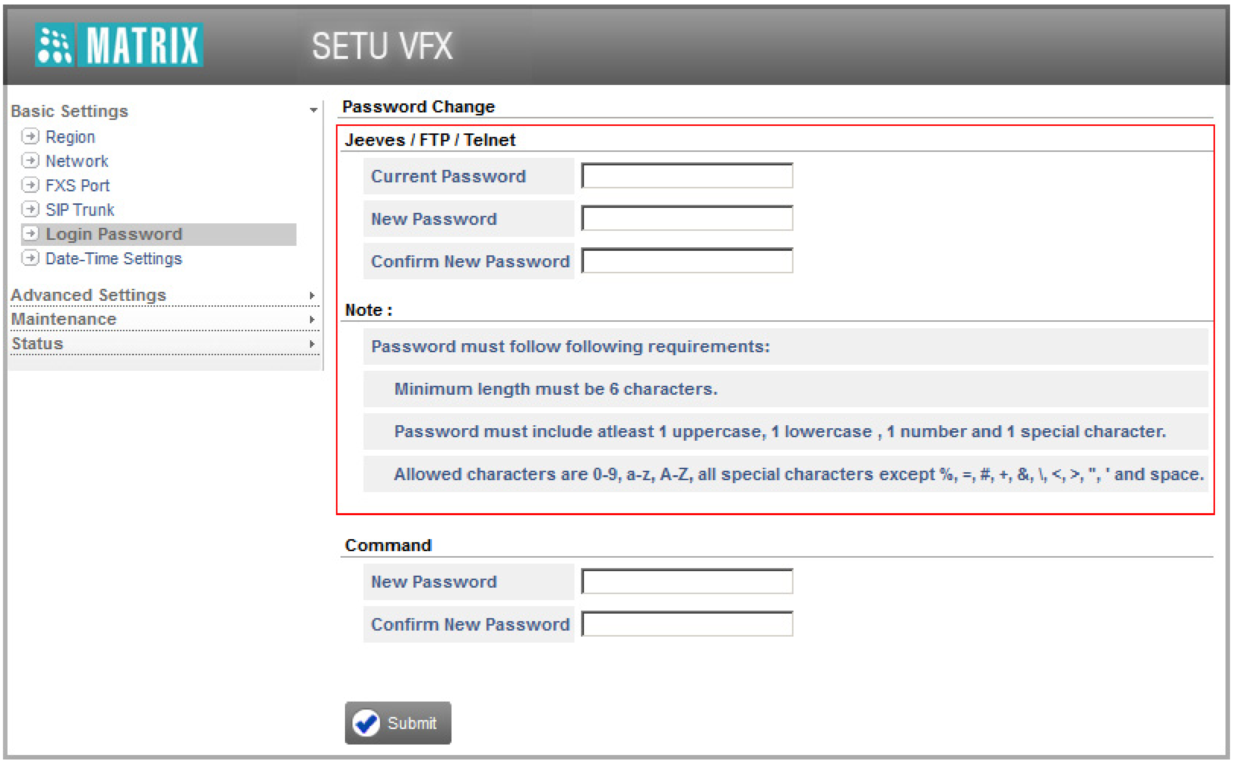
The Jeeves Password must fulfill the following requirements.

- 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.**



Under **Jeeves/FTP/Telnet**,

- Enter **Current Password.**

- **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

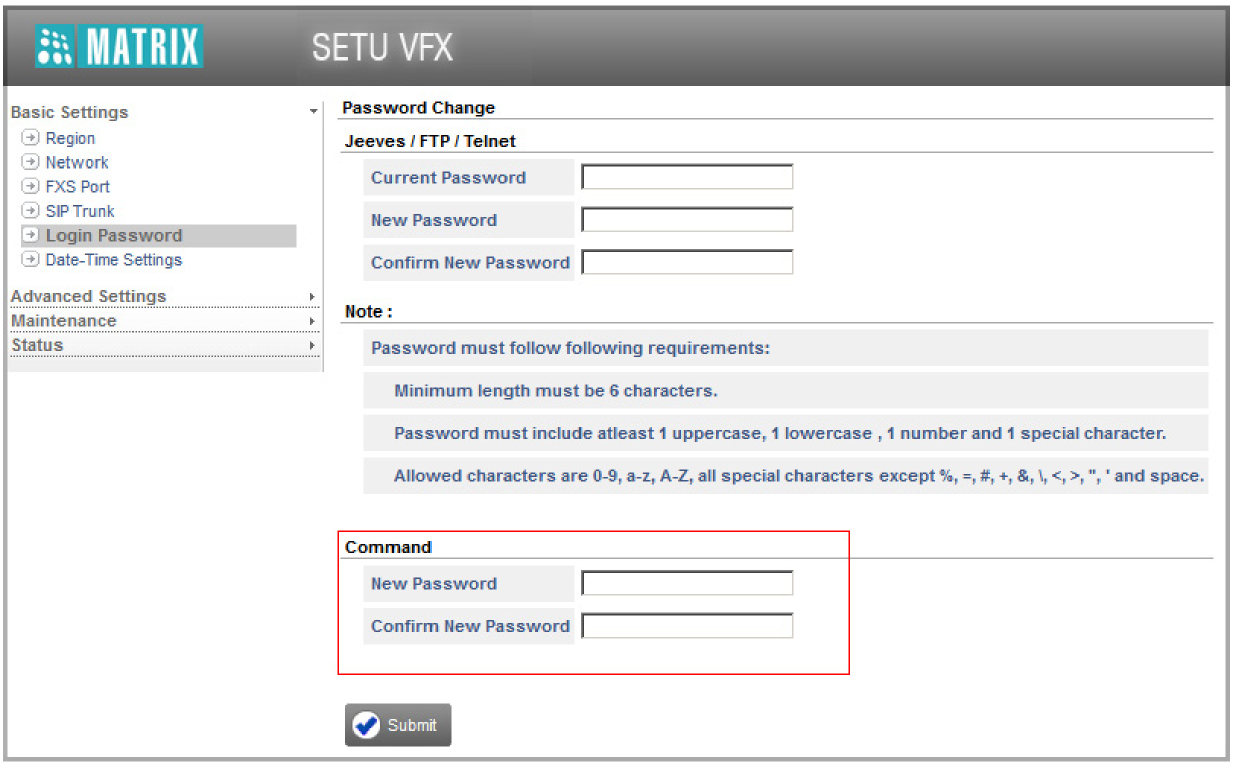
To configure system by dialing System Commands, you must enter the SE mode using the Command Password. This Password must not be less than 4 digits and can be of upto 8 digits. Digits 0-9 are allowed. The default Command Password is **1234.** You may change this Password using Jeeves.

To change the Command Password:

- Log into Jeeves.

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

- Click **Login Password.**



Under **Command**,

- Enter the new password in the **New Password** field.
- Type the new password again for confirmation in the **Confirm New Password** field.
- 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/Command Password (1234) and are unable to recall or locate it, you must restore the default Jeeves/Command Password.

Restoring Default Login Password

Restoring the Default Jeeves/Command Password requires you to change the Jumper Settings on the PCB.

To do this,

- Switch off the power supply
- Remove the top cover of the enclosure.
- Locate and change the position of the Jumper **J3** from ***BC*** to ***AB***.
- Replace the cover of the enclosure.
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system and remove the enclosure cover.
- Change the Jumper position from ***AB*** to the original position ***BC***.
- Replace the enclosure cover.
- Switch ON the system.

The Jeeves/Command Password will be restored to the default value, **1234.**

When you change the jumper positions to restore default Jeeves/Command Password (1234),a few other parameters will also be set to default. See *"Restoring Default Settings by changing the Jumper Position"* *for details.*

Date and Time Settings

Real Time Clock

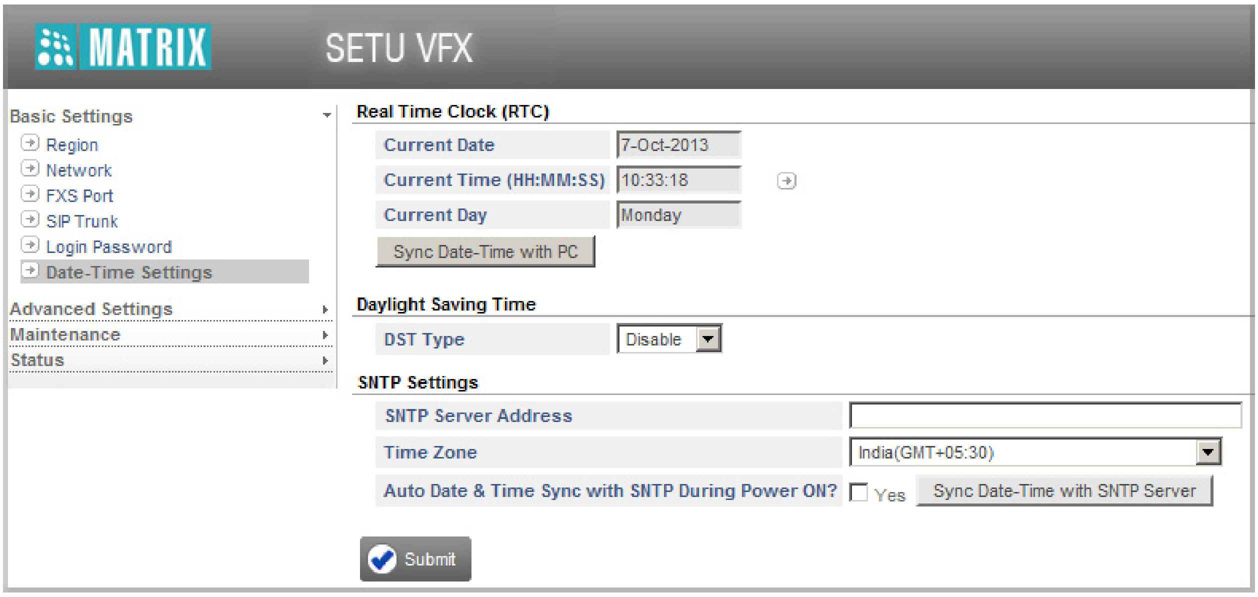
SETU VFX has its own Real Time Clock (RTC) to store date and time. When you select the Region, The RTC parameters will be set automatically.

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.

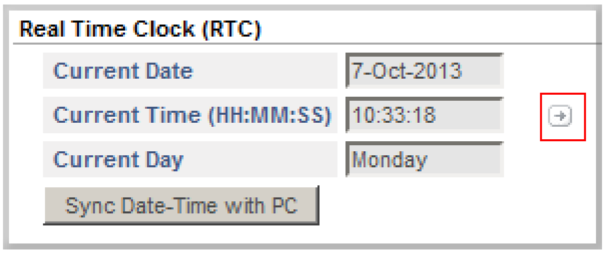
To set the Real Time Clock,

- Under **Basic Settings**, click the **Date-Time Settings** link.

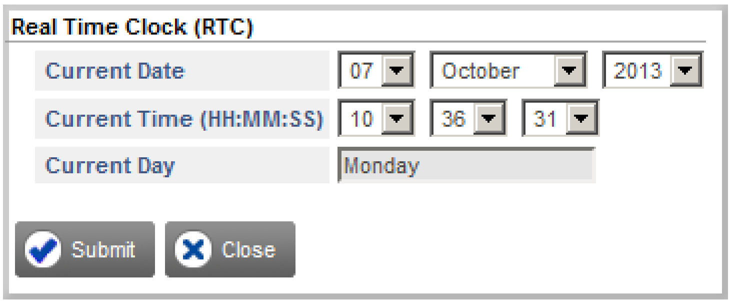
- The **Real Time Clock** parameters appear on your screen.



- Under **Real Time Clock (RTC)**, click **Settings** of the **Current Time (HH:MM:SS)**.



- 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[14] use it, though the start and end dates of DST vary by location and year.

SETU VFX supports Daylight Saving Time adjustment to enable you to set the Date and Time[15] of SETU VFX forward and backward according to the DST convention followed in your country.

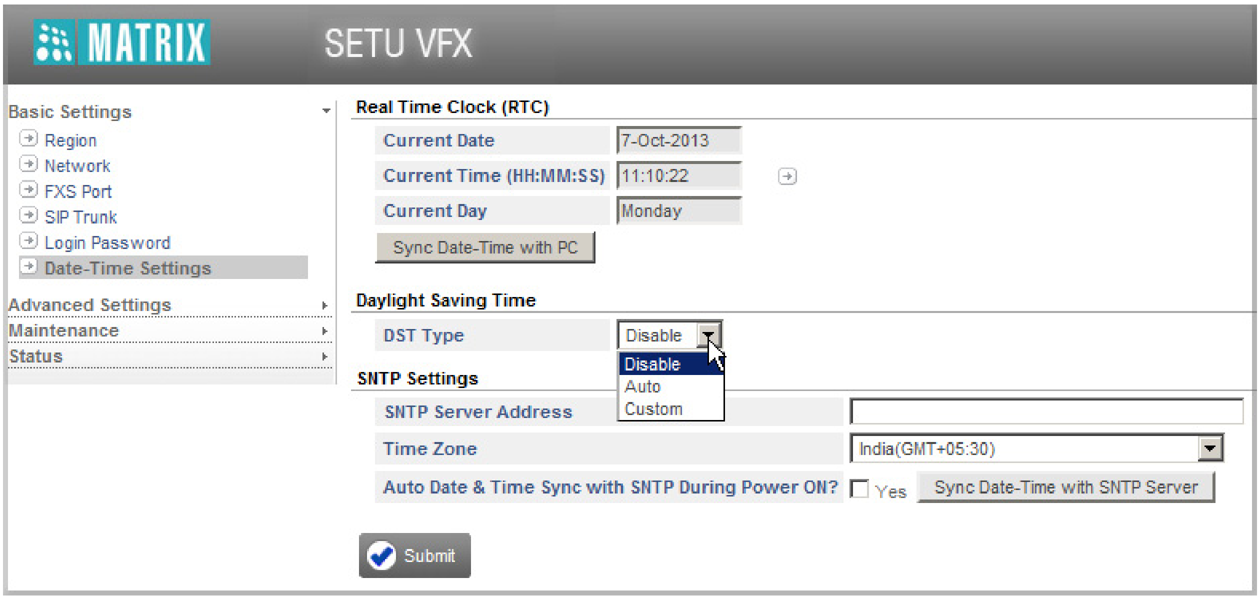
You can set DST by: **Day and Month** or **Date and Month**.

When SETU VFX is set to default, your DST settings will remain unchanged.

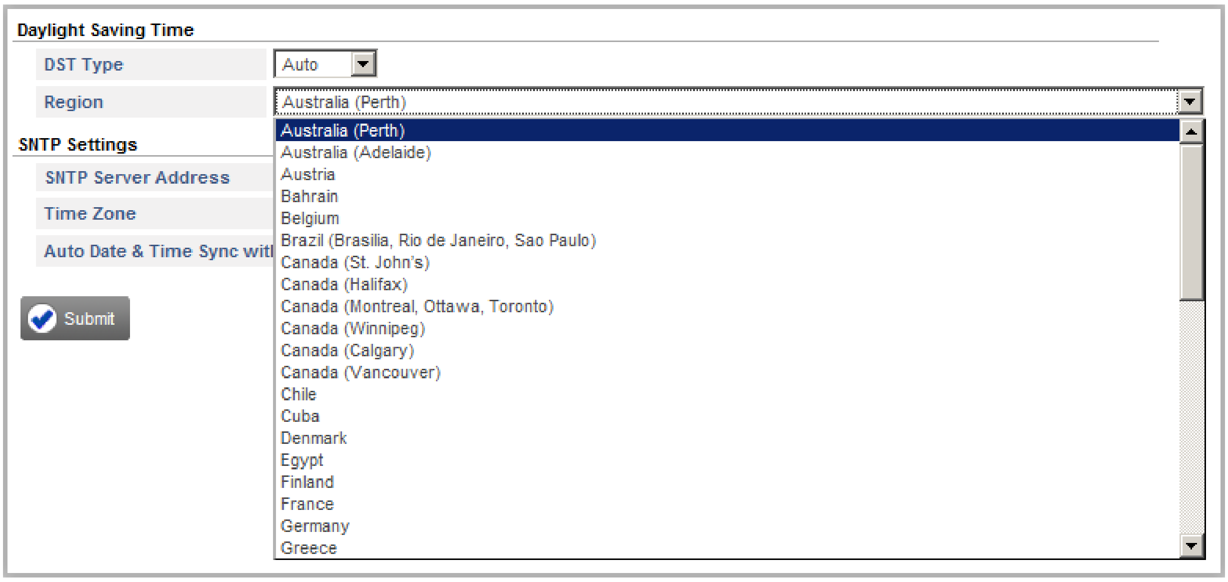
14. *In most countries in Asia and Africa, and in certain countries of South America, DST is not observed.*
15. *SETU VFX sets its Date and Time according to the* ***Time Zone*** *you selected, and synchronizes the time according to the* ***SNTP Server*** *you selected. See “Region”.*

To configure DST,

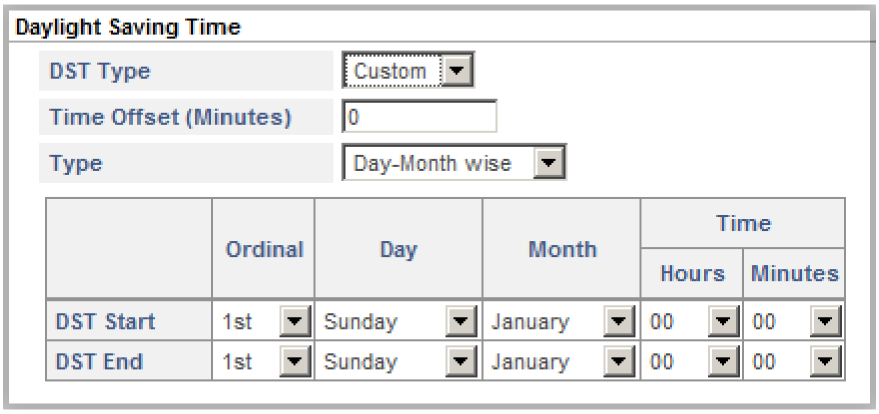
- Under **Basic Settings**, click the **Date and Time Settings** link.



- Go to **Daylight Saving Time** and do the following:
  - Select the **DST Type**. You may select **Auto** or **Custom**. If you do not want to apply DST, select Disable.
  - If you select **Auto**, you must select the **Region**. DST will be set automatically for the region you select.



- 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.



- In the **Time Offset** field, 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 VFX 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 VFX 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**[16].

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 VFX will set the clock forward by 1 hour. On Sunday 30 October, SETU VFX sets the clock back by 1 hour at 02:59:59.

SNTP Settings

To use SNTP for synchronizing with the Real Time Clock,

- Under **Basic Settings**, click the **Date and Time Settings** link.

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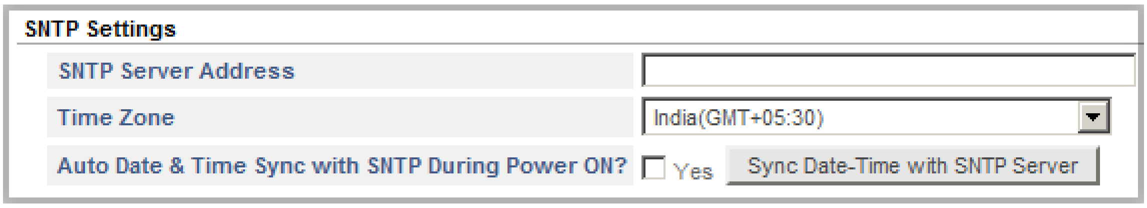
16. *You can also select Day-Month-wise as Type.*

- Go to **SNTP Settings** on this page.



- In the **SNTP Server Address** field, enter the Time Server Address. The SNTP Server address can be of a maximum of 40 characters. Default: Blank.

- By default, the time zone for the country/region where SETU VFX 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 VFX 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 VFX with the SNTP server whenever required, click the **Sync Date and Time Server with SNTP** button.

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

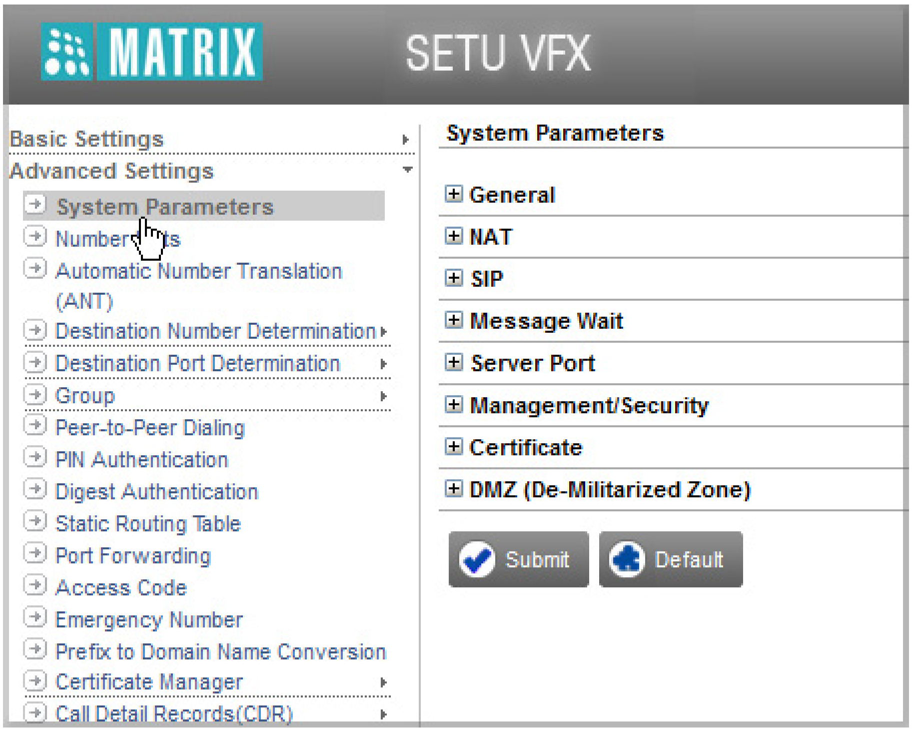
CHAPTER 5 **Advanced Settings**

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, Server Port, Certificates, DMZ etc.

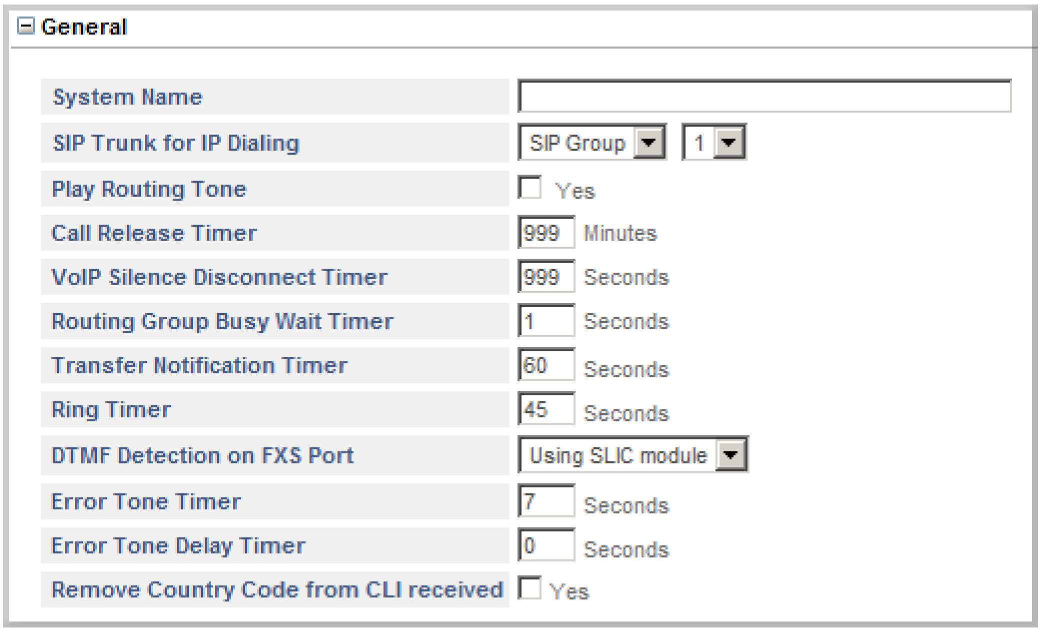
To change the settings of System Parameters,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **System Parameters** link to open the page.



General Parameters

- Click **General** to expand.



- Configure the following parameters.

  - **System Name:** You can assign a name to SETU VFX as 'System Name'. This name can serve as an identifier, when there is more than one SETU VFX 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 (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 have 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 is 1 to 9 and 1 to 9 for SIP Group. 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 Trunks" under *Basic Settings*.

    When you assign a SIP Group, you must configure the SIP Group first. See "Group" for instructions.

  - **Play Routing Tone:** 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. Select this check box, to enable the routing tone. Default: Disabled.

  - **Call Release Timer:** This is the timer used to release the ports involved in a call after a definite period of time, if they are not released due to any reason. This timer is loaded as soon as a call gets matured. This timer is stopped, if one of the ports involved in a call is released. The valid range of Call Release Timer is 001 to 999 minutes. Default: 999 minutes.

  - **VoIP Silence Disconnect Timer:** It is the duration (in seconds) after which SIP call is disconnected, if continuous silence (no RTP Packets) is detected for the set time period. The VoIP Silence Disconnect Timer is loaded as soon as silence is detected during an IP call. The IP call is 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.

The valid range of the VoIP Silence Disconnect Timer is 001 to 999 seconds. Default: 999 seconds.

- **Routing Group Busy Wait Timer:** It is the duration for which SETU VFX 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.

  The valid range of the Routing Group Busy Wait Timer is 1 to 99 seconds. Default: 1 second.

- **Transfer Notification Timer (Seconds):** It is the duration for which SETU VFX will wait for notification of the status of a transferred call, whether transfer target is busy, has answered, has disconnected.

  This timer is loaded as soon as a user performs a transfer activity and the user (transferrer) is notified of the status of the transfer activity within this timer.

  The valid range of the Transfer Notification Timer is 1 to 999 seconds. Default: 60 seconds.

- **Ring Timer (Seconds):** It is the duration for which SETU VFX 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.

  The valid range of the Ring Timer is 1 to 99 seconds. Default: 45 seconds.

- **DTMF Detection on FXS Port:** Select an appropriate method to detect DTMF digits on the FXS Port. SETU VFX can detect the DTMF digits either using SLIC Module or using DSP. Default: Using SLIC Module.

- **Error Tone Timer:** The time for which the system plays the Error Tone. The valid range of the Error Tone Timer is 0 to 9. Default: 7 seconds.

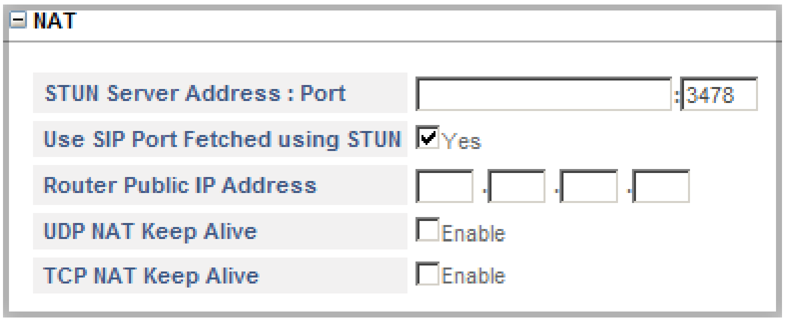
- **Error Tone Delay Timer:** It is the duration after which the system will play the Error Tone, if the call is disconnected during speech. The valid range of the Error Tone Delay Timer is 00 to 99 seconds. Default: 00 seconds.

- **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.

NAT

- Click **NAT** to expand.



- Configure the following NAT parameters.

  - **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 VFX is located behind a NAT router that is other than symmetric, use STUN.

    In the **STUN Server Address: Port** field, 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 VFX is located behind the NAT router and you have forwarded the SIP listening port of the SETU VFX 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** on the SIP Trunk as **STUN**. See “SIP Trunks”.*

  - **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 field 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** on the SIP Trunk as **Router’s IP Address**. See “SIP Trunks”.*

  - **UDP NAT Keep Alive:** When SETU VFX 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.

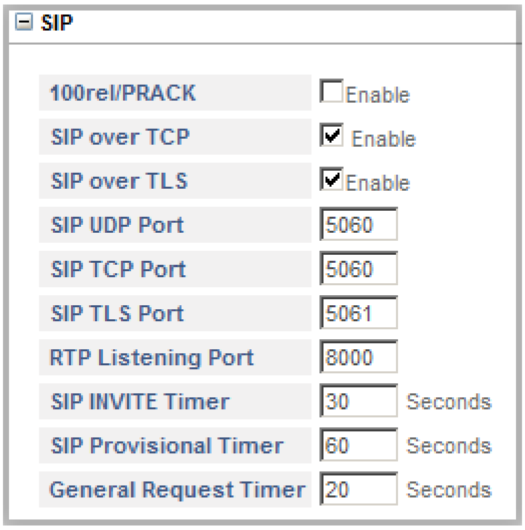
  - **TCP NAT Keep Alive:** When SETU VFX is connected behind a NAT router, and SIP messages are transported over TCP, NAT Keep Alive messages must be sent to refresh the binding in the NAT router.

    - 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.

SIP

- Click **SIP** to expand.



- Configure the following SIP parameters.

  - **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 VFX to use 100rel SIP extension for reliable transmission of SIP provisional responses and to use PRACK (Provisional Acknowledgement). Default: Disabled.

  - **SIP Over TCP:** SETU VFX 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 SIP messages.

    By default, SIP over TCP is enabled. If you want to receive SIP messages over TCP keep this option enabled.

You must also enable ‘TCP' or 'TCP (Fallback to UDP)' on the SIP Trunk.

- **SIP Over TLS:** SETU VFX supports transporting of SIP messages over TLS. TLS protects SIP signaling against loss of integrity, confidentiality and against replay.

  By default, SIP over TLS is enabled. If you want to receive SIP messages over TLS, keep this option enabled.

  You must also enable ‘TLS' on the SIP Trunk.

- **SIP UDP Port:** This is the port on which the SETU VFX 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:** This is the port on which the SETU VFX 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:** This is the port on which the SETU VFX 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.

- **RTP Listening Port:** This is the port on which the SETU VFX listens for RTP Packets. This port is also used as the source port for sending RTP packets to the remote peer. The valid range for this port is 1032–65535. Default: 8000.

- **SIP INVITE Timer:** This is the time in seconds for which SETU VFX 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 VFX terminates the call process and gives an error tone to the user. The range of the SIP INVITE Timer is 10–200 seconds. Default: 30 seconds.

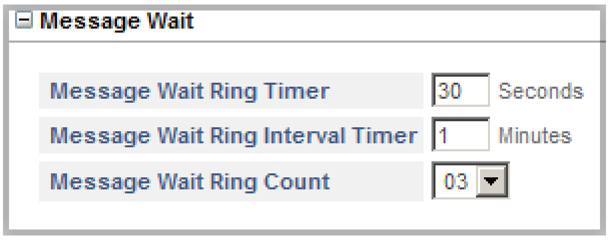
- **SIP Provisional Timer:** This is the time in seconds for which SETU VFX waits for final response after receiving the provisional response from the called party. This timer starts on the receipt of the provisional response from the called party and stops on receipt of the final response from the called party or when the user disconnects the call. On the expiry of the timer, the SETU VFX terminates the call process and gives error tone to the user. The range of the SIP Provisional Timer is 10–200 seconds. Default: 180 seconds.

- **General Request Timer:** This is the time in seconds for which the SETU VFX 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 VFX clears the transaction. The range of the General Request Timer is 10–60 seconds. Default: 20 seconds.

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.

Message Wait[17]

- Click **Message Wait** to expand.



- 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.

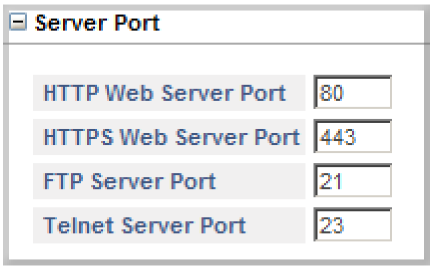
  - **Message Wait Ring Timer:** This is the time duration for which the Message Wait Ring will be played on the FXS Port for Message Wait Notification. The range of this timer is 01–60 seconds. Default 30 seconds.

  - **Message Wait Ring Interval Timer:** This is the time after which SETU VFX will play the Message Wait Ring again on the FXS Port for Message Wait Notification, if the previous ring remains unanswered by the user. The range of this timer is 001–999 minutes. Default: 1 minute.

  - **Message Wait Ring Count:** This is the number of times SETU VFX will play the Message Wait Ring on the FXS Port, until it is answered by the FXS Port user. The range of this count is 1–10. Default: 3.

Server Port

- Click **Server Port** to expand.



- Configure the following:

  - **HTTP Web Server Port:** SETU VFX has an embedded web server called *Jeeves*, for system configuration. You can access *Jeeves* using HTTP. By default, HTTP Web Server Port is 80. You can change it as per your requirement. Valid range of the port is: 80, 1031-65535.

  - **HTTPS Web Server Port:** You can access *Jeeves* of SETU VFX using HTTPS. By default, HTTPS Web Server Port is 443. You can change it as per your requirement. Valid range of the port is: 443, 1031-65535.

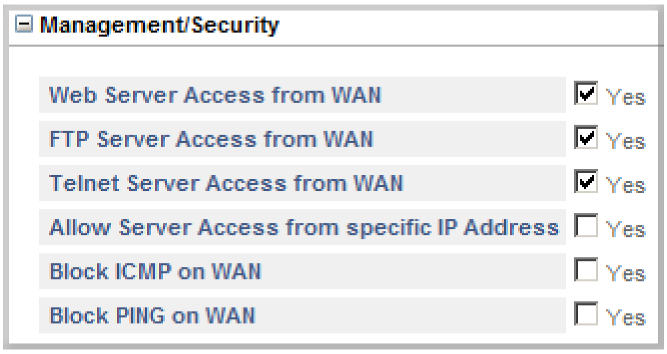
17. *Applicable only for VoIP-FXS Gateway.*

- **FTP Server Port:** SETU VFX has an embedded FTP server for Software Upgrade. By default, FTP Server Port is 21.You can change it as per your requirement. Valid range of the port is: 21, 1031-65535.

- **Telnet Server Port:** You can access SETU VFX using Telnet. By default, Telnet Server Port is 23. You can change it as per your requirement. Valid range of the port is: 23, 1031-65535.

Management/Security

- Click **Management/Security** to expand.



- Configure the following.

  - **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.

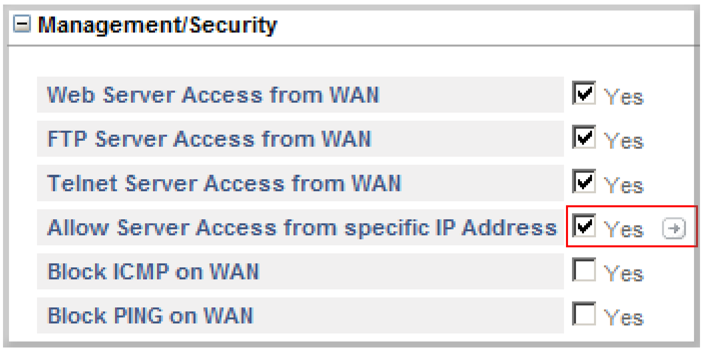
    You may clear this check box, if required. Default: Enabled.

  - **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.

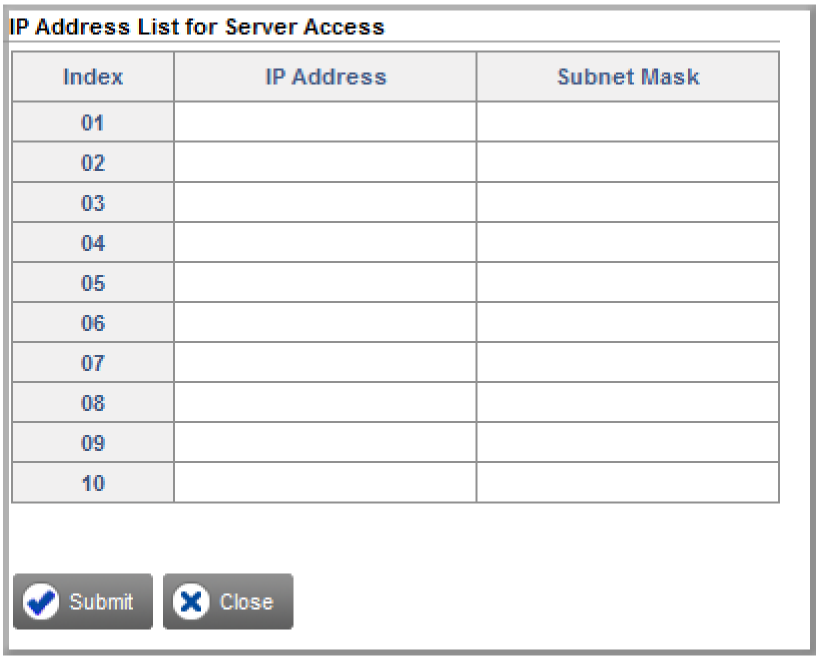
    You may clear this check box, if required. Default: Enabled.

  - **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 **IP Address List for Server Access** opens in a new window. You can store 10 entries in this table.



- Enter the IP Addresses and their respective Subnet Mask in the table.
- Click Submit and close the window.

SETU VFX will allow system access only to those users whose IP Address matches with the one configured in the IP Address List for Server Access.

When you change the Jumper position (J3) to restore the default IP Address or to restore default Jeeves/ Command Password, the following Server Access parameters will also be set to default.

- *Web Server Access from WAN*
- *FTP Server Access from WAN*
- *Telnet Server Access from WAN*

- **Block ICMP on WAN:** Enable this check box, if you want the system to discard the ICMP packets received on WAN. Default: Disabled.

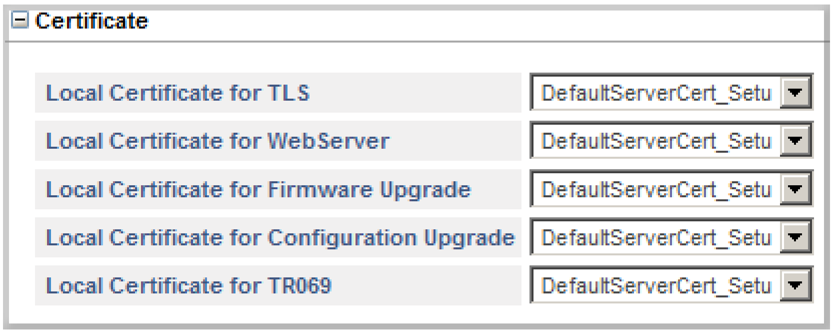
- **Block PING on WAN:** Enable this check box, if you want the system to discard the PING request received on WAN. Default: Disabled.

  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.



- Select the Certificate for each of the following.
  - 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.
  - In **Local Certificate for TR069**, select the desired certificate to be used by the system for TR069.

  To create and Upload /Download Certificates, see "Certificate Manager".

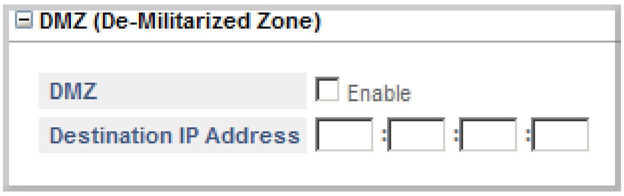
DMZ (De-Militarized Zone)

Demilitarized Zone (DMZ) is a type of network security. It is a small sub-network residing between the private network and the public network to protect the devices located in the private network.

Usually, the private network contains the host devices, mostly servers (Mail servers, FTP servers, VoIP servers, Web servers) that generally need to be accessed from the public network. These servers are most vulnerable to attacks from the public network. Thus, there is a need to separate these servers from the rest of the private network.

Using DMZ, you can configure multiple servers—Web Server, DNS Server, FTP Server, SMTP Relays—on a single computer and make these available to the general public. Thus, you can secure your private network from the threats of the public network.

- Click **DMZ (De-Militarized Zone)** to expand.



- Configure the following.
  - Select the **DMZ Enable** check box.
  - In the **Destination IP Address** field, enter the IP Address of the internal host (connected behind your SETU VFX) where the packets are to be forwarded. The Destination IP Address may be of maximum 15 characters. Valid characters are 0 to 9 and dot(.). Default: Blank.
  - *Port Forwarding, if enabled, will be given priority over DMZ.*
  - *If LAN IP Address or Subnet mask is changed, the DMZ parameters will be set to default.*
- Click **Submit** to save changes.

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 below.

SETU VFX 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 following features. 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—FXO, FXS, SIP Trunks—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 VFX 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.
- When following options is selected in **Route all Incoming Calls (with CLI)**:
  - 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 Allowed and Denied Lists are assigned to each port type:

Port Type	Default Allowed Numbers List	Default Denied Numbers List
FXS Ports	List 05	List 06
FXO 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, FXO, 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” in “FXS Ports”
  “Handling of Incoming Calls” in “FXO Port”
  “Handling of Incoming Calls” in “SIP Trunks”

  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 this 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. 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 full 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 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 Trunks”

Now, whenever there is an incoming call on the SIP Trunk you have applied this feature, the SETU VFX 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.

Make a list of numbers that you want to black list. Configure these numbers in a Number List. By default, 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.

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 VFX 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 sort 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 one 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 Records (CDR)".

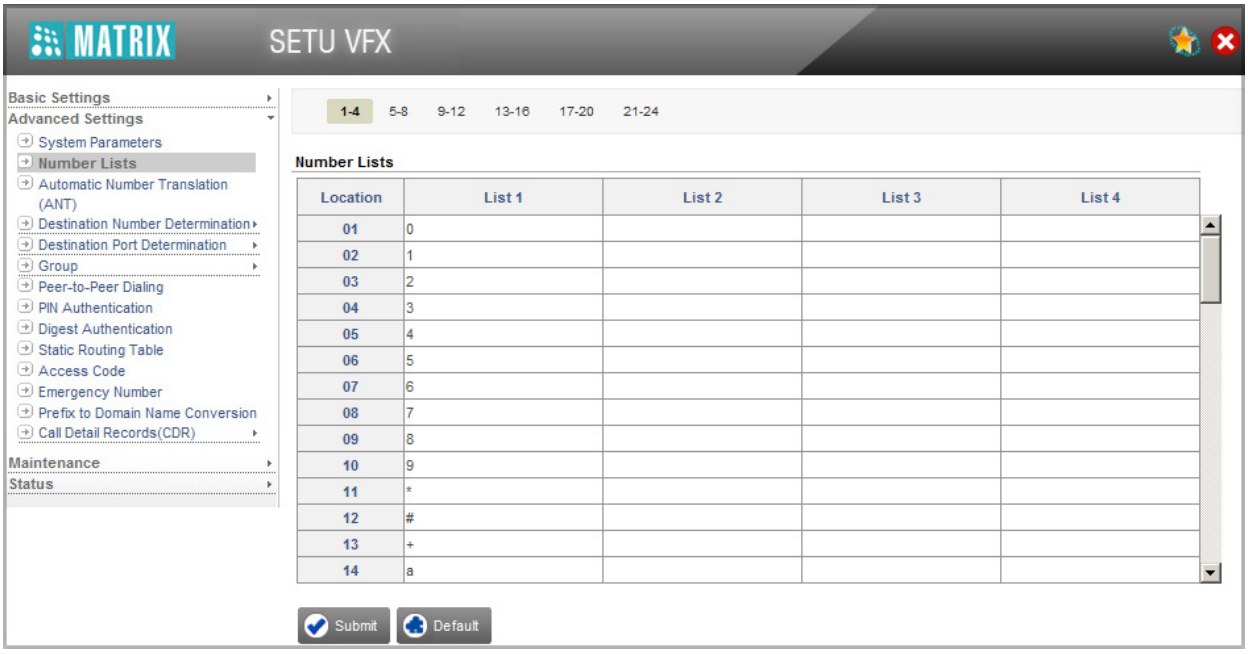
Configuring Number Lists

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.

- Under Advanced Settings, click the **Number List** link.



- 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** field, select **22**.
  - Click **Submit**.

You can also configure Number Lists on the respective SIP Trunk, FXO Port, FXS Port pages under the “Basic Settings” link of Jeeves.

Automatic Number Translation (ANT)

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, 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 "FXS Ports", "Handling of Outgoing Calls" under "SIP Trunks" and "Handling of Outgoing Calls" under "FXO Port".

Now, whenever there is a call on/from the Port on which you have applied this feature, SETU VFX 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.

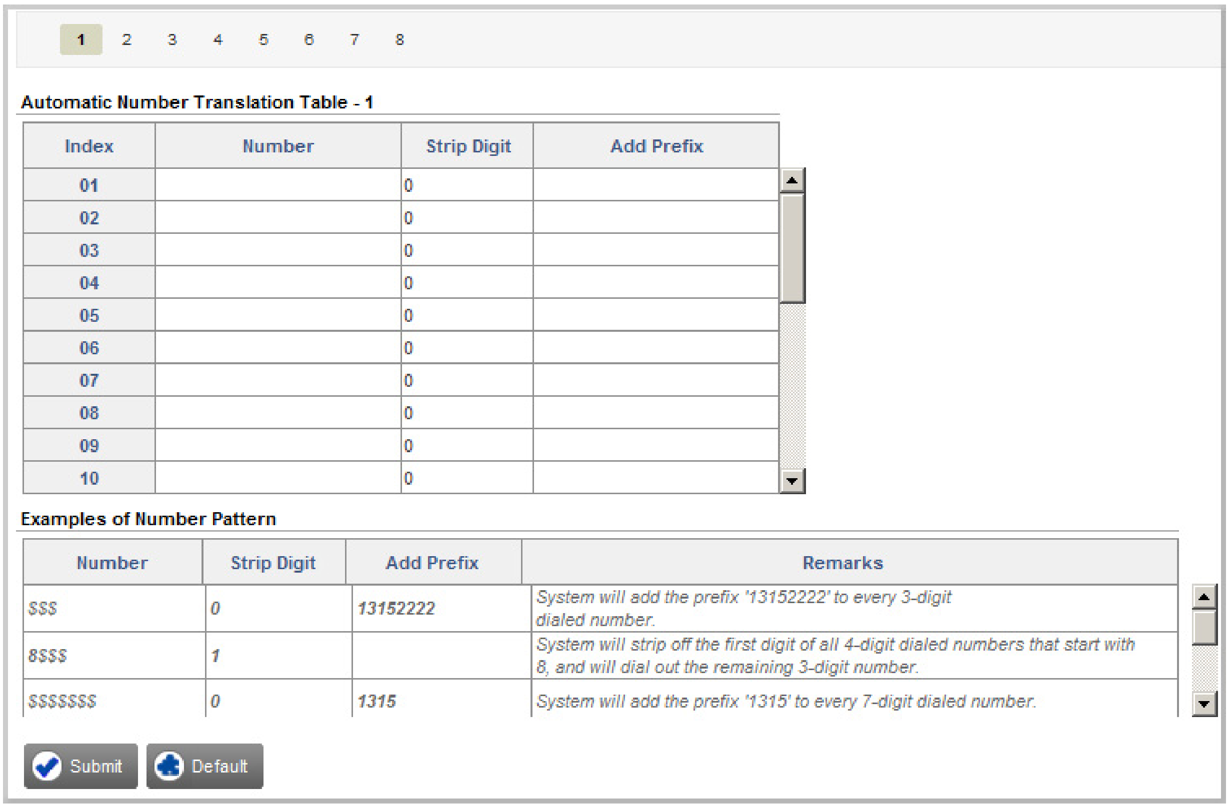
- The system matches the Called number with the Number configured in the ANT table. The number matches with the entry ‘00’ stored in the table.

- 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.
- Under Advanced Settings, click the **Automatic Number Translation (ANT)** link.



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	

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 the **Number** column, 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 VFX 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 the **Strip Digit** column, 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 the **Add Prefix** column, 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.

Destination Number Determination

The process of routing calls originated on FXO Ports, FXS Ports and SIP Trunks to the destination port in SETU VFX takes place in two steps:

- Determination of Destination Number
- Determination of Destination Port

SETU VFX supports different methods of determining the destination number for the calls originated on the FXO Ports and SIP Trunks.

Destination Number Determination on SIP Trunks

For SIP Trunks, 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
- 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 VFX 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 VFX 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 FXO Ports[18]

For FXO Ports, 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
- 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 VFX 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.

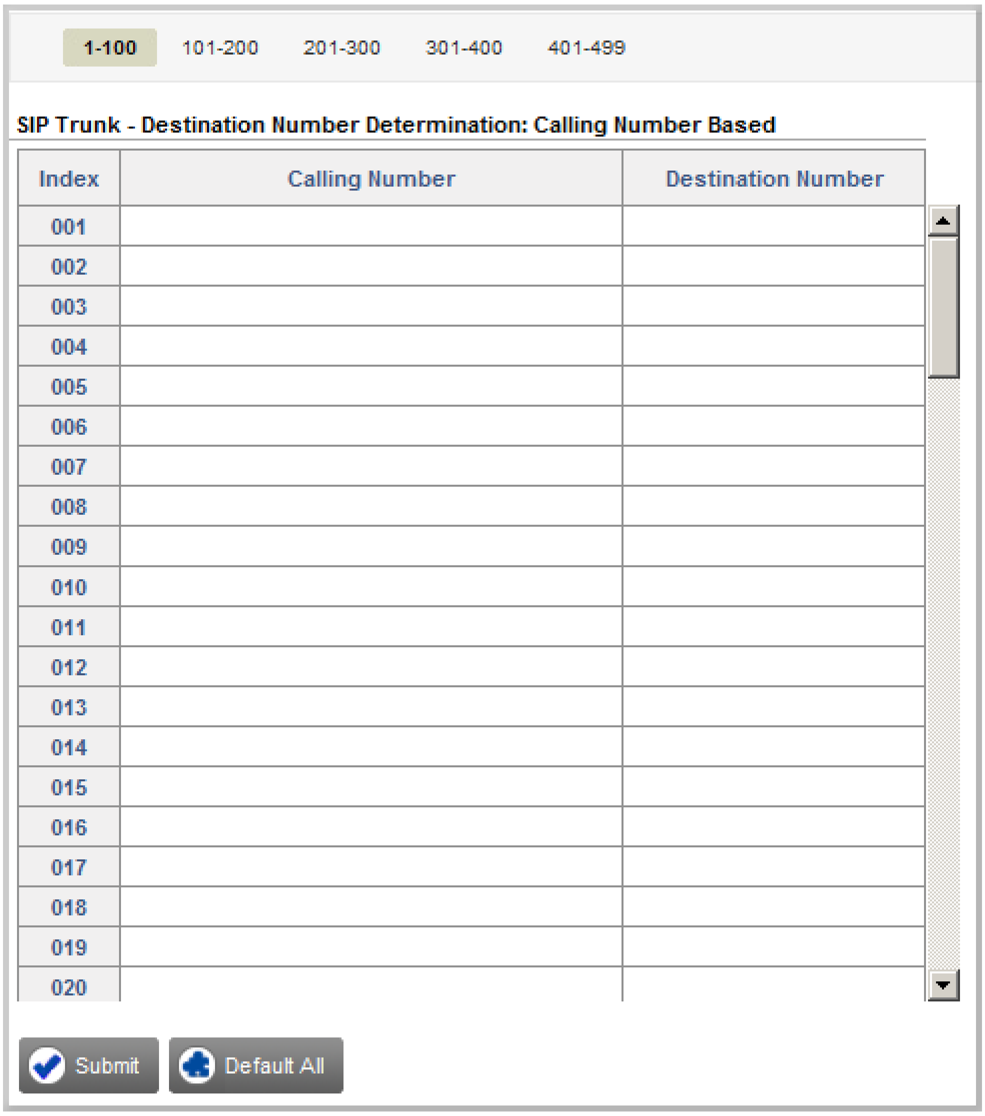
Configuring Calling Number Based Table

- Log into Jeeves.

- Click the **Advanced Settings** link.

18. Applicable only for VoIP-FXO Gateway.

- Click the **Destination Number Determination** link.



- To configure the Calling Number Based table for the SIP Trunk, click the **SIP-Calling Number Based** link.
- To configure the Calling Number Based table for the FXO Port, click the **FXO-Calling Number Based** link.

  The Calling Number Based Table page opens.

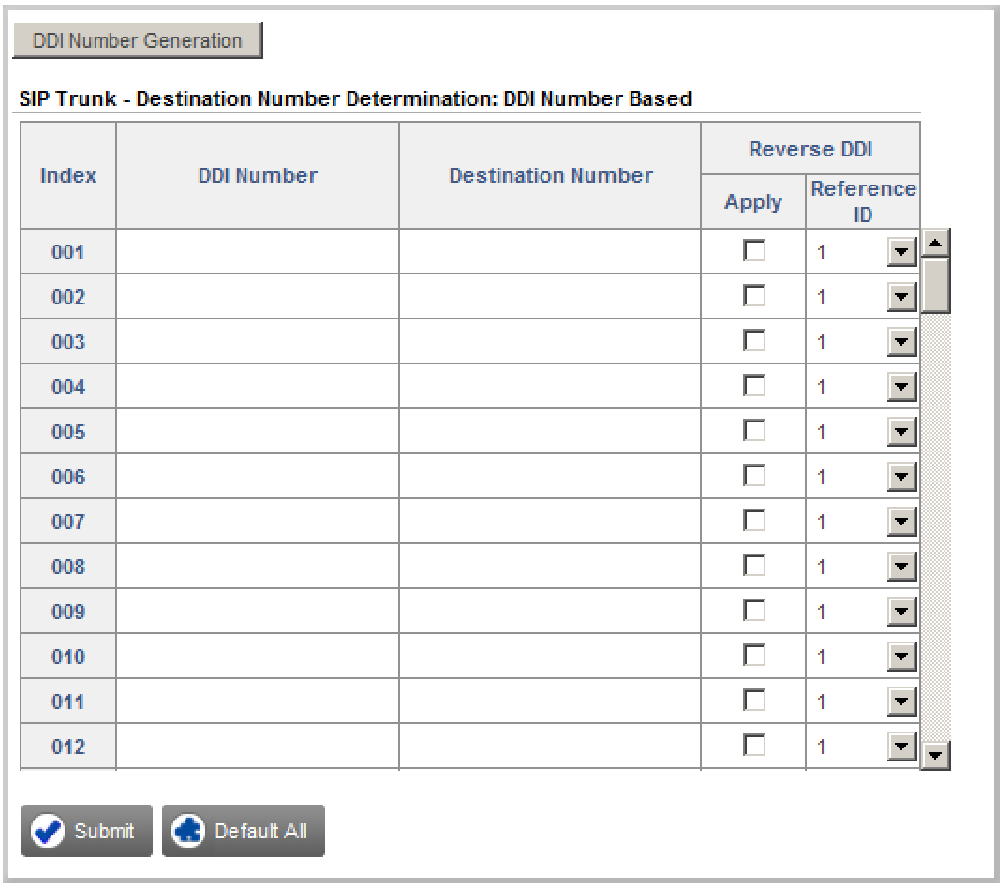
  - In the Calling Number Based Table, configure the following.
    - Enter the calling party numbers in the column **Calling Numbers**. Calling numbers may consist of a maximum of 24 characters. All ASCII characters are allowed. Default: Blank.
    - For each calling party number, enter a corresponding destination number in the column **Destination Numbers**. 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.
- Click **Default All** to clear all the entries.

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 DDI Number Based Table page opens.


DDI Number Generation

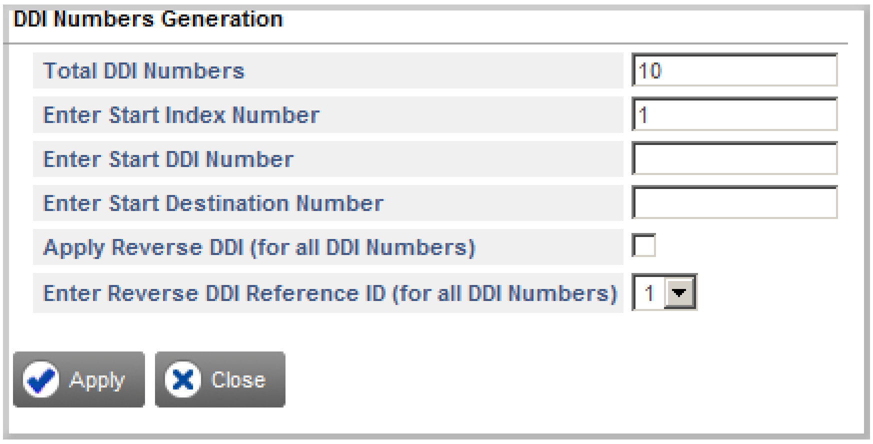
SIP Trunk - Destination Number Determination: DDI Number Based

Index	DDI Number	Destination Number	Reverse DDI	
			Apply	Reference ID
001				1
002				1
003				1
004				1
005				1
006				1
007				1
008				1
009				1
010				1
011				1
012				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:



  - **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

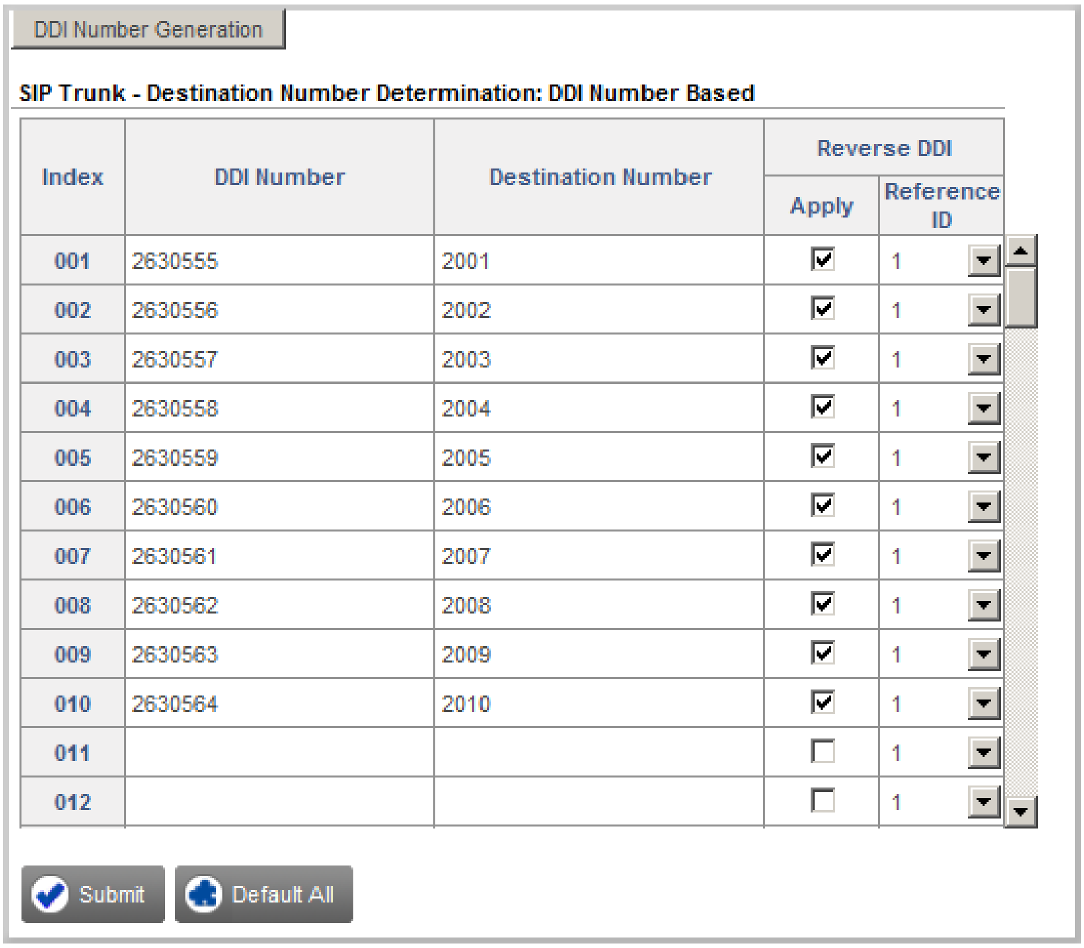
  - **Enter Start Index Number:** Enter the desired Index Number from where you want to start the DDI Number generation. Default: 1

  - **Enter Start DDI Number:** Enter the start DDI Number. DDI Number can be of maximum 24 characters. Characters 0-9, +, * and # are allowed in this field.

  - **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	☑	1
002	2630556	2002	☑	1
003	2630557	2003	☑	1
004	2630558	2004	☑	1
005	2630559	2005	☑	1
006	2630560	2006	☑	1
007	2630561	2007	☑	1
008	2630562	2008	☑	1
009	2630563	2009	☑	1
010	2630564	2010	☑	1
011			☐	1
012			☐	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 Reverse DDI?** check box.

      The Reverse DDI **Reference ID** for the DDI Number, will be applied on the DDI Number.

      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 FXO Ports, FXS Ports and SIP Trunks to the destination port in SETU VFX takes place in two steps:

- Determination of Destination Number
- Determination of Destination Port

SETU VFX supports different methods of determining the destination port for the calls originated on SIP Trunks, FXO Ports and FXS Ports.

Destination Port Determination on FXS Ports[19]

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 **Destination Port Determination: FXS-Destination Number Based** table.

To configure Destination Number based table for FXS Ports, see "Configuring Destination Number Based Table for FXS Ports".

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 **Destination Port Determination: SIP-Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **Destination Port Determination: SIP-Destination Number Based** table.

Destination Port Determination on FXO Ports[20]

For FXO 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 **Destination Port Determination: FXO-Calling Number Based** table.

To apply Destination Port Determination **on the basis of Destination Number**, you must configure the **Destination Port Determination: FXO-Destination Number Based** table.

19. *Applicable only for VoIP-FXS Gateway.*
20. *Applicable only for VoIP-FXO Gateway.*

To configure Calling Number based table for SIP/FXO Ports, see “Configuring Calling Number Based Table for SIP/FXO Ports”.

To configure Destination Number based table for SIP/FXO Ports, see “Configuring Destination Number Based Table for SIP/FXO Ports”.

Configuring Destination Number Based Table for FXS Ports

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Under **Destination Port Determination**, click the **FXS-Destination Number Based** link.
- The **FXS Port - Destination Port Determination - Destination Number Based** table opens.

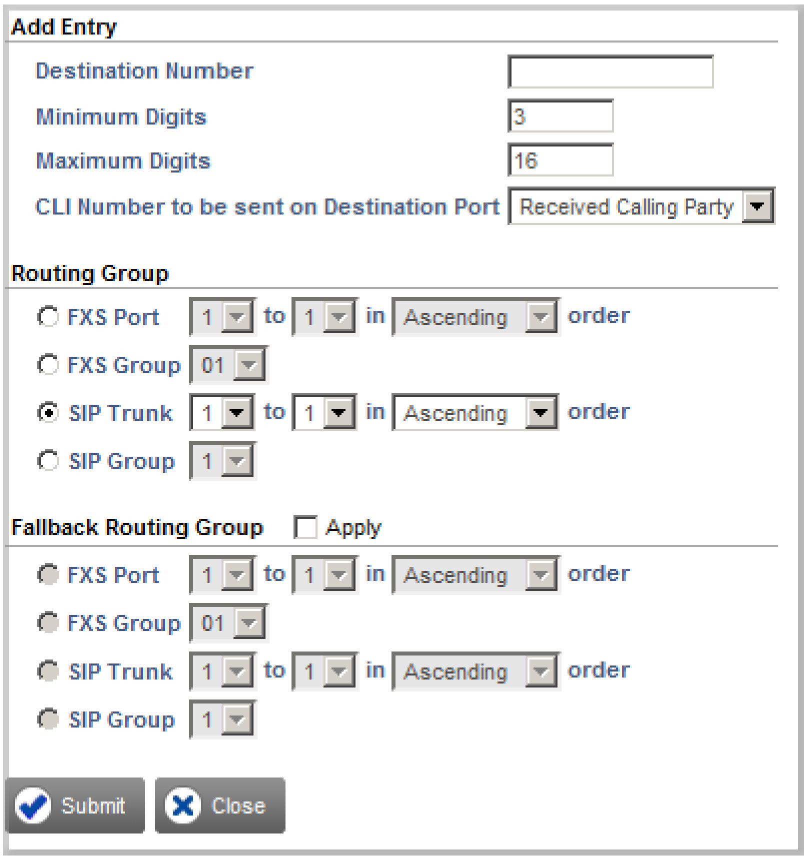
FXS Port - Destination Port Determination - Destination Number Based

☐	Edit	Destination Number	Minimum Digits	Maximum Digits	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
☐		2001	4	4	FXS Port 1 - 1 (Ascending)	None	Received Calling Party
☐		2002	4	4	FXS Port 2 - 2 (Ascending)	None	Received Calling Party
☐		2003	4	4	FXS Port 3 - 3 (Ascending)	None	Received Calling Party
☐		2004	4	4	FXS Port 4 - 4 (Ascending)	None	Received Calling Party
☐		2005	4	4	FXS Port 5 - 5 (Ascending)	None	Received Calling Party
☐		2006	4	4	FXS Port 6 - 6 (Ascending)	None	Received Calling Party
☐		2007	4	4	FXS Port 7 - 7 (Ascending)	None	Received Calling Party
☐		2008	4	4	FXS Port 8 - 8 (Ascending)	None	Received Calling Party

Total Records : 8   1

Add   Delete   Close

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



- In the **Destination Number** field, enter the number (24 characters, maximum) that you expect callers to dial. Valid digits: 0–9, *, #, (dot). Default: Blank.

- In the **Minimum Digits** field, enter the minimum number of digits of the destination number that the caller must dial for the system to route the call. Valid range: 01–24. Default: 3.

  If the dialed number string is less than the configured minimum length, the call will be rejected.

- In the **Maximum Digits** field, enter the maximum number of digits to be dialed by the caller for the system to consider it as end-of-dialing for routing the call. Valid range: 01–24. Default:16.

  If the number string dialed by the caller exceeds the maximum length configured, the system will strip off the extra digits, and route the call.

- Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
  - Received Calling Party
  - Received Called Party

  Default: Received Calling Party

***CLI Number to be sent on Destination Port** is applicable only when **FXS Port** or **FXS Group** is selected as the Destination Port (Routing Group).*

- 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.

    To start hunting from the first to the last member FXS Port, select **Ascending**.

    To start hunting from the last to the first member FXS Port, select **Descending**.

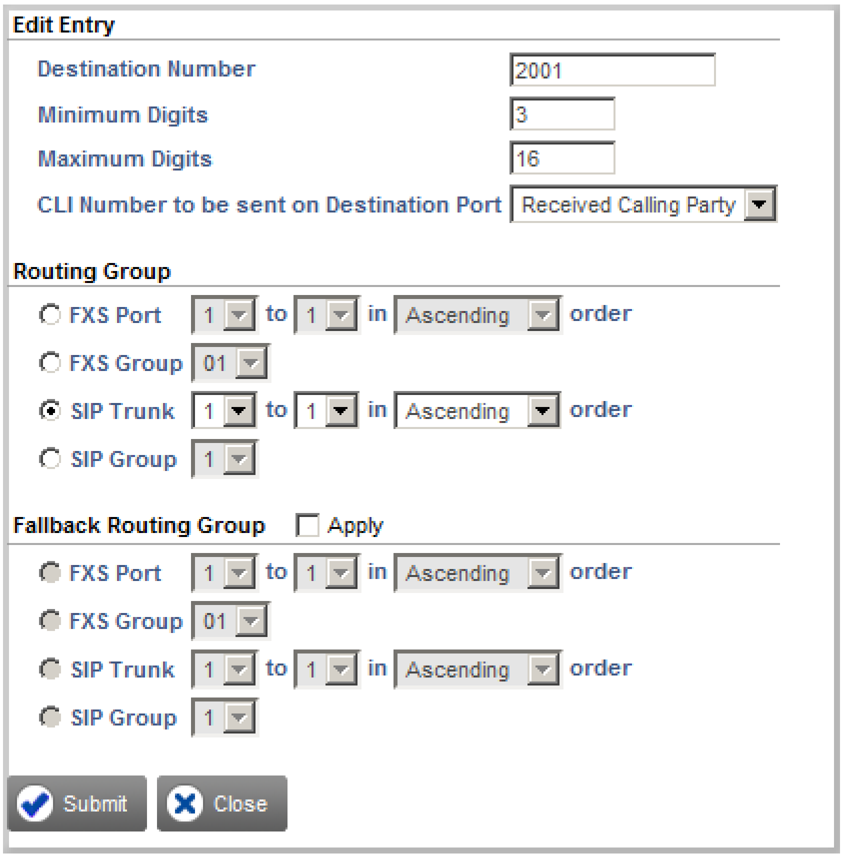
    Default: Ascending.
- To create a group of *not-sequential* **FXS Ports** as members,
  - Select an **FXS Group**.
  - Select **FXS Group** number. Default:1.
  - Click **Settings** . The **FXS Groups** window opens. Create the FXS Group. For detailed instructions, see "Group".
- Similarly, you can create group of *sequential* and *not-sequential* FXO Ports, SIP Trunk and Mobile Ports as members.

- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the same instructions provided earlier to create *sequential* and *not-sequential* groups of FXS Ports, FXO Ports, SIP Trunks and Mobile Ports.
  - Click **Submit** to save changes. The **Add Entry** window closes.
- The entry you added appears in the **FXS Port - Destination Port Determination - Destination Number Based** window.

  To edit the Routing Group and/or the Fallback Routing Group, do the following:

  - Under **Edit**, click **Settings** .

- The **Edit Entry** window opens.



- Create the **Routing Group** and **Fallback Routing Group** as described earlier.
- Click **Submit** and close the window.

- Follow the same steps as above to add/edit another entry in this table.
- To delete an entry, select the check box of the entry and click the **Delete** button.
- Close the window if you have finished adding/editing entries.

Configuring Destination Number Based Table for SIP/FXO Ports

- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **Destination Port Determination** link.

SIP Trunk - Destination Port Determination - Destination Number Based

	Edit	Destination Number	Minimum Digits	Maximum Digits	Routing Group	Fallback Routing Group	CLI Number to be sent on Destination Port
		No Match Found	3	16	FXS Port 1 - 1 (Ascending)	None	Received Calling Party

Total Records : 1 1

Add Delete

- To configure the table for the SIP Trunk, click the **SIP-Destination Number Based** link.
- To configure the table for the FXO Port, click the **FXO-Destination Number Based** link.

  The **Destination Number Based** Table page opens.

- Click **Add** to add an entry. A new window opens.
  - In the **Destination Number** field, enter the number (max. 24 characters) you expect callers to dial. Valid characters: 0 to 9, +, * and #. Default: Blank.
  - In the **Minimum Digits** field, enter the minimum number of digits of the destination number that the caller must dial for the system to route the call. Range: 01 to 24. Default: 03.

    If the dialed number string is less than the configured minimum length, the call will be rejected.

  - In the **Maximum Digits** field, enter the maximum number of digits of the destination number the caller must dial for the system to route the call. Maximum length range: 01 to 24. Default:16.

    If the number string dialed by the caller exceeds the maximum length configured, the system will strip off the extra digits, and route the call.

  - Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
    - Received Calling Party
    - Received Called Party

    Default: Received Calling Party

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

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

      To start hunting from the first to the last member FXS Port, select **Ascending**.

      To start hunting from the last to the first member FXS Port, select **Descending**.

Default: Ascending.

- 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 Groups** window opens. Create the FXS Group. See “Group” for detailed instructions.
- Similarly, you can create group of *sequential* and *not-sequential* FXO Ports and SIP Trunks as members.

- You may create the **Fallback Routing Group**. To do this,
    - Select the **Apply** check box.
    - Follow the same instructions given for creating *sequential* and *not-sequential* groups, for FXS Ports, FXO Ports and SIP Trunks.
    - Click **Submit** to save changes. The **Add Entry** window closes.
- The entries you added appear on the screen.
- To change the default Routing Groups assigned for No Match Found numbers entry,
    - For the **No Match Found** entry, under **Edit**, click **Settings** .
    - The **Edit Entry** window opens.
    - Create the **Routing Group** and **Fallback Routing Group**.
    - Click **Submit** and close the window.
- 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.
- Close the window if you have finished adding/editing entries.

Configuring Calling Number Based Table for SIP/FXO Ports

- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **Destination Port Determination** link.

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

- To configure the table for the SIP Trunk, click the **SIP-Calling Number Based** link.
- To configure the table for the FXO Port, click the **FXO- Calling Number Based** link.

  The **Calling Number Based** Table page opens.

- Click **Add** to add an entry. A new window opens. Configure the following parameters:
  - In the **Calling Number** field, enter numbers (max. 24 characters) from which you expect calls to be received. All ASCII characters are allowed. Default: blank.
  - Select the **CLI Number to be sent on Destination Port**. You may select from the following options:
    - Received Calling Party
    - Received Called Party

    Default: Received Calling Party

CLI Number 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** field, select the order in which the system should check for a free member FXS Port to route the call.

        To start hunting from the first to the last member FXS Port, select **Ascending**.

        To start hunting from the last to the first member FXS Port, select **Descending**.

        Default: Ascending.
    - 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 Groups** window opens. Create the FXS Group. See “Group” for detailed instructions.
- Similarly, you can create group of *sequential* and *not-sequential* FXO Ports and SIP Trunks as members.

- You may create the **Fallback Routing Group**. To do this,
  - Select the **Apply** check box.
  - Follow the instructions provided for creating *sequential* and *not-sequential* groups, for FXS Ports, FXO Ports and SIP Trunks.
  - Click **Submit** to save changes. The **Add Entry** window closes.
- The entries you added will appear on the screen.
- To change the default Routing Groups assigned for No Match Found numbers entry,
  - For the **No Match Found** entry, under Edit, click **Settings** .
  - The **Edit Entry** window opens.
  - Create the **Routing Group** and **Fallback Routing Group**.
  - Click **Submit** and close the window.
- 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.

Group

SETU VFX supports the following methods of determining the destination port for the calls originated on FXO Ports, FXS 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.*

For any of these methods that you select, you need to configure Routing Group and Fallback Routing Group.

A Routing Group may have *sequential* or *non-sequential* ports as members.

A Routing Group of *sequential* ports is to be formed when you select **FXO Port** or **FXS Port** or **SIP Trunk** as the destination port.

A Routing Group of non-sequential ports is to be formed when you select **SIP - Group** or **FXO - Group** or **FXS - Group** as the destination port. The **SIP/FXO/FXS Group** has members of the same port type, but not in a sequence. A SIP Group can have only SIP Trunks as members. Similarly, a FXO Group can have only FXO Ports as members.

Configuring Groups

To create a Group,

- Click **Advanced Settings** link.
- Click the **Group** link.

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	6	7	8	9
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

Submit Default

- To create Groups of SIP Trunks, click **SIP Group**. You can create 9 Groups with 9 members in each group.

- Select a SIP Group Number from **1 to 9.**
- Configure member ports - **Member 1** to **Member 9**.
  - For each **Member**, select a SIP Trunk number from **1 to 9** from the combo box.
  - 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 2, and set the remaining members in the group to None.
- Define the **Member Selection Method.** To route a call the system checks availability of a free port. There are two options, namely:
  - **First Free:** In this method, the first free port will be used for routing the call each time. For example, SIP Group Number 1 has four members SIP Trunk 1 (Member 1), SIP Trunk 2(Member 2), SIP Trunk 3 (Member 3) and SIP Trunk 6 (Member 4). For every incoming call, SETU VFX will check the status of Member 1 first. If free, the call will be routed to this port else system will check status of Member 2 and so on.
  - **Rotation:** In this method, the first call will be routed to the first member port and the subsequent call to the next member port and so on. For example, SIP Group Number 2 has four members SIP Trunk 6 (Member 1), SIP Trunk 7(Member 2), SIP Trunk 8 (Member 3) and SIP Trunk 9 (Member 4). For the first incoming call, SETU VFX will check the status of Member 1 (SIP Trunk 6). If free, the call will be routed to this port else system will check status of Member 2 (SIP Trunk 7) and so on. For the next call, system will check status of Member 2 (SIP Trunk 7) first. If free, call will be routed to this port else the system will check the status of next member in the group.

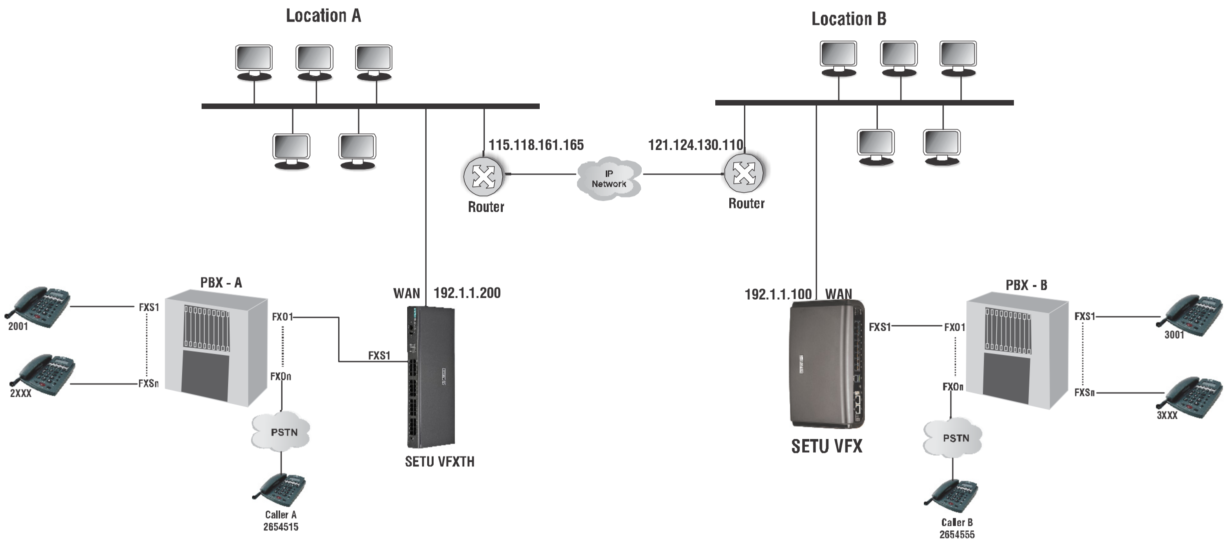
    Default: **First Free**.

- Click **Submit** to save the group.
- Similarly you can create FXO Groups and FXS Groups.
- To create Groups of FXO Ports, click **FXO Group**. You can create 12 Groups with 8 members in each group.
- To create Groups of FXS Ports, click **FXS Group**. You can create 12 Groups with 8 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 VFX is installed at Location B.
- Peer-to-Peer calls can be made between the two locations with suitable configuration of SETU VFX 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.

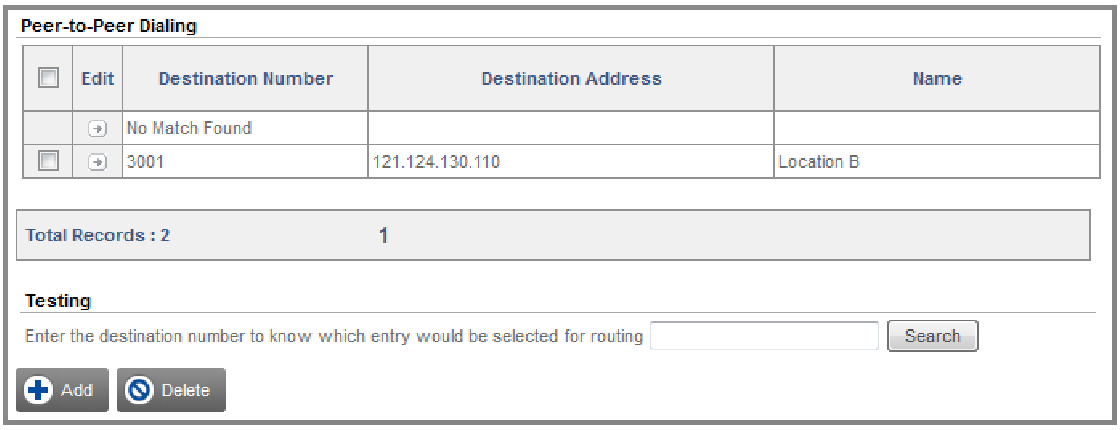
- Now, configure the **Peer-to-Peer Table**.

In this example, you would have to configure the Peer-to-Peer table as follows:

- At Location A, in the Number field of the Peer-to-Peer table, enter the Number you want to dial to call the phone at Location B. In this case, 3001.

- For the number you entered, in the Destination Address field in the table, 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:



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 '3' to the IP Address 121.124.130.110.

Destination Number	Destination Address	Name
No Match Found		
3	121.124.130.110	Location B

- At **Location B**, you need to do the following configuration in SETU VFX:

  - 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** field of the SIP Trunk **blank**.

In the Router, you must configure the same SIP and RTP Ports as configured in the SETU VFX. 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 B, you must configure the IP Address of the Router at Location A.

- 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.

For instructions on configuring SIP Trunk parameters, see “SIP Trunks” under *Basic Settings.*

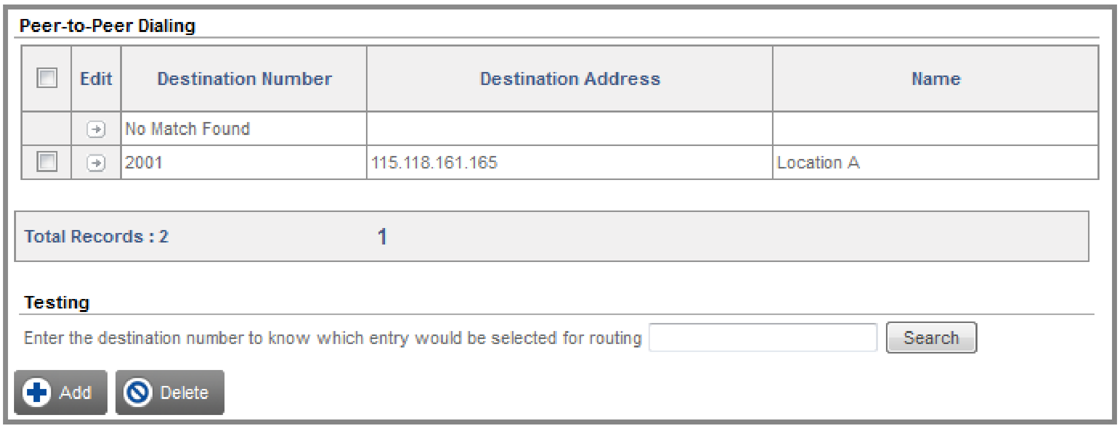
- Now, configure the **Peer-to-Peer Table**.

In this example, you would have to configure the Peer-to-Peer table as follows:

- At Location B, in the Number field of the Peer-to-Peer table, 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 field in the table, 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 VFX at Location B would look like this:



- 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 '3', 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 3001, 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 VFX at location B routes this call through the FXO Port of the SETU VFX to the FXS Port user 3001.

- Similarly, when the user 3001 of location B calls 2001, the call is received on the SIP Trunk of the SETU VFX 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.

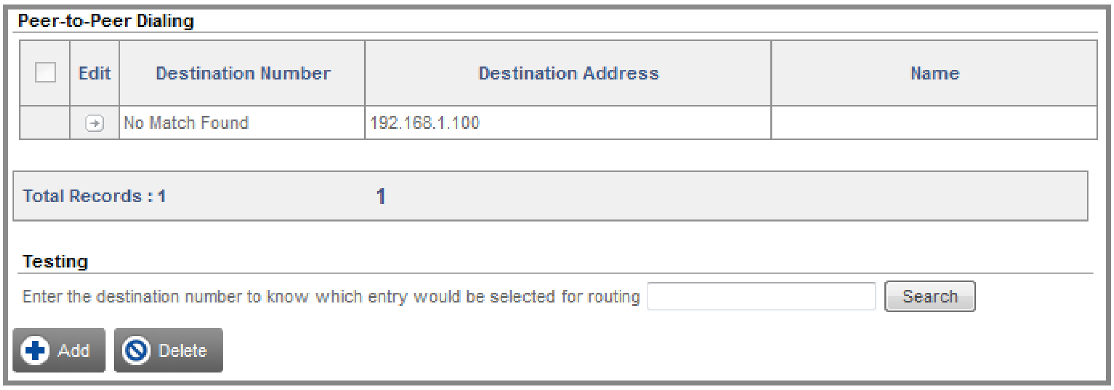
How to Configure

For instructions on configuring the SIP Trunk parameters for the Peer-to-Peer application—SIP Trunk Mode, Peer-to-Peer Table, SIP ID, Handling of Incoming Calls —see “SIP Trunks” under *Basic Settings*. You can also configure the Peer-to-Peer Table from the SIP Trunk page under *Basic Settings*.

The Peer-to-Peer table stores upto 500 entries. Each entry consists of the parameters —Destination Number, Destination Address and Name.

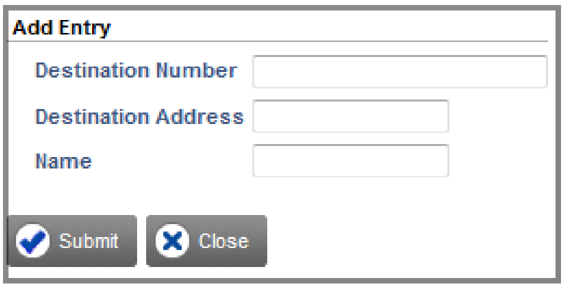
To configure the Peer-to-Peer Table,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Peer-to-Peer Dialing** link. The Peer-to-Peer table opens.



In the Peer-to-Peer table, the first entry is reserved for No Match Found.

- Click the **Add** button. A new window opens.



- In the **Destination Number** field, enter the peer-to-peer number string—prefix or entire number—that will be dialed. The number string must not exceed 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 in this field.

Wildcard Characters

SETU VFX supports following characters.

Character	Description
X (letter X)	X represents any single digit from 0 to 9.
#	When # is configured in a number string, it will not be considered as End of Dialing.
*****	When * is configured in a number string, it will not be considered as End of Dialing.
+	+ (plus) can be configured as a first character of the Destination Number string in the *SIP Trunk-Destination Port Determination-Destination Number Based* 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.

- In the **Destination Address** field**,** enter the domain name or IP Address to where the call is to be placed. The Destination Address may consists of 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 the **Name** field, 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.

PIN Authentication

PIN Authentication is a necessary security feature to restrict access to the system and prevent possible misuse of resources.

You can use PIN Authentication on the source port (FXO Port) to establish the identity of callers before their call is processed by SETU VFX.

PIN Authentication can be used on the source port only if the incoming call routing for the source port is set to ***After Answering the Call and Collecting 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 up to 500 PIN Numbers and their corresponding authentication Passwords.

When you enable PIN Authentication on the source port, SETU VFX answers the incoming call on the port and plays the prompt tone. It waits for the caller to dial the PIN Number and the Password. It collects the digits dialed by the caller and matches them with the PIN Authentication table.

When a match is found in the table, SETU VFX 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 VFX allows the caller to make two more attempts to dial a valid PIN Number and Password. If the caller fails to dial the correct PIN and Password in all attempts, the system disconnects the call.

Configuring PIN Authentication

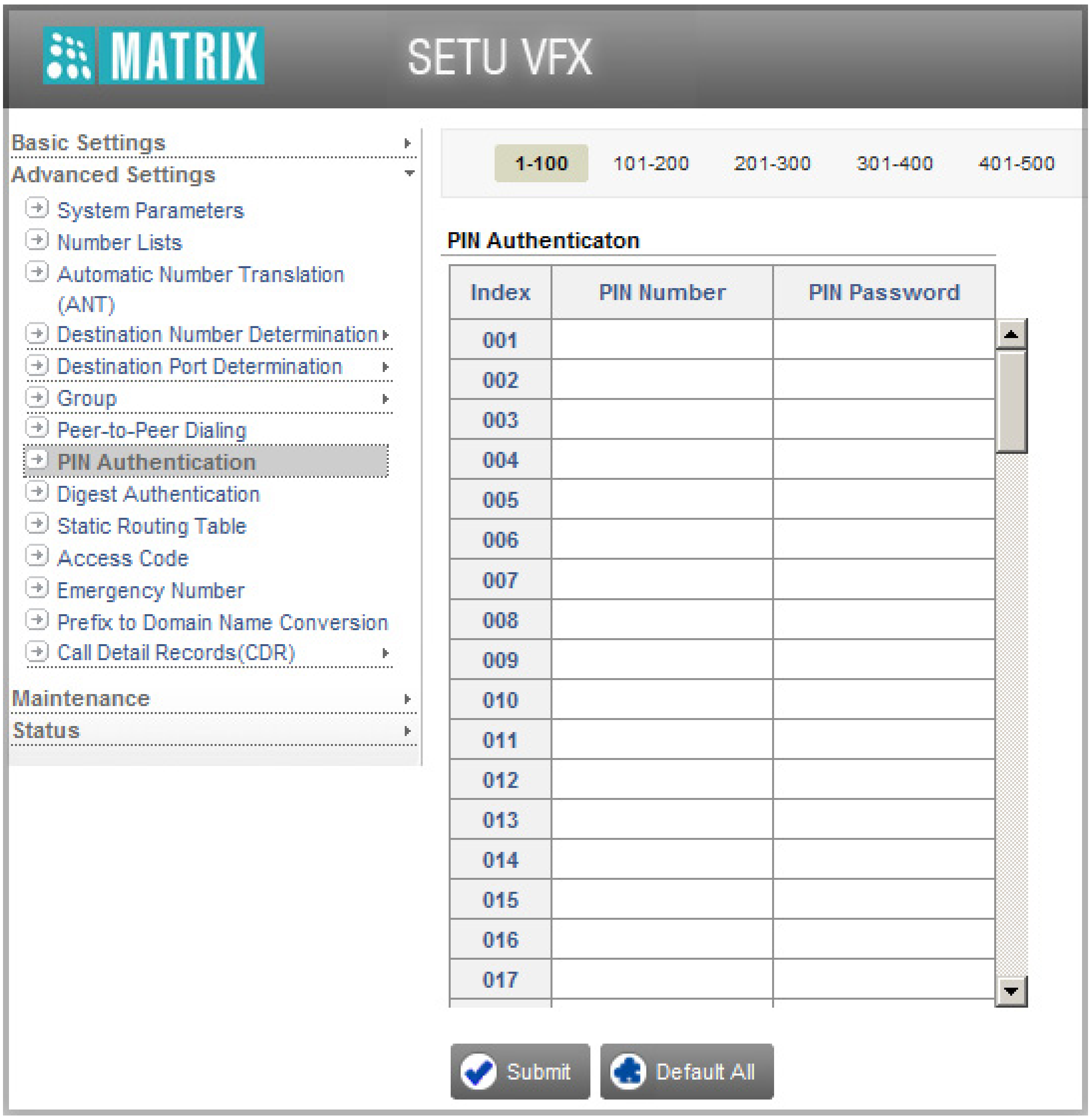
You can also enable PIN Authentication and configure the PIN Authentication Table on the FXO Port page under *Basic Settings*.

If you have not configured the PIN Authentication Table on FXO Port, you may configure the PIN Authentication table now.

To configure PIN Authentication table,

- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **PIN Authentication** link.



- Now, configure the **PIN Authentication** table.
    - In the **PIN Number** column, enter the numbers with which callers will authenticate themselves. Default: Blank. The digits 0 to 9, * and # are allowed in 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 an authenticating password in the **PIN Password** field. 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.

- Now, enable PIN Authentication on the desired FXO Ports on which you have selected the incoming call routing option **After Answering the Call and Collecting Digits** under *Handling of Incoming Calls*. Enable **Prompt caller to enter PIN** on the port. For instructions, see "FXO Port" under *Basic Settings*.

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 VFX supports Digest Authentication. The Digest Authentication feature works on the basis of the Digest Authentication Table, in which the credentials, namely the User Name and Passwords of trusted/authorised calling party SIP devices are stored. You must enable the Digest Authentication on the SIP Trunk and configure the Digest Authentication table.

SETU VFX 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 VFX 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 VFX 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 VFX will consider it as invalid authentication information and reject the call.

You may use Digest Authentication to:
- restrict access to SETU VFX to specific callers.
- prevent unwanted or malicious calls.

Configuring Digest Authentication

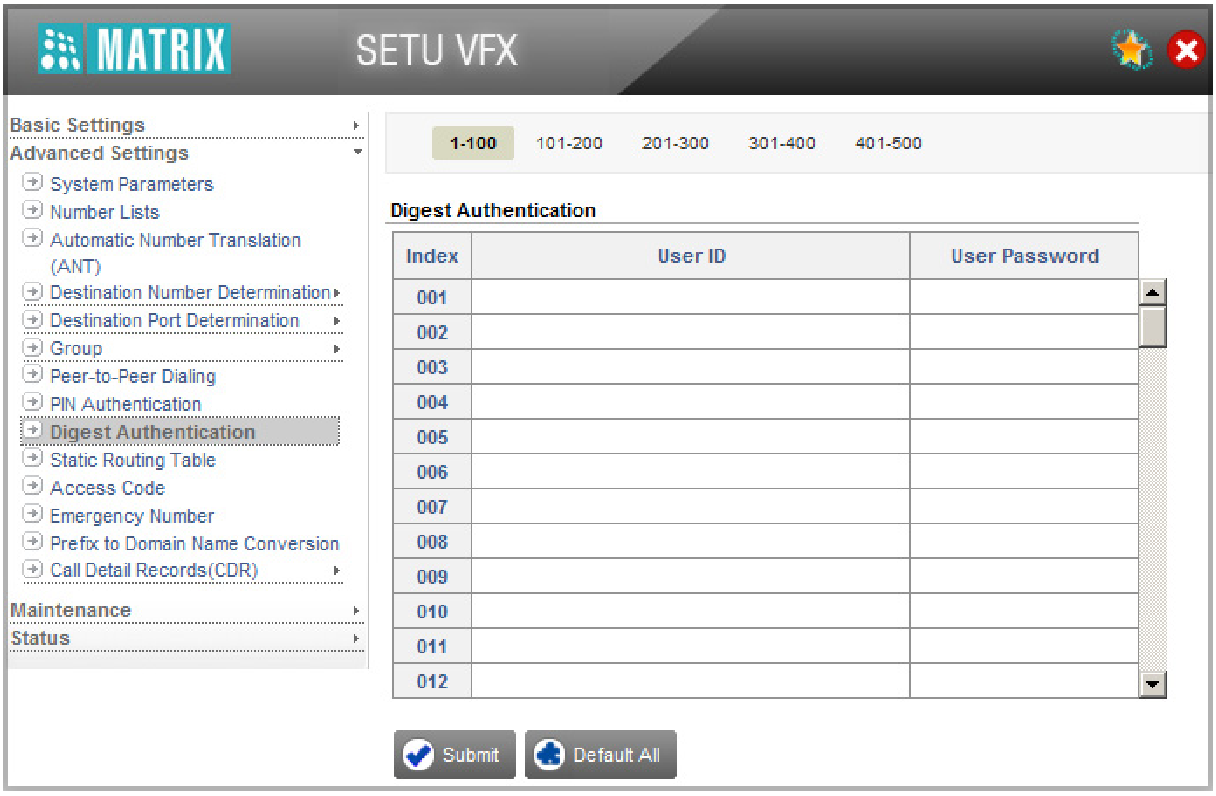
To use this feature, you must enable **Digest Authentication** on the desired SIP Trunk and configure the Digest Authentication Table.

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.



- Enter the user name assigned to the caller/calling device in the **User ID** field. SETU VFX 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.

- Enter the password to authenticate the user ID in the **User Password** field. 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.

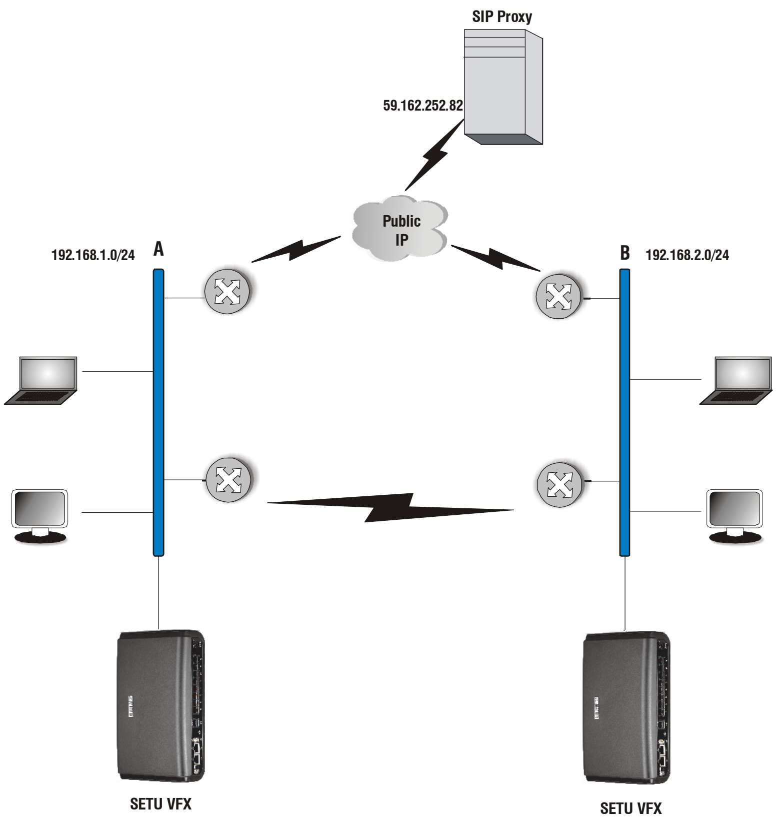
- Click **Submit** to save the entries.
- Now, enable Digest Authentication on the desired SIP Trunk. See “SIP Trunks” under *Basic Settings* for instructions.

Static Routing Table

Static Routing Table is required when you have more than one router (gateway) in your network and you want SETU VFX 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 VFX 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 VFX sends packets, if the final destination IP Address and SETU VFX are not in the same Subnet, the system will check the Static Routing Table.

If a perfect match is found, SETU VFX will start sending the IP packets to the corresponding Gateway Address configured in the table.

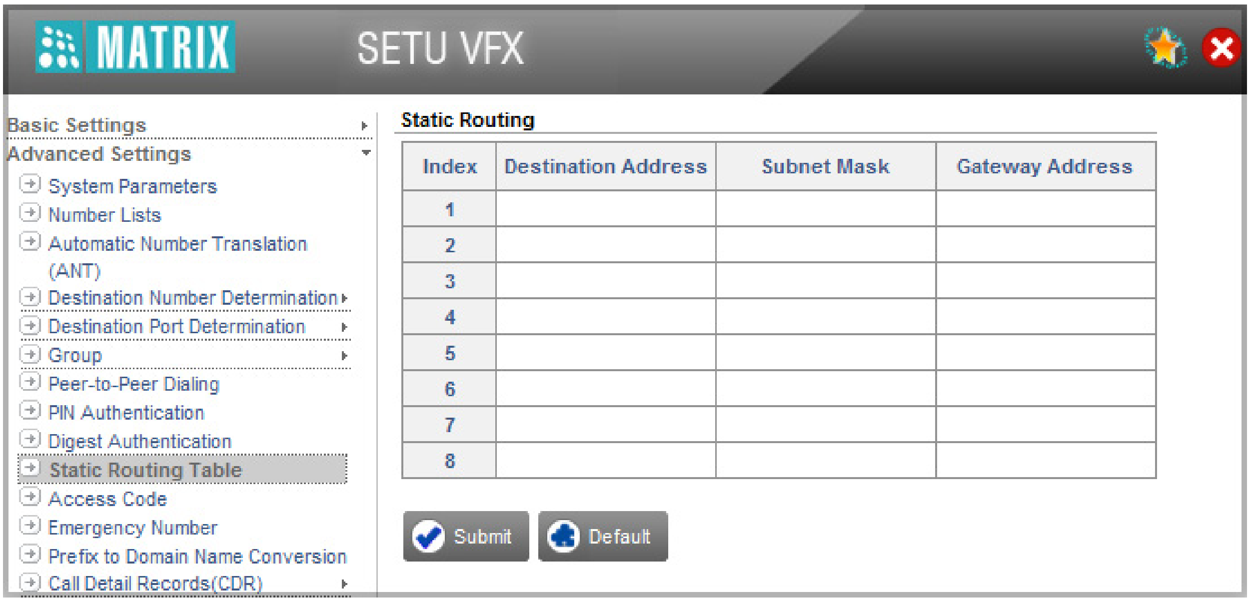
If no match is found, SETU VFX will send the IP Packets to the **Default Gateway Address** (Network Connection Type) you configured in "Network Parameters" page.

The Static Routing Table is common for all SIP Trunks.

Configuring Static Routing Table

The Static Routing Table must be configured at each location where SETU VFX is installed. To configure the Static Routing Table,

- Log into Jeeves.
- Click the **Advanced Settings** link on the left navigation bar to expand.
- Click the **Static Routing** link. The Static Routing Table page opens.



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 fields:

- **Destination Address:** This is the address of the final destination where the call is to be made. This can be a device IP Address or Network Address.
- **Subnet Mask**: This is the mask to be applied on 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 VFX.

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

  To take the above example further, the Static Routing Table of SETU VFX 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 VFX at location A.

    With the Static Routing Table configured thus, all calls made by SETU VFX 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 VFX to address other than 192.168.2.0/ 24 will be routed through the Default Gateway.

    Similarly, configure the Static Routing Table in SETU VFX at location B to enable calling from Location B to Location A.

Port Forwarding

Port Forwarding is redirecting of packets sent via the public network to their intended destination within a private network.

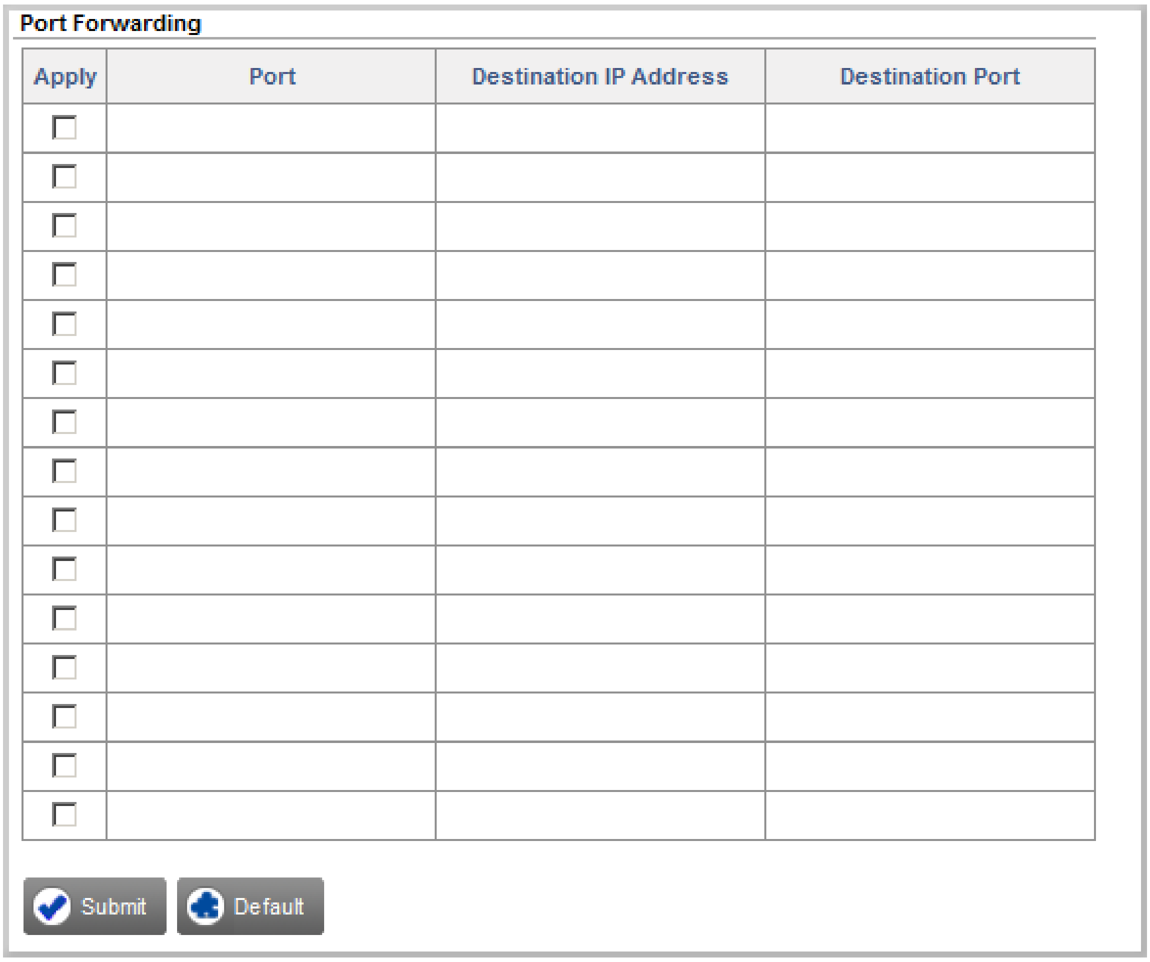
Port Forwarding functionality allows the remote computers on the public network to get connected with specific computers in the private network. In this way the data or services of your computer such as FTP server, Web server, File server or Gaming applications can be shared with the devices located in the public network.

Port Forwarding in SETU VFX helps in distributing the network traffic received on the WAN Port, between specific LAN devices.

Configuring Port Forwarding Table

To configure Port Forwarding,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Port Forwarding** link.



The Port Forwarding Table allows you to configure upto 15 entries.

For each entry, you must configure the following parameters:

- Select the **Apply** check box to enable the Port Forwarding rule.
- In **Port**, enter the WAN Port number whose packets are to be forwarded. Valid port range is 1–65535. Default: Blank.
- In **Destination IP Address**, enter the IP Address of the destination device within the private network, where you want to forward the packets received on a specific WAN Port. The Destination IP Address can be of maximum 15 characters. Valid characters are 0-9 and dot (**.**). Default: Blank.
- In **Destination Port**, enter the destination port number to which you want to forward the packets received on a specific WAN Port. Valid port range is 1–65535. Default: Blank.

If LAN IP Address or Subnet mask is changed, the Port Forwarding parameters will be set to default.

Access Codes

Access Code is a string of digits dialed to use a feature. SETU VFX users, can access the following features and facilities by dialing the Access Codes assigned to them from a phone.

Feature/Function	Default Access Code
System Engineer (SE) Programming	#19[a]
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[a]
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
Access Life Line Port	#*

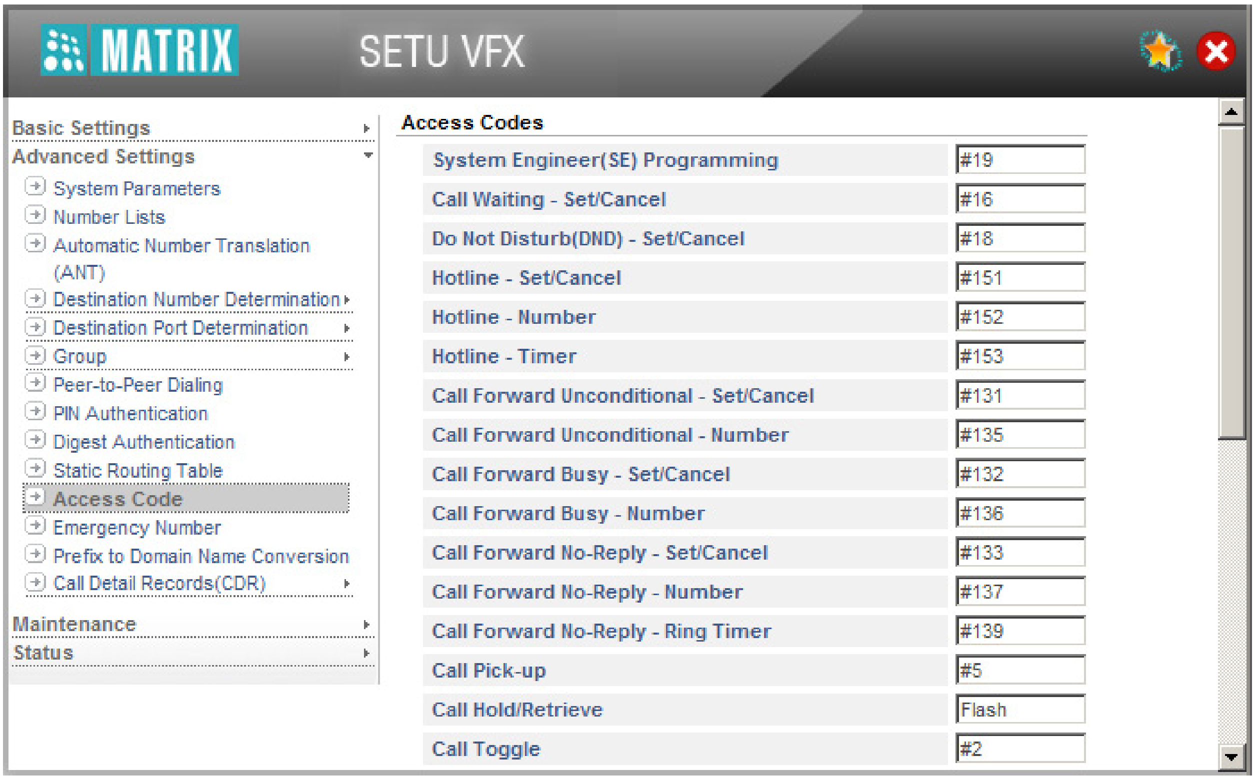
a. Non-programmable.

You can change the default access codes assigned 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.



- Change the default access code for the feature/facility, as required. Access Codes can be of maximum 4 digits and digits 0-9 and the characters *, # , ^ are allowed.
- Click **Submit** to save changes.

- *Make sure that Access Codes you have configured do not conflict with the Emergency Numbers.*
- *Access Codes have priority over the Destination Numbers.*
- *Emergency Numbers have priority over Access Codes.*
- *Access Codes can be dialed from the FXS Port, even if outgoing calls are disabled.*
- *Access Codes can be dialed only from the phones connected to the FXS Ports, except for 'Making New Call' and 'Disconnect Call' which can be dialed from any port.*
- *If the configuration you have does not support FXS Ports, you will only be able to use the Access Codes for 'Making New Call' and 'Disconnect Call'.*

Emergency Number Dialing

SETU VFX 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 more numbers of emergency services as per your requirement and assign Routing Group for the numbers in this table.

The Emergency Number Table stores up to 10 numbers, including those loaded by default.

- *For a few Regions, the system may not load the 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 VFX fails.*
- *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 VFX will dial out an emergency number only if the FXO Ports/SIP Trunks included in the Routing Group for the number are enabled.

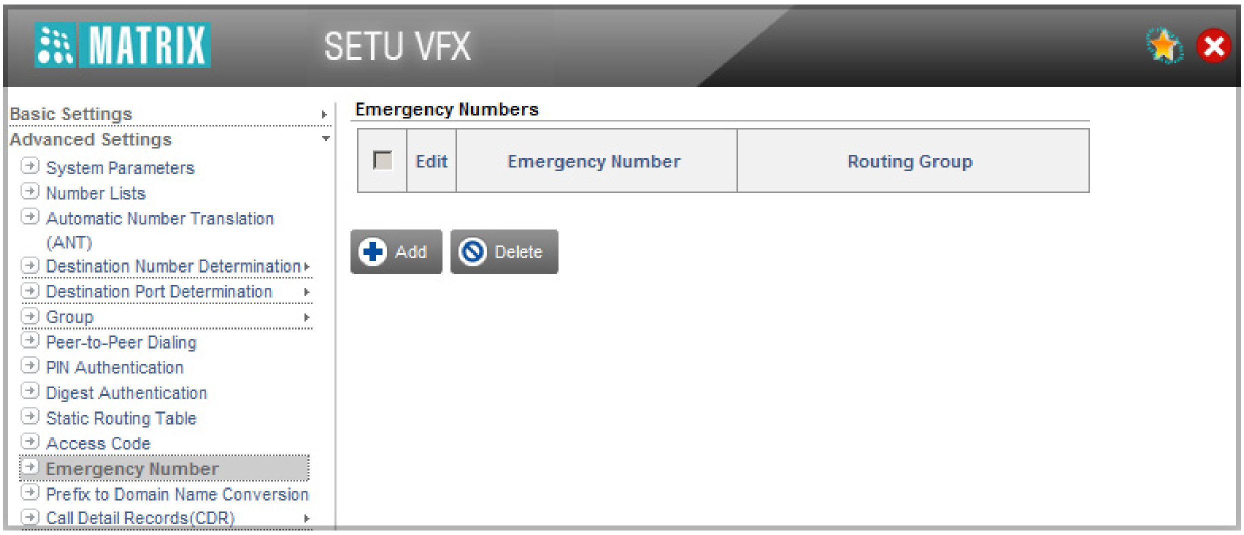
Emergency Number can be dialed even when the option **Block all calls through this FXS Port** is enabled on the FXS Ports.

Configuring Emergency Numbers

To configure the Emergency Number Table,

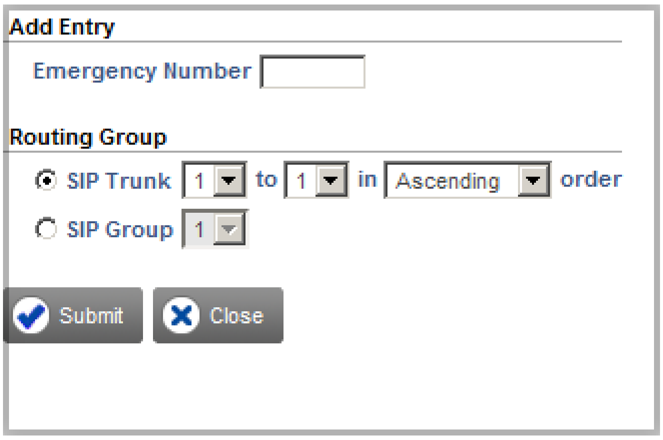
- Log into Jeeves.
- Click the **Advanced Settings** link.

- Click the **Emergency Number** link.



- To **Add** an Emergency Number to the table, click the **Add** button.
- To **Edit** an Emergency Number and or assign a Routing Group, click the settings icon of that number.

  A new window opens, to allow you to add/edit the entry.



  - In the **Emergency Number** field, 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* **SIP Trunks** as members,
    - Select the desired **SIP Trunk** numbers as members.
    - In the **in-order** box, select the order in which the system should hunt for a free member to route the call.

To start hunting from the first to the last member, select **Ascending**.

To start hunting from the last to the first member, select **Descending**.

Default: Ascending.

- To create a group of *non-sequential* **SIP Trunks** as members,
    - Select a **SIP Group**.
    - Select **SIP Group** number.
    - Click **Settings** . The **SIP Trunk** window opens. Create the SIP Group. For detailed instructions, see “Group”.
- Similarly, you can create group of *sequential* and *non-sequential* FXO Ports as members.

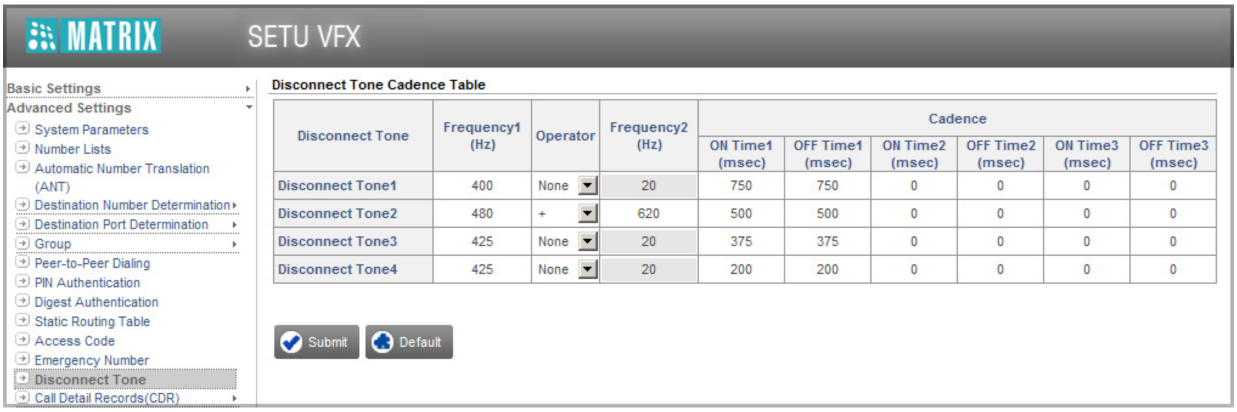
- Click **Submit** to save changes you made. Close the **Add Entry/Edit Entry** window.
- The entries you added appear on the Emergency Numbers page.

Disconnect Tone

If Call Disconnection is signalled by your CO Network in the form of Disconnect Tone, configure **Disconnect Tone** 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 your CO network. To do this,

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Disconnect Tone** link. The Disconnect Tone Cadence Table opens.



- For each Disconnect Tone, set the following parameters:
  - **Frequency 1 (Hz):** Frequency 1 is from 300 to 1400 Hz.
  - **Operator:** This parameter has 3 options: No operator, Modulation (*), Addition (+)

    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 Time3 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 VFX supports multiple SIP Trunks and FXS Ports. When a FXS Port user dials a SIP number, SETU VFX routes the call to the IP network using the SIP Trunk determined by the routing mechanism. The FXS user can dial only numbers, not domain names. Therefore, it becomes necessary that the domain names be assigned Prefix codes which the FXS user can dial.

Now, it is necessary that the number string dialed by SETU VFX 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, 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 VFX is configured to route calls made to domain 'abc.com' from the FXS Port through the SIP Trunk subscribed with the ITSP 'Pulver.com'.
- Since the FXS 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'.
- Now, when FXS user wants to dial the SIP ID 9874@abc.com, the user must dial *234 followed by 9874.
- SETU VFX 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 user sets Call Forward to *2349874 (i.e. 9874@abc.com).
- When an external caller calls the FXS 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 VFX 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.

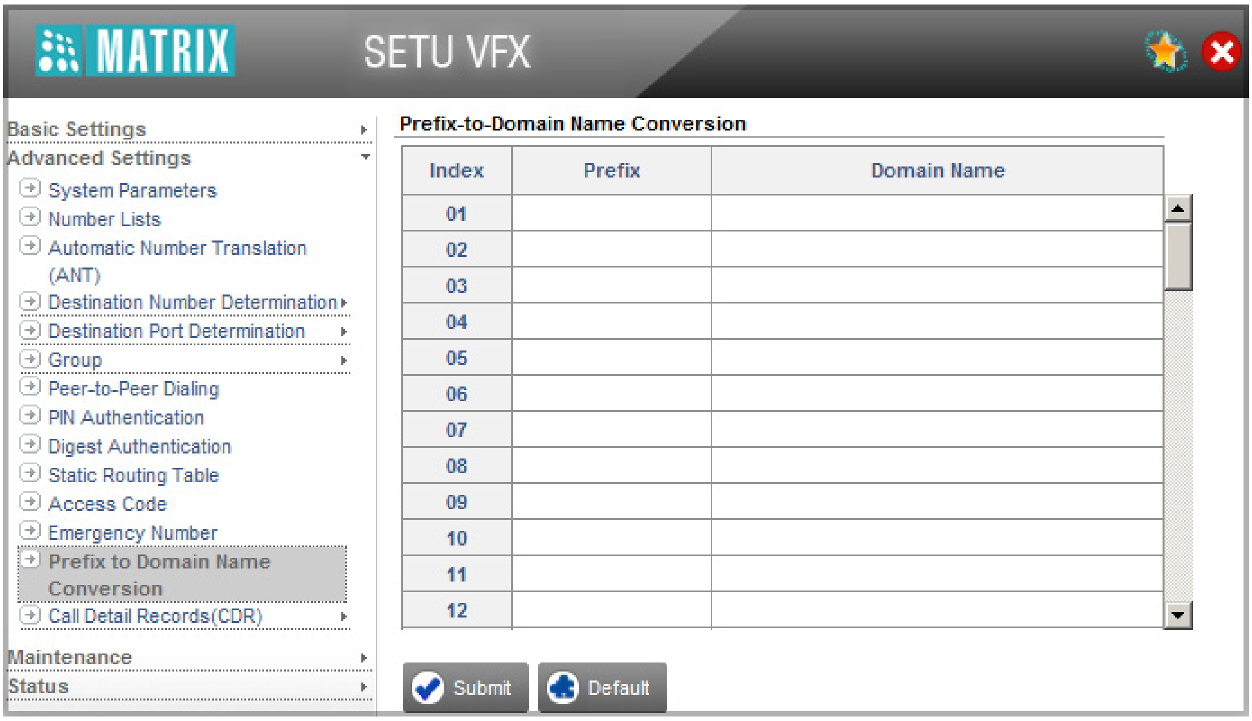
To configure Prefix to Domain Name Conversion,

- Log into Jeeves.

- Click the **Advanced Settings** link.

- Under **Advanced Settings**, click **Prefix to Domain Name Conversion**' link.

- The Prefix to Domain Name Conversion table opens. You can store up to 64 entries in this table.



- In the **Prefix** column, 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 in the **Domain Name** field. The Domain Name may consist of a maximum of 40 characters.

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

- You may log out of Jeeves.

Certificate Manager

SETU VFX supports certification for TLS, Web Server, Firmware Upgrade, Configuration Upgrade and TR-069.

SETU VFX 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.

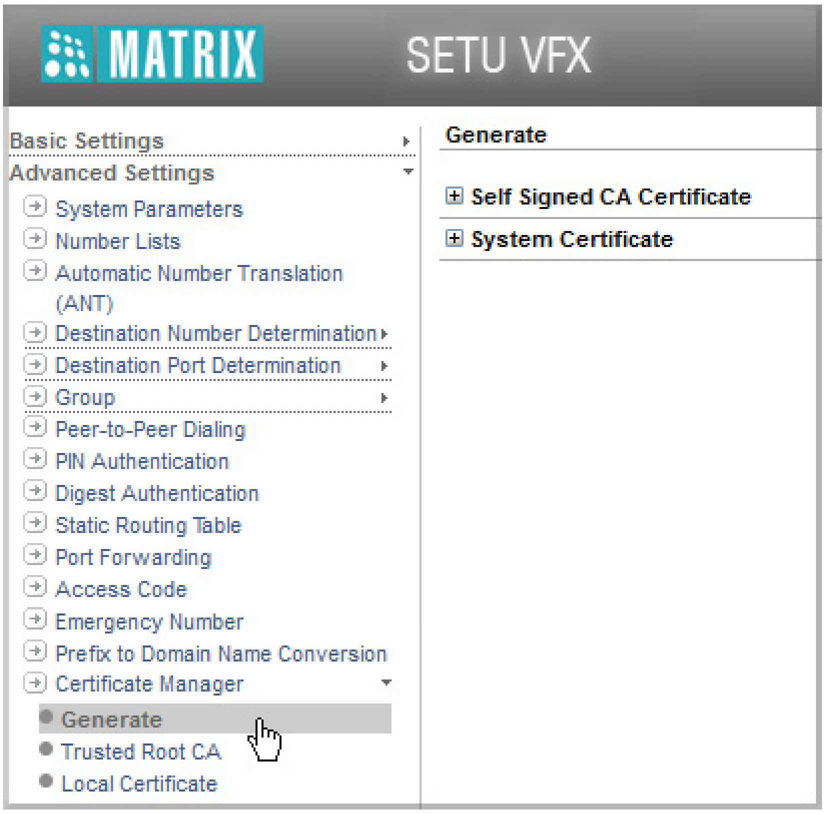
If you select **Self-Signed Certificate**, you need to do the following:

1. Create a Self-Signed CA Certificate.
2. Create a System Certificate (Self-Signed Certificate).

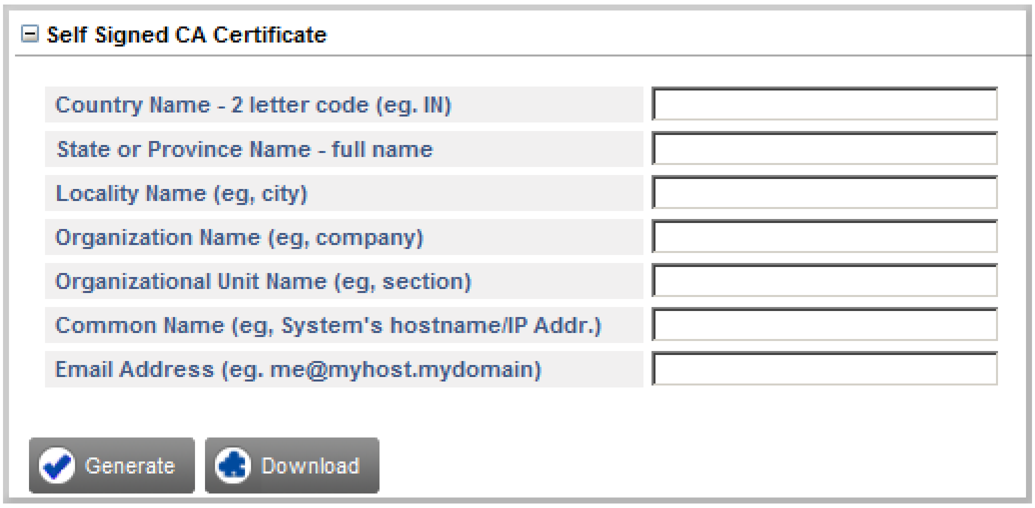
Generating a Self-Signed CA Certificate

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Certificate Manager** link.

- Click the **Generate** link.



- Click **Self Signed CA Certificate** to expand and configure the following parameters.



- 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 VFX is installed.

- In **Organizational Unit Name (eg. section)**, enter the name of the unit or section or domain of your organization, where SETU VFX is installed.

- In **Common Name (eg. System's hostname/IP Addr.)**, enter your Server's (SETU VFX) 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.

- 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)

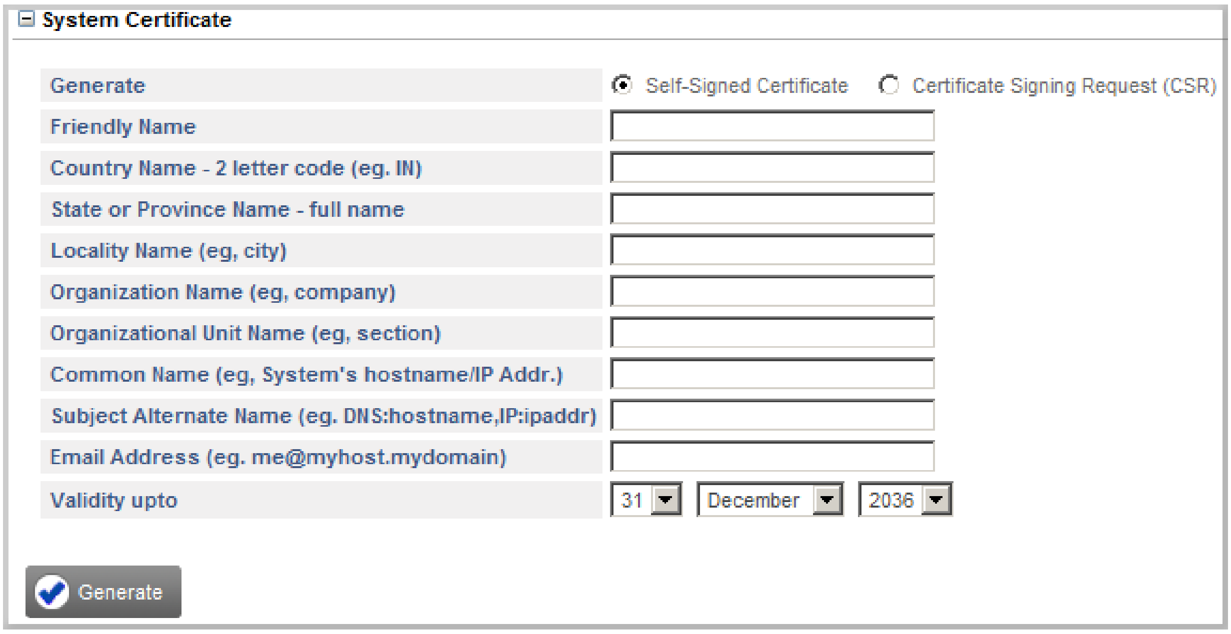
After creating a Self-Signed CA Certificate, you can either,

- 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,

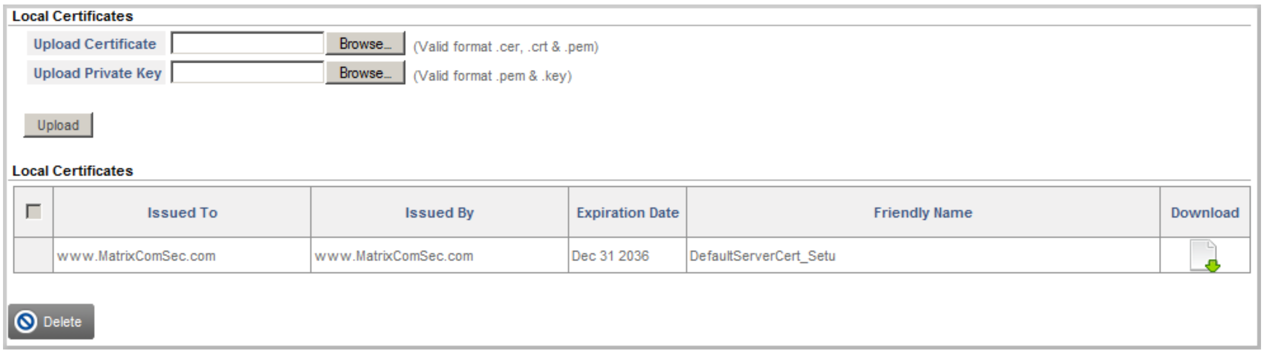
- Click the **Certificate Manager** link.
- Click the **Generate** link.
- Click **System Certificate** to expand and configure the following parameters.



- In **Generate**, select the type of certificate you want to create. You must select **Self-Signed Certificate**.
- In **Friendly Name**, enter the name you want to assign to the certificate.
- In **Country Name - 2 letter code (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 VFX is installed.
- In **Organizational Unit Name (eg. section)**, enter the name of the unit or section or domain of your organization, where SETU VFX is installed.
- In **Common Name (eg. System's hostname/IP Addr.)**, enter your Server's (SETU VFX) 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. The generated certificate appears in the **Local Certificates** table.



- If you want to upload other System Certificates, in **Upload Certificate** browse the location at which the certificate is saved. Along with the certificate you also need to upload the Private Key, in **Upload Private Key** browse the location at which the key is saved and click **Upload**.

  The System Certificate you uploaded appears in the **Local Certificates** table.Valid formats for certificate are .cer, .crt and .pem. Valid format for key are .pem and .key (Base64 encoded ASCII file).

- To delete a System Certificate, select the check box of the respective Certificate and click **Delete**.
- To download the System Certificate, click **Download** .

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:
```

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

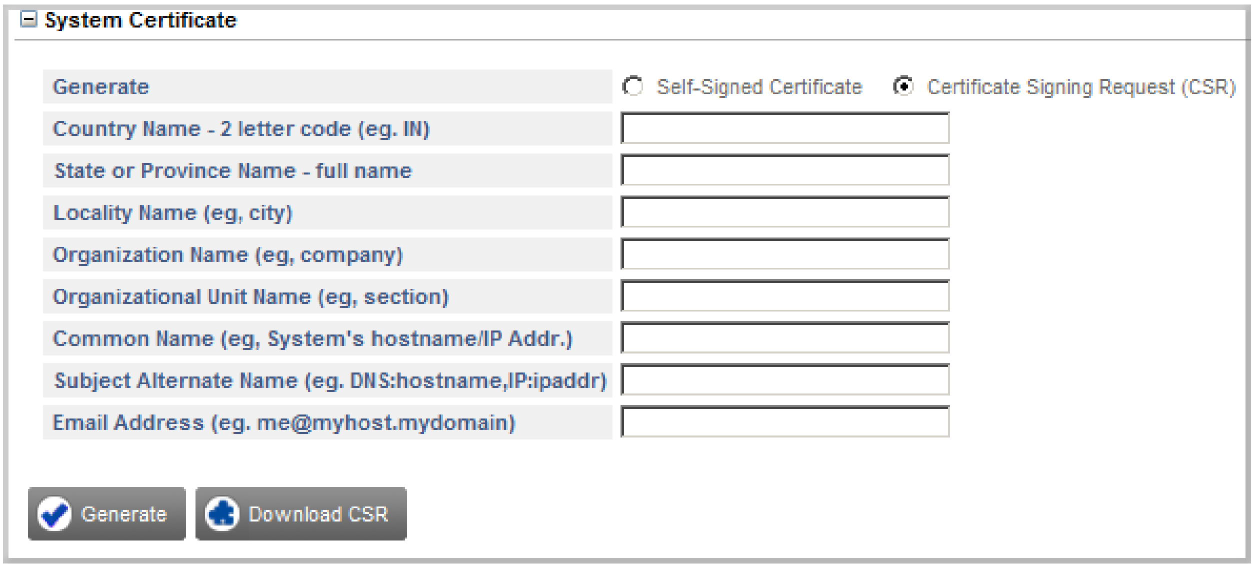
Certificate Authority (CA) is a trusted organization which creates and sells TLS Certificates to websites. *CA Signed Certificates* are the TLS Certificates which are created by such trusted CAs, signed and sold to any applicant. These certificates contains a public key and the identity of the owner; 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.
- Click the **Generate** link.
- Click **System Certificate** to expand and configure the following parameters.



- In **Generate**, select the type of certificate you want to create. You must select **Certificate Signing Request (CSR)**.
- In **Country Name - 2 letter code (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 VFX is installed.
- In **Organizational Unit Name (eg. section)**, enter the name of the unit or section or domain of your organization, where your SETU VFX is installed.
- In **Common Name (eg. System's hostname/IP Addr.)**, enter your Server's (SETU VFX) 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-----
MIIDLDCCAhQCAQAwgaoxCzAJBgNVBAYTAk1OMRAwDgYDVQQIEwdHdWphcmF0MREw
DwYDVQQHEwhWYWRvZGFyYTEgMB4GA1UEChMXTUFUUk1YIENPTVNFQyBQV1QuIExU
RC4xDDAKBgNVBAsUA1ImRDEdMBsGA1UEAxMUd3d3Lk1hdHJpeENvbVN1Yy5jb20x
JzAlBgkqhkiG9w0BCQEWGFN1cHBvcnRATWF0cml4Q29tU2VjLmNvbTCCASIwDQYJ
KoZIhvcNAQEBBQADggEPADCCAQoCggEBAPutA1/cZcz/qZe3soIITiVpPI8PIZ6d
9RvInx4haqVob7Ml10dYWvN2rLmFod3ZtEu9dX645crC4NXn9pxKXmkp5iNdBVca
rm1qedZ63S1cR3m4YhL2dUc7DQ9T1GNTPbXLr1A4sQk+nVwO+C+XU/jP1pqiR0sn
Idh2/eLWVOauRgY3qdGjPaN8ndq8xVieY+v1/XpLQa4Oyd6aP+xn+z4pSWK4YLeP
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YjoC/gbrhSUwgbR4Jh+13guInViTyJ5YDt9pLc8xzJe23MV2XDv4ImSSUSkRojcg
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MbwSosS/QM7ZjE147b13m9Lk69jdzfSAPmCW4Adu1Be7PENGGI+MMzfAVyYSwdkw
-----END CERTIFICATE REQUEST-----
```


Enrolling the Certificate Signing Request with CA

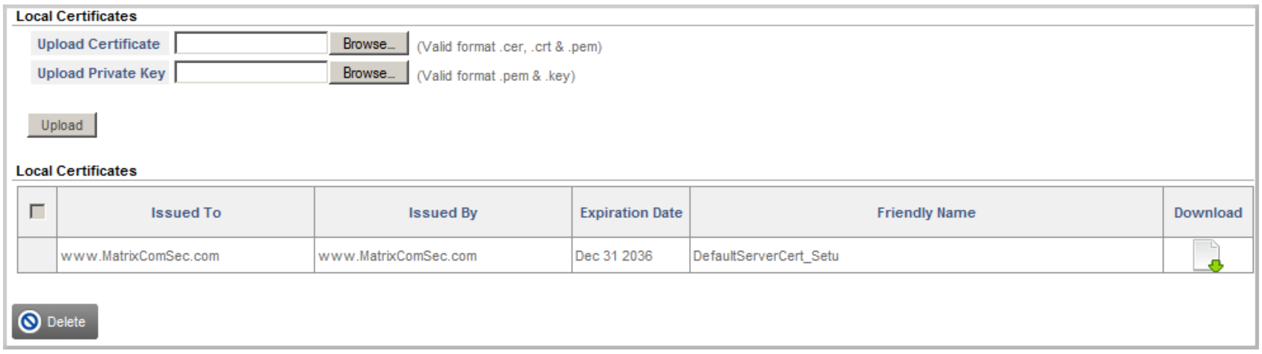
Enrollment is a process of obtaining a certificate from any trusted third party (CA). After you have generated the Certificate Signing Request (CSR), you must contact any authorized third party that issues TLS Certificates to companies or web owners, such as Thawte, VeriSign, etc. and enroll the Certificate 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.



- 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 Records (CDR)

SETU VFX 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 date
- 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 those 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 VFX supports maximum 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.

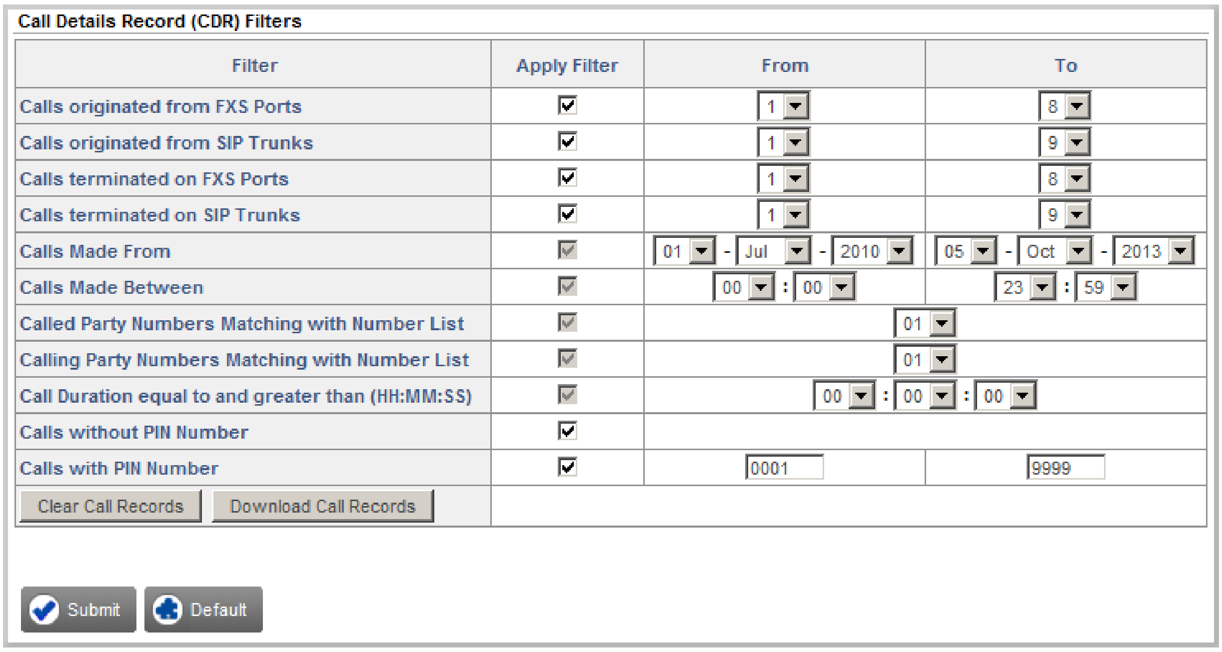
Call records can be cleared manually at any time.

Configuring Call Detail Record Filters

- Log into Jeeves.
- Click the **Advanced Settings** link.
- Click the **Call Detail Record (CDR)** link.

Setting Filters

- To set filters, click the **Filters** link under Call Detail Record.



Call Details Record (CDR) Filters

Filter	Apply Filter	From	To
Calls originated from FXS Ports	☑	1	8
Calls originated from SIP Trunks	☑	1	9
Calls terminated on FXS Ports	☑	1	8
Calls terminated on SIP Trunks	☑	1	9
Calls Made From	☑	01 - Jul - 2010	05 - Oct - 2013
Calls Made Between	☑	00 : 00	23 : 59
Called Party Numbers Matching with Number List	☑	01	
Calling Party Numbers Matching with Number List	☑	01	
Call Duration equal to and greater than (HH:MM:SS)	☑	00 : 00 : 00	
Calls without PIN Number	☑		
Calls with PIN Number	☑	0001	9999
Clear Call Records / Download Call Records			

Submit Default

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 will generate report of the calls originated from FXS Ports, that is, the calls that were received on the FXS Ports of SETU VFX for further routing. To generate report using this filter, set the range of the FXS Ports in the From and To fields.

  You can also generate report of a single FXS Port, by setting the same port number in the **From** and **To** fields.

- **Calls originated from SIP Trunks:** The system will generate report of the calls originated from SIP Trunks, that is, the calls that were received on the SIP Trunks of SETU VFX for further routing. To generate report using this filter, set the range of the SIP Trunks in the **From** and **To** fields.

  You can also generate report of a single trunk, by setting the same trunk number in the **From** and **To** fields.

- **Calls originated from FXO Ports:** The system will generate report of the calls originated from the FXO Ports. Set the range of the FXO Ports in the **From** and **To** fields.

  You can generate report of a single FXO Port, by setting the same port number in the **From** and **To** fields.

- **Calls terminated on FXS Ports:** The system will generate report of the calls terminated on FXS Ports. To generate report using this filter, set the range of the FXS Ports in the **From** and **To** fields.

You can also generate report of a single FXS Port, by setting the same port number in the **From** and **To** fields.

- **Calls terminated on SIP Trunks:** The system will generate report of the calls terminated on the SIP Trunks. To generate report using this filter, set the range of the SIP Trunks in the **From** and **To** fields.

  To generate report of calls terminated on a single SIP Trunk, set the same trunk number in both fields.

- **Calls terminated on FXO Ports:** The system will generate report of the calls terminated on the FXO Ports. Set the range of the FXO Ports in the **From** and **To** fields.

  You can generate report of a single FXO Port, by setting the same port number in the **From** and **To** fields.

- **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 of calls that were made using PIN Authentication. You can generate report of calls of specific PIN Numbers.

  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 it 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.

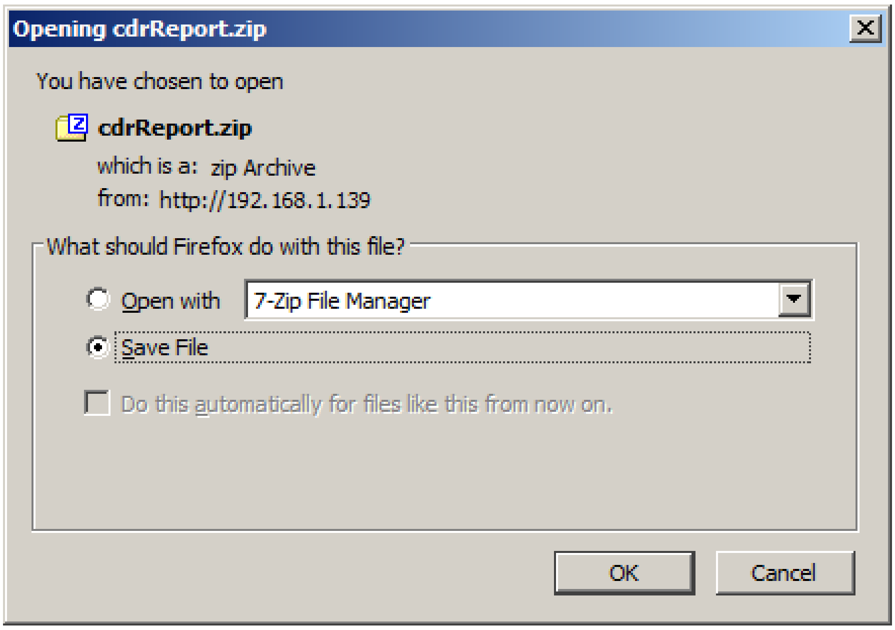
When call records are cleared, the **From** field of the filter **Calls Made Between** will change to the date of clearing of the records.

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.

A sample **Call Detail Record Report** is presented at the end of this topic.

Viewing Call Detail Report

- To view the report generated by the system for the filters you set, click the **Report** link under Call Detail Record.

Call Detail Record(CDR) Report

Sr. No.	Date	Start Time	Calling Number	Called Number	Duration (sec)	Source Port	Destination Port
0001	31-Aug-2013	13:05	2001	2536	00:00:27	FXS-1	SIP-1
0002	31-Aug-2013	13:04	2001	1235	00:01:40	FXS-1	SIP-1
0003	31-Aug-2013	18:40	1000@192.168.101.123		00:00:01	SIP-5	FXS-1
0004	31-Aug-2013	21:40	2001	9	00:00:17	FXS-1	FXS-2
0005	31-Aug-2013	21:41	2003	2222	00:00:00	FXS-3	FXS-4
0006	31-Aug-2013	21:43	2001	9	00:00:08	FXS-1	FXS-2
0007	31-Aug-2013	21:48	2003	2222	00:00:09	FXS-3	FXS-4
0008	31-Aug-2013	21:57	2003	2222	00:00:00	FXS-3	FXS-4
0009	31-Aug-2013	21:59	2003	2222	00:01:27	FXS-3	FXS-4
0010	31-Aug-2013	21:59	2001	9	00:01:48	FXS-1	FXS-2
0011	31-Aug-2013	22:03	2001	9	00:00:10	FXS-1	FXS-2
0012	31-Aug-2013	22:03	2003	2222	00:00:01	FXS-3	FXS-4
0013	31-Aug-2013	22:08	2001	9	00:00:46	FXS-1	FXS-2
0014	31-Aug-2013	22:08	2003	2222	00:00:44	FXS-3	FXS-4
0015	02-Sep-2013	09:16	2001	9	00:00:08	FXS-1	FXS-2

Total Records : 181 1 2 3 4 5 6 7 8 9 10 >>

- Call Detail Record Report generated as per the filters you set will appear in the following columns:

  - **Date:** Calls made between particular dates.
  - **Start Time:** Calls made between a particular time period.
  - **Calling Number:** Calls received from specific numbers.
  - **Called Number:** Calls made to specific numbers.
  - **Duration:** Calls of a specific time duration.
  - **Source Port:** Calls originated from the SIP Trunks/ FXO Ports/ FXS Ports.
  - **Destination Port:** Calls terminated on the SIP Trunks/ FXO Ports/ FXS Ports.
  - **Disconnected By:** The port/ channel that disconnected the call.
  - **Cause:** The cause for disconnection.
  - **PIN Number:** Calls made using PIN Authentication, the PIN Number dialed by the caller.
  - **Remarks:** The type of call. A for Anonymous, U for Unanswered and N for Normal.
  - **By Port:** Displays the FXS Port number that uses the features as Call Forward, Blind Transfer, Attended Transfer.
  - **By Number:** Displays the number assigned to the FXS Port.

  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.

CHAPTER 6 Features

SETU VFX 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
- Message Wait Indication on SIP Trunks
- Call Pick-up
- Hotline
- Supplementary Services of Service Providers
- Making a new call using Access Code
- Disconnecting a call using Access Code

*Except **Making a New Call using Access Code** and **Disconnecting a Call using Access Code**, all other features are applicable only for VoIP-FXS Gateway.*

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 VFX offers users IP Dialing and Fax over IP (FoIP).

Call Hold[21]

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 while using the following features:

- Retrieve Held Call
- Make a Second Call
- Call Toggle
- Call Conference

21. Applicable only for VoIP-FXS Gateway.

- 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 Ports” 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[22]

You can make a second call, by putting the current call on hold.

*To use this feature, make sure that **Call Hold** is enabled in the **Class of Service** of the FXS Port. For instructions, see "FXS Ports" 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.

22. *Applicable only for VoIP-FXS Gateway.*

Call Toggle[23]

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 Ports" under Basic Settings.*

How to use 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.

23. *Applicable only for VoIP-FXS Gateway.*

Call Transfer - Attended[24]

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 VFX enables the transferrer to know the result of the transfer activity, whether the call has been transferred successfully or not.

As soon as the transferrer dials the Attended Transfer access code to transfer the call, SETU VFX loads the Transfer Notification Timer.

SETU VFX indicates to the transferrer of the result of the transfer activity within this Timer. The Transfer Notification Timer is configurable.

This is how Attended Transfer works:

- A (transferrer) 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 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 is not stopped even if the transferrer 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 transferrer (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 Ports" 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, see "System Parameters".

24. *Applicable only for VoIP-FXS Gateway.*

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[25]

Blind Transfer is when you transfer the call to the desired party, without informing the party of the transfer.

SETU VFX enables the transferrer to know the result of the transfer activity, that is, whether the call has been transferred successfully or not. As soon as the transferrer dials the Blind Call Transfer access code, SETU VFX loads the Transfer Notification Timer. This timer stops if the transferrer goes On-hook. SETU VFX indicates to the transferrer of the result of the transfer activity within this Timer. The Transfer Notification Timer is configurable.

This is how Blind Call Transfer works:

- A (transferrer) 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 (transferrer) 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 (transferrer) gets busy tone, and speech is established between A and the transferee B.

- If no message is received during the Transfer Notification Timer, A (transferrer) 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 Ports” 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.

---

25. *Applicable only for VoIP-FXS Gateway.*

When you transfer a call and remain Off-hook, that is, do not replace your handset after dialing the transfer target'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, that is, 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[26]

When you are away from your phone[27], but would like to answer your calls, you can use the Call Forward feature of SETU VFX to forward incoming calls received on the FXS Ports to another number.

SETU VFX supports the following Call Forward options, which you can set on your phone:

- **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 VFX waits for the duration of the **Call Forward-No Reply Timer** for the phone to answer. This timer is configurable, and is set to 45 seconds as default.

  If the phone does not answer before the timer expires, SETU VFX 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 i.e. when the FXS Port is in speech with another port.*

Configuring Call Forward

To use this feature, 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 Ports” under *Basic Settings*.

For this feature to work, the Destination Port Determination method should be selected as Fixed, and either **FXS Port** or **FXS Group** should be selected as **Routing Group** and **Fallback Routing Group**.

How to use Call Forward

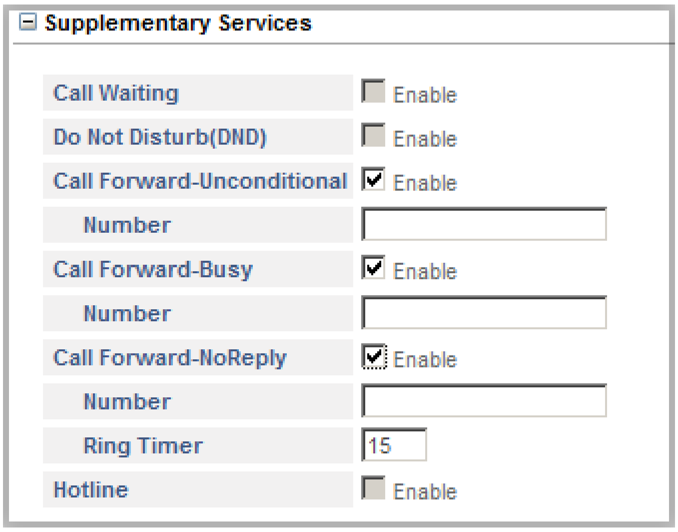
You can set/cancel Call Forward from Jeeves and from the phone connected to the FXS Port.

To set or cancel Call Forward from Jeeves,

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

26. *Applicable only for VoIP-FXS Gateway.*
27. *Connected to the FXS Port of SETU VFX.*

- On the FXS Port page, click the **Supplementary Services** link to expand.



- Choose a Call Forward option, **Call Forward-Unconditional**, **Call Forward-Busy**, **Call Forward-No Reply**, and select the respective check box to set the desired Call Forward. Default: Disabled.

- For the Call Forward option you selected, in the corresponding **Number** field that appears, enter the desired destination number (up to 40 characters) you want the calls to be forwarded to.

- If you selected **Call Forward- No Reply**, you may also change the duration of the no-reply **Ring Timer**, if required. The range of the Call Forward No-Reply Timer is 01 to 99 seconds. Default: 15 seconds.

- Click **Submit** to save.

  To cancel any of the Call Forward options, clear the check boxes.

- Click **Submit** to save.

- Log out of Jeeves.

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[28]

Conference-3 Party (also referred to as Three-Way Calling) is a telephone call, in which the calling party 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 are 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 Vocoder selected on SIP should be same.*

Configuring Conference

To use this feature, 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 Ports” 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.

28. *Applicable only for VoIP-FXS Gateway.*

DND[29]

If you do not want to receive any calls on your phone, you may set the Do-Not-Disturb feature on your phone. When you set DND, all incoming calls on your phone will be rejected, but you can continue to make outgoing calls.

- *If Do Not Disturb (DND) and Call Forward-Unconditional are both 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

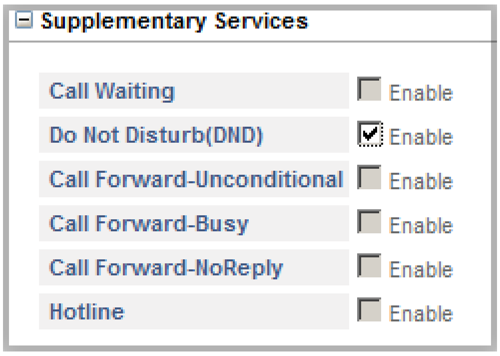
To use this feature, you must enable **Do Not Disturb** in the **Class of Service** of the FXS Port. For instructions on enabling features in the Class of Service of the FXS Port, see "FXS Ports" under *Basic Settings*.

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.
- Under **Basic Settings**, click the **FXS Port** link.
- Click the FXS Port 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.



- 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.

29. *Applicable only for VoIP-FXS Gateway.*

To set or cancel DND from the Phone,

- Lift handset of your phone.
- Dial ***#18-1*** to set.
- Dial ***#18-0*** to cancel.
- Replace the handset.

Call Waiting[30]

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 VFX 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, that is, when the FXS Port is in speech with another port.*
- *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 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 Ports" under *Basic Settings*.

How to use Call Waiting

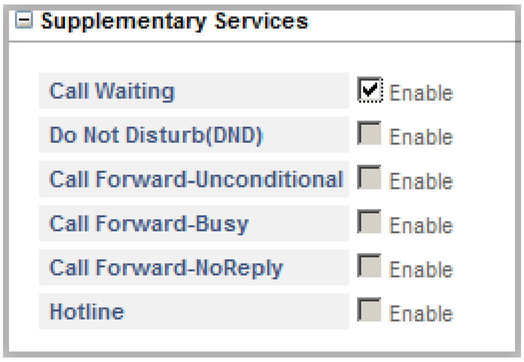
You can set/cancel Call Waiting from Jeeves and from the phone connected to the FXS Port.

To set or cancel Call Waiting from Jeeves,

- Log into Jeeves.
- Under **Basic Settings**, click the **FXS Port** link.
- Click the FXS Port on which you want to use this feature. The page of the selected port opens.

30. *Applicable only for VoIP-FXS Gateway.*

- On the FXS Port page, click **Supplementary Services** link to expand.



- 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[31]

SETU VFX 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 to the 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 the FXS Port:

On the SIP Trunk,

- Enable **Subscribe for MWI** flag.
- Configure the Message Retrieval Number.
- 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 VFX 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.

  For configuration instructions, see “Message Wait Indication (MWI)” under “SIP Trunks”.

On the FXS Port selected as Destination for MWI,

- Select the **Message Wait Notification Type.** Whenever a new message arrives in the Mailbox of the SIP Trunk, SETU VFX gives notification to the destination FXS Port according to the type of Message Wait Notification set on the FXS Port. For instructions to select notification type, under the topic “FXS Ports”, see “General”.

  SETU VFX supports the following types of Message Wait Notification:

  - **Stuttered Dial Tone:** To indicate new message, a stuttered dial tone will be played to the user when the phone connected to the FXS Port goes off-hook. You can change the frequency and cadence of the Stuttered Dial tone, if required. For instructions, see “Call Progress Tones”.
  - **LED Lamp:** If the phone connected to the FXS Port is equipped with a 'Message Wait' lamp, you can set this type of Message Wait Notification. The lamp will blink continuously to indicate arrival of new message. The lamp will be turned off after you have retrieved all the waiting messages.
  - **Ring:** To indicate the arrival of a new message, the *Message Wait Ring* (Short, Fast) will be played on the FXS Port. You can change the Message Wait Ring type, if required. For instructions, see “Region”.

31. Applicable only for VoIP-FXS Gateway.

The FXS Port will ring for the duration of the *Message Wait Ring Timer* (configurable; default: 30 seconds). If the ring is not answered within this timer by the FXS Port user, the system will ring again for as many times as the *Message Wait Ring Count* (configurable; default: 3 times), and at the interval set as the *Message Wait Ring Timer Interval* (configurable; default: 1 minute).

You can change the *Message Wait Ring Timer*, the *Message Wait Ring Timer Interval*, and the *Message Wait Ring Count* to the desired value. For instructions, in the *Advanced Settings* chapter, under "System Parameters", see "Message Wait".

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 Trunks", 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 VFX.
- SETU VFX 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 VFX 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**' field, 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[32]

Call Pick-up feature allows you to answer calls ringing on other phones connected to SETU VFX 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 VFX will store the number of the FXS Port from which the call was answered (from which access code for Call Pick-up was dialed and the speech was connected) as the *Destination port* in the Call Detail Record (CDR). To know more, see "Call Detail Records (CDR)".

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 an 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, in the *Basic Settings* chapter, under the topic *FXS Port*, see "General".

- enable **Call Pick-up** in the **Class of Service** of the FXS Ports you have assigned to Call Pick-up groups. For instructions on enabling features in the Class of Service of the FXS Port, in the *Basic Settings* chapter, under the topic FXS Port, see "Class of Service".

You can also change, if required, the default access code for Call Pick-up, **#5**, to the desired value. For instructions, in the *Advanced Settings* chapter, see the topic "Access Codes".

How to use Call Pick-up

To use this feature, you must know which of the phones connected to SETU VFX are in the same Call Pick-up group as yours. Also, make sure Call Pick-up is included in your Class of Service.

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.

32. Applicable only for VoIP-FXS Gateway.

Hotline[33]

The Hotline feature connects the FXS Port user immediately to a particular number, when the user goes Off-hook. You can use Hotline to dial the number you call most frequently.

Let us understand how this feature works with an example,

- A is Sales Manager who frequently dials the number of the B, the Sales Coordinator.
- Instead of dialing B's number each time, A sets Hotline on his phone.
- A 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 VFX plays dial tone and waits for the duration of the Hotline Timer.
- If A does not dial any digit during this Timer, on the expiry of the Timer, SETU VFX dials out B's number.

Allowed-denied number logic is applied for the hotline number.

Configuring Hotline

To use this feature, 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 Ports" under *Basic Settings*.

How to use Hotline

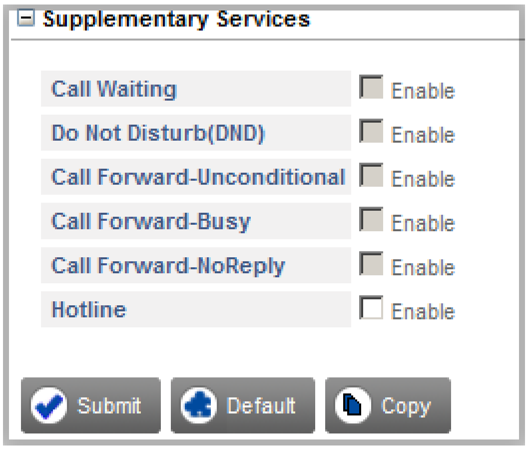
You can set/cancel Hotline from Jeeves and from the phone connected to the FXS Port.

To set or cancel Hotline from Jeeves,

- Log into Jeeves.
- Under **Basic Settings**, click the **FXS Port** link.
- Click the FXS Port on which you want to use this feature. The page of the selected port opens.

33. *Applicable only for VoIP-FXS Gateway.*

- On the FXS Port page, click **Supplementary Services** link to expand.



- Select the **Hotline Enable** check box. Default: Disabled.

  To cancel, clear the check box.

- In the **Number** field, 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.

Supplementary Services of Service Provider[34]

When SETU VFX interfaced with a service provider server (ITSP, an ATA, 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[35], you may choose to access the features of the service provider, or to access primarily the features of SETU VFX.

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 VFX.

If you want to use the features of SETU VFX, you must select ***Gateway*** as the Subscriber Type for the FXS Port of SETU VFX. However, you will be able to access the features of the service provider in the Gateway mode, by dialing Flash, followed by **#4**, the Access Code for Using Supplementary Services of Service Provider.

When the SETU VFX 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 VFX will start an 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 VFX.

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 Ports” under *Basic Settings*.

How to use Supplementary Services of Service Provider

If you have set SETU VFX 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 providers server.

If you have set SETU VFX 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 providers server within 10 seconds.

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34. Applicable only for VoIP-FXS Gateway.
35. To be able to use features of the service provider SETU VFX supports dialing of Flash during speech on SIP.

Making a New Call using Access Code

This feature enables callers to disconnect the current call and make a new call using SETU VFX without getting disconnected from the system.

This feature is useful when you want to allow users to make multiple calls without getting disconnected each time their call ends.

Let us understand this feature with an example:

- A Cyber-Cafe has installed SETU VFX for providing international calling service to the home users.
- The Cyber-Cafe provides home users who have subscribed for this service, a number to call the SETU VFX, a PIN Number and a Password.
- To make international calls, home users must call SETU VFX, dial the PIN Number and Password and the international number. Thus each time they want to make a call, they must repeat this process of dialing over and over again.
- Making a New Call feature eliminates repeated dialing of these numbers.
- After calling SETU VFX, home users can dial their PIN number and Password once and call the international number. At the end of the call, they can dial the Access Code for Making a New Call. They will remain connected to the system and can make another call.
- If the remote end disconnects the call during speech, SETU VFX will play error tone for 4 seconds followed by dial tone. The users 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.*
- *However, if you have enabled **Connect Source Port when number is outdialed** on the FXO 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 FXO Ports. For instructions, under *Basic settings*, see “FXO Port”.

How to make a New Call using Access Code

- When you are in speech during the current call.
- Dial **#91**. Current call will disconnect.
- 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 VFX enables users to disconnect a call using an access code. When the call disconnect access code is dialed, SETU VFX releases the port engaged in the call.

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 and FXO Ports. For instructions, under *Basic Settings*, see “FXO Port” and “SIP Trunks”.

How to Disconnect a Call using Access Code

- You are in speech or are at the end of the current call.
- Dial **#92**

IP Dialing

SETU VFX supports direct dialing of IP Addresses from the source port. To provide IP Dialing facility to the users, you must configure a SIP Trunk or a SIP Group for IP Dialing.

When a number is dialed out from the source port, SETU VFX 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 VFX, the system does not check the Destination Port Determination method you have configured for that port, instead it routes the dialed IP Address through the SIP Trunk or SIP Group you 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 VFX interprets the * you dialed as a '.' (dot/period).

Configuring IP Dialing

To provide this feature to users,

- you need to select a SIP Trunk or a SIP Group through which the dialed IP Address are to be routed.

  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. See the topic “Group” for instructions.

  When you assign a SIP Trunk, make sure it is enabled and has the necessary configuration done. See “SIP Trunks” under *Basic Settings* for instructions.

- assign the SIP Trunk you want or the SIP Group you configured to **SIP Trunk for IP Dialing** in the System Parameters. See “System Parameters” under *Advanced Settings* for instructions. By default, SIP Group 1 is selected for IP Dialing in the System Parameters.

If SIP Trunk Group for IP Dialing is programmed as 'None', SETU VFX will give error tone to the caller i.e. the call will be rejected.

CHAPTER 7 **Maintenance**

Firmware Upgrade

You can upgrade Firmware of SETU VFX:

1. From a Provisioning Server
2. From a Personal Computer

Firmware Upgrade from Provisioning Server

Auto Firmware Upgrade

Using Auto-Firmware Upgrade, SETU VFX 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 VFX 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 VFX on the Provisioning Server.
   - matrix_firmware.html file
   - SETU VFX_VxRy.Zip file

2. The following parameters must be configured in the SETU VFX.
   - 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 VFX 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 VFX decide which firmware it should upgrade to.

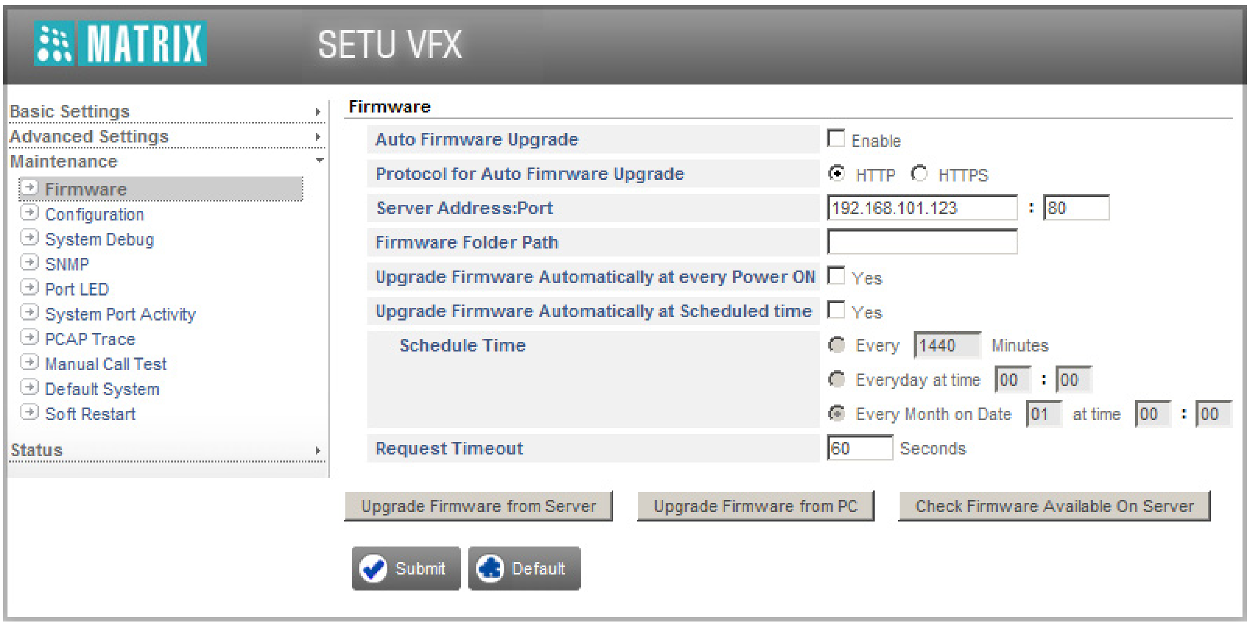
4. After SETU VFX decides the Firmware Version/Revision to upgrade to, it will send the request for the firmware files to the Provisioning Server. Once the respective firmware files are received, SETU VFX will upgrade its current firmware with the new firmware without the intervention or assistance of a technician.

The table below describes a few possible cases and the corresponding action taken by SETU VFX.

Version-Revision of your SETU VFX	Version- Revision in the matrix_firmware.html file received from the Provisioning Server	Action Taken by SETU VFX
V1R5	V1R4	SETU VFX will accept and upgrade its current firmware with V1R4.
	V1R5	SETU VFX will discard the upgrade process.
	V1R6 and V1R6.1	SETU VFX will accept and upgrade its current firmware with V1R6.1.
	V1R4, V1R5 and V1R6	SETU VFX will accept and upgrade its current firmware to the highest version, V1R6.
	V2R2_V2R1, V2R1, V1R8	Highest Version available is V2R2, however, V2R2 has a benchmark of V2R1. Therefore, SETU VFX will first upgrade with V2R1 and then with V2R2.

To configure Auto Firmware Upgrade parameters,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **Firmware** link.



- Select the **Auto Firmware Upgrade** check box. Default: Disabled.

- Select the **Protocol for Auto Firmware Upgrade** to be used by the Provisioning Server to upgrade the firmware of SETU VFX. SETU VFX 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 VFX are stored.

  The Provisioning Server Address can also be obtained by SETU VFX 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 VFX 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 VFX 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 VFX 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 VFX should check for firmware updates.

  - **Everyday at HH:MM:** The time in **Hours(00-23)** and **Minutes(00-59)** when SETU VFX 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 VFX should check for firmware updates every month.

*If SETU VFX has to upgrade itself with the benchmark firmware and you have selected **Upgrade Firmware Automatically at Scheduled Time**, SETU VFX 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 VFX tries to connect to the Provisioning Server for TCP/TLS binding. This timer specifies for how long SETU VFX 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 VFX 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 VFX, whenever you want.

To manually upgrade firmware of SETU VFX from server,

- Click the **Upgrade Firmware from Server** button on the Firmware page. SETU VFX will automatically upgrade its firmware with the latest firmware available on the server.

Checking Firmware Availability

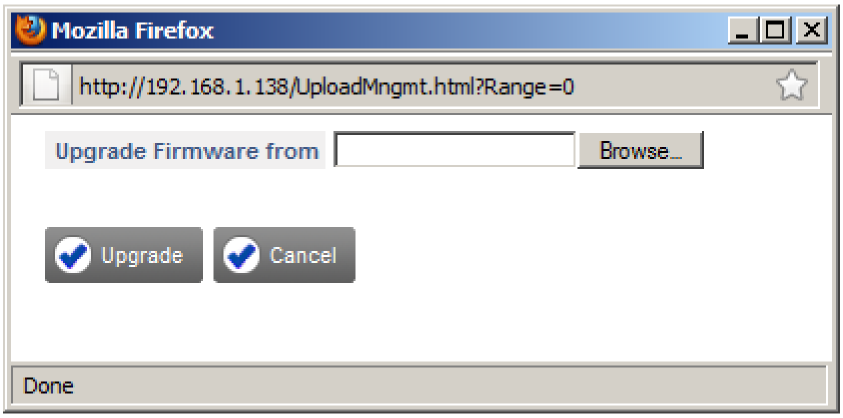
You can check the firmware files available on the server and then decide whether you want to upgrade SETU VFX. Before upgrading Firmware from server, you can also choose the firmware with which you want to upgrade your SETU VFX.

- 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 VFX with the desired Firmware, select the Firmware and click the **Submit** button.
- SETU VFX will upgrade itself with the firmware you select.

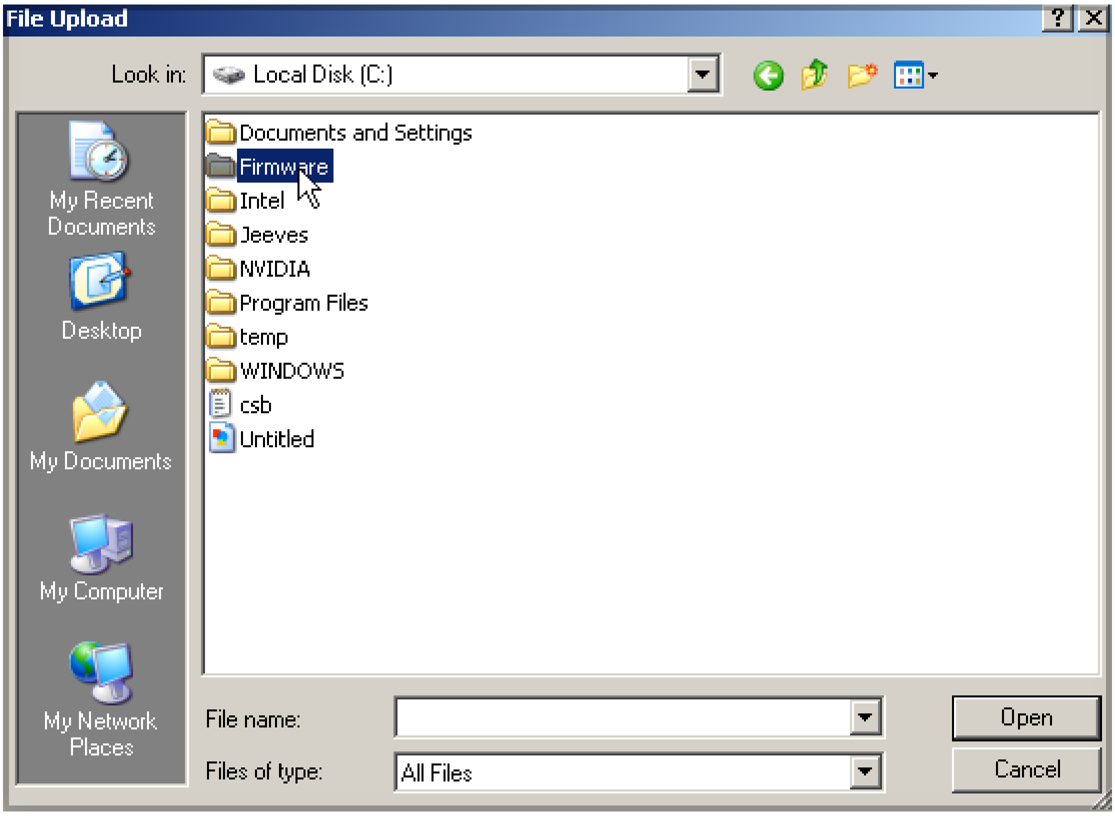
Firmware Upgrade from Personal Computer

You can also upgrade firmware of SETU VFX 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.



- The path to the file will appear in the **Firmware Upgrade From** box. Click the **Upgrade** button.

Configuration Upgrade

You can upgrade Configuration of SETU VFX:

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 VFX 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 VFX. ITSPs can store the configuration files of each SETU VFX 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 VFX 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 VFX.
   - 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.

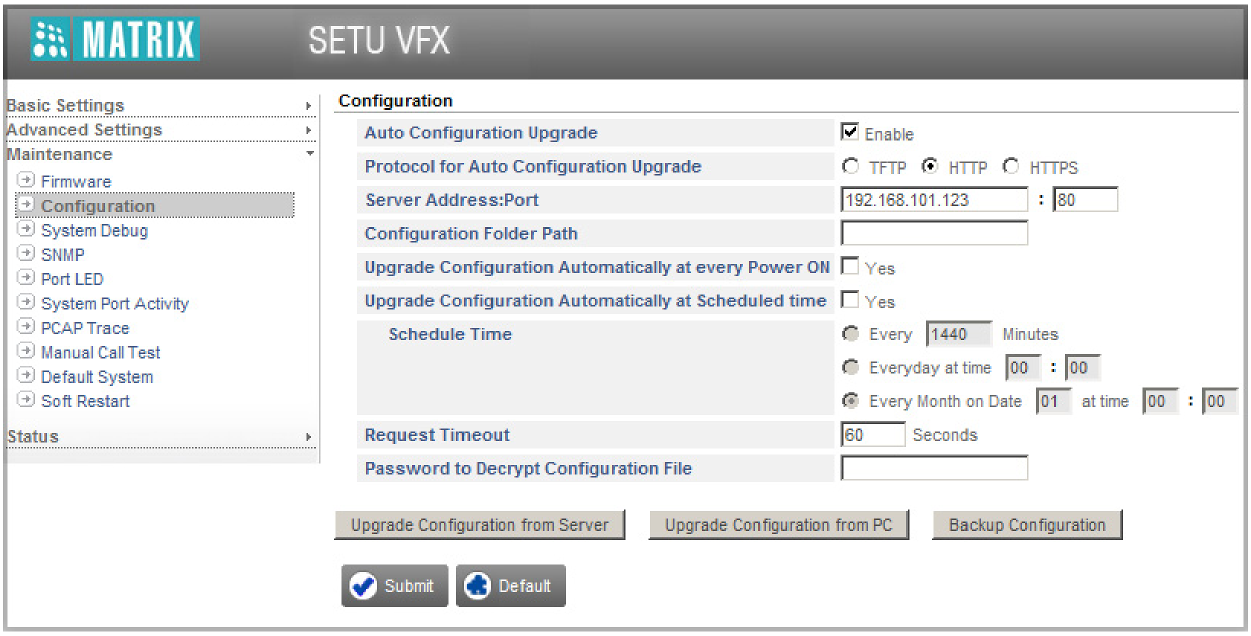
4. When SETU VFX installed at a customer site connects to the ITSP network, it will automatically download its configuration file stored on the Auto-Configuration Server (ACS), without the intervention or assistance of a technician.

To configure Auto Configuration parameters,

- Log into Jeeves.

- Click the **Maintenance** link.

- Click the **Configuration** link.



- By default, **Auto Configuration Upgrade** 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 VFX 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 VFX are stored.

  The Auto Configuration Server Address can also be obtained by SETU VFX 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 VFX 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 VFX 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 VFX should check for configuration updates.
- **Everyday at HH:MM:** The time in **Hours(00-23)** and **Minutes(00-59)** when SETU VFX 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 VFX should check for configuration updates every month.

- **Request Timeout:** Request Timeout is the time for which SETU VFX will try to connect to the Auto Configuration Server for TCP/TLS binding using HTTP or HTTPS. This timer specifies for how long SETU VFX 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 VFX 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 retry 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 VFX 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.

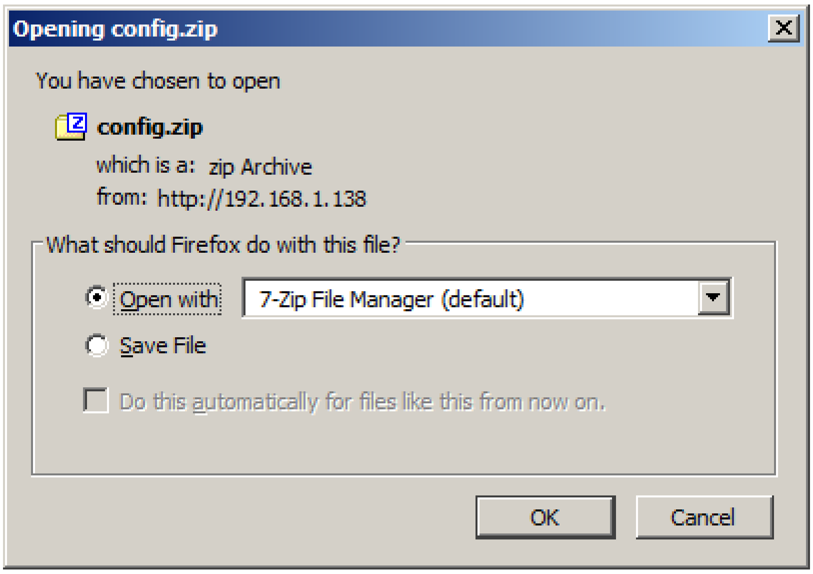
Manual Configuration Upgrade

To manually upgrade configuration of SETU VFX, click the **Upgrade Configuration from Server** button.

Backup Configuration

- To save the existing configuration files as backup, click the **Backup Configuration** button.

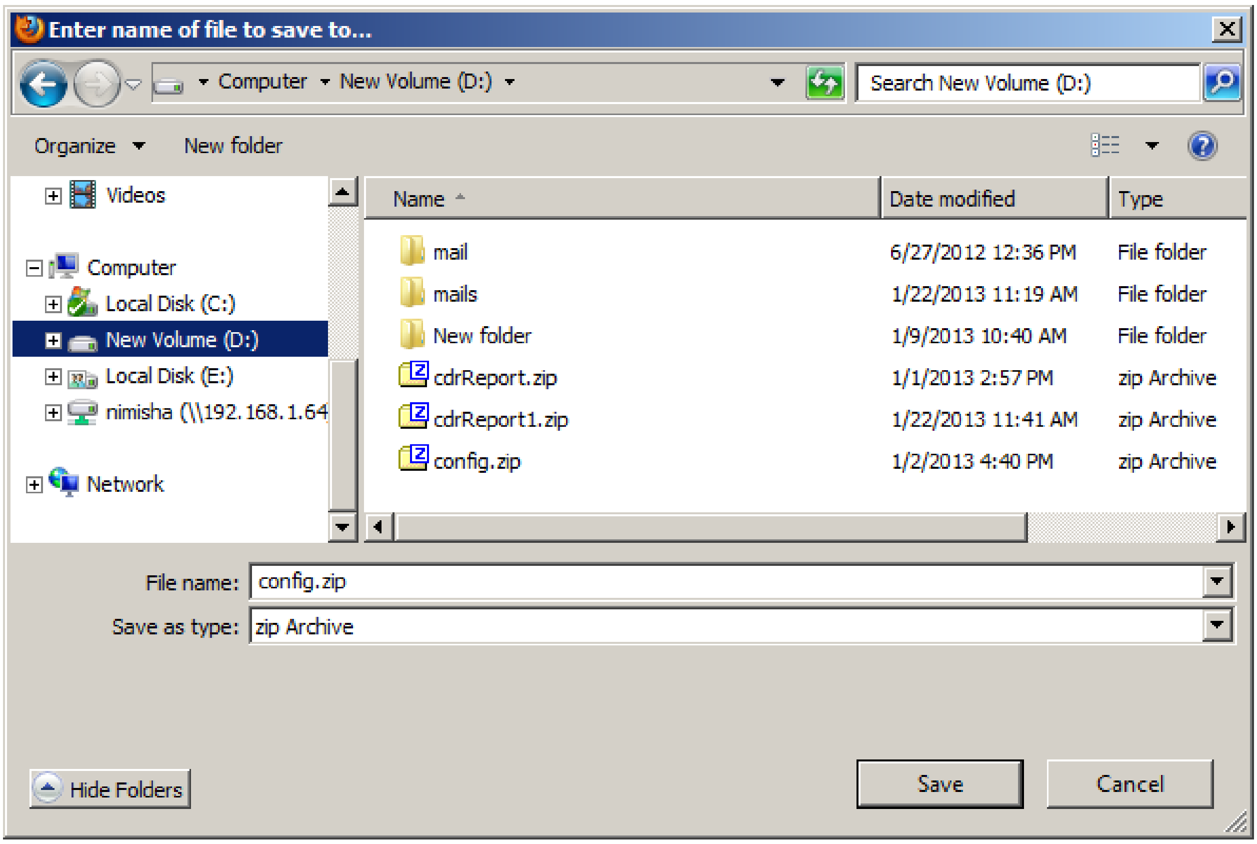
A **Opening config.zip** window will open.



- 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.



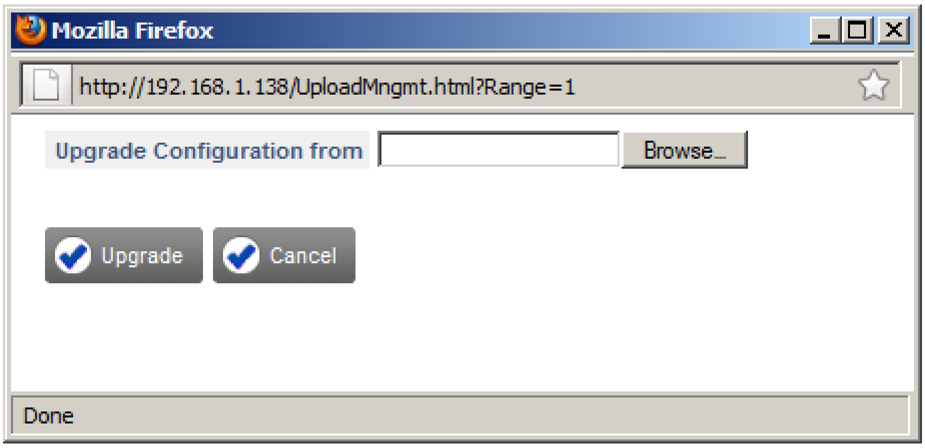
Save the back up configuration files by tagging the file name with the Version-Revision of the Firmware and tag the name of the backup folder on your computer with the date. This will help you at the time of restoring the back up configuration files.

- Open the configuration file (.zip) from the location you saved. The zip file contains all the system configuration files.
- Keep this folder as a backup. In case there is a problem with the system configuration files these backup files can be restored back in the system.

Upgrading Configuration from a Personal Computer

You can upgrade configuration of SETU VFX 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.
- The path to the file will appear in the **Configuration Upgrade From** box.
- 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

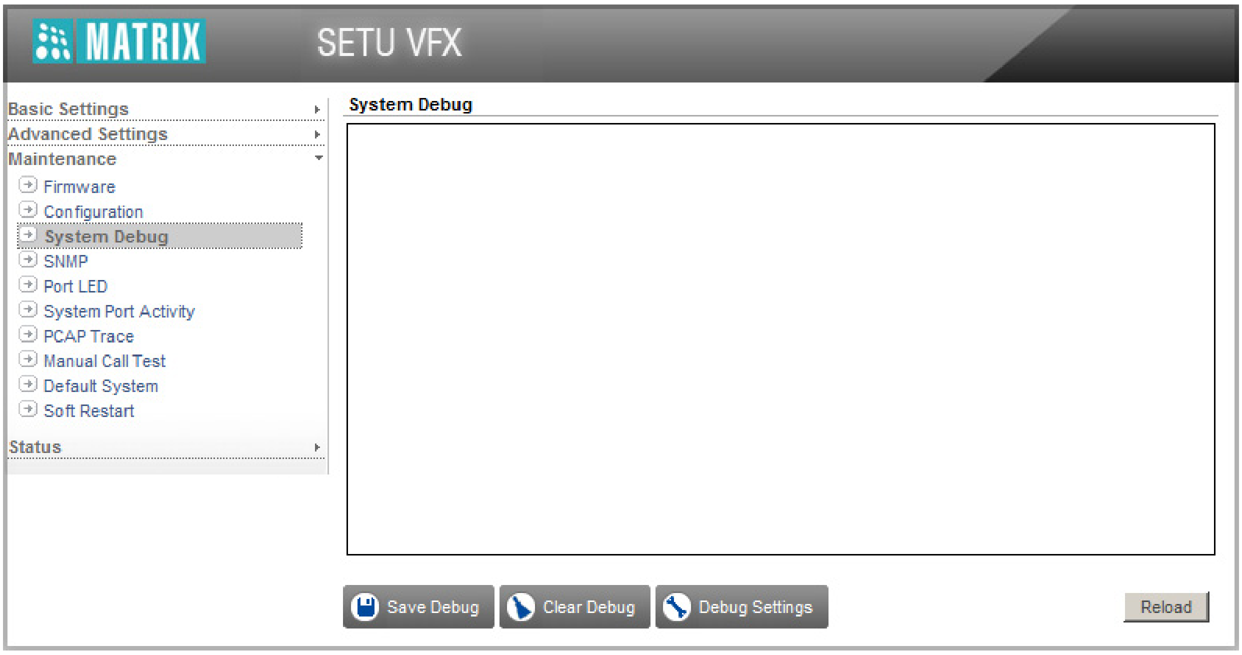
Debugging is a method used for recording actions and events of the system. Debugs are the primary record keepers of the system and network activity. Debugging has several benefits which include troubleshooting, security and system administration.

SETU VFX supports Syslog Client for sending debug messages to the remote syslog server on the IP network.

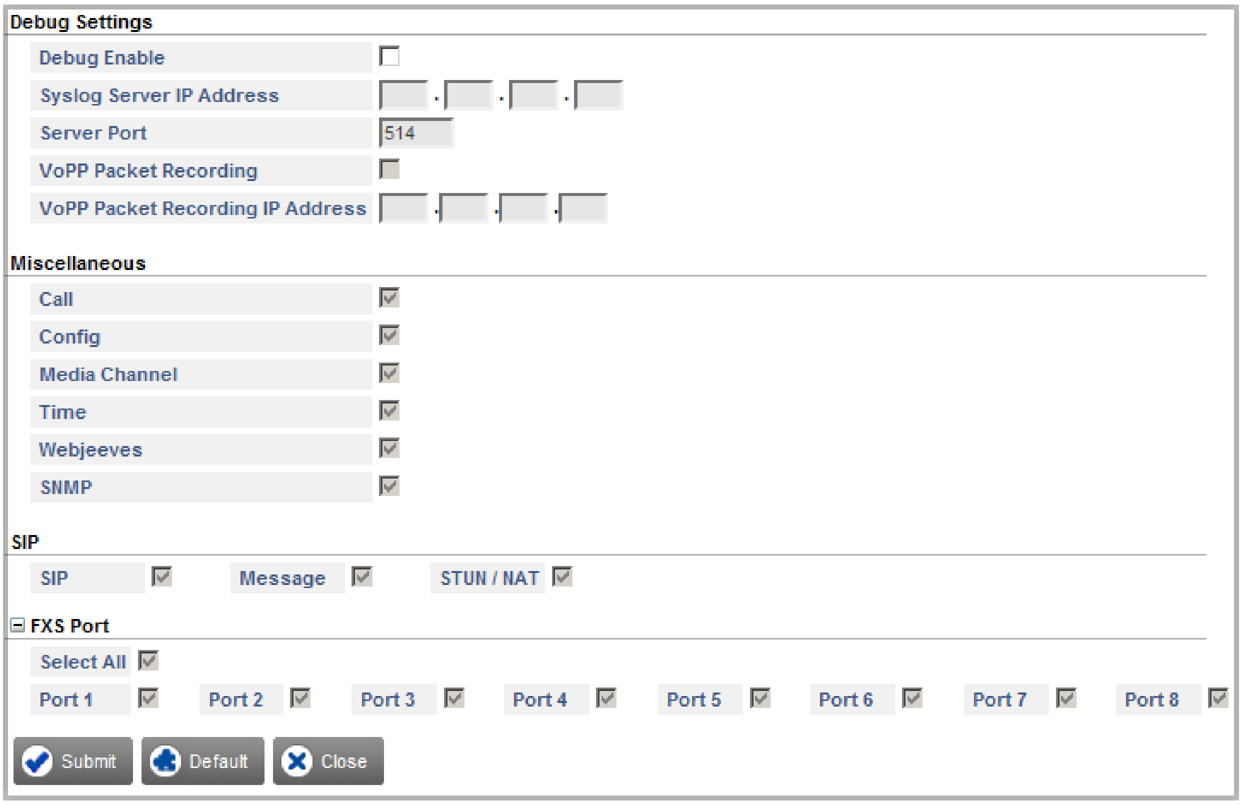
Configuring System Debug

Debugging is a continuous process and you can view the Debug events on the screen. To do so,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **System Debug** link.



- Click the **Debug Settings** button and configure the settings.



- Select the **Debug Enable** check box to enable system debug. Default: Disabled.
- In the **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.
- If you have enabled **VoPP Packet Recording** debug in *System Debug*, configure the **VoPP Packet Recording IP Address**.
- For **Miscellaneous**, select the desired debug level:
  - Call
  - Config
  - Media Channel
  - Time
  - Webjeeves
  - SNMP

  Default: All debug levels, are enabled. To disable a debug level, clear the respective check box.
- For **SIP Port**, select the desired debug level:
  - SIP
  - SIP message
  - STUN/ NAT

  Default: all are enabled.

- To debug the **FXS Port 1** to **8**, keep the check boxes enabled.
- To debug the **FXO Port 1** to **8,** keep the check boxes enabled.

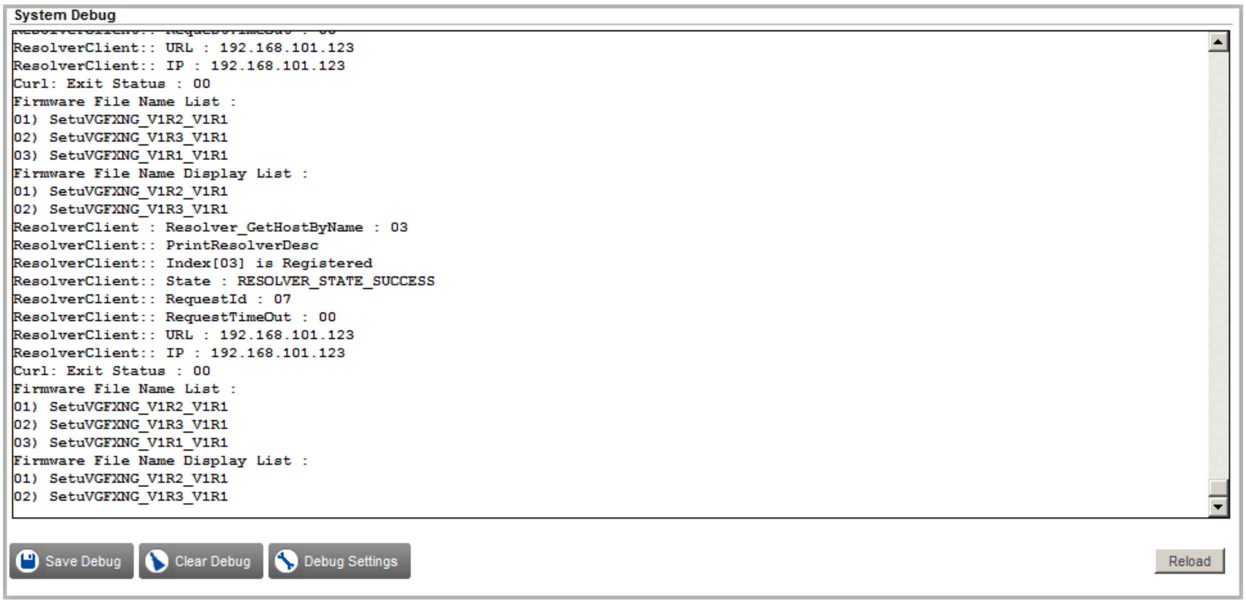
- Click **Submit** to save changes.

If debug is enabled, atleast one debug level should be selected. If no debug level is selected, SETU VFX 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.

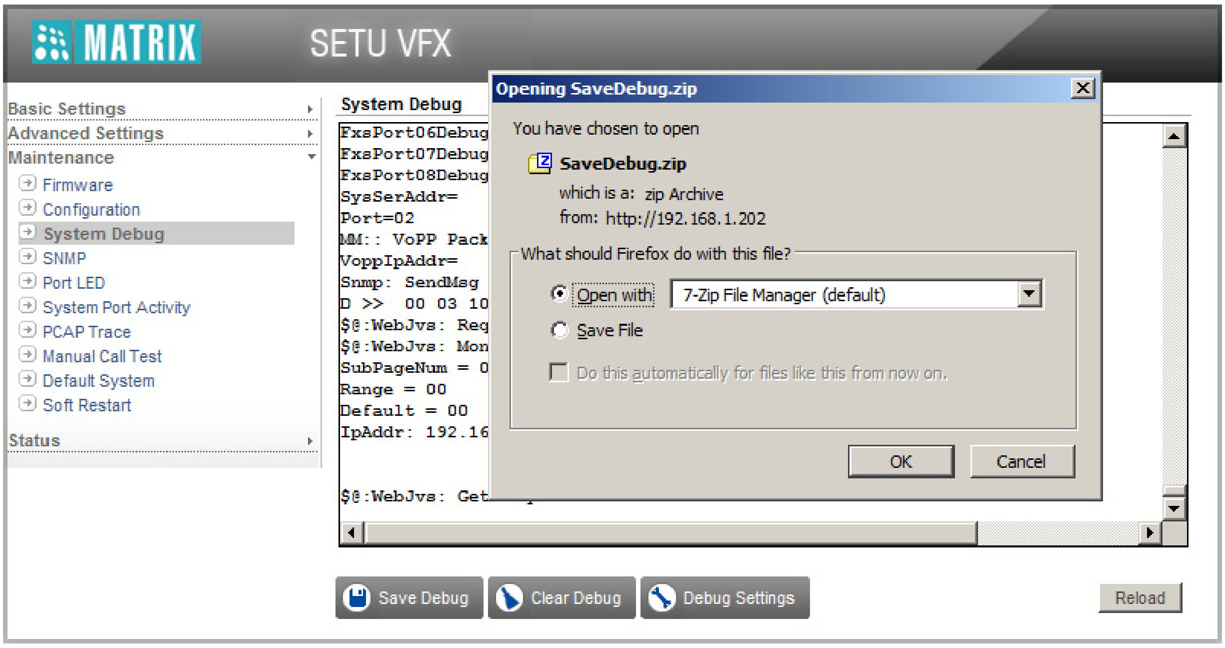
Events will be displayed only if you enable Debug.

- Click the **Reload** button, if you want the system to fetch an updated debug report, whenever required.



- Click the **Save Debug** button, to save all the events.

- 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)

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.

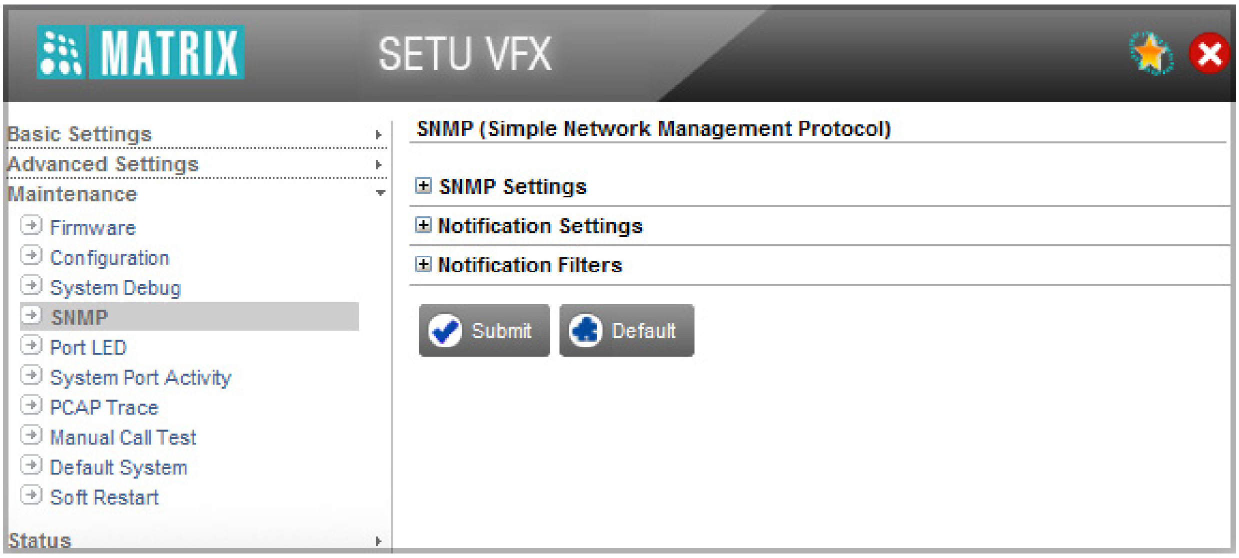
- **Management Information Base** is the commonly shared database between the Agent and the Manager.

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,

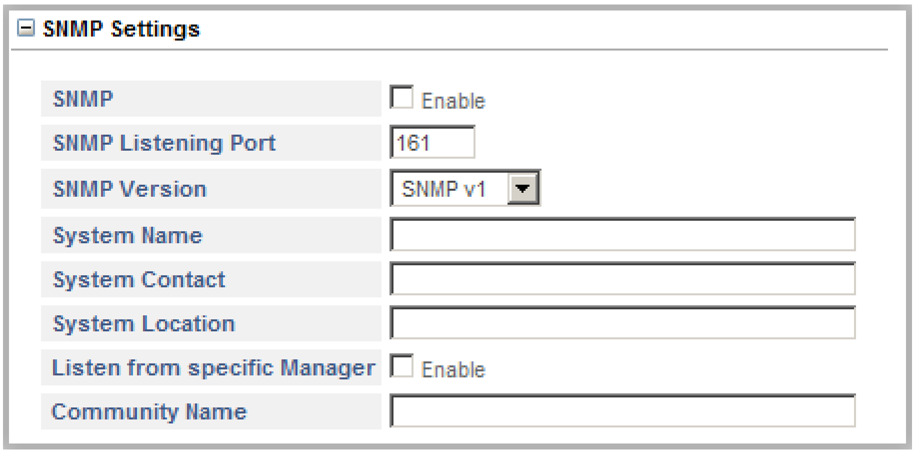
- Log into Jeeves.

- Under **Maintenance**, click the **SNMP** link.



SNMP Settings

- Click **SNMP Settings** to expand.



- Select the **Enable SNMP?** check box. 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 VFX. 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.**

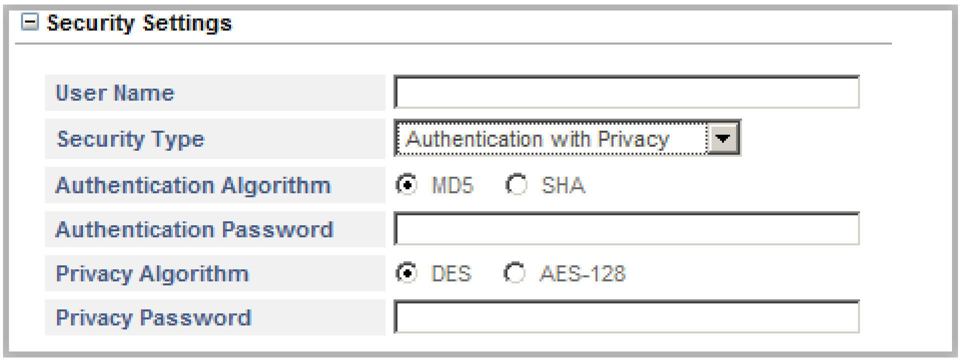
    The Manager's Address can be a Domain Name or an IP Address. It can be a maximum of 64 characters. Default: Blank.
- If SNMP version is set as **SNMPv1** or **SNMPv2c,** configure **Community Name**.

  Community Name identifies the SNMP community in which the sender and recipient of the message are located. It enables communication between SETU VFX 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.

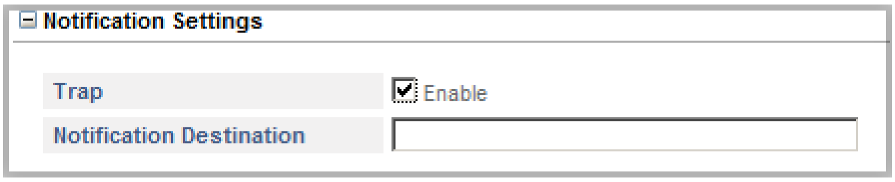
    - Enter **Authentication Password** for the User Name you have assigned. The Authentication Password must be a minimum of 8 characters and may have upto 24 characters. Default: Blank.

    - 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 VFX 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 VFX 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 VFX 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 VFX will send the notification (error message) to the destination configured.

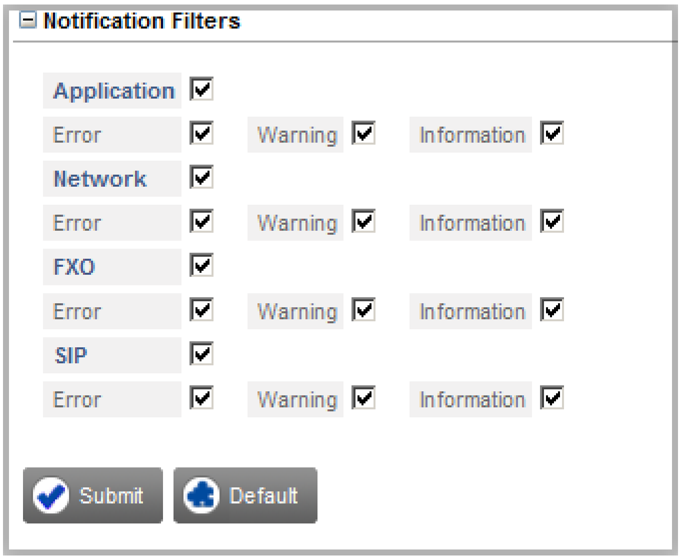
  The Notification Destination can be an IP Address or a Domain Name and the Port of the Manager or of any other device where you want to receive the trap messages. IP Address/Domain Name can be a maximum of 64 characters. Valid range of the port is 0-65535. Default port is 162.

- 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 shipped with the documents of SETU VFX in your SNMP Manager to get the status and notifications for SNMP.

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
	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	
	SIP Trunk disabled	

FXO[36]

Error	Warning	Information
	FXO - Line Disconnected	FXO - Line Connected

36. *Applicable only for VoIP-FXO Gateway.*

Port LED

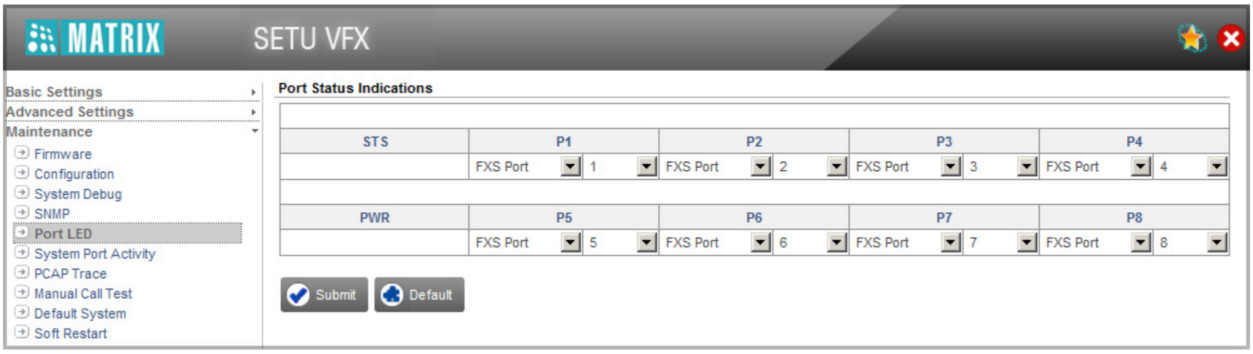
SETU VFX has 8 LEDs for port status indication. These LEDs indicate the various events occurring on the ports and error conditions.

By default, the LEDs labeled as **P1** to **P8** show the status of the FXO/FXS Ports.

It is possible to reassign a few or all of these LEDs to the SIP Trunks, if required.

To reassign LEDs,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **Port LED** link.

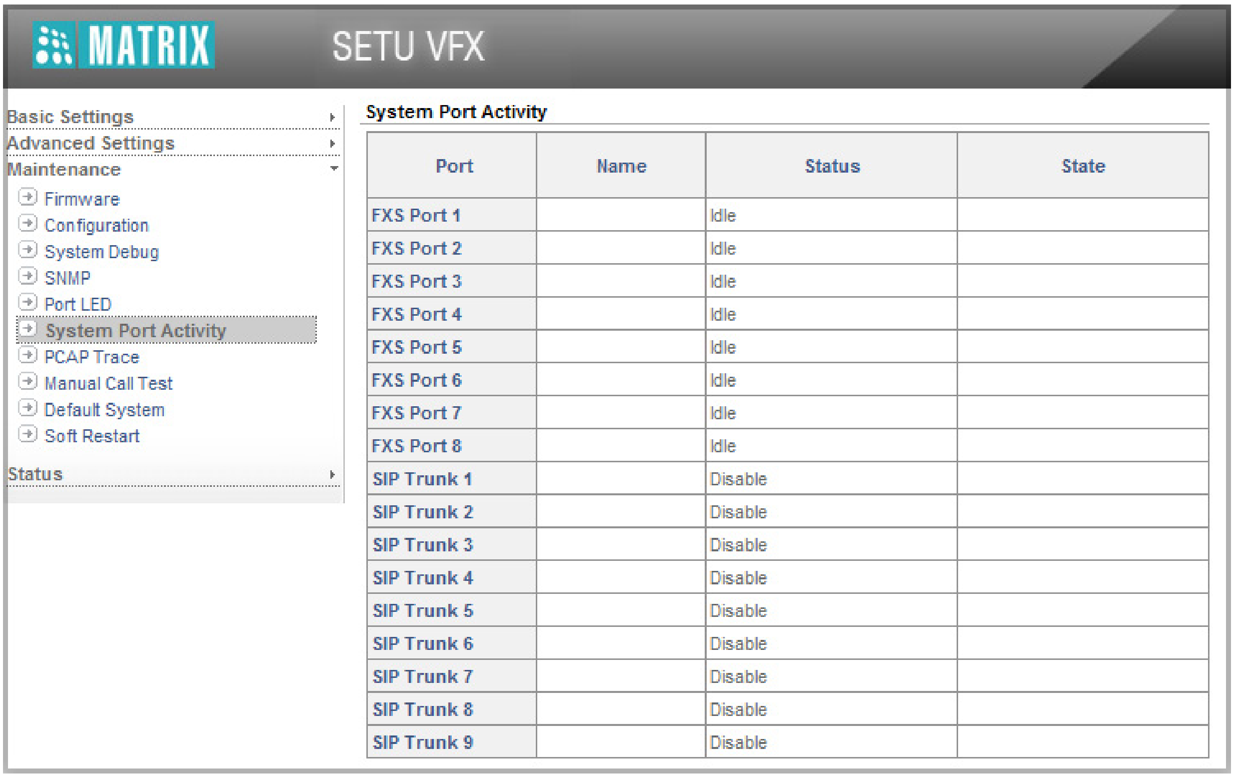


- For each LED labeled **P1, P2, P3, P4, P5, P6, P7, P8,**
  - Select the port type—**FXS Port** / **FXO Port**, **SIP Trunk**—which you want to assign to this LED from the list.
  - For the port type you selected, select the number of the FXS Port/ FXO Port/ SIP Trunk that you want to assign to the LED from the list.
- Click **Submit** to save.
- You may log out of Jeeves.

System Port Activity

You can view the state and activity on each Port of SETU VFX.

- 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.



Port	Name	Status	State
FXS Port 1		Idle	
FXS Port 2		Idle	
FXS Port 3		Idle	
FXS Port 4		Idle	
FXS Port 5		Idle	
FXS Port 6		Idle	
FXS Port 7		Idle	
FXS Port 8		Idle	
SIP Trunk 1		Disable	
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	

- 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 are 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, but there is no call present on this port currently.
  - **Active**, when the port is enabled, in use and a call is present on the port.
- In the **State** column, the states of **Active** ports are displayed as:

- **Dial**, when the port is in Dial state, that is, 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 Port).
- **Speech**, when source port and destination port are in speech.
- **Ringing**, when Ring event is detected on the FXS Port.
- **Incoming Call Proceeding**, when Ring event is detected on the FXO Port or SIP Trunk.
- **Remote Held**, when Hold message is received on the SIP Trunk or the FXS Port has put the call on Hold.
- **Error,** when the other party disconnects the call.

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 stream 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 VFX supports PCAP Trace, which you can use to detect and diagnose network related problems; for example, when the SIP Trunk is not getting registered, or a SIP related feature is not functioning.

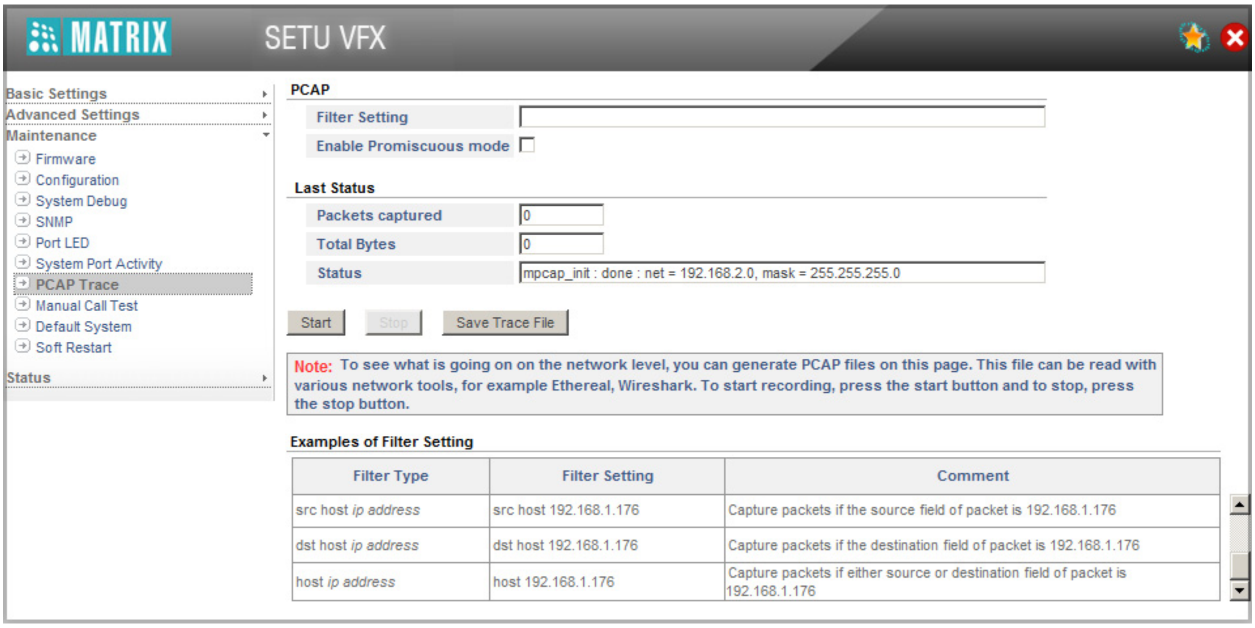
Packets traveling over a network are captured and saved in the system. You can save these trace files (packets captured by the system) on a computer and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

A maximum of 2 MB of packets can be captured and stored in the system.

SETU VFX 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 **PCAP.**



- Decide the type of packets to be captured and set the Filter accordingly. The Filter Settings parameter must be within 60 characters. By default, this field 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 VFX 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 VFX will be picked up by the PCAP Trace.

'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

- Click the **Start** button to begin the capturing of the packets.

- Click the **Stop** button to stop packet capture.

  OR

  Wait for the system to stop packet capturing. The system stops packet capturing once the maximum allotted memory of 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** field 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 again.

- You may log out of Jeeves.

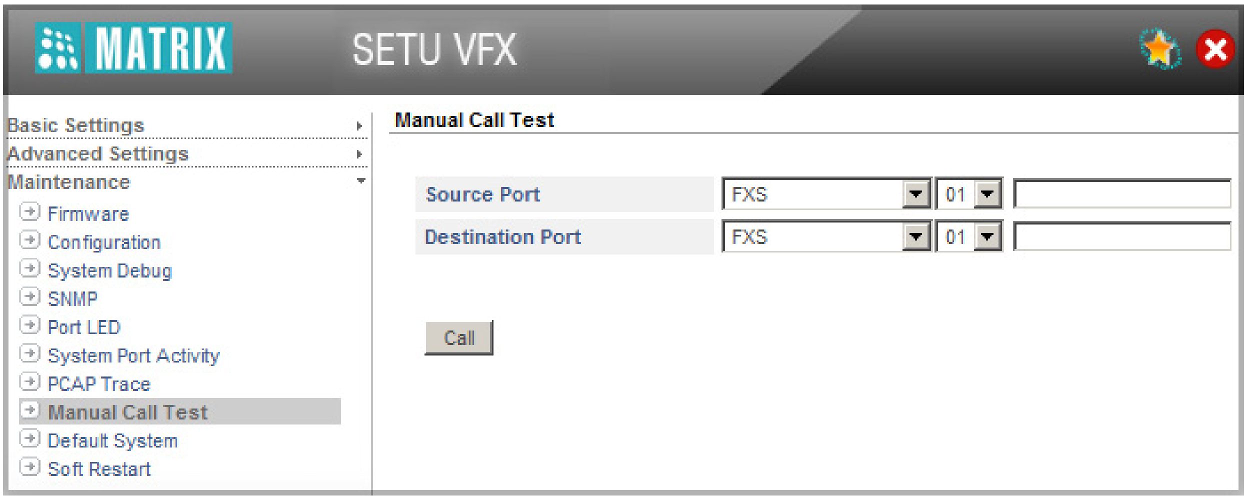
- Now, you can open the trace files using Wireshark/Ethereal or any other software which supports opening of trace files.

Manual Call Test

Manual Call Test enables you to check the quality of Speech between two ports—Source Port and Destination Port—of SETU VFX 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.



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 VFX 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.

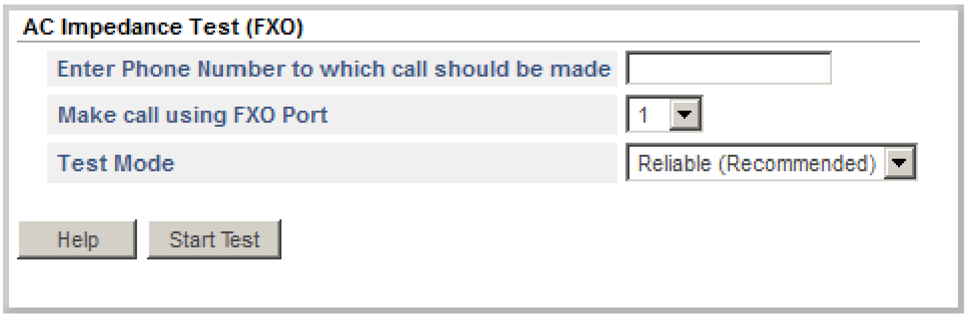
For more information on Call States and Port Status, see “System Port Activity”.

AC Impedance Test (FXO)

SETU VFX 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 VFX 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.

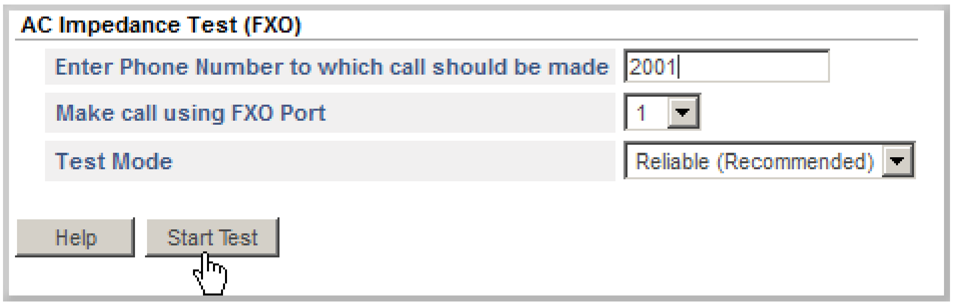
If you are using a mobile phone number, make sure the handset of the configured mobile number supports the Mute function.

- 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.



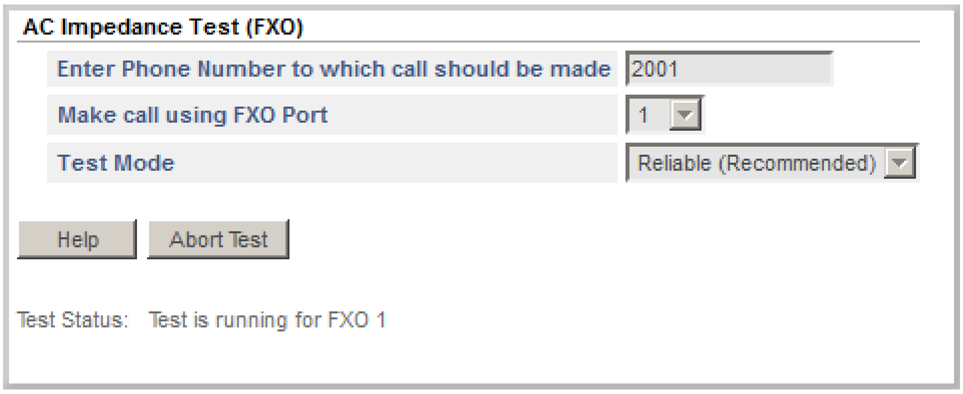
While the test is being conducted, you will hear pulsating tone on all the ports of the system.

- 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.

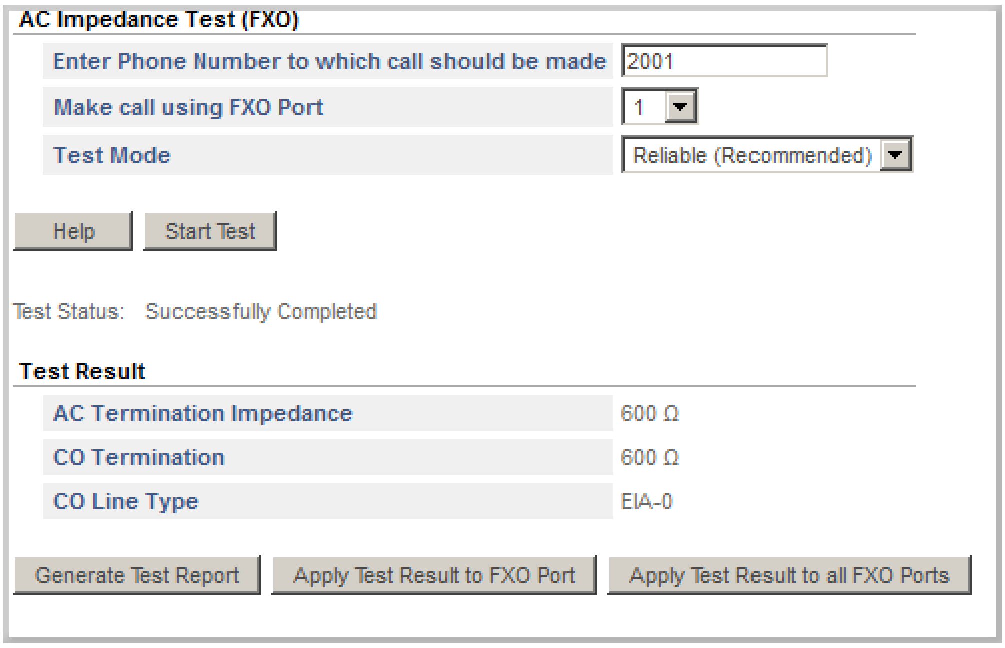
- After approximately 5 seconds, you will hear the test signals being transmitted by the system for the duration of the test. The message '*Test is running for FXO 1...*' appears on your screen.



  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.



- 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.

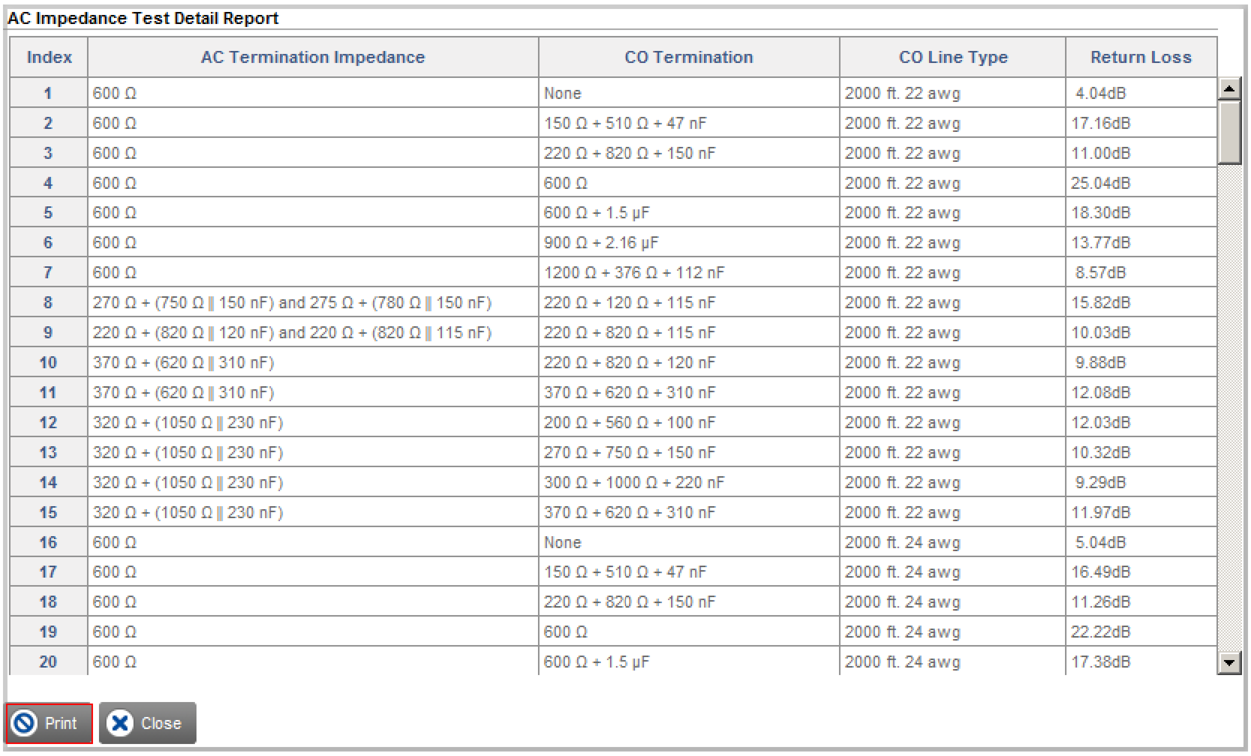
  If there is no echo/mild echo, and the volume level is low/high, you may adjust the **Rx Gain** and **Tx Gain** of the FXO Port manually. See “Hardware Settings” under the “FXO Port” for details.

  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.

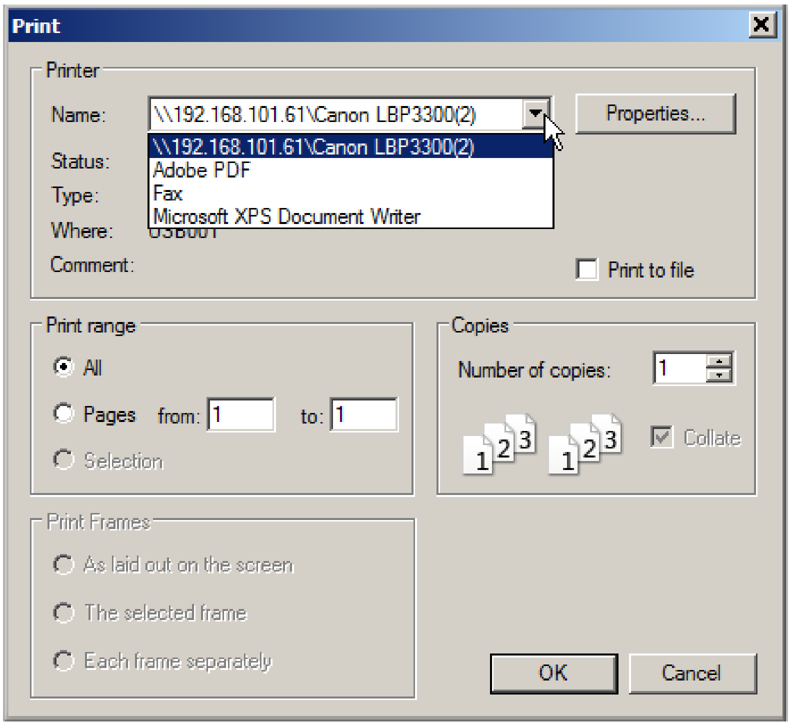
The detailed test report appears in a new window.



AC Impedance Test Detail Report

Index	AC Termination Impedance	CO Termination	CO Line Type	Return Loss
1	600 Ω	None	2000 ft. 22 awg	4.04dB
2	600 Ω	150 Ω + 510 Ω + 47 nF	2000 ft. 22 awg	17.16dB
3	600 Ω	220 Ω + 820 Ω + 150 nF	2000 ft. 22 awg	11.00dB
4	600 Ω	600 Ω	2000 ft. 22 awg	25.04dB
5	600 Ω	600 Ω + 1.5 μF	2000 ft. 22 awg	18.30dB
6	600 Ω	900 Ω + 2.16 μF	2000 ft. 22 awg	13.77dB
7	600 Ω	1200 Ω + 376 Ω + 112 nF	2000 ft. 22 awg	8.57dB
8	270 Ω + (750 Ω ‖ 150 nF) and 275 Ω + (780 Ω ‖ 150 nF)	220 Ω + 120 Ω + 115 nF	2000 ft. 22 awg	15.82dB
9	220 Ω + (820 Ω ‖ 120 nF) and 220 Ω + (820 Ω ‖ 115 nF)	220 Ω + 820 Ω + 115 nF	2000 ft. 22 awg	10.03dB
10	370 Ω + (620 Ω ‖ 310 nF)	220 Ω + 820 Ω + 120 nF	2000 ft. 22 awg	9.88dB
11	370 Ω + (620 Ω ‖ 310 nF)	370 Ω + 620 Ω + 310 nF	2000 ft. 22 awg	12.08dB
12	320 Ω + (1050 Ω ‖ 230 nF)	200 Ω + 560 Ω + 100 nF	2000 ft. 22 awg	12.03dB
13	320 Ω + (1050 Ω ‖ 230 nF)	270 Ω + 750 Ω + 150 nF	2000 ft. 22 awg	10.32dB
14	320 Ω + (1050 Ω ‖ 230 nF)	300 Ω + 1000 Ω + 220 nF	2000 ft. 22 awg	9.29dB
15	320 Ω + (1050 Ω ‖ 230 nF)	370 Ω + 620 Ω + 310 nF	2000 ft. 22 awg	11.97dB
16	600 Ω	None	2000 ft. 24 awg	5.04dB
17	600 Ω	150 Ω + 510 Ω + 47 nF	2000 ft. 24 awg	16.49dB
18	600 Ω	220 Ω + 820 Ω + 150 nF	2000 ft. 24 awg	11.26dB
19	600 Ω	600 Ω	2000 ft. 24 awg	22.22dB
20	600 Ω	600 Ω + 1.5 μF	2000 ft. 24 awg	17.38dB

- You may print the report by clicking the **Print** button in the test report window.
- Select your Printer in the Printer options.



- 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.
- by changing the Jumper position.

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:

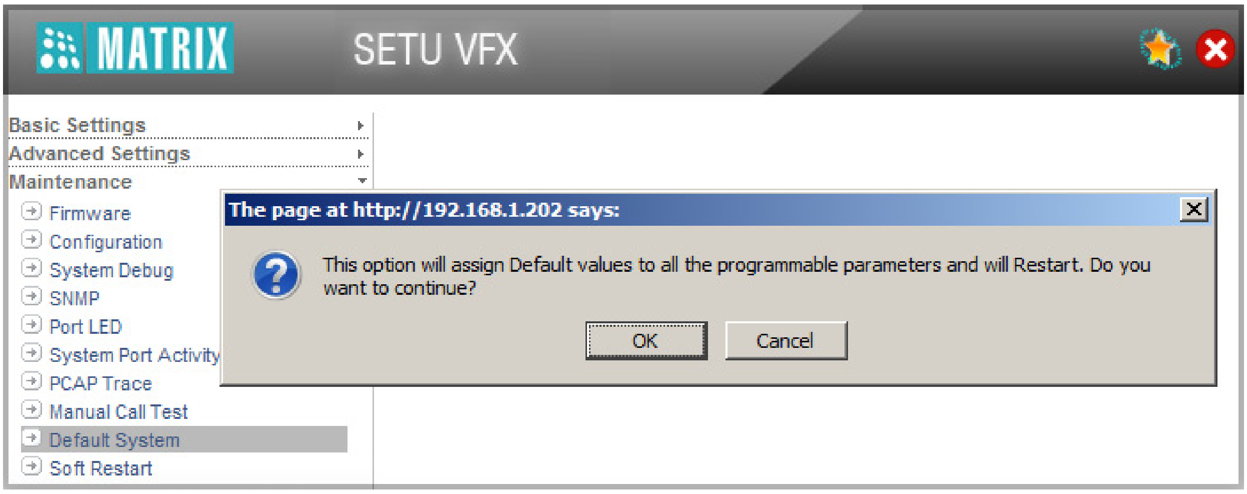
- Real Time Clock
- Call Detail Records
- Region
- Language
- Network
  - Connection Type
  - DNS Settings
  - DYN DNS
- System Parameters - NAT
  - Route Public IP Address
  - STUN Server Address
  - STUN Server Port
- System Parameters - Server Ports
  - HTTP Web Server Port
  - HTTPS Web Server Port
  - FTP Server Port
  - Telnet Server Port
- Firmware Parameters
- Configuration Parameters
- Login Password (Jeeves and Command)

To restore the default settings using the Web Jeeves,

- Log into Jeeves.

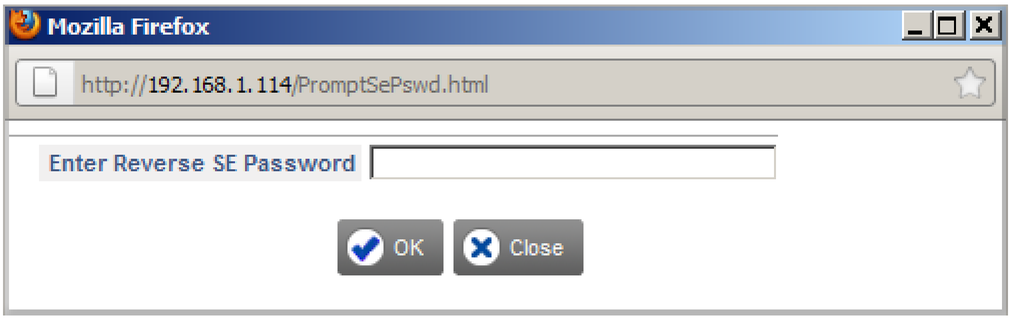
- Click the **Maintenance** link.

- Click **Default System.**



- An alert message will be 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[37]

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

37. *Applicable only for VoIP-FXS Gateway.*

- 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
- 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-#****

- 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
    - DHCP Server parameters
    - DMZ parameters
    - Port Forwarding parameters
- 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 VFX will restart.

Restoring Default Settings by changing the Jumper Position

By changing the position of **Jumper J3** on the PCB, you can restore the following parameters to default values:

- SE Password
- LAN Port Parameters
  - IP Address
  - Subnet Mask
  - DHCP Server parameters
  - DMZ parameters
  - Port Forwarding parameters
- 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 by changing the position of **Jumper J3** on the PCB,

- Make sure, you are wearing an electrostatic discharge preventive wrist strap or belt and have a grounding mat.
- Switch off the power supply
- Unscrew and remove the top cover of the enclosure.
- Locate and change the position of the **Jumper J3** from ***BC*** to ***AB***.
- Switch ON the system and wait for 15 seconds.
- Switch OFF the system and remove the enclosure cover.
- Change the Jumper position from ***AB*** to the original position ***BC***.
- Replace the enclosure cover.
- Switch ON the system.

Soft Restart

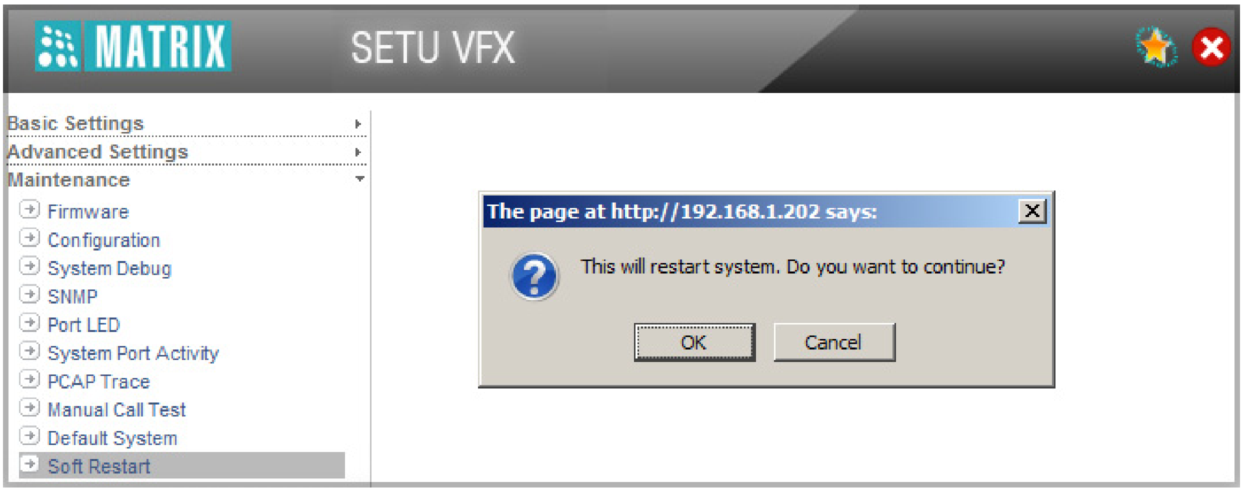
If you need to restart SETU VFX, you may do it by

- 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 not be affected.

To restart the system using Jeeves,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click **Soft Restart.**



- 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.

TR-069

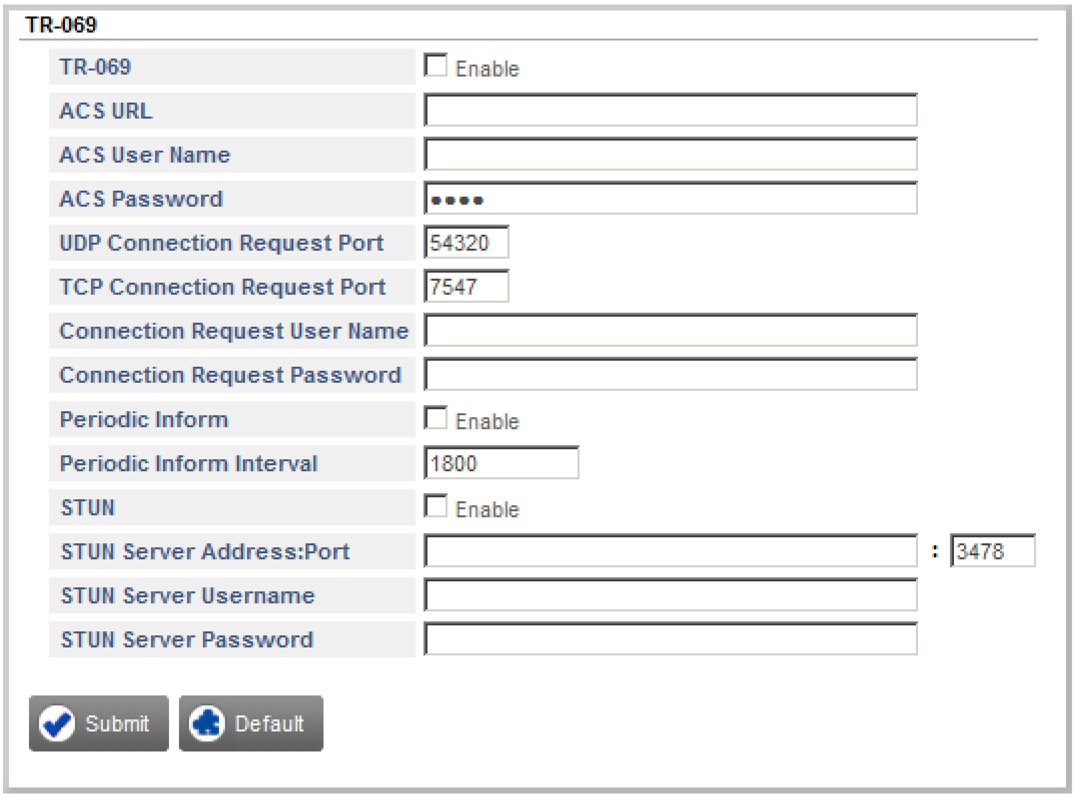
TR-069, also known as CPE WAN Management Protocol (CWMP), is a remote management protocol used for secure communication between the Customer Premises Equipment (CPE) and an Auto-Configuration Server (ACS) for various functionalities such as:

- Auto-configuration and dynamic service provisioning
- Firmware Management
- Status and performance monitoring
- Diagnostics

SETU VFX supports TR-069. Using TR-069, service providers can manage SETU VFX remotely for the functions described above.

To configure TR-069 parameters,

- Log into Jeeves.
- Click the **Maintenance** link.
- Click the **TR-069** link.



- Select the **TR-069 Enable** check box to use TR-069. Default: Disabled.
- In the **ACS URL** field, enter the URL of the ACS. SETU VFX will connect and send message to this server address. Default: Blank.

- In the **ACS Username** field, enter the username used by SETU VFX for HTTP authentication. Default: Blank.

- In the **ACS Password** field, enter the password used by SETU VFX for HTTP authentication. Default: Blank.

- In the **UDP Connection Request Port** field, enter the port on which the ACS will make a connection request to SETU VFX using UDP connection. The valid Port range is from 1031-65535. Default: 54320.

- In the **TCP Connection Request Port** field, enter the port on which the ACS will make a connection request to SETU VFX using TCP connection. The valid Port range is from 1031-65535. Default: 7547.

- In the **Connection Request Username** field, enter the username used by SETU VFX to authenticate the incoming connection request made by ACS. Default: Blank.

- In the **Connection Request Password** field, enter the password used by SETU VFX to authenticate the incoming connection request made by ACS. Default: Blank.

- Select the **Periodic Inform** check box, if you want SETU VFX to check updates available on ACS periodically. Default: Disabled.

- In the **Periodic Inform Interval** field, enter the time in seconds after which SETU VFX must attempt to connect with the ACS to check for updates. Default: 1800.

- Select the **STUN Enable** check box, if your SETU VFX is located behind the NAT Router and SIP messages need to be forwarded to the public network. Default: Disabled.

  STUN specifies the mechanism required for NAT traversal in SIP messages.

STUN server facilitates traversing through most NATs, except symmetric NATs. If your router has symmetric NAT, do not enable STUN.

- In the **STUN Server Address: Port** field, enter the STUN Server Address and the Listening Port of the STUN Server.

  The STUN Server Address can be a maximum of 256 characters. All ASCII characters are allowed. The valid range of the STUN Server Port is from 1025–65535. Default: 3478.

- In the **STUN Server Username** field, enter the username provided by the STUN server to authenticate the STUN Request. Default: Blank.

- In the **STUN Server Password** field, enter the password provided by the STUN Server to authenticate the STUN Request. Default: Blank.

- Click **Submit** to save changes.

- You may log out of Jeeves.

CHAPTER 8

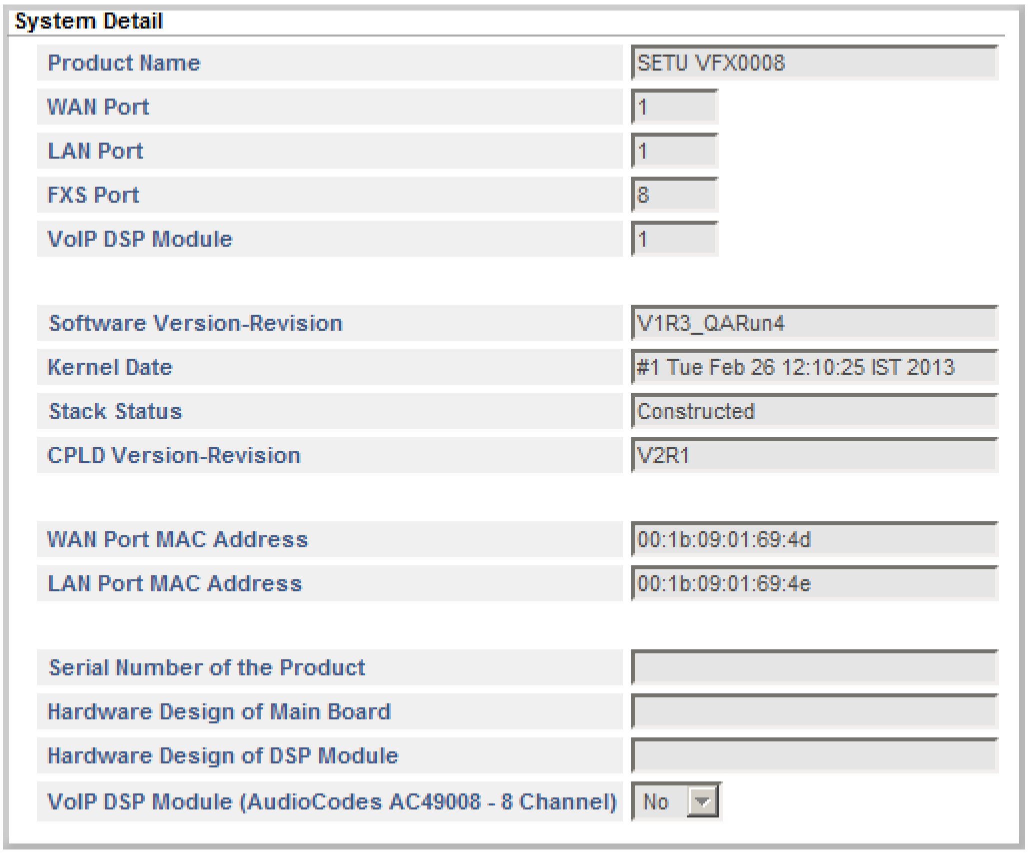
Status

You can view the System Details and Auto-Firmware upgrade, Auto-Configuration upgrade, the LAN Port, the WAN (Ethernet) Port, DHCP Clients, the SIP Trunks, the FXO Ports from Jeeves.

- Log into Jeeves.
- Click the **Status** link.

System Details

- Click the **System Detail** link.



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 Port in the system.
- **FXS Port[38]:** This field displays the number of FXS Port in the system.
- **FXO Port[39]:** This field displays the number of FXO Port in the system.
- **VoIP DSP Module:** This field displays the number of VoIP DSP Modules present in the system.
- **System Software Version-Revision:** This field displays the current version and revision of the firmware of SETU VFX.
- **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 DSP Module:** This field displays the Hardware Design of the DSP Module.
- **VoIP DSP Module (AudioCodes AC49008 - 8 Channel):** This field displays the VoIP DSP Module 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	Invalid Parameters

The following information related to Auto-Firmware upgrade will appear on your screen.

- **Last Upgraded On:** This field displays the firmware with which SETU VFX last upgraded itself through the provisioning server, along with the date (DD:MM:YYYY) and time (HH:MM) of the upgradation.

38. *Applicable only for VoIP-FXS Gateway.*
39. *Applicable only for VoIP-FXO Gateway.*

- **Next Upgrade On:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VFX 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 VFX 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.

<table>
<tr><th>Possible Responses</th><th>Event</th></tr>
<tr><td>Invalid Parameters</td><td>When parameters are not valid.</td></tr>
<tr><td>Local Failure</td><td>When internal error occurs, like Thread Creation failed.</td></tr>
<tr><td>Resolving Server Address</td><td>When IP Address is not found using DNS query.</td></tr>
<tr><td>Server Not Found</td><td>When server is not connected after the expiry of Retry Timer and Retry Counter.</td></tr>
<tr><td>Send Request Failed</td><td>When there is Curl Internal Error</td></tr>
<tr><td>Connecting to Server</td><td>When system is establishing TCP connection with server until the expiry of Retry Timer and Retry Counter.</td></tr>
<tr><td>TCP Connection Failed</td><td>When no response is received for TCP connection until expiry of Retry Timer and Retry Counter.</td></tr>
<tr><td rowspan="2">Connection Failed</td><td>When no response is received for TCP connection after expiry of Retry Timer and Retry Counter.</td></tr>
<tr><td>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.</td></tr>
<tr><td>Permission Denied</td><td>When access is denied.

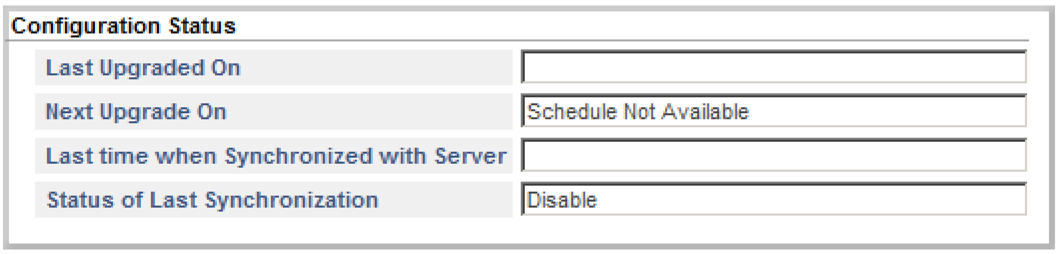
When there is permission problem on the server.

When login fails.</td></tr>
<tr><td>Downloading Firmware Index File</td><td>When the system is retrieving Firmware Index file.</td></tr>
<tr><td>Downloading Firmware</td><td>When the system is retrieving Firmware zip file.</td></tr>
<tr><td>File Not Found</td><td>When the remote file is not found.</td></tr>
<tr><td>Waiting for Firmware File Name</td><td>When Check Firmware Available on Server button is clicked manually and the list of available firmware is presented.</td></tr>
</table>

Possible Responses	Event
No File Found for Up-gradation	When selected firmware benchmark is not found. When user does not select the firmware name manually. When matrix_firmware.html file is received but current product name is not found from this file. Single firmware name is received in matrix_firmware.html but this benchmark file does not match with current firmware benchmark. Multiple firmware names are received but all files are below the current firmware.
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	When the firmware zip file decryption has failed. When the firmware file name does not match or benchmark is less than the current firmware version-revision in the text file.
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.



The following information related to Auto-Configuration upgrade will appear on your screen.

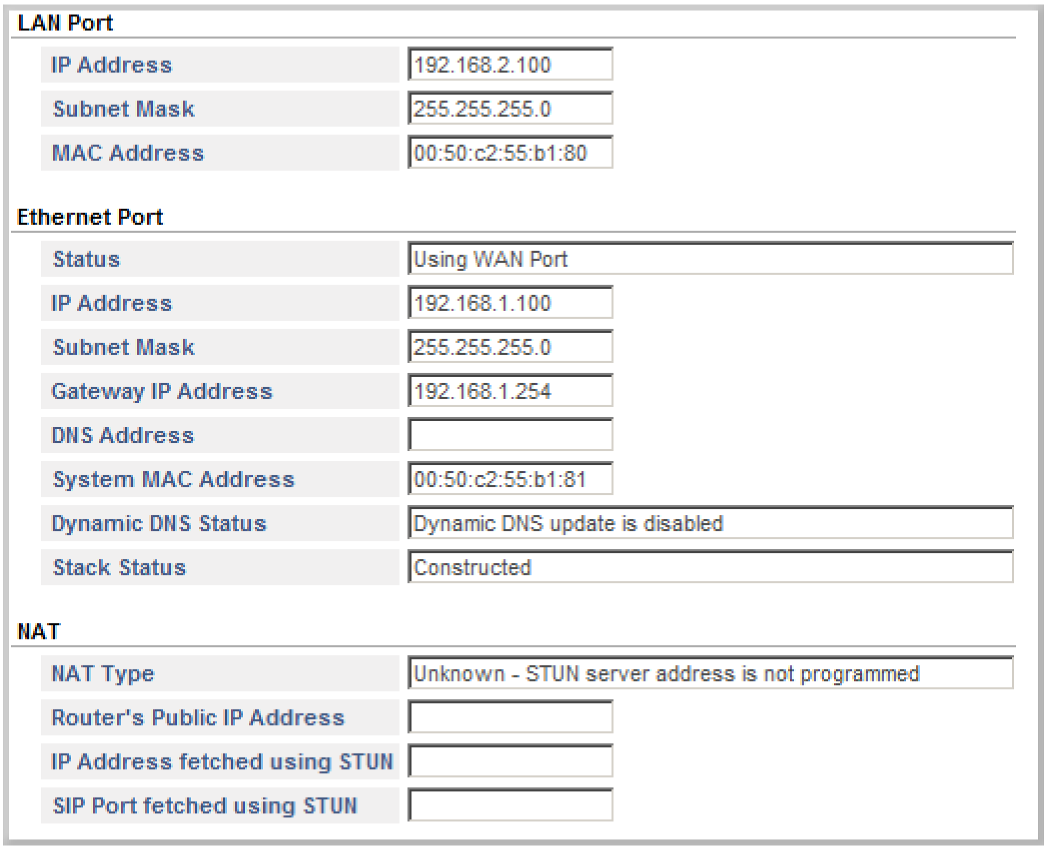
- **Last Upgraded On:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VFX last upgraded its configuration through the server.

- **Next Upgrade On:** This field displays the date (DD:MM:YYYY) and time (HH:MM), when SETU VFX 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 VFX 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.
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.



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 VFX.
- **Subnet Mask:** This field displays current Subnet Mask assigned to the LAN Port of SETU VFX.
- **MAC Address:** This field displays the MAC Address assigned to the LAN Port of SETU VFX.

Ethernet (WAN) Port

- **Status:** This field displays the status of the Ethernet Port of SETU VFX.
- **IP Address:** This field displays the IP address assigned to the Ethernet Port of SETU VFX.
- **Subnet Mask:** This field displays the Subnet Mask assigned to the Ethernet Port of SETU VFX.
- **Gateway IP Address:** This field displays the Gateway Address assigned to the Ethernet Port of SETU VFX.
- **DNS Address:** This field displays the DNS address.

- **System MAC Address:** This field displays the MAC Address assigned to the Ethernet Port of SETU VFX.

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 VFX. The commonly used NAT types are:
  - Unknown
  - Open
  - Conenat
  - Restrictednat
  - Portrestrictednat
  - Symmetricnat
  - Symmetricfirewall
  - Blocked

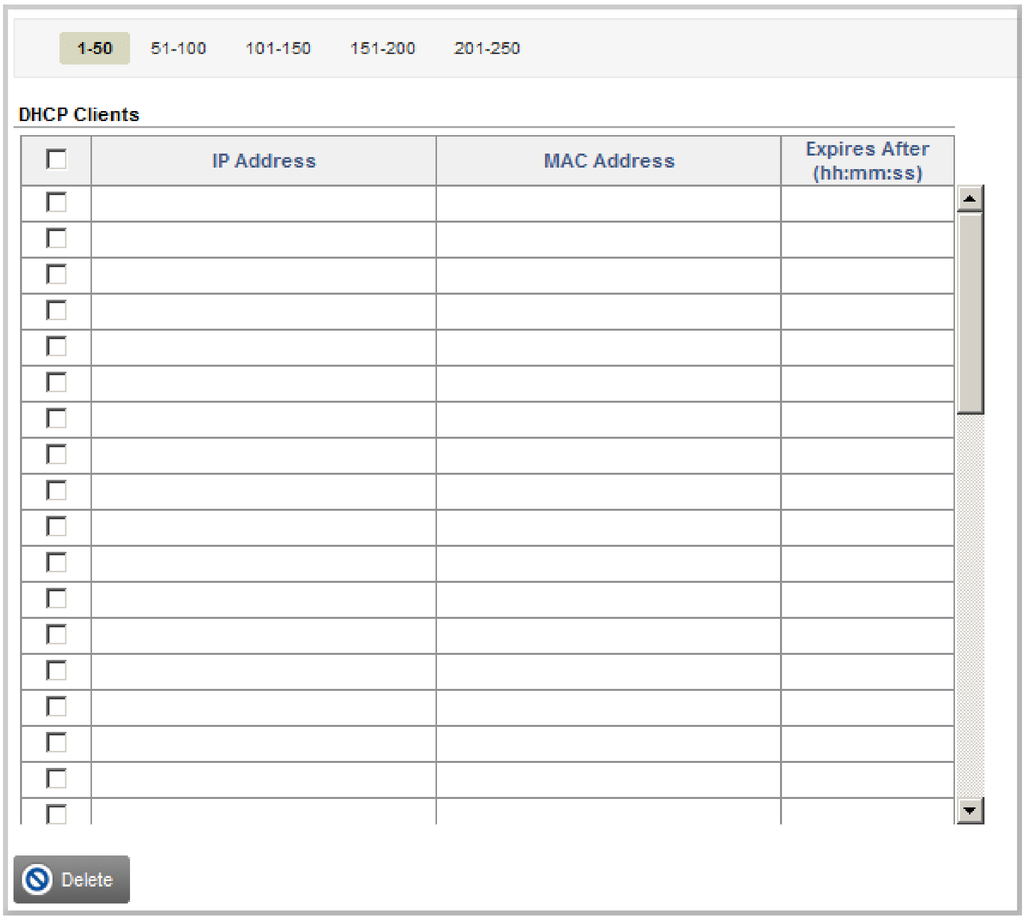
- **Router's Public IP Address:** This field displays the Router's Public IP address programmed in the System Parameters. See "NAT" under *System Parameters.*

- **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.

DHCP Clients

- Click the **DHCP Clients** link.



Following parameters will be displayed for each DHCP Client to which the IP Addresses have been leased by SETU VFX.

- **IP Address:** This field displays the IP Address assigned to the DHCP Client by the SETU VFX.

- **MAC Address:** This field displays the MAC address of the LAN device (DHCP Client) to which the SETU VFX has assigned an IP Address.

- **Expires After (hh:mm:ss):** This field displays the time after which the lease expires for the DHCP Client.

FXO Port[40]

- Click the **FXO Port** link.

FXO Port Status

FXO Port Number	Status
1	Line not Connected
2	Line not Connected
3	Line not Connected
4	Line not Connected
5	Line not Connected
6	Line not Connected
7	Line not Connected
8	Line not Connected

- The status of the FXO Ports appear on this page, as FXO Port *Line not Connected* or *Line Connected*.

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.

40. *Applicable only for VoIP-FXO Gateway.*

- **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.
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).
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 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	Response timeout
	Registering	0	2	Local Failure
	Registering	0	2	Local Failure
2	Active	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. The following status indications will appear for the MWI subscription on SIP Trunks.

SIP Trunk Status

SIP Trunk Number	MWI						
	Subscription Status	New Messages	Old Messages	Urgent New Messages	Urgent Old Messages	Message Notificaiton on	Failed Reason
1	Disabled	0	0	0	0	No	
2	Disabled	0	0	0	0	No	
3	Disabled	0	0	0	0	No	
4	Disabled	0	0	0	0	No	
5	Disabled	0	0	0	0	No	
6	Disabled	0	0	0	0	No	
7	Disabled	0	0	0	0	No	
8	Disabled	0	0	0	0	No	
9	Disabled	0	0	0	0	No	

- **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.

Status Message	Description
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 new messages waiting for the SIP Trunk.
- **Old Messages**: It displays the number of old messages for the SIP Trunk.
- **Urgent New Messages:** It displays the urgent new messages waiting on 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

Acronyms

ASCII	American Standard Code for Information Technology
ANT	Automatic Number Translation
CDR	Call Detail Record
CLI	Caller Line Identification
CLIP	Caller Line Identification and Presentation
CoS	Class of Service
CPT	Call Progress Tone
DHCP	Dynamic Host Control Protocol
DND	Do Not Disturb
DNS	Domain Name Service
DTMF	Dual Tone Multi-Frequency
FDWT	First Digit Wait Timer
FoIP	Fax over IP
FXO	Foreign Exchange Office
FXS	Foreign Exchange Subscriber
GMT	Greenwich Mean Time
ICMP	Internet Control Message Protocol
IDWT	Inter Digit Wait Timer
IP	Internet Protocol
ITSP	Internet Telephony Service Provider
LAN	Local Area Network
LED	Light Emitting Diodes
ms/msec	Millisecond
MAC	Media Access Control
MWI	Message Wait Indication

NAT	Network Address Translation
NTP	Network Time Protocol
PBX	Private Branch Exchange
PIN	Personal Identification Number
PPPoE	Point-to-Point Protocol over Ethernet
PSTN	Public Switched Telephone Network
PWR	Power
RTC	Real Time Clock
RTP	Real Time Protocol
SE	System Engineer
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol
SNTP	Simple Network Time Protocol
SOHO	Small Office Home Office
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Universal Reference/Resource Locator
VoIP	Voice over IP

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, Indiapolis, 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 VFX 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 VFX 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 VFX. 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 VFX

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

CPTG Type	Country	Dial tone		Ring Back Tone		Busy Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
6	Brazil	425	Continuous	425	1.0on 4.0 off	425	0.25on 0.25off
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

CPTG Type	Country	Dial tone		Ring Back Tone		Busy Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
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
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

CPTG Type	Country	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
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
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

CPTG Type	Country	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
35	UAE	400	0.4on 0.35off 0.225on 0.525off	400	1on 1off	350+440	0.1on 0.1off
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

CPTG Type	Country	Feature / Programming / Prompt Tone		Routing Tone		IntrusionTone	
		Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)	Freq.(Hz)	Cadence (Seconds)
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
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 Number	CADENCE (In Seconds)						Countries
	TON1	TOFF1	TON2	TOFF2	TON3	TOFF3	
01	Continuous						
02	0.75	0.75					
03	0.50	1.5					
04	0.75	2.25					
05	1.5	0.5					
06	1.0	4.0					Brazil, Greece, Italy, Netherland, Switzerland, Finland, Germany
07	2.0	4.0					Egypt, USA, Canada, Namibia
08	0.4	0.2	0.4	2.0			Australia, India, Singapore, South Africa, UK, Ireland, Malaysia
09	0.4	0.2	0.4	0.2	0.4	2.0	
10	1.0	2.0					Japan
11	1.0	3.0					China, korea, Russia, Belgium, Taiwan
12	1.0	5.0					Portugal, Sweden
13	1.5	3.0					Spain
14	1.5	3.5					France,
15	2.0	3.0					Israel, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3.5	5.5	0.79	1.1			

The Ring Type is not set to default, if the system is set to default.

Features at Glance

Feature Description	Feature Code
To Enter Programming Mode	***#19-Command Password (Default=1234)***
To Exit Programming Mode	***00#****
To Set Hotline	***#151-1***
To Cancel Hotline	***#151-0***
To Enable Call Waiting	***#16-1***
To Disable Call Waiting	***#16-0***
To Set DND	***#18-1***
To Cancel DND	***#18-0***
To Set Call Forward Unconditional	***#131-1***
To Cancel Call Forward Unconditional	***#131-0***
To Set Call Forward Busy	***#132-1***
To Cancel Call Forward Busy	***#132-0***
To Set Call Forward No Reply	***#133-1***
To Cancel Call Forward No Reply	***#133-0***
To Program Hotline Number	***#152-Destination Number-End-of-Dialing***[a]
To Program Hotline Timer	***#153-X (X is the timer value)***
To Program Call Forward Unconditional Number	***#135-Destination number-End-of-Dialing***[a]
To Program Call Forward Busy Number	***#136-Destination Number-End-of-Dialing***[a]
To Program Call Forward No Reply Number	***#137-Destination Number-End-of-Dialing***[a]
To Program No-Reply Timer	***#139-XX (XX is time in seconds)***
For Call Pick-up	***#5***
For Call Hold	***Flash***
To Retrieve Held Call	***Flash***
For Call Toggle (Call Split)	***#2***
To Reject the Waiting Call and Speech with Current Call	***#31***
To Ignore the Waiting Call and Speech with Current Call	***#32***
To Accept the Waiting Call and Hold Current Call	***#33***
To Accept the Waiting Call and Release Current Call	***#34***
For Blind Transfer	***#6***
For Conference	***#8***
For Using Supplementary Services of Service Provider	***#4***
For Using Voice Mail of the Service Provider	***#7***

Feature Description	Feature Code
For Attended Transfer	**^**
For Making a New Call	***#91***
To Disconnect Call	***#92***

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.

Product Specifications

Port Description

Port Name	Application
FXS Port	To connect analog phone or fax machine
FXO Port	To connect with PSTN or to FXS Ports of PBX
SIP Trunks	To connect to Internet Network for VoIP
LAN Port	Computer Connectivity
WAN Port	VoIP Connectivity

Different Configurations supported in the SETU VFX

Sr. No.	Configuration	VoIP Channels	FXO Ports	FXS Ports	WAN Port	LAN Port	Life Line (FXO) Port
1	SETU VFX-808	8	0	8	1	1	1
3	SETU VFX-880	8	8	0	1	1	0
2	SETU VFX-404	4	0	4	1	1	1
4	SETU VFX-440	4	4	0	1	1	0

FXS Port Parameters[41]

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/25Hz and Sinusoidal 52VRMS/25Hz
REN	3
DTMF Detection	ITU-T Q.24
CLI Presentation	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage 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[42]

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, SNTP, STUN, HTTP, TLS
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, iLBC-30ms and iLBC-20ms
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	WAN Port RJ45 Auto MDIX 10/100 Base T
Security	Password Protected Administration

41. *Applicable only for VoIP-FXS Gateway.*
42. *Applicable only for VoIP-FXO Gateway.*

Telephony features:

- Voice Calls using SIP proxy and Voice calls without using SIP proxy (Peer-to-Peer Calling)
- Battery Reversal- Useful when some billing machine is connected to SETU VFX
- 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 10 LEDs)

- Power = 1
- Status = 1
- Port = 8

Packing

- **Dimension (W x H x D):** 231 x 162.5 x 55 mm
- **Unit Weight:** 540 Gms
- **Shipping Weight:** 1.25 Kg
- **Mounting:** Wall Mounting and Table-Top

Power Supply

- **External Adaptor:** 12V DC @ 2A
- **Power Consumption:** 15W Typical
- **Connector:** DC Power Jack

Environmental

- **Operating Temperature:** 0°C to 40°C (32°F to 104°F)
- **Storage Temperature:** -40°C to 85°C (-40°F to 185°F)
- **Operating Humidity:** 5-95% RH (Non-Condensing)
- **Storage Humidity:** Max. 0-95% RH (Non-Condensing)

System Commands

Certain parameters of SETU VFX 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 VFX.

- 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 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-#****

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.

Matrix warranties the replacement or repair of any product or component(s) found to be defective during the applicable period and return the same, or grant a reimbursement credit with respect to the product or component. Parts repaired or replaced will be under warranty throughout the remainder of the original warranty period only. In case of software program design defect(s) that prevents the program from performing the specified functionality, affecting service and beneficial use of the product, Matrix reserves the right to incorporate solutions in its new release of the software and make it available to the customer within a reasonable period of time. The above said with regard to the software design defect, constitutes the sole obligation of Matrix and its authorized installer with respect to the product.

Matrix does not, however, affirm or stand for that the functions or features contained in the system will satisfy its end-user's particular purpose and /or requirements or that the operation of the program will be uninterrupted or error free.

This warranty is voidable by Matrix:

1. If the product is used other than under normal use and is not properly serviced and maintained by qualified technicians.
2. If the product is not maintained under proper environmental conditions.
3. If the product is subjected to abuse, damage, misuse, neglect, fire, power flow, acts of God, accident.
4. If the product is installed or used in combination or in assembly with the products that are not supplied or authorized by Matrix or are of inferior quality or design than Matrix supplied products, which may cause reduction or degradation in functionality.
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.

In no event will Matrix be liable for any damages, including lost profits, lost business, lost savings, downtime or delay, labor, repair or material cost, injury to person, property or other incidental or consequential damages arising out of use of or inability to use such product, even if Matrix has been advised of the possibility of such damages or losses or for any claim by any other party.

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This warranty is not transferable and applies only to the original user of the Product. All legal course of action subjected to Vadodara (Gujarat, India) jurisdiction only.

Disposal of Products/Components after End-Of-Life

Main components of Matrix products are given below:

- **Soldered Boards:** At the end-of-life of the product, the soldered boards must be disposed through e-waste recyclers. If there is any legal obligation for disposal, you must check with the local authorities to locate approved e-waste recyclers in your area. It is recommended not to dispose-off soldered boards along with other waste or municipal solid waste.

- **Batteries:** At the end-of-life of the product, batteries must be disposed through battery recyclers. If there is any legal obligation for disposal, you may check with local authorities to locate approved batteries recyclers in your area. It is recommended not to dispose off batteries along with other waste or municipal solid waste.

- **Metal Components:** At the end-of-life of the product, Metal Components like Aluminum or MS enclosures and copper cables may be retained for some other suitable use or it may be given away as scrap to metal industries.

- **Plastic Components:** At the end-of-life of the product, plastic components must be disposed through plastic recyclers. If there is any legal obligation for disposal, you may check with local authorities to locate approved plastic recyclers in your area.

After end-of-life of the Matrix products, if you are unable to dispose-off the products or unable to locate e-waste recyclers, you may return the products to Matrix Return Material Authorization (RMA) department.

Make sure these are returned with:
- proper documentation and RMA number
- proper packing
- pre-payment of the freight and logistic costs.

Such products will be disposed-off by Matrix.

"SAVE ENVIRONMENT SAVE EARTH"

E-Waste Management and Handling Rules

E-waste is a popular, informal name for electronic products nearing the end of their useful life. E-wastes are considered dangerous, as certain components of some electronic products contain materials that are hazardous, depending on their condition and density. The hazardous content of these materials pose a threat to human health and environment. Discarded electronics products such as circuit boards, batteries, wires and other electronic accessories if improperly disposed can leach lead and other substances into soil and groundwater. Many of electronic products can be reused, refurbished or recycled in an environmentally sound manner so that they are less harmful to the ecosystem.

Benefits of E-waste Recycling leach

Electronics Recycling Conserves Natural Resources

There are many materials that can be recovered from old electronic products. These materials can be used to make new products, thus reducing the need for the new raw materials. For instance, various metals can be recovered from circuit boards and other electronics can be recycled.

Electronics Recycling Supports the Community

Donating your old electronics plays an important role in the provision of refurbished products which can be of great help to certain industries, small organizations and non-profitable organizations. It also helps individuals gain access to technology that they could not have otherwise afforded.

Electronics Recycling Creates Employment Locally

Considering that around 90 percent of electronic equipment is recyclable, electronics recycling can play a significant role in creating employment. This is because new firms dealing with electronics recycling will form and existing firms will look to employ more people to recover recyclable materials. This can be triggered by the increase in the demand for electronics recycling.

Electronics Recycling Helps Protect Public Health and the Environment

Many electronics have toxic or hazardous materials such as mercury and lead, which can be harmful to the environment if disposed in trashcans. Reusing and recycling electronics safely helps in keeping the hazardous materials from harming humans or the environment. For example, certain electronic components and batteries are hazardous since they have lead in them. Printed circuit boards contain harmful materials such as cadmium, lead, mercury and chromium.

Instead of keeping old electronics or dumping them in landfills, recycling or reusing them is an appropriate option that should be supported by individuals and organizations. Considering the benefits of electronics recycling, it is very important that people in various parts around the world embrace this concept.
Creates Jobs

E-waste recycling creates new jobs for professional recyclers and creates a second market for the recycled materials.

Do's & Don'ts

Do's:

- Always look for information on the catalogue with your product for end-of-life equipment handling.
- Ensure that only Authorized Recyclers/Dismantler handle your electronic products.
- Always call at our toll-free No's to Dispose products that have reached end-of life.
- Always drop your used electronic products, batteries or any accessories, when they reach the end of their life at your nearest Authorized E-Waste Collection Points.
- Always disconnect the battery from product and ensure any glass surface is protected against breakage.

Don'ts:

- Do not dismantle your electronic Products on your own.
- Do not throw electronics in bins having "Do not Dispose" sign.
- Do not give e-waste to informal and unorganized sectors like Local Scrap Dealer/ Rag Pickers.
- Do not dispose your product in garbage bins along with municipal waste that ultimately reaches landfills.

E-Waste Management Plan

M/s. MATRIX COMSEC PVT LTD has partnered with **E-Waste Recyclers India (EWRI)** to comply with the new India E-Waste management and handling rules in providing drop-of centers and environmentally sound management of end of life electronics.

EWRI has obtained authorizations from the appropriate governmental agency for their processing facilities. EWRI will receive and recycle customer returned equipment, including all the e-waste. Customers can drop their e-waste in the drop-box provided at various collection centers of EWRI.

A list of collection centers along with the address is mentioned below.

The customers can also call on the following toll free number (1800-102-5679) from Monday to Friday between 10:00 AM to 5:30 PM to get details about the collection centers.

Collection Centers:

State/ City	Location	Logistic	Address	Toll-Free Number
Delhi	Rangpuri	Professional Logistics	Rangpuri, Milakpur Kohi Rangpuri, Rangpuri, New Delhi - 110037	1800-102-5679
Gurugram	Gurugram	Professional Logistics	295, LIG Colony, Sector 31, Gurugram, Haryana - 122022	1800-102-5679
Jharkhand	Dhanbad	Professional Logistics	Sardar Patel Nagar, Dhanbad, Jharkhand - 826004	1800-102-5679
Noida	Salarpur Khadar	Professional Logistics	2, Gejha Rd, Goyal Colony, Salarpur Khadar, Sector 102, Noida, Uttar Pradesh - 201304	1800-102-5679
Mumbai	Vashi	Professional Logistics	Plot-92,gala no 01,Sector 19C Vashi Navi, Mumbai - 400705	1800-102-5679

State/ City	Location	Logistic	Address	Toll-Free Number
Pune	Vallabh Nagar	Professional Logistics	No.3/20,Near Ashok Sah Bank, Vallabh Nagar, S.T.Stand Road, Pimpri, Pune - 302021	1800-102-5679
Odisha	Cuttack	Professional Logistics	Cuttack, Odisha	1800-102-5679
Hyderabad	Secunderabad	Professional Logistics	4,Block-3,4th Shatter at 179, MPR Estates Near Old Check Post Old Bowaenpally Secunderabad, Hyderabad - 500011	1800-102-5679
Bangalore	Yeshwanthpur	Professional Logistics	No.44 1st floor 2nd main D.D.U.T.T.L. Yeshwanthpur, Bangalore - 560022	1800-102-5679
Mangalore	Bhathery Road Boloor	Professional Logistics	Opp. Hindustan Lever Ltd, Sulthan, Bhathery Road Boloor, Mangalore (KA) - 575003	1800-102-5679
Jharkhand	Ranchi	Professional Logistics	Ranchi, Jharkhand	1800-102-5679
Chennai	Sennerkuppam	Professional Logistics	27,Sakthi Nagar Phase-II, Sennerkuppam, Near Bisleri Water Plant, Chennai - 600056	1800-102-5679
Rajasthan	Jaipur	Professional Logistics	A-81, 200 ft. By Pass, Heerapura, Jaipur, Rajasthan - 302021	1800-102-5679
Bokaro	Odisha	Professional Logistics	Cuttack, Odisha, India	1800-102-5679
Guwahati	Kundil	Professional Logistics	HN-34, Kundil Nagar Basistha Chariali, Near Parbhat Apartment, Guwahati - 781029	1800-102-5679
Lucknow	Kanpur Road	Professional Logistics	S-175,Ist Floor Transport Nagar Near RTO Kanpur Road Lucknow - 226004	1800-102-5679
Madhya Pradesh	Indore	Professional Logistics	284 AS-3 Scheme No.-78,Vijay Nagar, Indore, Madhya Pradesh	1800-102-5679
Ahmedabad	Pushp Penament	Professional Logistics	Shop No D-18, Pushp Penament, Behind Mony Hotel, Isanpur, Ahmedabad	1800-102-5679
Patna	Malyanil buddha	Professional Logistics	Dr. A.K Pandey (IPS) Malyanil buddha Colony, Patna (Bihar) - 800001	1800-102-5679
Andhra Pradesh	Vishakapatnam	Professional Logistics	Shop No.8, New Gajuwaka, Opp. High School Road, Vishakapatnam, Andhra Pradesh - 530026	1800-102-5679
Chandigarh	Pharbhat Road	Professional Logistics	Shop no:-19, Pharbhat Road, Opp:-Tennis Academy, Zirakpur, Chandigarh, Punjab	1800-102-5679

State/ City	Location	Logistic	Address	Toll-Free Number
Kolkata	B.T. ROAD DUNLOP	Professional Logistics	156A/73, Northern Park, B.T. Road Dunlop, Kolkata -700108	1800-102-5679
Odisha	Bhubaneswar	Professional Logistics	Acharya Vihar - jaydev Vihar Rd, Bhubaneswar, Odisha	1800-102-5679
West Bengal	Asansol	Professional Logistics	Shop No-4 Asansol Station Bus Stand Road, Munshi Bazar, Asansol, West Bengal - 713301	1800-102-5679

Regulatory Information

Federal Communications Commission Statement

Part 15:

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:
(1) This device may not cause harmful interference, and
(2) this device must accept any interference received, including interference that may cause undesired operation.

CE and ROHS Certificate

MATRIX
TELECOM | SECURITY

Declaration of Conformity

Manufacturer's Name:	Matrix Comsec Pvt. Ltd.
Manufacturer's Address :	15 & 19-GIDC, Waghodia, Dist: Vadodara 391760, Gujarat, India
Declares that the product/s Product:	SETU VFX
Model Type:	SETU VFX880, SETU VFX808, SETU VFX440 & SETU VFX404
Trade Name:	MATRIX

The designated products are in conformity with the below mentioned directive ;

EMI/EMC Standard:

EN 55032 : 2015
EN 61000-3-2 : 2014
EN 61000-3-3 : 2013
EN 55035 : 2017
EN 61000-4-2 : 2009
EN 61000-4-3 : 2006+ A1: 2008+ A2: 2010
EN 61000-4-4 : 2012
EN 61000-4-5 : 2014
EN 61000-4-6 : 2014
EN 61000-4-8 : 2010
EN 61000-4-11 : 2004

SAFETY Standard:

IEC 62368-1: 2014 (Second Edition)

Supplementary information:

The Product herewith complies with the following directives ;

EMC	2014/30/EU
Low Voltage Directive	2014/35/EU
RoHS Recast (RoHS 2)	2011/65/EU (as per standard EN 50581:2012)

Matrix Comsec Pvt. Ltd. ★ Vadodara ★

Mr. Ganesh Jivani
Director
Date: 18.03.2019

CE

MATRIX COMSEC PVT. LTD.
Registered/Head Office: 394-GIDC, Makarpura, Vadodara-390 010, India. Ph: +91 265 2630555, Email: Inquiry@MatrixComSec.com • **www.MatrixComSec.com**
Manufacturing Unit: 15 & 19-GIDC, Waghodia, Dist. Vadodara-391 760, India. Ph: +91 2668 263172/73 • **CIN:** U72200GJ1998PTC034047

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- The source of the open source software used in this product is available on CD, upon written request from:
  R&D Team
  Matrix Comsec Pvt Ltd
  394, Makarpura GIDC,
  Vadodara - 390 010
  Gujarat
  India.
  Customer shall bear the shipping and handling charges.

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