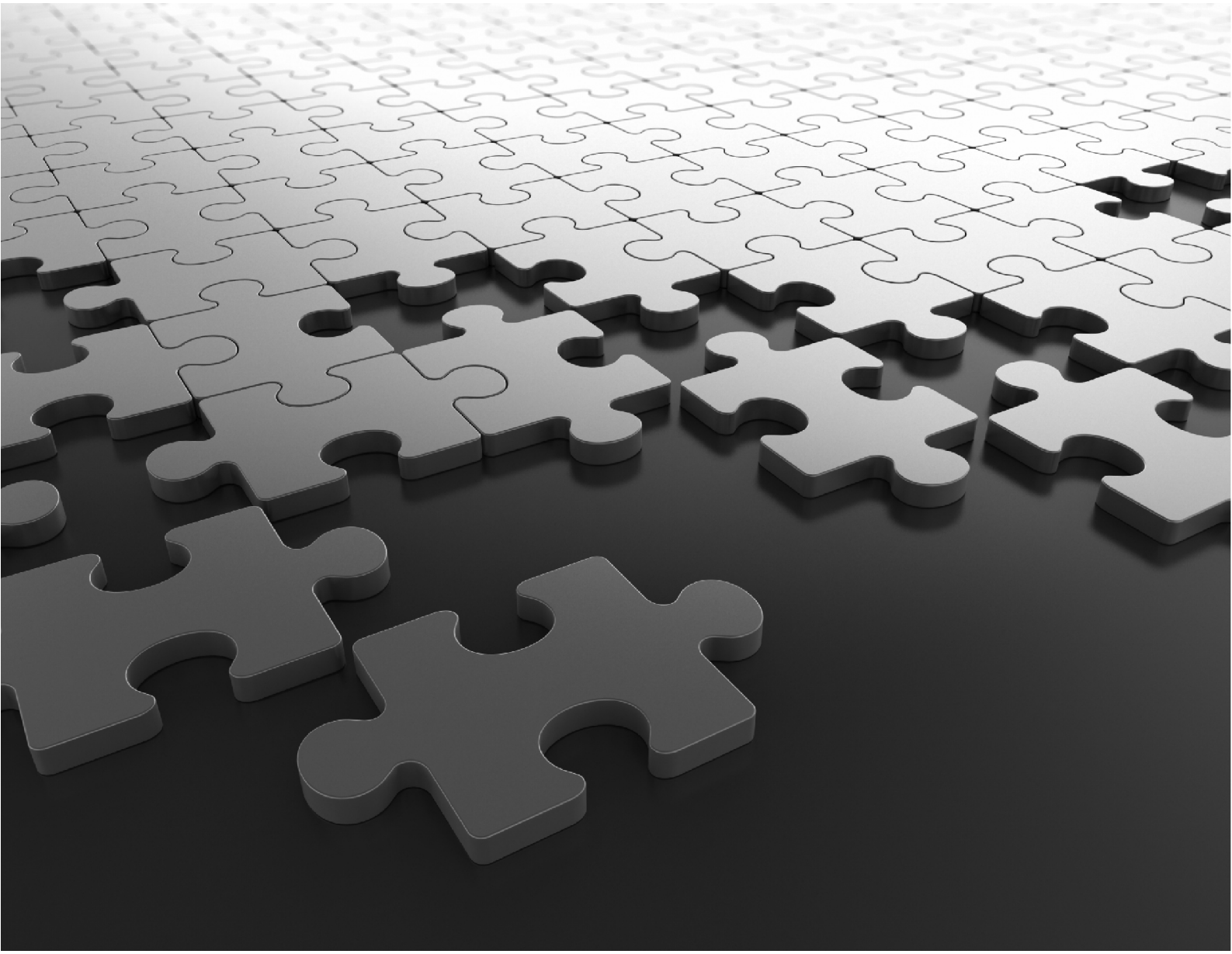
SETU VFXTH
System Manual



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SETU VFXTH

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 Factory-set configurations 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 VFXTH! This product is designed to give you the highest performance, combined with real ease of use. We hope you will make optimum use of this intelligent, intuitive, feature-packed Multi-Port VoIP-FXO-FXS Gateway. Please read this document carefully before installing your SETU VFXTH.

About this System Manual

This System Manual provides detailed instructions for installing, configuring and using SETU VFXTH as well as the information on protecting and maintaining it. We have made the best efforts to let you know your SETU VFXTH so that it becomes fun for you to use it.

Those who want instructions on quick installation and operation of SETU VFXTH are advised to refer SETU VFXTH Quick Start and User Card shipped with the product.

Intended Audience

This System Manual is aimed at:

- **Network and System Engineers**, who will install, configure and maintain the SETU VFXTH. 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 programming VoIP-FXO-FXS Gateways and are familiar with telecom wiring technology, how it works, and the various technical terms and functions associated with it.

- **Users**, who will actually use the SETU VFXTH. It is assumed that the users have some previous experience in operating Multi-Port VoIP-FXO-FXS Gateways.

  Users are not expected to configure or program the features of SETU VFXTH. However, it is anticipated that some of them may have to or want to do it. Therefore, this document provides instructions on installation and configuration in a lucid manner.

Organization of this Document

This System Manual contains following sections:

- **Introduction:** Gives an overview of this document, its purpose, intended audience, organization, terms and conventions used to present information and instructions etc.
- **Know Your SETU VFXTH:** Helps you to know your SETU VFXTH and its various applications.
- **Getting Started:** Contains information for installing and switching ON the SETU VFXTH.
- **Features of SETU VFXTH:** Gives detailed instructions for understanding the features of SETU VFXTH. It also shows steps for programming and using various features like Automatic Number Translation, PIN Authentication, Emergency Number Dialing, IP Dialing, Peer to Peer Calling, etc.
- **Appendices:** Contains information such as Acronyms, Features at Glance, System Commands etc.

How to Read this System Manual

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

Using table of contents and the index: You may use the table of contents and Index to navigate through this document to the relevant topic or information you want to look up.

Cross-references: It is given in blue shaded text. You will be able to navigate easily through the System Manual with the help of Cross-references.

Relevant Topics: It is provided under each feature description. Under this sub-heading, you will find references to topics that are related to the feature/facility, you are currently referring to.

Conventions used in this System Manual

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

Note: *A note provides additional helpful information. It indicates something that requires your special attention or reminds you of something you need to do when you are using SETU VFXTH.*

Caution: *It indicates an action or condition that is likely to cause injury to you or to others.*

Warning: *It indicates a hazard or an action which can result in malfunction or damage to the SETU VFXTH or other equipments.*

Terminology used in this System Manual

Following terms are used interchangeably throughout this system manual:

1. '**SETU VFXTH**', '**SETU VFXTH1616', 'System**' and '**Gateway**'
2. '**System Engineer (SE)**' and '**you**'
3. '**Caller**' and '**Calling Party**'
4. '**Callee**' and '**Called party**'

Some of the terms used in this System Manual are defined below:

System Engineers (SE): A person who install, configure and maintains SETU VFXTH.

User: A person who uses SETU VFXTH.

Caller/ Calling party: A person who makes calls.

Callee/ Called party: A person who receives call/ to whom call is made.

Source/ Originating Port: A port from which a call originates.

Destination/ Terminating Port: A port on which a call terminates.

SETU VFXTH is available in different configurations depending on the requirement of the user/organization. To see various configurations supported by SETU VFXTH, please refer “Product Specifications” in the Appendix.

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

CHAPTER 2

Know Your SETU VFXTH

Overview of SETU VFXTH

SETU VFXTH is a gateway that provides voice services over IP network using SIP protocol. It is an effective and flexible solution for accessing internet based telephone services and corporate intranet systems across established LAN.

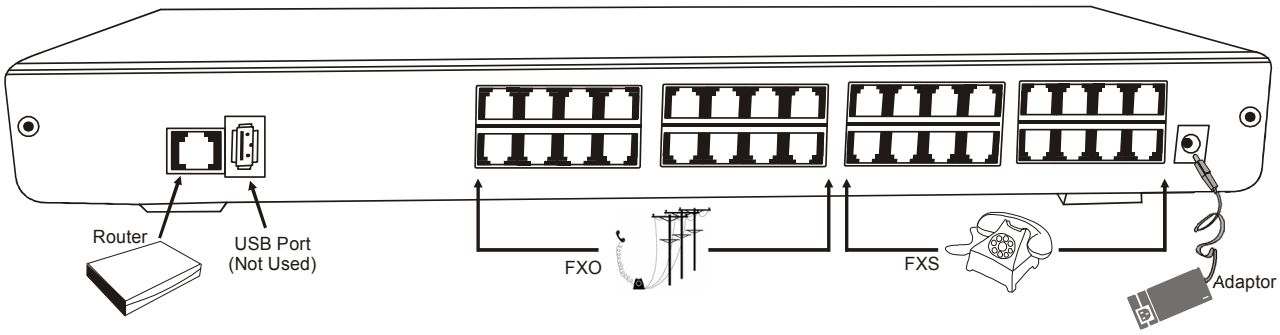
SETU VFXTH is a gateway and not a PBX. It is an innovative enterprise gateway that offers excellent functionality and sound quality. It supports different voice ports viz. FXO, FXS and SIP for providing voice services. It is developed to fulfill the requirements of SOHO (Small Office-Home Office) users and Small and Medium scale Enterprises.

SETU VFXTH is available in following different configurations and physical port assignment:

Sr. No.	Configuration	VoIP Channels	FXO Ports	FXS Ports	Physical Ports Label	
					FXO	FXS
1	SETU VFXTH0016	16	0	16	0	P01-P16
2	SETU VFXTH0024	24	0	24	0	P01-P24
3	SETU VFXTH0032	32	0	32	0	P01-P32
4	SETU VFXTH0800	8	8	0	P01-P08	0
5	SETU VFXTH1600	16	16	0	P01-P16	0
6	SETU VFXTH2400	24	24	0	P01-P24	0
7	SETU VFXTH3200	32	32	0	P01-P32	0
8	SETU VFXTH0808	16	8	8	P01-P08	P09-P16
9	SETU VFXTH1212	24	12	12	P01-P12	P13-P24
10	SETU VFXTH1616	32	16	16	P01-P16	P17-P32

The explanation in this system manual is given keeping in mind the 10th configuration (i.e. SETU VFXTH1616) of the above table. Readers are requested to manipulate the instructions as per their system configuration.

SETU VFXTH1616 has 1 WAN Port, 16 FXO Ports, 16 FXS Ports, 32 SIP Trunks, a Power Socket and 34 LEDs. Depending on the configuration the number of FXO and FXS Ports may vary.



Ethernet (WAN) Port

The Ethernet (WAN) port labeled as ETHERNET is used to connect SETU VFXTH1616 to the Public or Private network. (Refer “Applications of SETU VFXTH”)

FXO Ports

The FXO Ports labeled as P01, P02, P03…..upto P16 are used to connect SETU VFXTH1616 to the PSTN network or to the PBX.

FXS Ports

The FXS Ports labeled as P17, P18, P19….upto P32 are used to connect telephone instruments to SETU VFXTH1616.

Power Socket

A Power socket labeled as 24V DC is used to power SETU VFXTH1616 using 24V DC, 2.5A power adaptor (supplied by Matrix).

SIP Trunks

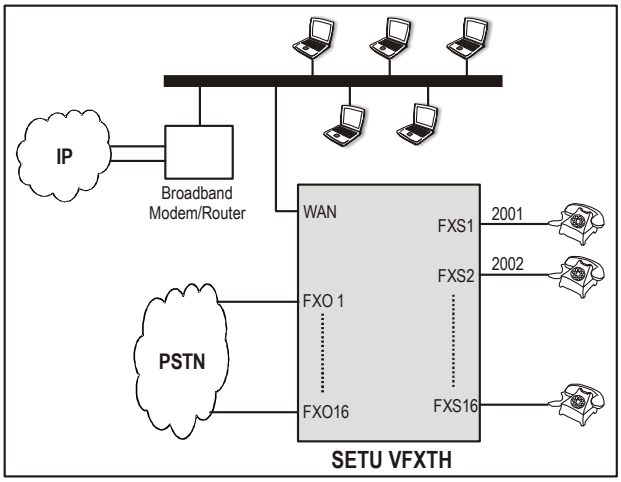
SETU VFXTH1616 supports 32 SIP Trunks. You can register these SIP Trunks either with one ITSP or with different ITSPs.

LEDs

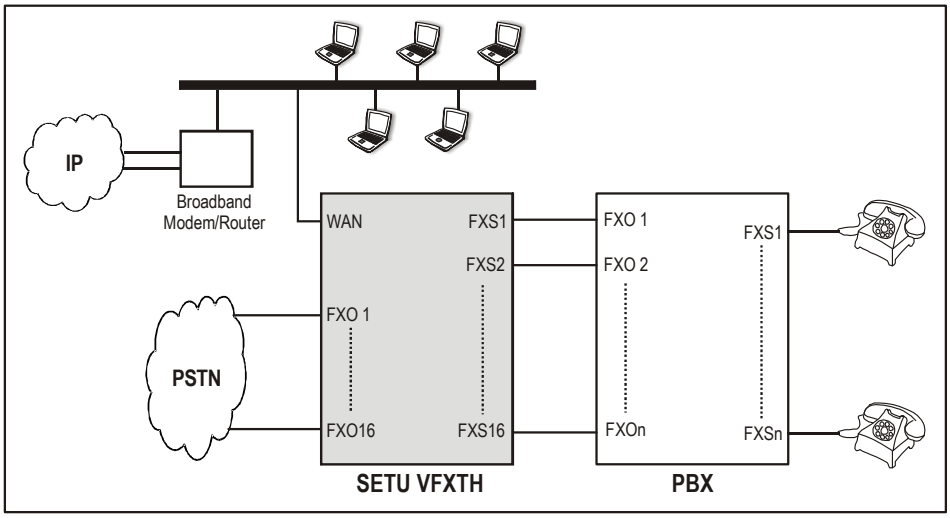
There are total 34 LEDs in SETU VFXTH1616. Power LED is of single colour (Green). All Port LEDs are of single colour (Red). Status LED is of double colour (Red/Green). These LEDs indicate the status of ports, various events occurring on the ports and also the error conditions.

Applications of SETU VFXTH

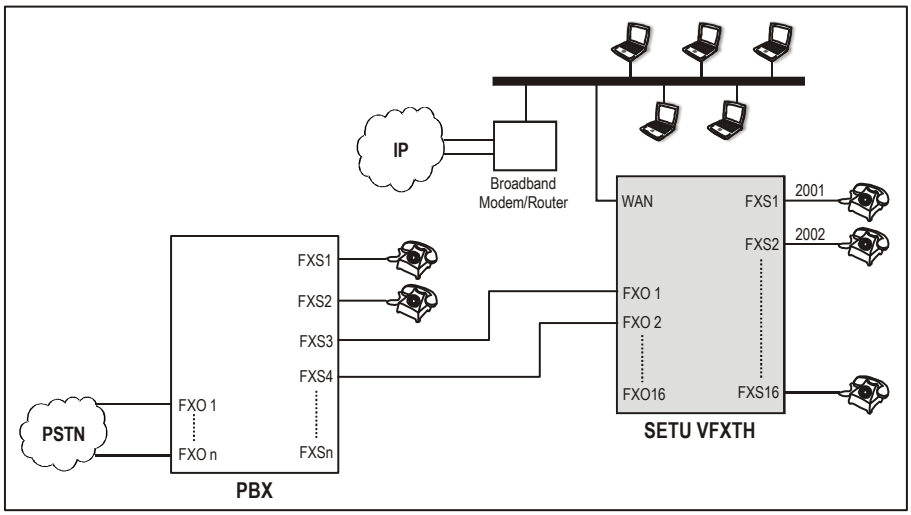
Case 1: Standalone



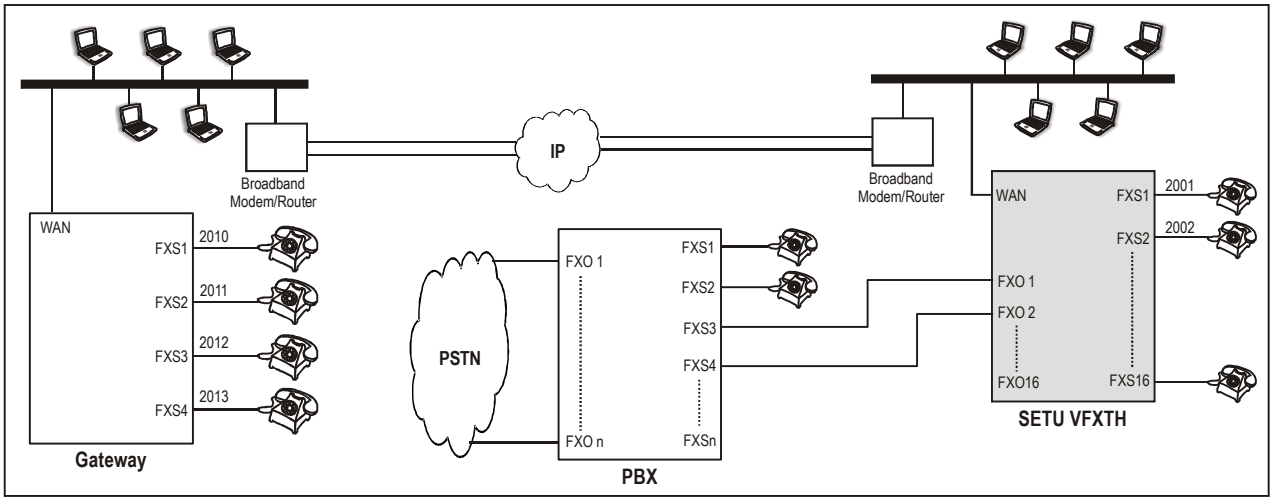
Case 2: Infront of PBX



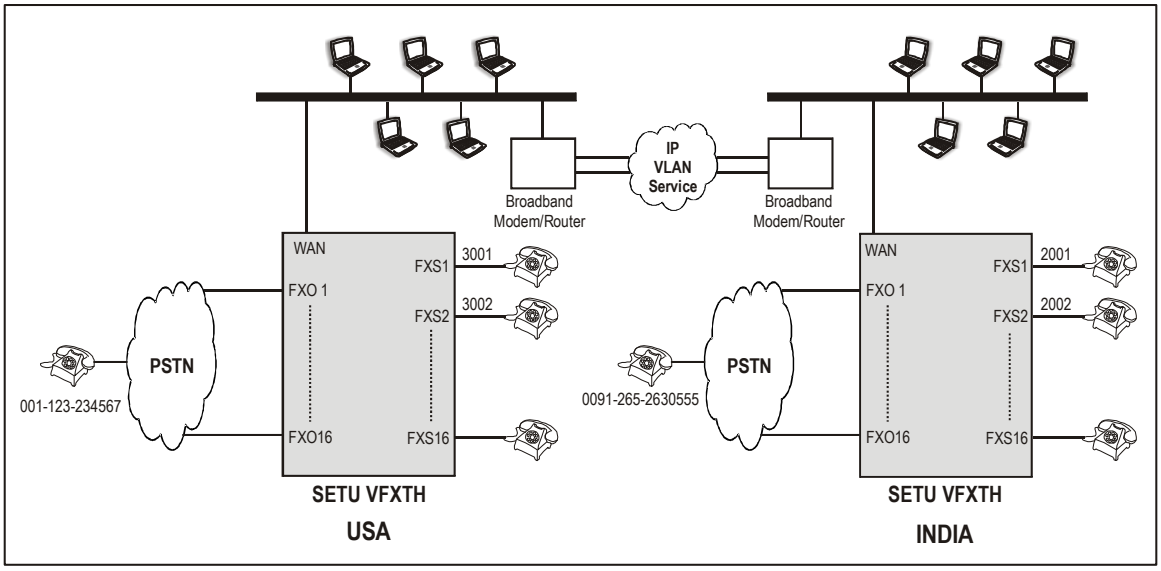
Case 3: Behind the PBX



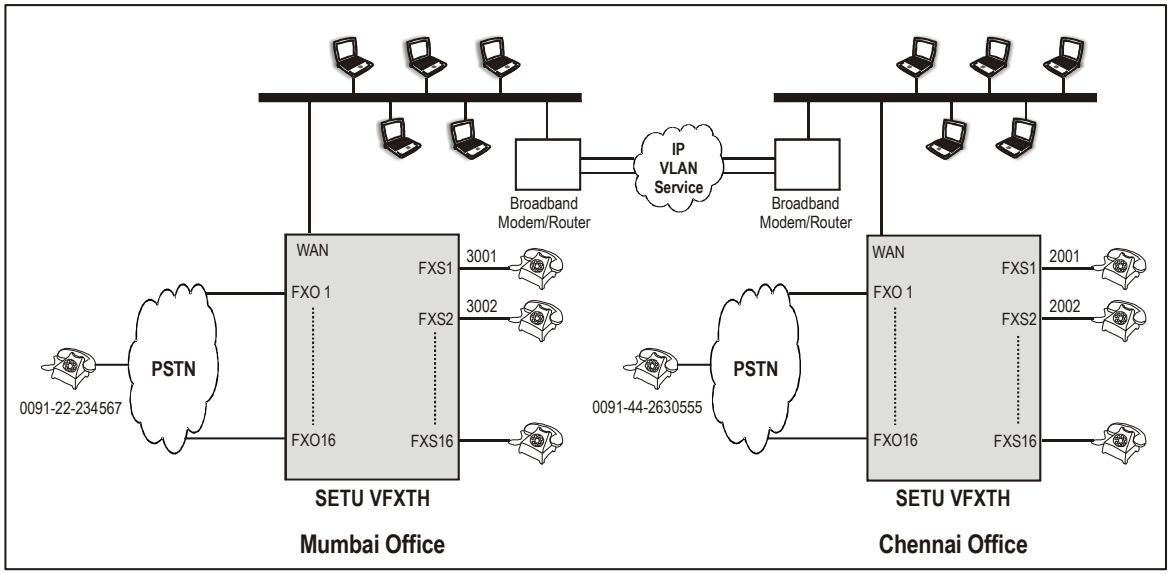
Case 4: Analog Extension of PBX over IP (Remote Extension)



Case 5: PSTN Call over IP (Long Distance converted to Local Call)



Case 6: Peer-to-Peer Calling



CHAPTER 3 **Getting Started**

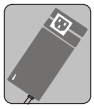
Installing SETU VFXTH

Pre-Installation Tips

You are advised to follow the tips given below before you start installing SETU VFXTH.

1. Verify the package contents. The sales kit of SETU VFXTH contains:


SETU VFXTH


Adaptor 24VDC, 2.5A
(Country Specific)


Quick Start and User Card


CD containing System Manual,
Quick Start and User Card


Ethernet Cable (RJ45)

- Two Screws M 7/30 with Grips
- External Antenna SMA with cable
- A Mounting Template
- A Warranty Card Set

Make sure that all the above mentioned components are present when you open the Sales kit of SETU VFXTH. In case any of the part is missing or damaged, contact the vendor/ System administrator from whom you have purchased it.

2. You should take following measures to protect your SETU VFXTH:

- Read and understand all the instructions given in the manual.
- Do not turn on the power supply until the installation is complete.
- Do not open SETU VFXTH in power ON condition.
- SETU VFXTH should be operated within recommended power supply voltage range.
- Install SETU VFXTH at the place where there is enough open space for ventilation.
- Do not install SETU VFXTH in direct sunlight and the place where there is excessive cold or humid atmosphere.
- Do not install SETU VFXTH at dusty places or places where it may come in direct contact with oil or water.
- Do not place the product at the place from where it can fall and serious damage may be caused to it.
- Do not install SETU VFXTH at the places where shocks or vibrations are frequent or strong.

- Reduce the risk of electric shock or damage to SETU VFXTH by taking it to a qualified serviceman for repair work.

3.3V Battery: *SETU VFXTH contains 3.3V DC Manganese Lithium Battery (ML 2032- Rechargeable) of diameter 20 (-0.2)mm and height 3.2 (-0.3)mm. Battery location is marked as 'B1' near J1 jumper on the CPU Board. Battery should be replaced only by authorized dealers of Matrix.*

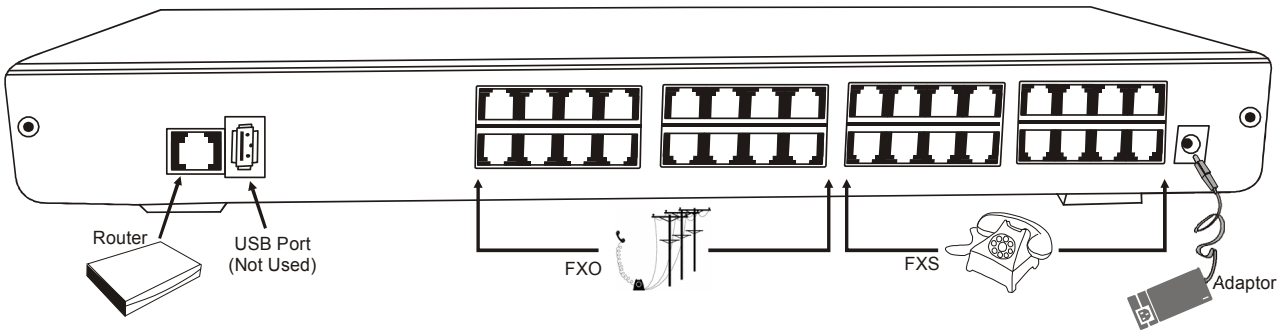
3. Contact qualified service personnel under following conditions:

   - When the power supply cord or plug is damaged or frayed.
   - 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 by following the operating instructions.
   - If it exhibits an erratic behavior.

Mounting SETU VFXTH on Wall

- Select a suitable place on the wall for mounting SETU VFXTH.
- Put the mounting template on the wall and mark the nail hole on the wall.
- Drill a hole of appropriate size.
- Insert the screw and tighten it leaving the screw head a few millimeters protruded of the wall.
- Check the strength of the nail.
- Hang SETU VFXTH on the wall.

Connecting SETU VFXTH



- Place SETU VFXTH at a suitable place where it can be provided with proper power supply.
- Connect Ethernet Port of SETU VFXTH to the LAN Switch/Network of the Enterprise. You can also connect a computer to the Ethernet Port.
- Connect PSTN line to the FXO Port of SETU VFXTH.
- Connect telephone instruments to the FXS Ports of SETU VFXTH.
- Connect Power Socket of SETU VFXTH to the power supply using 24V DC, 2.5A power adaptor.

Switching on the System

After connecting SETU VFXTH1616 as shown above, switch ON the power supply.

At Power ON, Power LED will turn ON (Continuous Green). Initialization process will start within 15 to 20 seconds and LED sequence of all other LEDs during initialization process is shown in the table given below:

System Status	STS	P01	P02	P03	P04	P05	P06	….	P32	Time (ms)
Application Load	OFF	**ON**	**ON**	**ON**	OFF	OFF	OFF	OFF	OFF	
All Init Done, System goes Live	**ON**	**ON**	**ON**	**ON**	**ON**	**ON**	**ON**	**ON**	**ON**	**1000 ms**
	OFF	**OFF**	**OFF**	**OFF**	**OFF**	**OFF**	**OFF**	**OFF**	**OFF**	**1000 ms**

*Red color text **(Bold)** indicates that the led will glow **red**.*

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

LED Status	**Colour**	**Comment**
Continuous On	Red	VoPP program download fail
1sec On - 1sec Off	Green	SETU VFXTH1616 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 and 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
400ms On - 400ms Off (continuous)	Red	Off-Hook Event (Dialing State)
Continuous On	Red	Speech

During Off-Hook state, FXS Port LED glows Red, 400ms On - 400ms Off (continuous). However, if the system is in programming mode, the LED of the port used to enter programming mode will glow continuous Red. After exiting programming mode, it again displays Off-Hook event.

After switching on the system, access Web JEEVES and configure the system as per your requirement.

Accessing Web JEEVES

You can program all the parameters of SETU VFXTH1616 using Web JEEVES. A few network port parameters such as IP Address, Subnet Mask, Connection Type, etc. necessary to set up the system, can also be programmed by using a telephone instrument.

You might need to change the network port parameters in most cases to make SETU VFXTH1616 a part of your network and to access the Web JEEVES. In such cases, it is recommended that you connect telephone instrument to the FXS Port and change network port parameters using telephone instrument first (Refer "Network Port Parameters" topic for details) and subsequently access Web JEEVES to configure rest of the system.

You can access Web JEEVES of SETU VFXTH1616 in two different ways.

- To access Jeeves of SETU VFXTH1616, it must be connected to a computer. The computer may be a standalone PC or a PC in a LAN.
- Also SETU VFXTH1616 and the computer must be in the same Subnet and their IP Addresses should be different.
- Obtain from your LAN administrator, the IP Address and the Subnet mask of the standalone PC/ LAN PC from which you wish to access the Web JEEVES.
- ***The default IP Address and Subnet mask of SETU VFXTH1616 are 192.168.001.136 and 255.255.255.000 respectively.***

Follow the steps given below:

1. Change the IP Address of SETU VFXTH1616, if it is conflicting with the IP Address of the PC to which it is connected or with any device on the LAN. Similarly, assign the Subnet of the PC to SETU VFXTH1616, if it is not the same. To do this:

   a. Pick up the handset of analog phone connected to SETU VFXTH1616 and dial #19-1234. You will get programming tone. (#19 is the default programming access code and 1234 is the default SE Password.)

   b. Change IP address by issuing command: ***11-IP Address-#****
   Where,
   IP address is of 12 digits in XXX.XXX.XXX.XXX format. Each octet is of three digits ranging from 001 to 254. For example, to program IP address 192.168.1.120, enter the command ***11-192168001120-#****.

   c. Change Subnet Mask by issuing command: ***12-Subnet Mask-#****
   Where,
   Subnet Mask is of 12 digits in XXX.XXX.XXX.XXX format. Each octet is of three digits. Valid range is 0, 128, 192, 224, 240, 248, 252, 254 and 255. For example, to program Subnet Mask 255.255.254.0, enter the command ***12-255255254000-#****.

2. Exit programming mode by issuing command: ***00#****

3. Open web browser of the computer connected to SETU VFXTH1616.

4. Enter the IP Address (default or changed) of SETU VFXTH1616 in the address bar of the web browser. Login using the default SE Password 1234 into the JEEVES and program necessary parameters by clicking on links given on the left side.

   OR

1. Change IP Address and Subnet Mask of the computer connected to SETU VFXTH1616 in such a way that the WAN Port of SETU VFXTH1616 and the computer are in the same subnet and their IP Addresses are different.

2. Open Web Jeeves of SETU VFXTH1616 by entering the default IP Address in the URL field of the web browser of your computer.

3. Login using default SE password 1234 into the JEEVES and go to 'Network Port Parameters' page. Change the IP Address, Subnet Mask and other necessary parameters to make it a part of your network. The system shall reboot.

4. Once the necessary changes are made, restore the earlier settings of the computer. Open Web JEEVES of SETU VFXTH1616 and program necessary parameters of SETU VFXTH1616 by clicking on links given on the left side.

Detailed instructions for programming various features of SETU VFXTH1616 are provided in chapter: “Features of SETU VFXTH”.

After switching on the SETU VFXTH1616, you are recommended to program it in the following sequence:

1. “Region Selection”
2. “Date and Time Settings”
3. “Network Port Parameters”
4. “Port Parameters-SIP”
5. “Port Parameters-FXS”
6. “Port Parameters-FXO”
7. “Routing Group”
8. “Routing Mechanism on SIP”
9. “Routing Mechanism on FXS”
10. “Routing Mechanism on FXO” and so on...

Checking the Status

You can check the status of various ports in following ways:

1. Through LEDs
2. By Issuing Command
3. Through Web JEEVES

Checking Status through LEDs

SETU VFXTH1616 has 34 LEDs viz. PWR, STS, P01, P02, P03, P04... P32. By default, LEDs labeled as P01, P02 upto P16 show status of FXO Ports and P17, P18 upto P32 show status of FXS Ports.

Checking Status by issuing Command

No LED indication is given for SIP Trunks in SETU VFXTH1616. As a result, you cannot check the status of SIP Trunks through LEDs. However, you can check the status of SIP Trunks on the LCD of the conventional phone by issuing command.

Use following command to display the status of SIP Trunk:
27-SIP Trunk-#*
Where,
SIP Trunk is from 1 to 32.
Following table shows various Error/Event/Status of SIP Trunks that can be displayed on the LCD of the telephone instrument:

Number Field Display (For DTMF and FSK CLI type) Code	**Name Field Display (For FSK CLI type only) Description**
001	Disable
002	Registering
003	Active
004	Failed - Auth.
005	Failed - Other

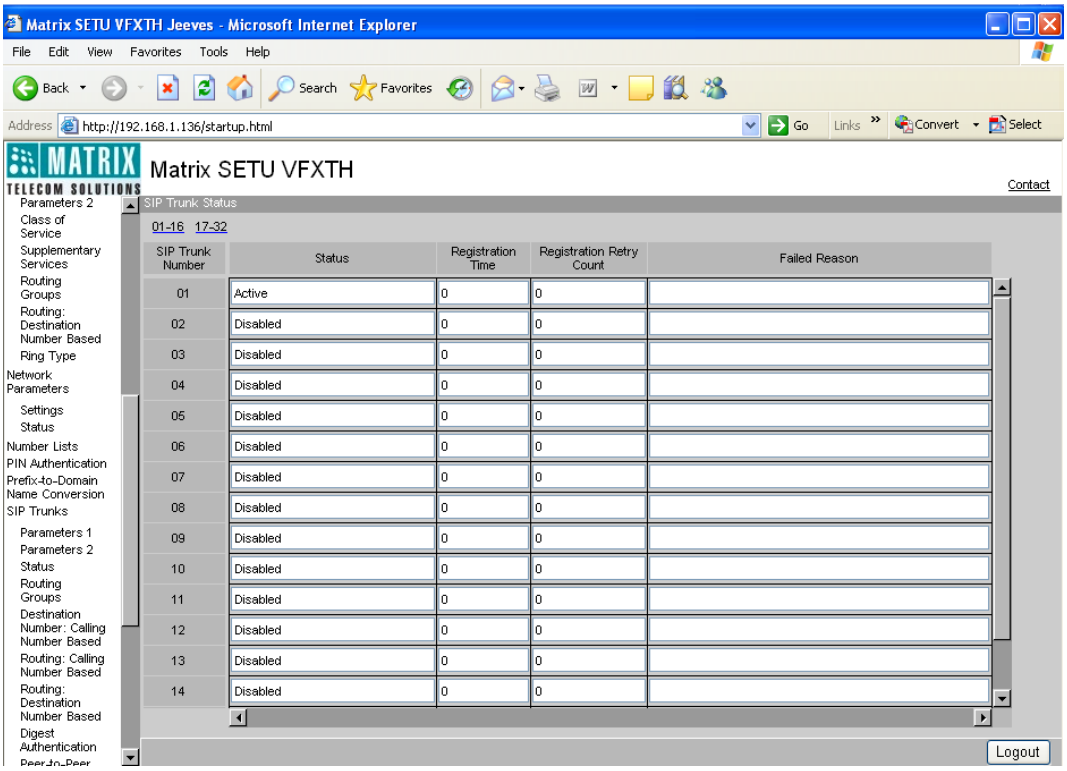
Only code will be displayed on the LCD of conventional phone, if CLI Type programmed on FXS Port is DTMF and it should be interpreted using above table. Both code and description will be displayed if CLI Type programmed on FXS Port is FSK.

Checking Status through Web JEEVES

SETU VFXTH1616 provides you the facility to check status of SIP Trunks and Network Parameters using Web JEEVES.

SIP Trunks Status

Open Web JEEVES of SETU VFXTH1616 and click on the **'SIP Trunk Status'** link. Following page will be displayed:



Following is a brief description of the parameters displayed on the 'SIP Trunk Status' page.

1. **SIP Trunk Number:** This field displays the SIP Trunk number.

2. **Status:** This field displays the status of SIP Trunk. Different status option which can appear in this field is explained below:

Status	Description
Disable	Shows that SIP Trunk is disable
Registering	Shows that SIP Trunk is enable and waiting for response from the SIP server
Active	Shows that SIP Trunk is registered with the SIP server.
Failed	Shows that some error has occurred in the SIP Trunk and no calls can be made using it (applicable only in case of Proxy Account).

3. **Registration Time:** This field displays the time left to re-register the SIP Trunk after it gets registered successfully. This time is provided by the registrar server. When the registration period gets over, SIP Trunk would again get registered automatically.

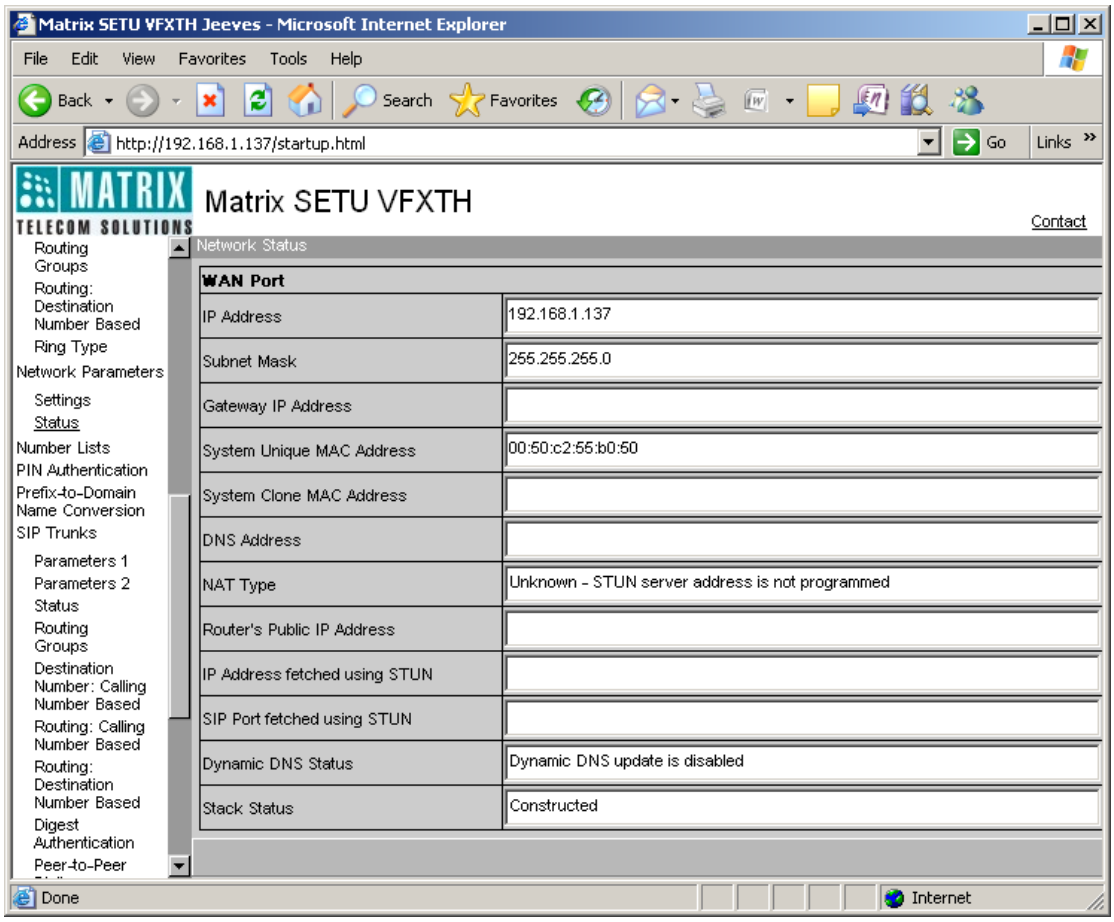
4. **Registration Retry Count:** This field displays the total number of register message which is sent to the registrar server for registering SIP Trunk.

5. **Failed Reason:** This field displays the reason for registration failure, if registration of SIP Trunk fails with the registrar server. Different registration failure reasons which can appear in this field are shown below:

Failed Reason	Description
Message send fail	This reason is displayed when registration request sent to registrar server fails
Failed to create Register client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ number of SIP client to register is more than programmed in the stack
Failed to detach Register client	This reason is displayed when SIP stack has memory constraint/ resource limitation/ number of SIP client to register is more than 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 fail
Response timeout	This reason is displayed after the expiry of General Request Timer
Error Response- 4xx to 6xx	This is error response code
No contact header in 2xx	This reason is displayed when no contact address is received in 2xx response from SIP server
Authentication Failed	This reason is displayed when 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 query to STUN server fails
Outbound address is not programmed	This reason is displayed when Outbound is enable 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 address is not programmed.

Network Status

Open Web JEEVES of SETU VFXTH1616 and click on the '**Network Status**' link. Following page will be displayed:



Following is a brief description of the parameters displayed on the 'Network Status' page.

1. **IP Address:** This field displays IP address currently assigned to SETU VFXTH1616.
2. **Subnet Mask:** This field displays Subnet Mask currently assigned to SETU VFXTH1616.
3. **Gateway Address:** This field displays Gateway Address assigned to SETU VFXTH1616.
4. **MAC Address:** This field displays MAC Address assigned to SETU VFXTH1616.
5. **DNS Address:** This field displays the DNS address of SETU VFXTH1616.
6. **NAT Type:** This field displays NAT Type, if STUN is enabled in SETU VFXTH1616. Commonly used NAT types are:
    - Unknown
    - Open
    - Conenat
    - Restrictednat
    - Portrestrictednat
    - Symmetricnat
    - Symmetricfirewall
    - Blocked

7. **Router's Public IP Address:** This field displays Router's Public IP address programmed in the Network Port Parameters.

8. **IP Address fetched using STUN:** This field displays the IP address fetched using STUN if STUN server address is programmed.

9. **SIP Port fetched using STUN:** This field displays the SIP port fetched using STUN if STUN server address is programmed.

10. **Dynamic DNS Status:** This field displays the response received from DDNS server while sending the IP Address update request to the server. Different responses that can appear in this field are shown below:

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.	

11. **Stack Status:** In this field, strings such as Idle, DHCP Response wait, PPPoE response wait, NAT checking response wait, construct and error is displayed.

Restart the System

Sometimes it becomes necessary for the user to restart SETU VFXTH1616. Many a times it is difficult to walk to the system and restart it. Thus, SETU VFXTH1616 gives facility to restart the system remotely through Web JEEVES. Restarting SETU VFXTH1616 gives the same effect as switching OFF the system and switching it ON again. (Refer “Restart SETU VFXTH” for more details.)

Default the System

User can default SETU VFXTH1616 and assign default values to all programmable parameters except:

1. Call Detail Records
2. Date and Time
3. Call Progress Tones

4. Ring Type
5. Region/Country

(Refer “Default SETU VFXTH” feature for more details.)

You can also change or default the SE password in case it is disclosed to an unauthorized person or is lost or forgotten. To change or to default SE password, refer “SE Password” feature.

Software Upgrade

Due to continuous feedbacks from the customers and their requirements, a software change becomes inevitable. SETU VFXTH1616 provides facility to upgrade the software to latest version through Web JEEVES. (Refer “Software Upgrade” feature for more details)

Test Calls

After completing basic programming and switching on the system, make test calls.

Making an Outgoing Call:

To make an outgoing call to a remote SIP user from FXS Port, ensure that the Destination Port Determination Method programmed for the FXS Port is 'Fixed' and Port/Group Type programmed is 'SIP Trunk'.

1. Pick up handset of the telephone instrument connected to FXS Port.
2. You will get a dial tone.
3. Dial 123@abc.com. The call will be routed to the dialed destination through the SIP Trunk.
4. Talk when the remote SIP user answers the call.
5. Replace the handset or dial Call Disconnect Access Code to disconnect the call.

Receiving an Incoming Call:

To receive an incoming call on SIP Trunk, ensure that Destination Number Determination Method programmed for the SIP Trunk is 'Number Not Required' and 'Destination Port Determination Method' is 'Fixed'. Ensure that the Port/Group Type programmed for routing the call landed on the SIP Trunk is 'FXS Port'. When remote caller calls SIP Trunk of SETU VFXTH1616, the call is routed to the FXS Port and the telephone instrument connected to that FXS Port starts ringing.

1. Pick up handset of the telephone instrument connected to the FXS Port.
2. Talk
3. Replace the handset after the remote caller disconnects the call.

CHAPTER 4

Features of SETU VFXTH

Access Codes

Access Code is a string of digits dialed by SE/User to:

- Enable/Disable a feature
- Access Supplementary feature
- Enter into the programming mode

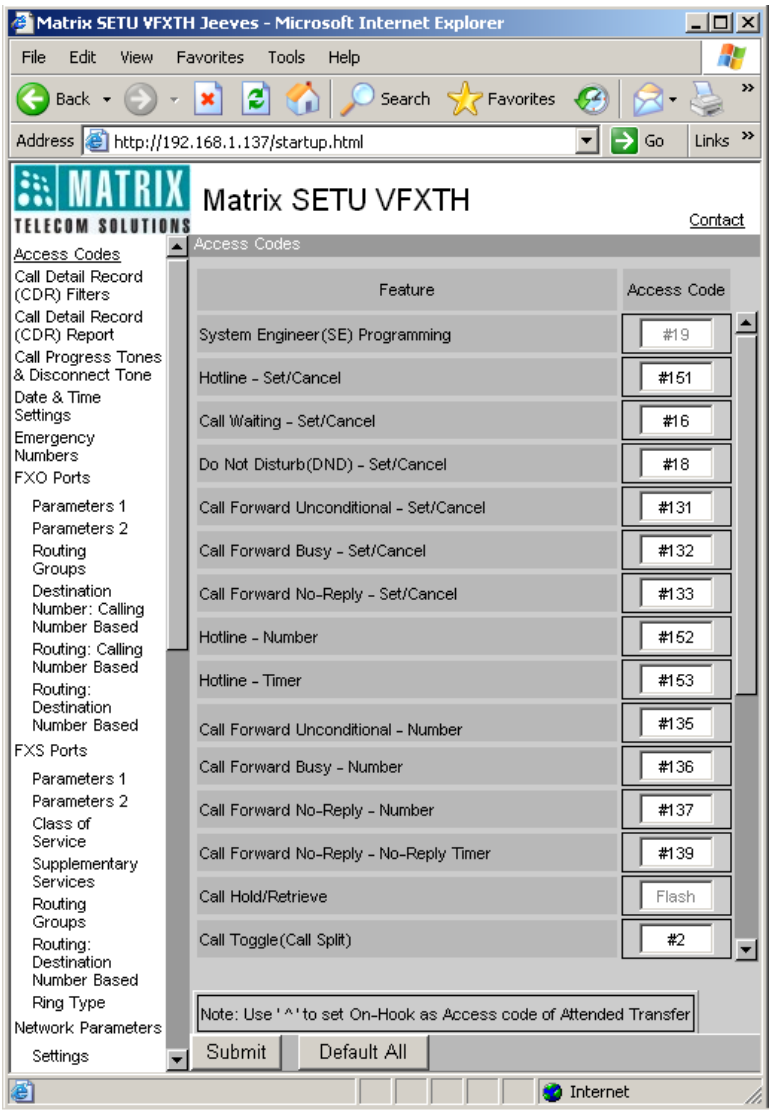
SETU VFXTH provides default Access Codes for all features. You can either use the default Access Codes or change them to suit your preferences. However, access codes for SE Programming and Call Hold/Retrieve functions are non-programmable i.e. you cannot change them.

How to Program?

Follow the steps given below to change the default Access Codes in the Access Code table:

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Access Codes'** link to open the Access Code page.



- Change the Access Codes as per your choice. Access Codes can be of maximum 4 digits and digits 0-9, *, # and ^ are allowed. (^ is for 'On hook')

- Click on **'Submit'** link at the bottom of the page to apply changes. SETU VFXTH does not allow programming of conflicting numbers in the Access Code table. If you enter a code that is already assigned to some other feature, following message will appear "Conflict number is not allowed" and the value will remain unchanged.

- Click on **'Default All'** link to default all the Access Codes.

a. *Emergency Numbers have priority over Access Codes and Access Codes have priority over the Destination Numbers.*

b. *User can dial Access Code from the FXS Port even if outgoing calls are disabled for that FXS Port.*

c. *Except 'Making New Call' and 'Disconnect Call' all other Access Codes can be dialed from FXS Port only.*

d. *If your SETU VFXTH does not have FXS Port then you will not be able to use any Access Codes except that of 'Making New Call' and 'Disconnect Call'. In other words, you will be able to access only 'Making New Call' and 'Disconnect Call' feature.*

Default Access Codes table as shown below:

Features	Access Code
System Engineer (SE) Programming	***#19***
Hotline-Set/Cancel	***#151***
Call Waiting-Set/Cancel	***#16***
Do Not Disturb(DND)-Set/Cancel	***#18***
Call Forward Unconditional-Set/Cancel	***#131***
Call Forward Busy-Set/Cancel	***#132***
Call Forward No Reply-Set/Cancel	***#133***
Hotline-Number	***#152***
Hotline-Timer	***#153***
Call Forward Unconditional-Number	***#135***
Call Forward Busy-Number	***#136***
Call Forward No Reply-Number	***#137***
Call Forward No Reply-No Reply Timer	***#139***
Call Hold/Retrieve	***Flash***
Call Toggle (Call Split)	***#2***
Reject Waiting Call and Speech with Current Call	***#31***
Ignore Waiting Call and Speech with Current Call	***#32***
Accept Waiting Call and Hold Current Call	***#33***
Accept Waiting Call and Release Current Call	***#34***
Blind Transfer	***#6***
Conference	***#8***
Using Supplementary Services of Service Provider	***#4***
Attended Transfer	***^***
Making New Call	***#91***
Disconnect Call	***#92***

Relevant Topics:

1. “Allowed-Denied Numbers” 28
2. “Class of Service” 55
3. “Call Processing” 47
4. “Emergency Number Dialing” 66
5. “Routing Mechanism on FXO” 143
6. “Routing Mechanism on FXS” 155
7. “Routing Mechanism on SIP” 161

Allowed-Denied Numbers

Allowed-Denied Number feature provides flexibility to allow or deny dialing of a particular number or a set of numbers from a particular port or from all ports. This feature restricts users from dialing numbers that are programmed in the Denied Number List.

This feature is applicable on source port only. You should program allowed numbers and denied numbers in different Number Lists and assign these Number Lists to the source port. (Refer "Number Lists" topic for feature explanation of number lists.)

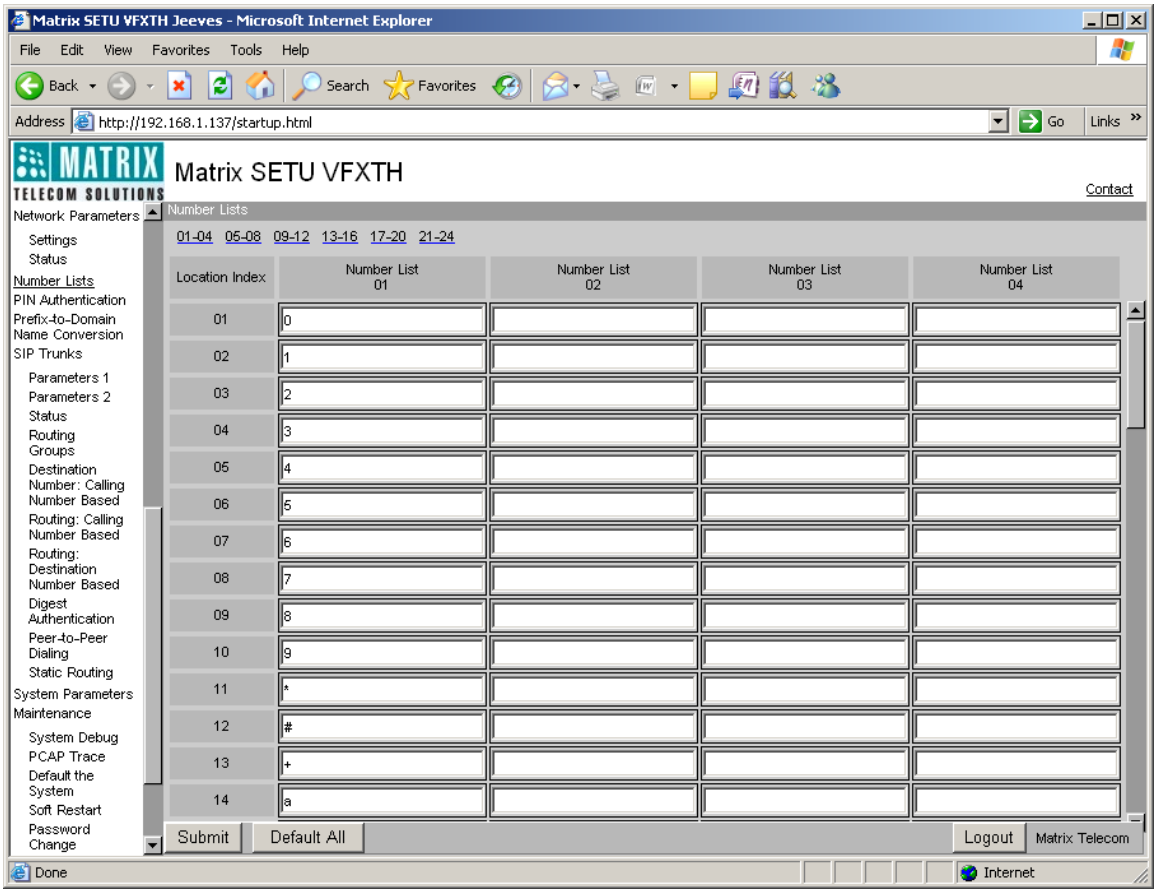
Allowed-Denied Number feature is not applicable in following cases:

1. Destination number string matches with any Access Code.
2. Destination number string matches with any Emergency Number.
3. For Hotline Number programmed.
4. For Call Forward Number programmed.
5. For Fixed Destination Number programmed.
6. For Destination Number collected through CLI Based Destination Number Determination Method.

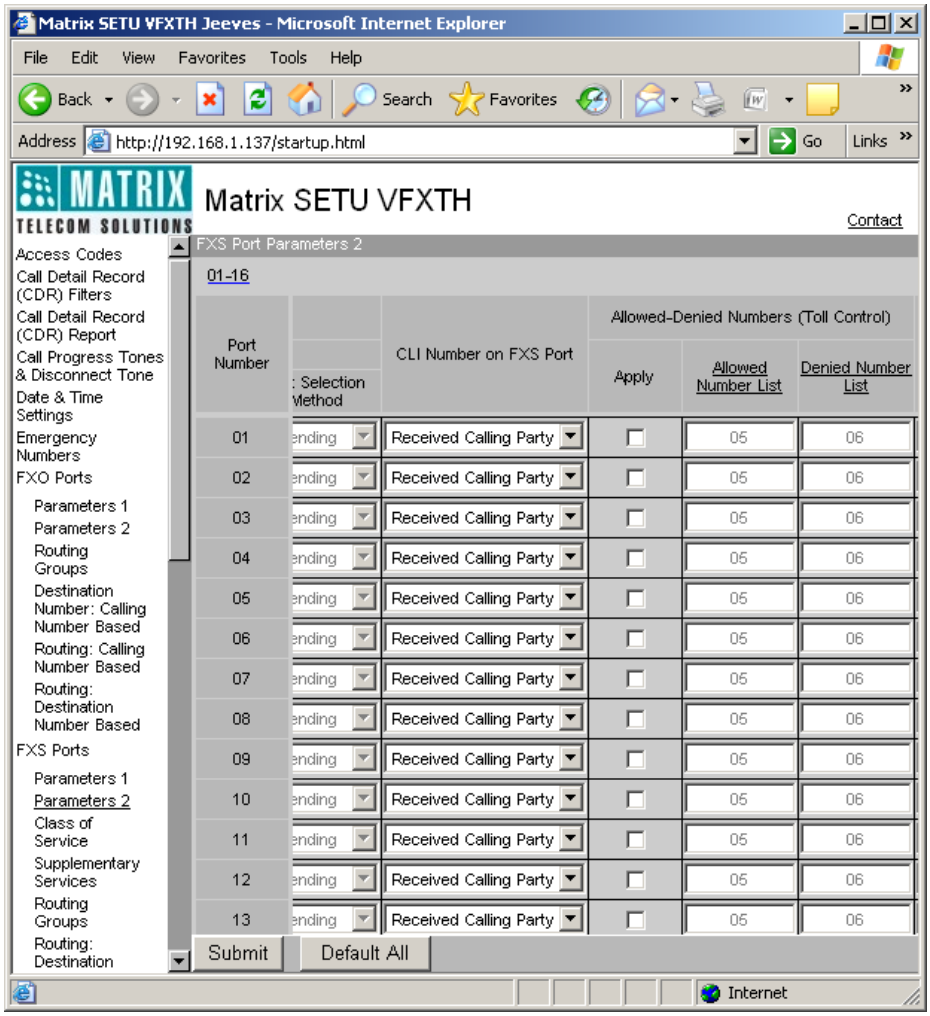
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")

- Click on **'Number Lists'** link. Program numbers to be allowed and those to be denied from FXS Port in different Number Lists. Similarly program allowed and denied numbers for FXO Ports and SIP Trunks in different Number Lists. (Refer "Number Lists" for more details)

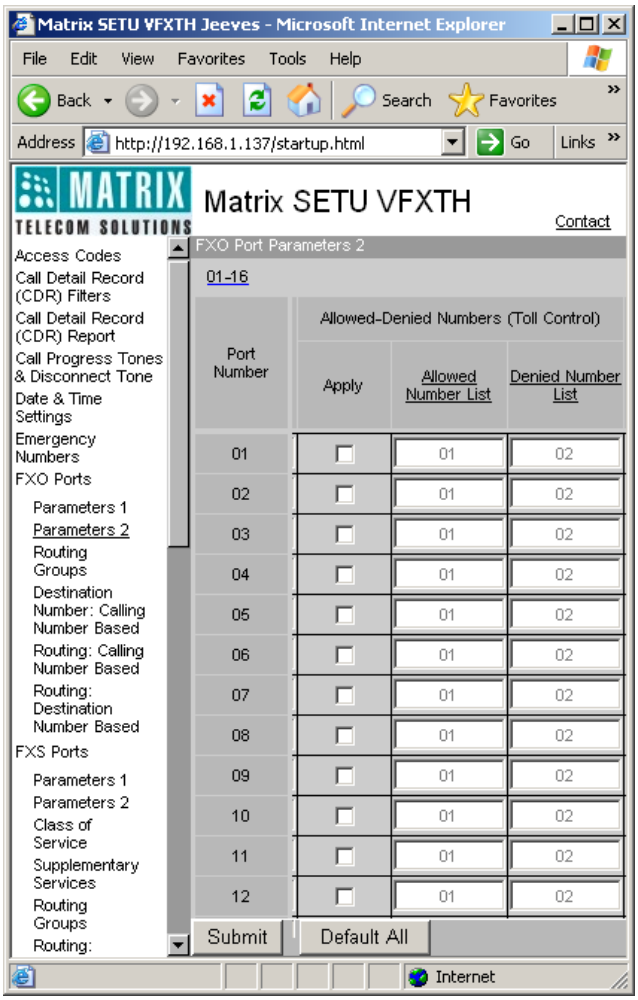


- Click on **'FXS Port Parameters 2'** link. Go to the Allowed-Denied Numbers (Toll Control) column.



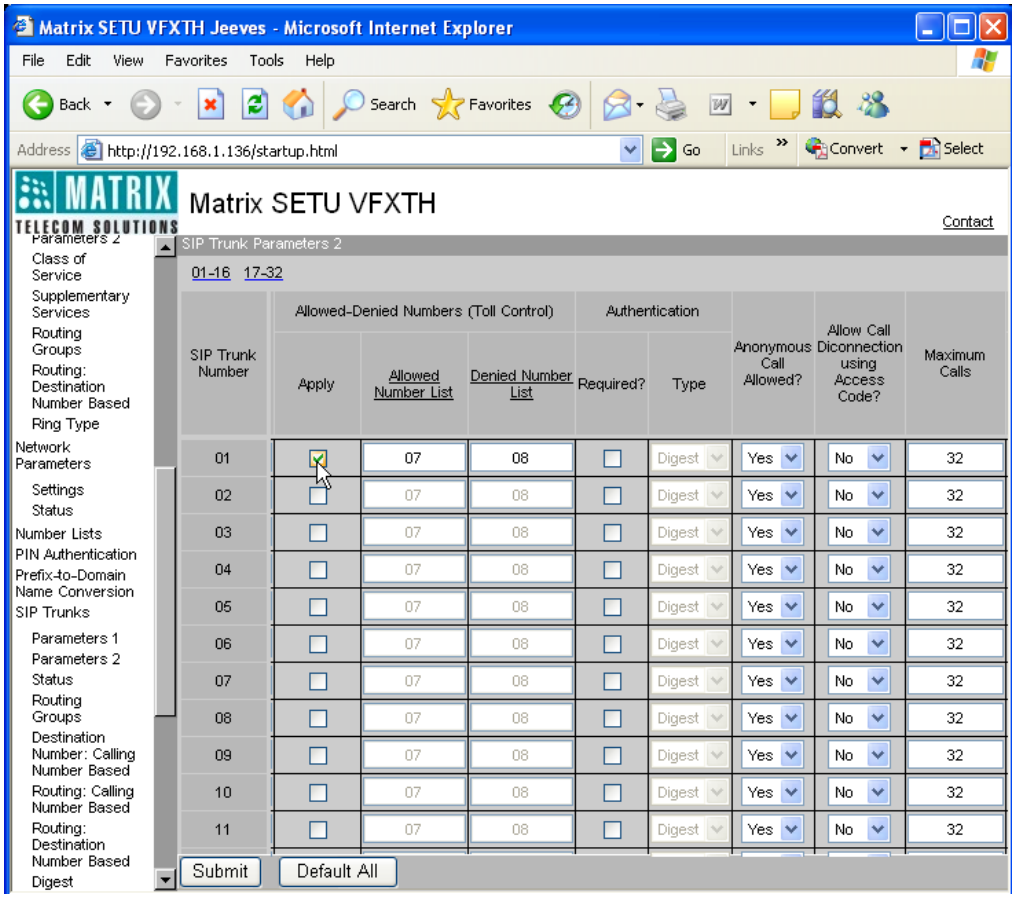
- **Apply:** Tick this checkbox to enable this feature for FXS Port. By default, it is Untick.
- **Allowed Number List:** Enter number of the Number List in which you have programmed numbers to be allowed from FXS Port in this column. By default, list number 05 is programmed for each FXS Port.
- **Denied Number List:** Enter number of the Number List in which you have programmed numbers to be denied from FXS Port in this column. By default, list number 06 is programmed for each FXS Port.
- Click on **'Submit'** link at the bottom of the page to apply the changes.

- Click on **'FXO Port Parameters 2'** link. Go to the Allowed-Denied Numbers (Toll Control) column.



- **Apply:** Tick this checkbox to enable this feature for FXO Port. By default, it is Untick.
- **Allowed Number List:** Enter number of the Number List in which you have programmed numbers to be allowed from FXO Port in this column. By default, list number 01 is programmed for each FXO Port.
- **Denied Number List:** Enter number of the Number List in which you have programmed numbers to be denied from FXO Port in this column. By default, list number 02 is programmed for each FXO Port.
- Click on **'Submit'** link at the bottom of the page to apply the changes.

- Click on **'SIP Trunk Parameters 2'** link. Go to the Allowed-Denied Numbers (Toll Control) column.



- **Apply:** Tick this checkbox to enable this feature for SIP Trunk. By default, it is Untick.
- **Allowed Number List:** Enter number of the Number List in which you have programmed numbers to be allowed from SIP Trunk in this column. By default, list number 07 is programmed for each SIP Trunk.
- **Denied Number List:** Enter number of the Number List in which you have programmed numbers to be denied from SIP Trunk in this column. By default, list number 08 is programmed for each SIP Trunk.
- Click on **'Submit'** link at the bottom of the page to apply the changes.

a. *SETU VFXTH uses best match found logic while comparing the numbers with Allowed and Denied Number List.*

b. *SETU VFXTH does not compare the dialed number string with Allowed- Denied Number Lists if it matches with any Emergency Number or Access Code.*

c. *If same number is programmed in both Allowed and Denied Number List then SETU VFXTH dials out that number.*

d. *If dialed number does not match any of the numbers programmed in Allowed and Denied Number List then SETU VFXTH dials out that number.*

Relevant Topics:

1. “Access Codes” 25
2. “Call Processing” 47
3. “Emergency Number Dialing” 66
4. “Number Lists” 84
5. “Port Parameters-FXO” 104
6. “Port Parameters-FXS” 112
7. “Port Parameters-SIP” 117
8. “Routing Mechanism on FXO” 143
9. “Routing Mechanism on FXS” 155
10. “Routing Mechanism on SIP” 161

Automatic Number Translation

SETU VFXTH supports multiple port types' viz. FXO, FXS and SIP. When a number is dialed from any of these ports, SETU VFXTH routes the call to the desired destination as per the routing mechanism determined for that port.

Each network understands unique numbering scheme. Sometimes, when user dials out a number, the dialed number string is not understood by the network through which the call is to be routed. Automatic Number Translation feature facilitates conversion of the dialed number string into the number that is understood by the destination network.

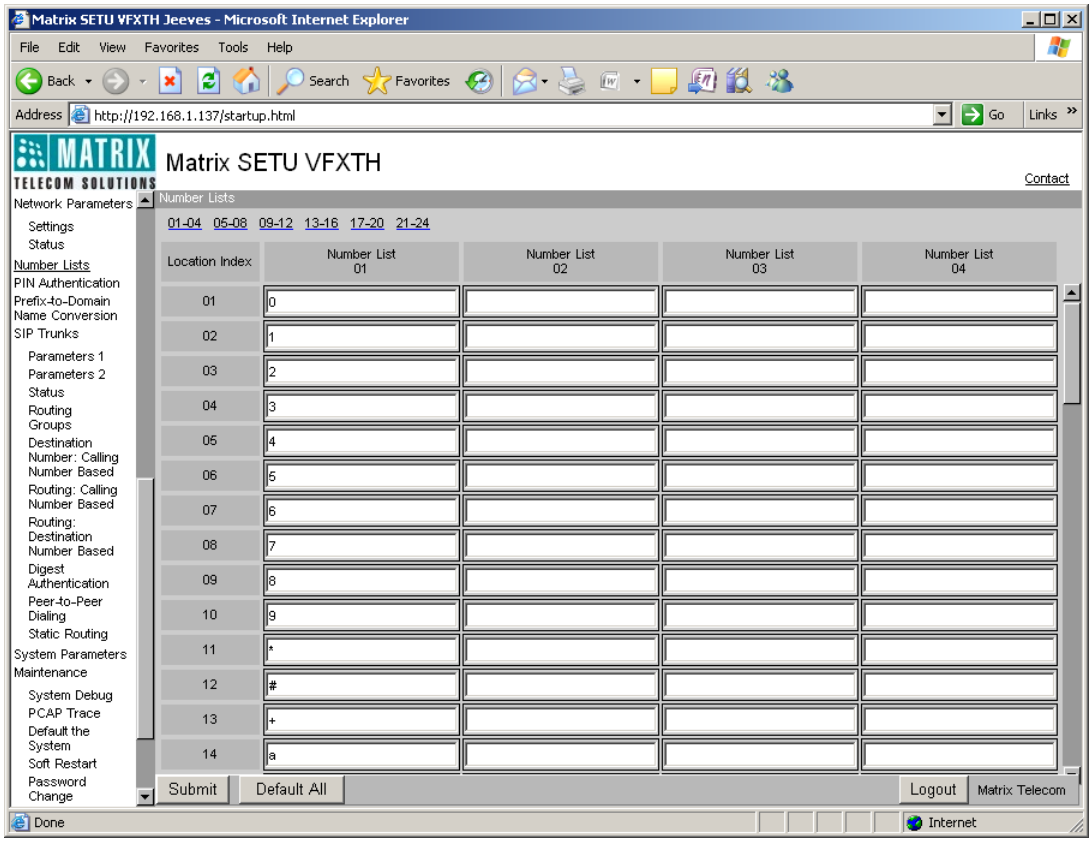
Let us understand this feature with the help of an example:

- Suppose an organization has registered itself with Pulver.com and Voiptalk.com.
- Pulver.com and Voiptalk.com suggests different prefixes for calling different domains. Pulver.com suggests its user to dial *777 to make calls to abc.com whereas Voiptalk.com suggests its users to dial *888 to make calls to abc.com.
- SE has suggested a set of prefixes to its users to make calls to other domains to help users remember only one prefix code per domain and ease the routing of calls. SE has suggested its users to dial *234 to make calls to abc.com.
- The user dials *2349874 (*234 being prefix code suggested by the SE to make call to abc.com and 9874 being the subscriber number of abc.com)
- The routing logic of SETU VFXTH is so programmed that this call shall be routed through Pulver.com. However, Pulver.com does not recognize the string *234. Thus, it shall not route the call correctly.
- Automatic Number Translation does the job.
- SE should program *234 in one of the index of one of the number list. For e.g. SE should program *234 in index 01 of Number list 01.
- SE should program *777 in index 01 of Number list 02.
- Now SE should program Number list 01 as dialed number list and Number list 02 as substitute number list.
- Doing so, when the user dials *2349874, the ANT logic shall translate this string to *7779874 and then dial it out on Pulver.com. Pulver.com shall recognize this string and route the call correctly.
- ANT feature can also be used as Memory dialing or Abbreviated dialing.
- ANT can also be used to strip off some digits before dialing. For e.g. A user dials UK number as 0044-xxxx whereas the ITSP expects 44-xxxx. ANT does this. ANT translated the number string '0044-xxxx' dialed by the user to '44-xxxx'.

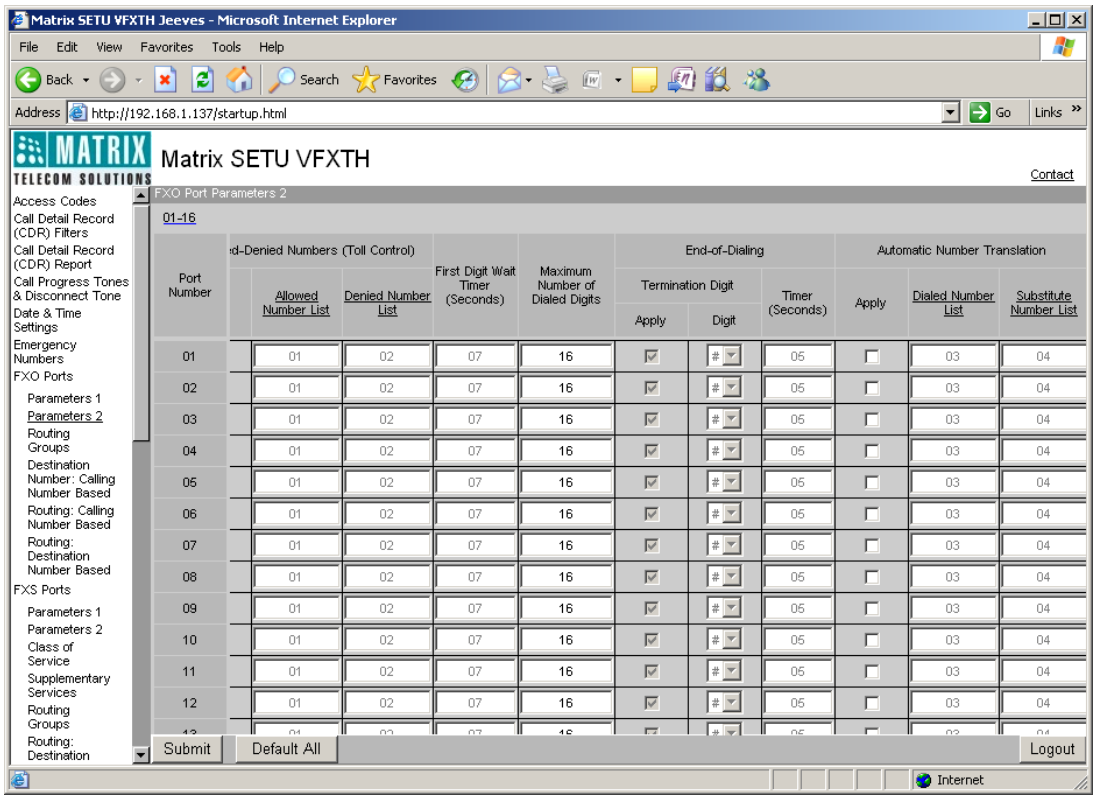
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Number Lists'** link. Program numbers expected to be dialed out and those to be substituted in place of the dialed numbers for FXO Port in different Number Lists. Similarly program dialed and substitute numbers for SIP Trunks in different Number Lists. (Refer "Number Lists" for more details)

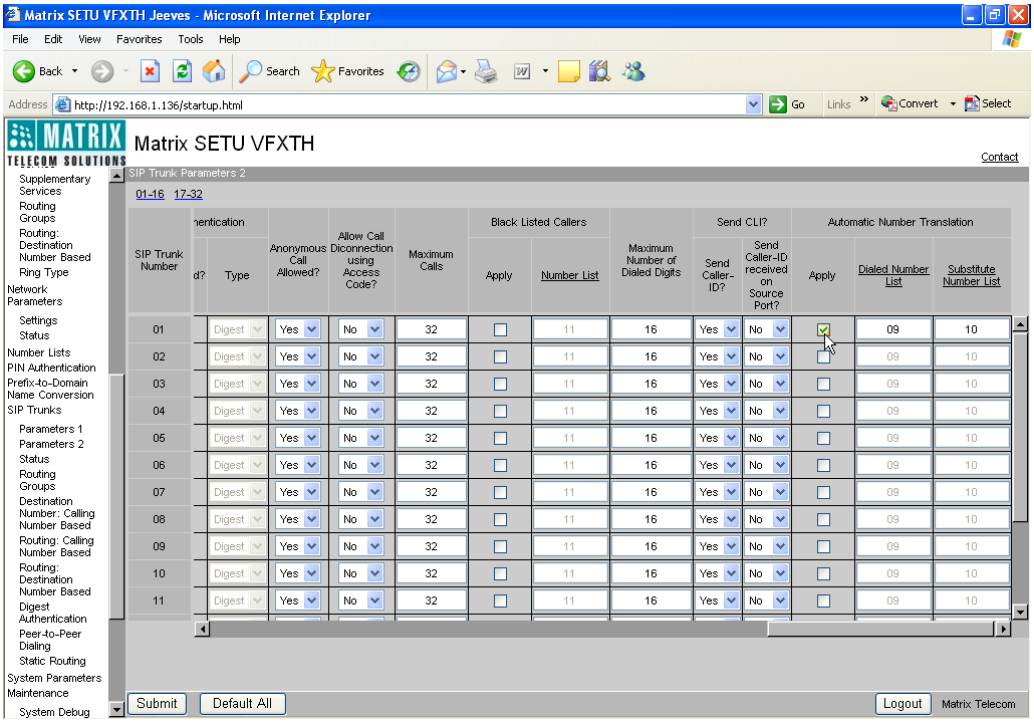


- Click on **'FXO Port Parameters 2'** link. Go to the Automatic Number Translation column.



- **Apply:** Tick this checkbox to enable this feature for FXO Port. By default, it is Untick.
- **Dialed Number List:** Enter number of the Number List in which you have programmed possible numbers that can be out dialed from FXO Port in this column. By default, list number 03 is programmed for each FXO Port.
- **Substitute Number List:** Enter number of the Number List in which you have programmed numbers to be substituted in place of the numbers that are out dialed from the FXO Port in this column. By default, list number 04 is programmed for each FXO Port.
- Click on **'Submit'** link at the bottom of the page to apply the changes.

- Click on **'SIP Trunk Parameters 2'** link. Go to the Automatic Number Translation column.



- **Apply:** Tick this checkbox to enable this feature for SIP Trunk. By default, it is Untick.
- **Dialed Number List:** Enter number of the Number List in which you have programmed possible numbers that can be out dialed from SIP Trunk in this column. By default, list number 09 is programmed for each SIP Trunk.
- **Substitute Number List:** Enter number of the Number List in which you have programmed numbers to be substituted in place of the numbers that are out dialed from the SIP Trunk in this column. By default, list number 10 is programmed for each SIP Trunk.
- Click on **'Submit'** link at the bottom of the page to apply the changes.

a. *SETU VFXTH dials out the same number string as dialed by the user, if the dialed number string does not match any of the numbers programmed in Dialed Number List.*

b. *Automatic Number Translation feature is not applicable while dialing Emergency Numbers.*

c. *Automatic Number Translation feature is not supported on the FXS Port.*

Relevant Topics:

1. “Emergency Number Dialing” 66
2. “Number Lists” 84
3. “Port Parameters-FXO” 104
4. “Port Parameters-SIP” 117

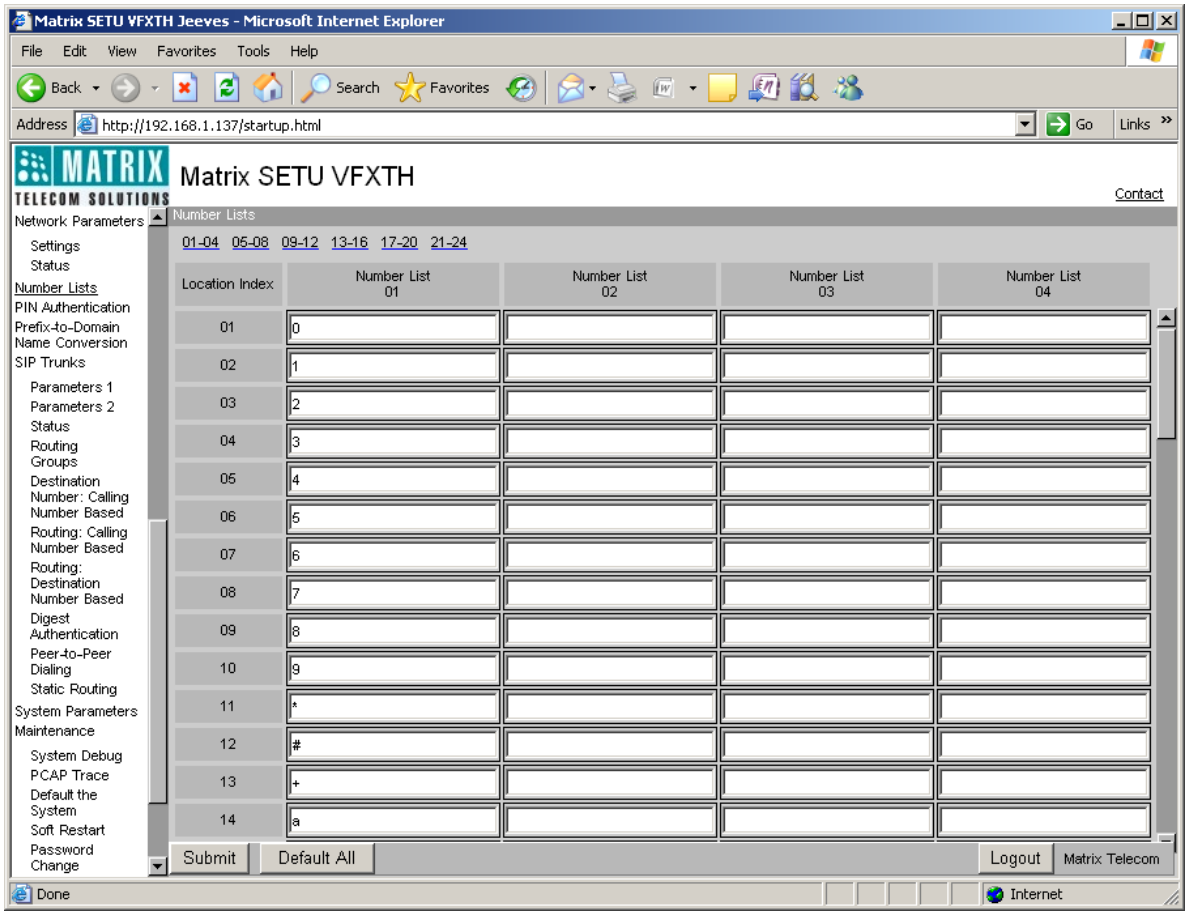
Black Listed Callers

Black Listed Callers feature allows you to block incoming calls from specific addresses/contacts, from which you do not wish to receive calls. This feature is applicable only for incoming calls on SIP Trunks.

When an incoming call lands on the SIP Trunk, SETU VFXTH match the caller's number with the entries in the list of 'Black Listed Callers'. If it matches with any of the entries on this list, the call will be automatically rejected. Thus all incoming calls from the numbers you have 'blacklisted' will be automatically rejected, shielding you from unwanted calls.

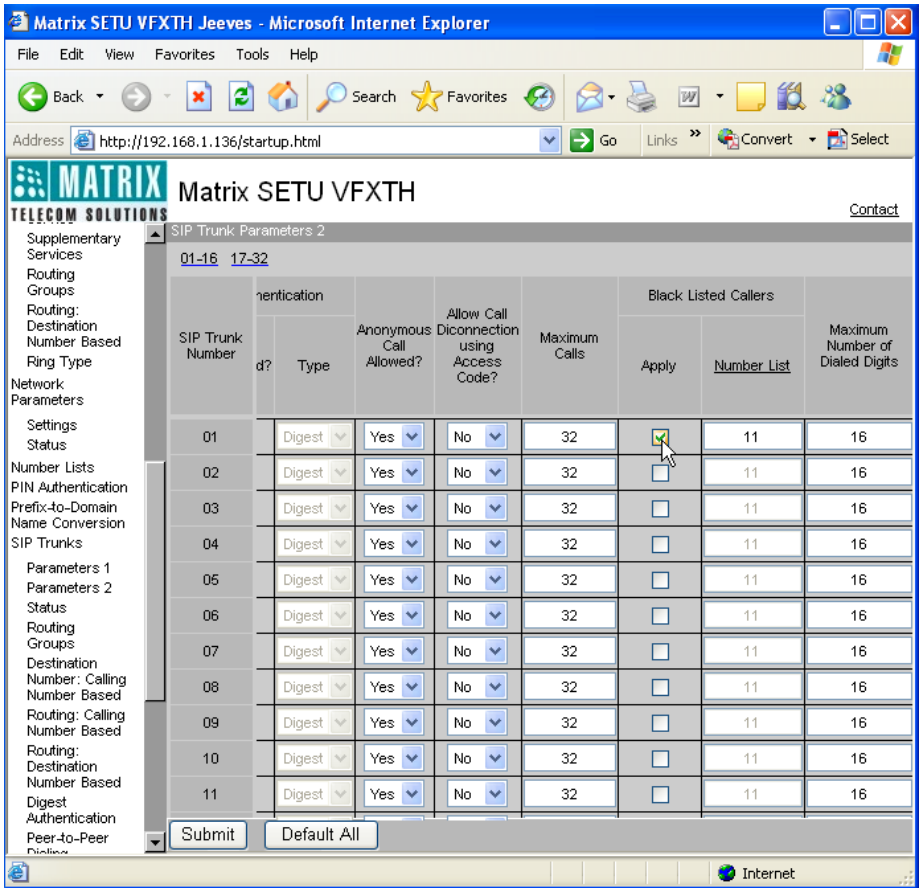
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")
- Click on **'Number Lists'** link.



In one of the Number List, program numbers/addresses of all the remote callers whose incoming calls you wish to restrict. In each number field maximum 24 characters can be entered and all ASCII characters are allowed.

- Click on **'SIP Trunk Parameters 2'** link.



- **Apply:** Tick this checkbox to enable this feature for SIP Trunks. By default, it is Untick.
- **Number List:** Enter number of the Number List in which you have programmed numbers of Black Listed Callers in this column. By default, list number 11 is programmed for each SIP Trunk.
- Click on **'Submit'** link at the bottom of the page to apply the changes.

Number of maximum 24 characters is allowed in each number list field. Therefore, if callers' number is more than 24 characters long then only first 24 characters of the number is checked. In case if first 24 characters of the callers' number perfectly matches with any of the number programmed in blacklisted callers list then that call will be rejected.

Relevant Topics:

1. “Call Processing” 47
2. “Number Lists” 84
3. “Port Parameters-SIP” 117

Call Detail Record

Call Detail Record feature is used to generate report of various call attributes such as:

- Call originating port
- Call terminating port
- Calling party number
- Called party number
- Date of call
- Time of call
- Duration of call
- PIN Number

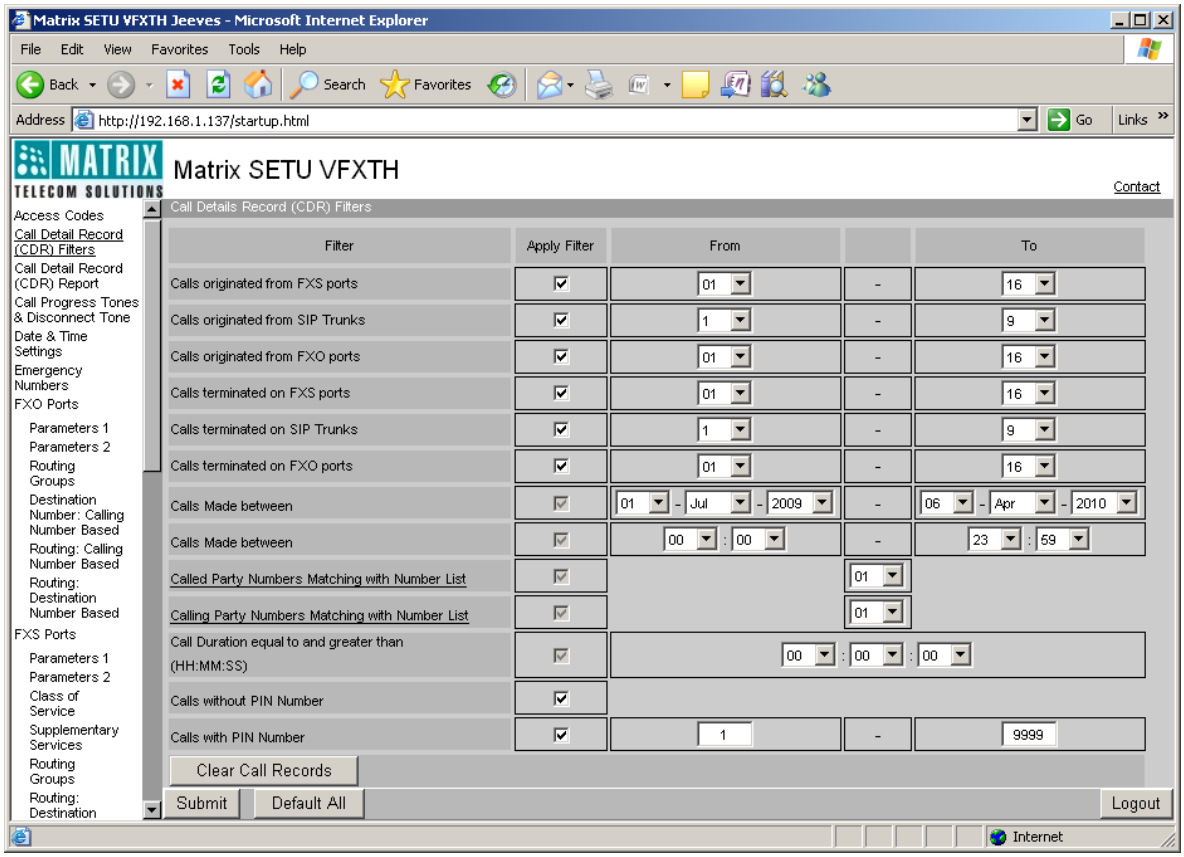
It is also possible to produce report of calls after applying filters such as calls made to specific numbers, calls received from specific numbers, calls with specific duration etc.

It is desirable for an organization to generate such reports for various reasons such as cost control, security and privacy. SETU VFXTH can store maximum 2000 call records and the entries are stored in First in First out (FIFO) method.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Call Detail Record (CDR) Filters'** link. Program following parameters:



- **Calls originated from FXS Ports:** Calls originated from FXS Ports includes numbers dialed from FXS Port and the hotline number out-dialed.
    - **Apply Filter:** Tick to enable report generation of calls originated from FXS Ports.
    - **From/To:** Select the range of FXS Ports in the corresponding 'From' and 'To' field whose report is to be generated.
- **Calls originated from SIP Trunks:** Calls originated from SIP Trunks are the calls received on the SIP Trunk of SETU VFXTH for further routing.
    - **Apply Filter:** Tick to enable report generation of calls originated from SIP Trunks.
    - **From/To:** Select the range of SIP Trunks in the corresponding 'From' and 'To' field whose report is to be generated.
- **Calls originated from FXO Ports:** Calls originated from FXO Ports are the calls (number) landing on FXO Port of SETU VFXTH for further routing.
    - **Apply Filter:** Tick to enable report generation of calls originated from FXO Ports.
    - **From/To:** Select the range of FXO Ports in the corresponding 'From' and 'To' field whose report is to be generated.
- **Calls terminated on FXS Ports:** Calls terminated on FXS Port are the calls placed on the FXS Port of SETU VFXTH.
    - **Apply Filter:** Tick to enable report generation of calls terminated on FXS Ports.
    - **From/To:** Select the range of FXS Ports in the corresponding 'From' and 'To' field whose report is to be generated.
- **Calls terminated on SIP Trunks:** Calls terminated on SIP Trunk are the calls routed using SIP Trunk of SETU VFXTH.
    - **Apply Filter:** Tick to enable report generation of calls terminated on SIP Trunks.
    - **From/To:** Select the range of SIP Trunks in the corresponding 'From' and 'To' field whose report is to be generated.
- **Calls terminated on FXO Ports:** Calls terminated on FXO Port are the calls routed using FXO Port of SETU VFXTH.
    - **Apply Filter:** Tick to enable report generation of calls terminated on FXO Ports.
    - **From/To:** Select the range of FXO Ports in the corresponding 'From' and 'To' field whose report is to be generated.
- **Calls made between- Date:** Use this filter to generate reports for specific days. Select the start date and end date in the corresponding 'From' and 'To' field to generate report of the calls made between those dates.

- **Calls made between-Time:** Use this filter to generate reports for a specific period of time. Select start time and end time in the corresponding 'From' and 'To' field to generate report of calls made between selected time period.

- **Called Party Numbers matching with Number List:** Use this filter to generate report of calls made to specific numbers. Select Number List containing remote party's numbers from the given combo box. Report would be generated of the calls made to specific numbers programmed in the selected Number List.

- **Calling Party Numbers matching with Number List:** Use this filter to generate report of calls received from specific numbers. Select Number List containing numbers of callers (who calls SETU VFXTH frequently) from the given combo box. Report would be generated of the calls received from specific numbers programmed in the selected Number List.

- **Call Duration equal to and greater than (HH: MM: SS):** Use this filter to generate report of calls having specific time duration. Select call duration in HH: MM: SS format. Report of calls having duration equal to or greater than the programmed time will be generated.

- **Calls without PIN Number:**

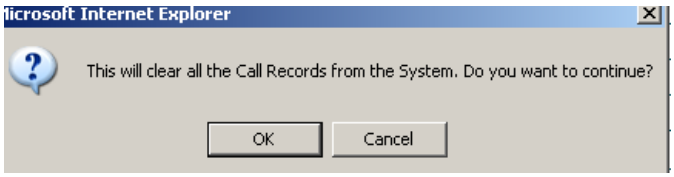
  - **Apply Filter:** Tick this combo box to generate report of calls without PIN Number.

- **Calls with PIN Number:**

  - **Apply Filter:** Tick this combo box to generate report of all calls having PIN Number.
  - **From/To:** Program range of PIN Number in the corresponding 'From' and 'To' field. PIN Numbers can be in the range of 0000 to 9999. Report of all calls having PIN Numbers within the programmed range will be generated.

- **Clear Call Records:** Call Detail Record will be cleared only if 'Clear Call Records' button is clicked.

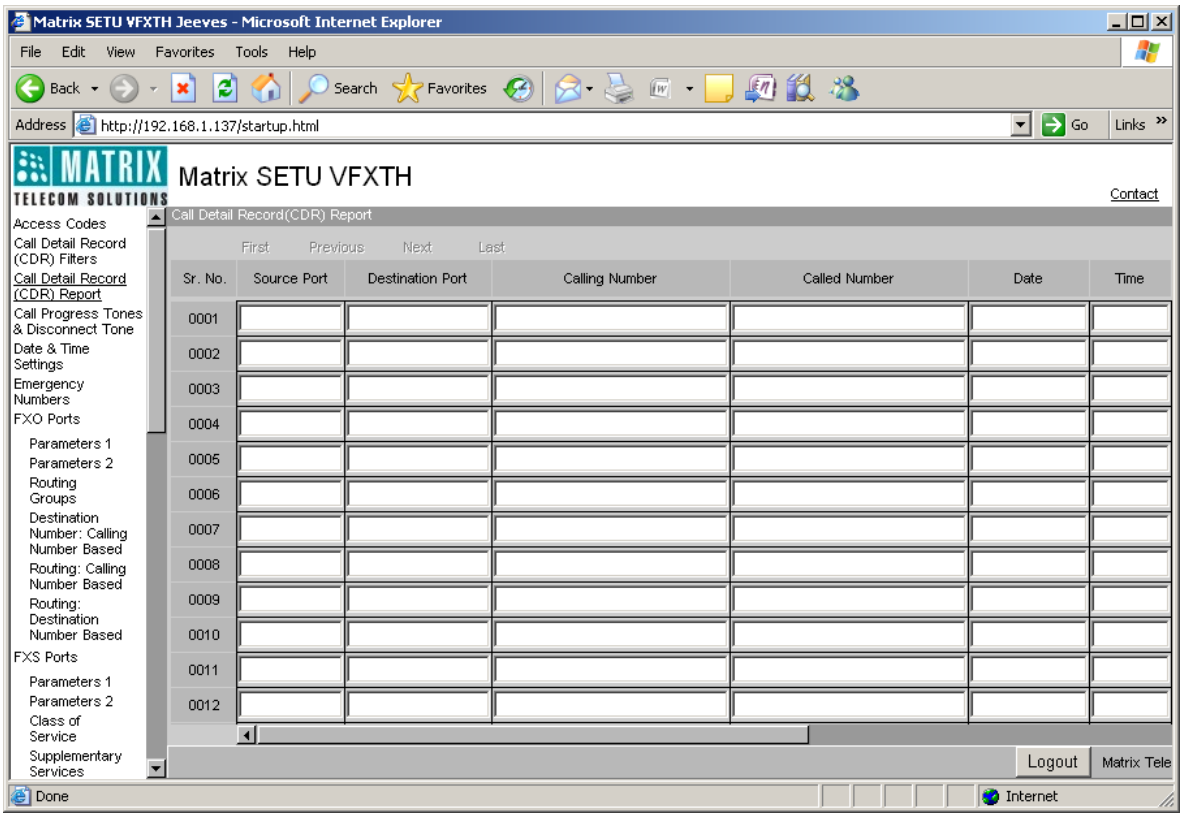
  - Click on **'Clear Call Records'** button. An alert message will appear on the screen.

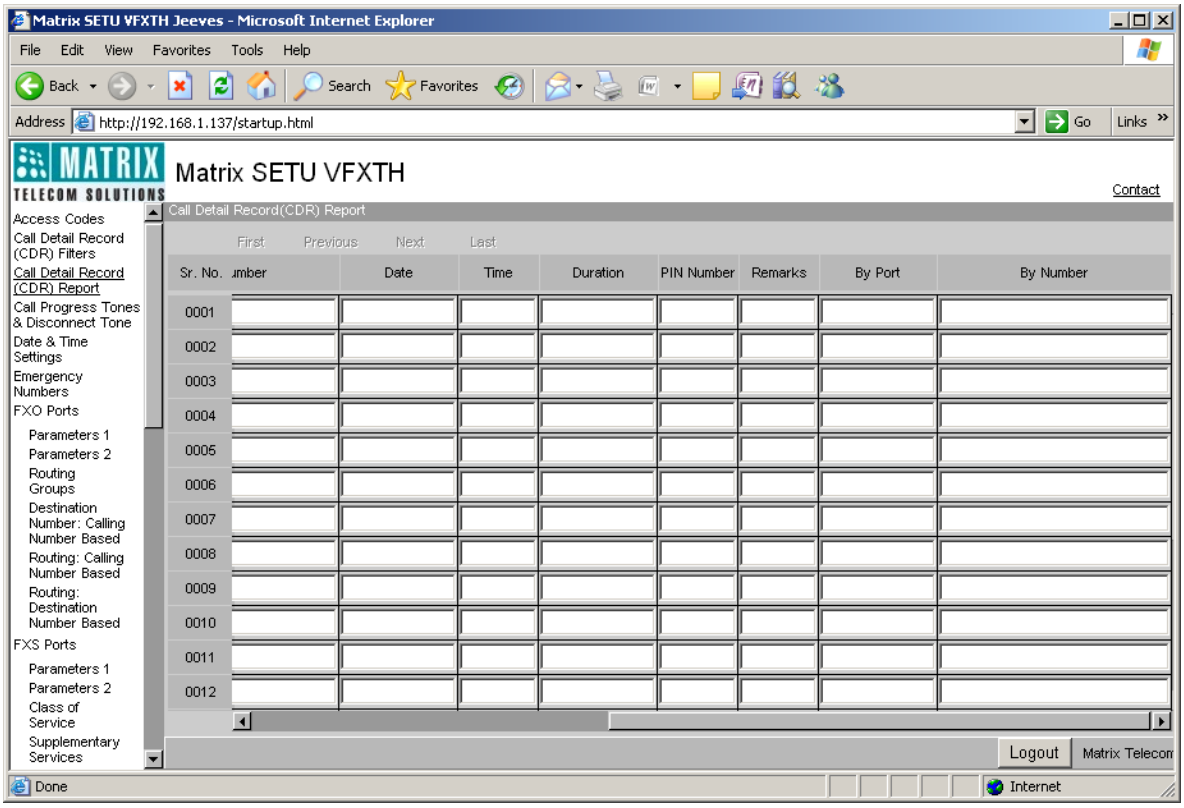


  - Click on **'OK'** to clear all call records. When call records are cleared, 'From' field of 'Calls Made Between' (Date) will change to the date of clearing of the records.

- Click on **'Submit'** link at the bottom of the page to apply the changes.

- Click on **'Call Detail Record Report'** link to check CDR report.





- Call Detail Record Report would be generated as per the filter settings.
- In one page total 20 records are displayed. To see the next 20 records, click on 'Next' button.

- Click on **'First'**, **'Previous'**, **'Next'** and **'Last'** button to check all call records.
- Alert message "No Calls to Display" will appear on the screen if there are no records to display.

a. *Only mature calls are stored in Call Detail Records.*

b. *Call records are cleared if Clear Call Records button is clicked or Software Version of the application is changed.*

c. *Defaulting SETU VFXTH will not clear Call Detail Records.*

CALL DETAIL RECORD REPORT as on 25-Feb-2010

SRN	S-PORT	D-PORT	Calling Number	Called Number	DATE	TIME	DUR	PIN	REM	By Port	By Number
1	FXS-01	SIP-1	2001	192.168.1.34	19-Feb-2010	14:42	00:00:32		N		
2	FXS-01	SIP-1	2001	3301	22-Feb-2010	16:29	00:00:46		N		
3	FXS-02	SIP-2	2002	3301	22-Feb-2010	16:30	00:00:31		N		
4	FXS-02	SIP-2	2002	3001	22-Feb-2010	16:31	00:01:33		N		
5	FXS-02	SIP-2	2002	3001	22-Feb-2010	16:37	00:00:47		N		
6	FXS-02	SIP-2	2002	3001	22-Feb-2010	16:40	00:00:41		N		
7	SIP-1	FXS-01	192.168.1.13		22-Feb-2010	17:00	00:00:07		N		
8	FXS-02	SIP-1	2002	192.168.1.13	22-Feb-2010	17:01	00:00:22		N		
9	FXS-02	SIP-1	2002	192.168.1.13	22-Feb-2010	17:02	00:02:11		N		
10	SIP-1	FXS-01	192.168.1.140		22-Feb-2010	17:05	00:00:36		N		
11	FXS-02	SIP-1	2002	192.168.1.140	22-Feb-2010	17:05	00:00:36		N		
12	FXS-02	SIP-1	2002	192.168.1.148	23-Feb-2010	11:06	00:09:39		N		
13	FXS-02	SIP-1	2002	192.168.1.148	23-Feb-2010	11:28	00:01:52		N		
14	FXS-02	SIP-1	2002	192.168.1.148	23-Feb-2010	11:39	00:00:43		N		
15	FXS-02	SIP-1	2002	192.168.1.156	23-Feb-2010	12:18	00:00:31		N		
16	FXS-02	SIP-1	2002	192.168.1.156	23-Feb-2010	12:19	00:00:25		N		
17	FXS-02	SIP-1	2002	192.168.1.156	23-Feb-2010	12:33	00:00:19		N		
18	FXS-02	SIP-1	2002	192.168.1.156	23-Feb-2010	12:47	00:00:09		N		
19	FXS-02	SIP-1	2002	192.168.1.148	23-Feb-2010	13:10	00:14:30		N		
20	FXS-01	SIP-1	2001	192.168.1.148	23-Feb-2010	13:17	00:07:01		N		
21	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:01	00:02:55		N		
22	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:05	00:00:25		N		
23	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:06	00:03:05		N		
24	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:26	00:00:25		N		
25	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:27	00:00:13		N		
26	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:28	00:00:49		N		
27	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:29	00:01:18		N		
28	FXS-01	SIP-1	2001	3005	24-Feb-2010	12:32	00:00:46		N		
29	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:34	00:01:20		N		
30	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:37	00:00:34		N		
31	FXS-02	SIP-2	2002	3001	24-Feb-2010	12:39	00:00:26		N		
32	SIP-2	FXS-02	3001@192.168.1.118		24-Feb-2010	12:40	00:00:15		N		
33	SIP-1	FXS-01	3305@192.168.1.118		24-Feb-2010	12:40	00:00:45		N		
34	FXS-01	SIP-1	2001	3005	24-Feb-2010	12:41	00:01:45		N		
35	SIP-1	FXS-01	3005@192.168.1.118		24-Feb-2010	12:43	00:02:08		N		
36	FXS-01	SIP-1	2001	3301	24-Feb-2010	12:45	00:00:37		N		
37	SIP-1	FXS-01	3302@192.168.1.118		24-Feb-2010	12:48	00:00:46		N		
38	FXS-01	SIP-1	2001	3302	24-Feb-2010	12:49	00:00:41		N		
39	SIP-1	FXS-01	2001@192.168.1.118		24-Feb-2010	12:50	00:00:18		N		
40	SIP-1	FXS-01	2005@192.168.1.118		24-Feb-2010	12:52	00:01:00		N		
41	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:53	00:00:19		N		
42	FXS-01	SIP-1	2001	3001	24-Feb-2010	12:54	00:00:42		N		
43	FXS-01	SIP-1	2001	3001	24-Feb-2010	15:13	00:01:08		N		
44	SIP-1	FXS-01	2001@192.168.1.118		24-Feb-2010	15:15	00:00:45		N		
45	SIP-1	FXS-01	3001@192.168.1.118		24-Feb-2010	15:17	00:00:41		N		
46	SIP-2	FXS-02	3001@192.168.1.118		24-Feb-2010	15:18	00:00:27		N		
47	FXS-02	SIP-2	2002	3304	24-Feb-2010	15:18	00:00:16		N		
48	SIP-1	FXS-01	3305@192.168.1.118		24-Feb-2010	15:18	00:00:17		N		
49	FXS-01	SIP-1	2001	3001	24-Feb-2010	15:19	00:00:20		N		
50	FXS-02	SIP-2	2002	3304	24-Feb-2010	15:19	00:00:20		N		
51	SIP-1	FXS-01	3305@192.168.1.118		24-Feb-2010	15:19	00:00:20		N		
52	FXS-01	SIP-1	2001	3001	24-Feb-2010	15:19	00:00:36		N		
53	FXS-01	SIP-1	2001	3302	24-Feb-2010	15:20	00:00:42		N		
54	SIP-1	FXS-01	2001@192.168.1.118		24-Feb-2010	15:21	00:01:16		N		
55	FXS-01	SIP-1	2001	3001	24-Feb-2010	15:23	00:00:05		N		
56	FXS-01	SIP-1	2001	3302	24-Feb-2010	15:23	00:00:13		N		
57	SIP-3	FXS-01	2011@192.168.1.189		24-Feb-2010	18:13	00:02:03		N		
58	SIP-3	FXS-01	2011@192.168.1.189		24-Feb-2010	18:15	00:03:23		N		
59	FXS-01	SIP-1	2001	3001	24-Feb-2010	18:26	00:00:16		N		
60	FXS-01	SIP-1	2001	3301	24-Feb-2010	18:27	00:00:20		N		

Matrix SETU VFXTH V1R1

Relevant Topics:

1. “Number Lists” 84
2. “PIN Authentication” 101
3. “Port Parameters-FXO” 104
4. “Port Parameters-FXS” 112
5. “Port Parameters-SIP” 117

Call Disconnect Tone

The POTS network plays call disconnection tone to signal far-end (remote party) disconnection. The call disconnection tone is played in both the following cases:

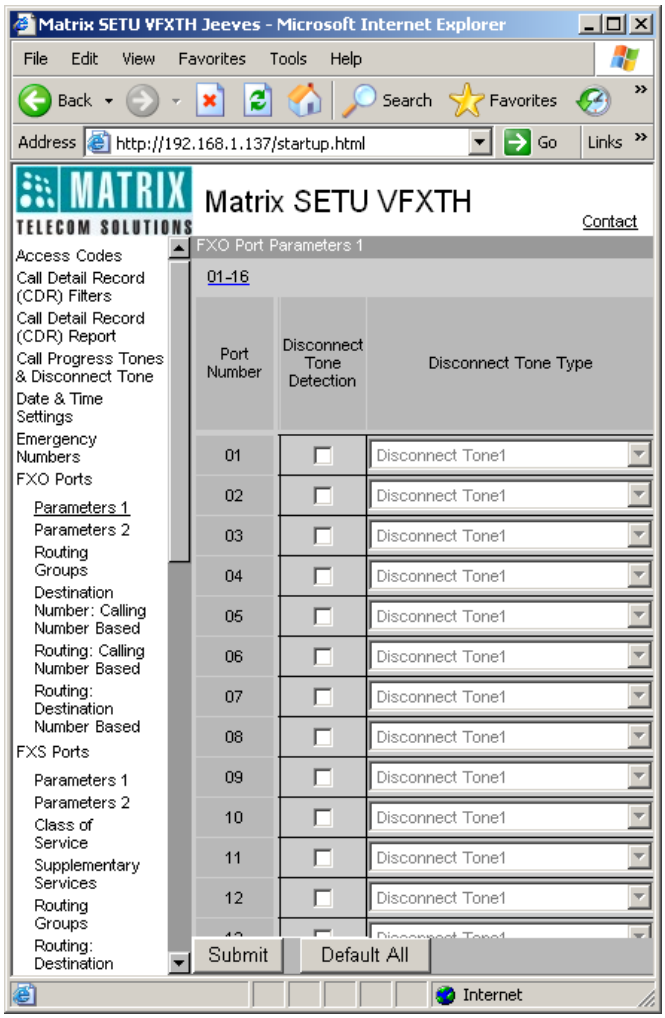
- When an outgoing call made by the user is disconnected by the far-end.
- When an incoming call received by the user is disconnected by the far-end.

Generally when two human users are talking, either one of them goes on-hook on listening to the call disconnection tone but in gateway type of products, it is required that the system itself detects the call disconnection tone and release the resources involved in the call.

SETU VFXTH supports detection of Call Disconnect Tones on its FXO Ports. You can also program frequency and cadence of each 'Disconnect Tone' to match it with the Call Disconnect Tone supported by your POTS network.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on '**FXO Port Parameters 1**' and program the following parameters:



- **Disconnect Tone Detection:** Tick 'Disconnect Tone Detection' flag to enable detection of Call Disconnect Tone sent by PSTN network, on the FXO Port.

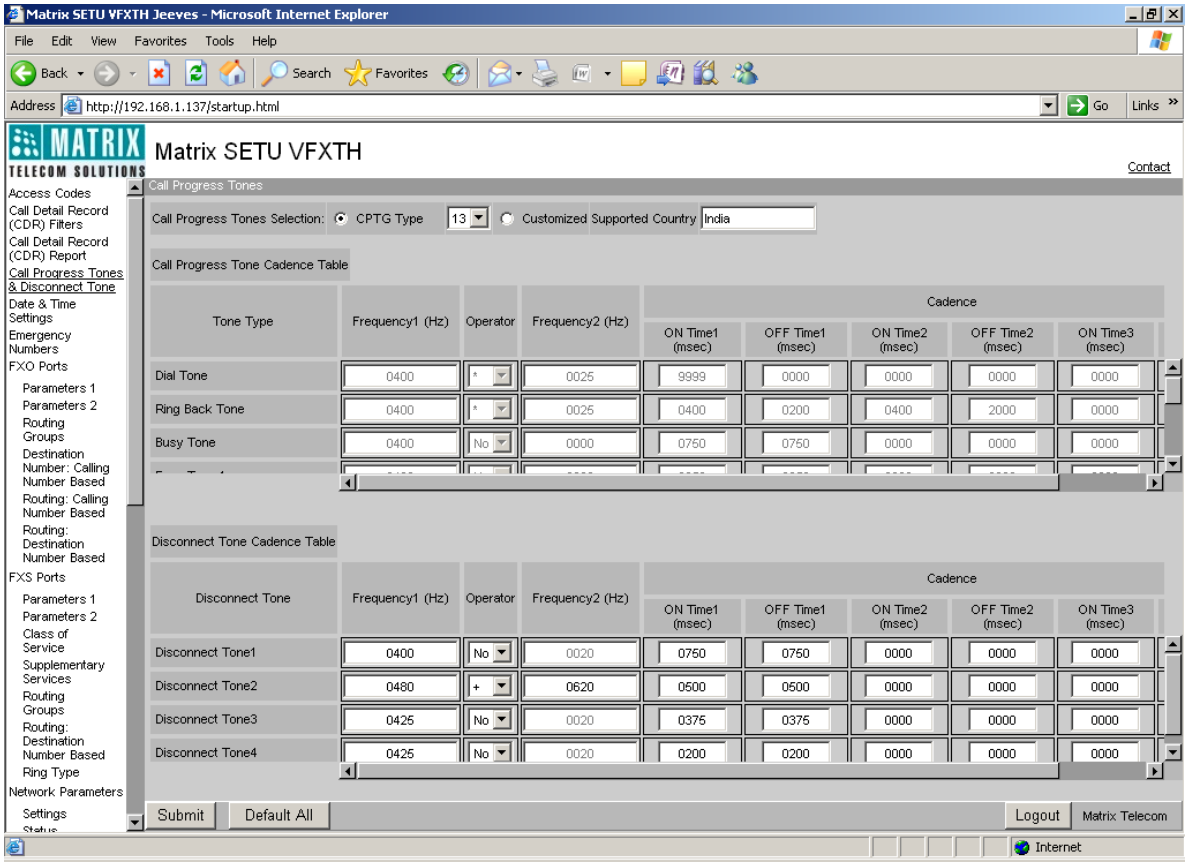
- **Disconnect Tone to be detected:** Select the type of tone to be considered as Call Disconnect Tone on each FXO Port. You can select one of the following tones:

  - 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

  By default, 'Disconnect Tone 1' is selected for all FXO Ports.

- Click on '**Submit**' link at the bottom of the page to save the changes.

- Click on '**Call Progress Tones**' link and program following parameters in Disconnect Tone Cadence Table for each Disconnect Tone:



- **Frequency1 (Hz):** Program frequency1 for all tones in this field. Range of frequency1 is 300-1400 Hz for all tones.

- **Frequency2 (Hz):** Program frequency2 for all tones in this field. Range of frequency2 is 20-1400 Hz for all tones.

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

  i. ***No:*** If 'No' is programmed then Frequency2 is not applicable.
  ii. **** (Modulation):*** If '*' (Modulation) is programmed then Frequency1 and Frequency2 are used as modulation i.e. F1* F2.
  iii. ***+ (Addition):*** If Addition (+) is programmed then Frequency1 and Frequency2 are used as addition i.e. F1 + F2.

- **Cadence:** Program Cadence ON Time1-OFF Time1, ON Time2-OFF Time2 and ON Time3-OFF Time-3 for each Disconnect Tone. Valid ON Time and OFF Time range is 0000-9999 msec.

- Click on '**Submit**' link at the bottom of the page to save the changes.

a. *When Operator is set to 'No', Frequency 2 is not applicable.*

b. *Disconnect tone's frequency and cadence value should be programmed similar to that of PSTN exchange from which trunk line service is subscribed.*

Relevant Topics:

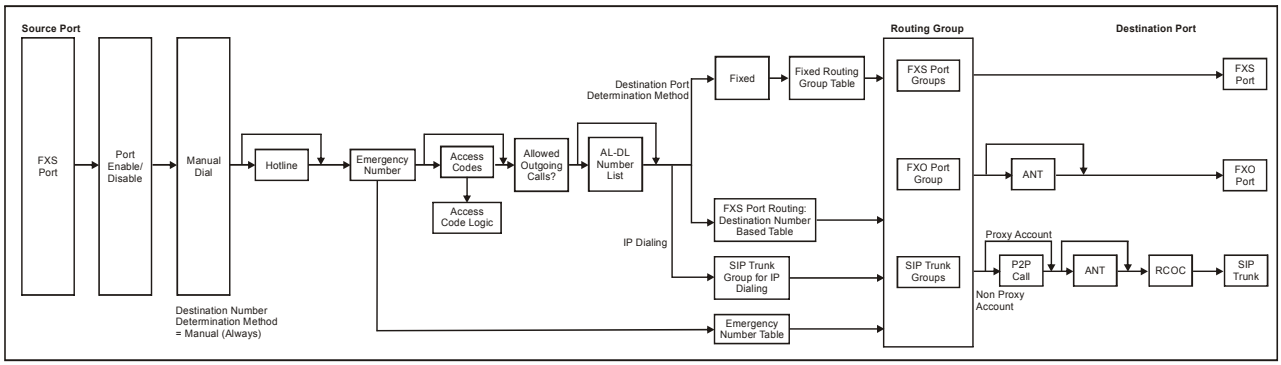
1. “Call Processing” 47
2. “Port Parameters-FXO” 104

Call Processing

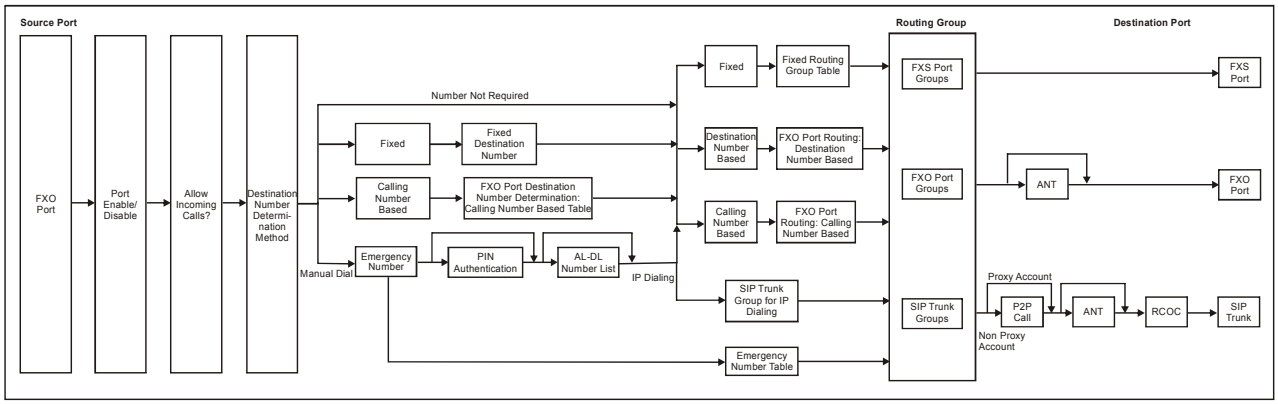
SETU VFXTH supports multiple port types' viz. FXO, FXS and SIP. When the call originates on the source port, it has to be routed to the destination port through some process. This process of routing the call from source port to the destination port is called Call Processing.

SETU VFXTH processes the call originated on different ports as shown below:

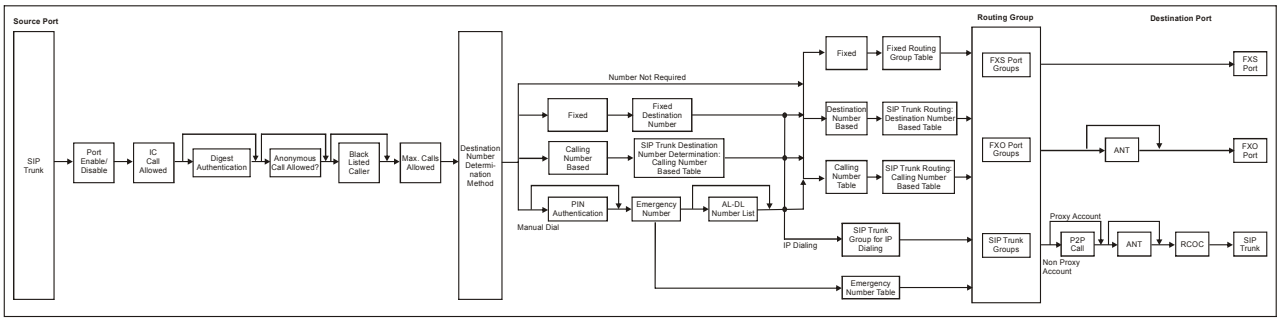
Call originated from FXS Port



Call originated on FXO Port



Call originated on SIP Trunk



Call Progress Tones

Call Progress Tones are audible tones sent by switching systems such as PSTN or PBX, to calling parties to show the status of the phone call. Different tones have distinctive frequency and cadence assigned to it, for which some standards have been established. On the basis of specific frequency, modulating frequency and cadence, CPTs 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
- Remote Hold Tone

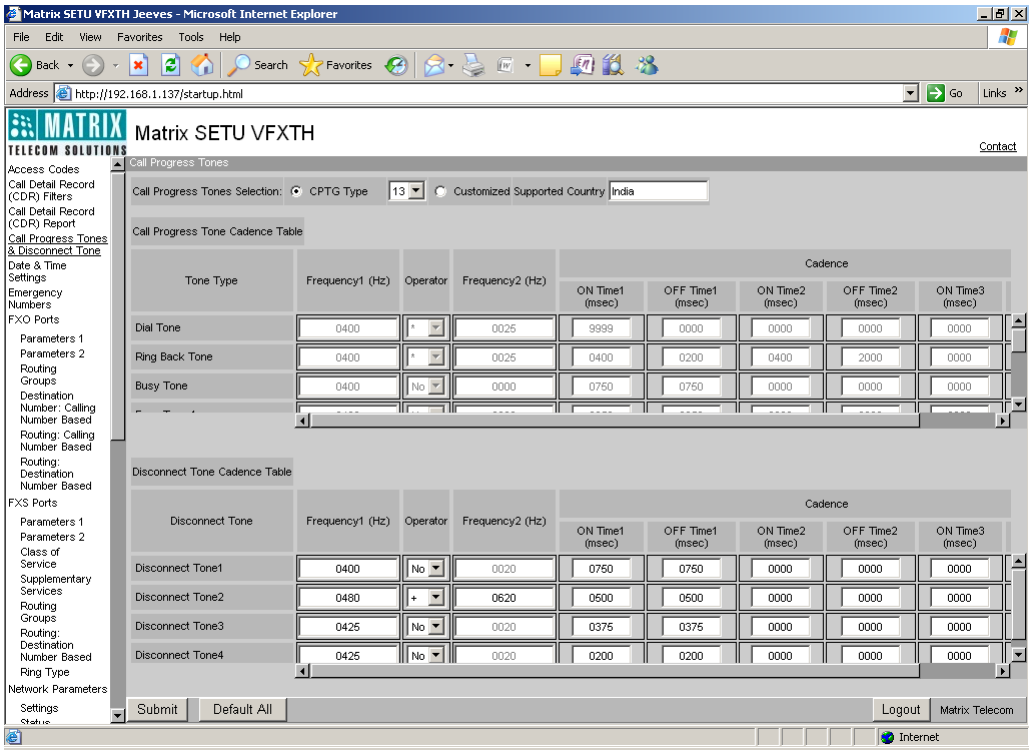
CPT standards for call progress tones are applied differently in different situations and in different countries. For example, as per ITU standard, the Dial Tone for India consists of 400 Hz modulated by 25 Hz, whereas it is 350+440 Hz, without modulation, for USA/ Canada.

You can match call progress tones of SETU VFXTH to that of the country standard in which it is installed.

SETU VFXTH also supports detection of Call Disconnect Tone on its FXO Port and you can match the Disconnect Tone of the system with that of the PSTN exchange from which you have subscribed for the trunk line services.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)



- Click on '**Call Progress Tones**' link. Select either CPTG Type or Customized option. By default, CPTG Type option is selected.

- If 'CPTG Type' is selected then the combo box for selecting the country becomes editable. Select the Code of the Country in which SETU VFXTH is installed (Refer CPTG table given at the end of this topic for codes of different countries).

  When you select this option, Call Progress Tone Cadence Table will become uneditable. By default, 13 (Code of INDIA) is selected. In 'Supported Country' field, the name of the country whose code is selected in this field is displayed.

- If 'Customized' is selected then countrywise combo box becomes uneditable and Call Progress Tone Cadence Table will become editable. Program frequency and cadence for each tone. By default, values for all tones are displayed as per INDIA.

  - **Frequency1 (Hz):** Program frequency1 for all tones in this field. Range of frequency1 is 300-1400 Hz for all tones.

  - **Frequency2 (Hz):** Program frequency2 for all tones in this field. Range of frequency2 is 20-1400 Hz for all tones.

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

    i. ***No:*** If 'No' is programmed then Frequency2 is not applicable.
    ii. **** (Modulation):*** If '*' (Modulation) is programmed then Frequency1 and Frequency2 are used as modulation i.e. F1* F2.
    iii. ***+ (Addition):*** If Addition (+) is programmed then Frequency1 and Frequency2 are used as addition i.e. 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.

- Also program Frequency 1, Operator, Frequency 2 and Cadence for each Disconnect Tone in Disconnect Tone Cadence Table. You need to program these parameters only when 'Disconnect Tone Detection' is enabled on FXO Port and the 'Disconnect Tone' is selected in 'Disconnect Tone to be detected' field. (Refer "Call Disconnect Tone" topic for details)

- Click on '**Submit**' link at the bottom of the page to save the changes.

a. *When you default SETU VFXTH, Call Progress Tones will not be defaulted. Even country will not be defaulted.*

b. *Remote Hold Tone is fixed for all the countries i.e. non-programmable.*

Default values for all Call Progress Tones for all the countries as per ETSI standard is programmed in the system.

Default Call Progress Tones Table is shown below:

CPTG Type	Region	Dial tone		Ring Back Tone		Busy Tone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
01	UCP/VFX TH Type1	440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	440	0.75on 0.75off

CPTG Type	Region	Dial tone		Ring Back Tone		Busy Tone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
02	UCP/VFX TH Type2	400	Continuous	400	0.6on 0.2off 0.2on 2.0off	400	0.5on 0.5off
03	UCP/VFX TH Type3	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
04	Argentina	425	Continuous	425	1.0on 4.0 off	425	0.3on 0.2off
05	Australia	425*25	Continuous	400*25	.4on .2off .4on 2.0off	425	0.375on 0. 375off
06	Brazil	425	Continuous	425	1.0on 4.0 off	425	0.25on 0.25off
07	Canada	350+440	Continuous	440+480	2.0on 4.0off	480+620	0.5on 0.5off
08	China	450	Continuous	450	1.0on 4.0off	450	0.35 on 0.36off
09	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

CPTG Type	Region	Dial tone		Ring Back Tone		Busy Tone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
27	Portugal	425	Continuous	425	1.0on 5.0off	425	0.5on 0.5off
28	Russia	425	Continuous	425	0.8on 3.2off	425	0.4on 0.4off
29	Saudi Arabia	425	Continuous	425	1.2on 4.6off	425	0.5on 0.5off
30	Singapore	425	Continuous	425*24	0.4on 0.2off 0.4on 2.0off	425	.75on .75off
31	South Africa	400*33	Continuous	400*33	0.4on 0.2off 0.4on 2.0off	400	.5on .5off
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	UCP/VFX TH 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	UCP/VFX TH Type5	350+440	Continuous	350+440	0.4on 0.2off 0.4on 2.0off	400	0.75on 0.75off

CPTG Type	Region	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
01	UCP/VFX TH Type1	440	0.25on 0.25 off	440	1on 1off	350+440	0.1on 0.1off
02	UCP/VFX TH Type2	400	0.25on 0.25 off	400	1on 1off	400	0.1on 0.1off
03	UCP/VFX TH Type3	440	0.25on 0.25 off	440	1on 1off	350+440	0.1on 0.1off
04	Argentina	425	0.3on 0.4off	425	1on 1off	425	0.1on 0.1off
05	Australia	425	0.375on 0.375off	425	1on 1off	425*25	0.1on 0.1off
06	Brazil	425	0.25on 0.25 off	425	1on 1off	425	0.1on 0.1off
07	Canada	480+620	0.25on 0.25off	480+620	1on 1off	350+440	0.1on 0.1off
08	China	450	0.7on 0.7off	450	1on 1off	450	0.1on 0.1off
09	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	425	0.24on 0.24off	425	1on 1off	425	0.1on 0.1off

CPTG Type	Region	Error Tone 1		Error Tone 2		Confirmation Tone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
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
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	UCP/VFX TH 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	UCP/VFX TH Type5	400	0.25on 0.25 off	400	1on 1off	350+440	0.1on 0.1off

CPTG Type	Region	Feature/Programming/ Prompt Tone		Routing Tone		IntrusionTone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
01	UCP/VFX TH Type1	350+440	0.1on 0.9off	350+440	0.1on 1.9off	440	0.1on 2.9off
02	UCP/VFX TH Type2	400	1.5on 0.1off	400	0.1on 1.9off	400	0.2on 4.8off
03	UCP/VFX TH Type3	350+440	0.1on 0.9off	350+440	0.1on 1.9off	440	0.1on 2.9off
04	Argentina	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
05	Australia	425*25	0.1on 0.9off	425*25	0.1on 1.9off	425	Continuous
06	Brazil	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
07	Canada	350+440	0.1on 0.9off	350+440	0.1on 1.9off	480+620	0.5on 0.5off
08	China	450	0.1on 0.9off	450	0.1on 1.9off	450	0.2on 0.2off 0.2on 0.6off
09	Egypt	425*50	0.1on 0.9off	425*50	0.1on 1.9off	450	0.5on 0.5off
10	France	440	0.1on 0.9off	440	0.1on 1.9off	440	0.1on 2.9off
11	Germany	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
12	Greece	425	0.1on 0.9off	425	0.1on 1.9off	425	0.15on 0.25off 0.15on 1.45off
13	India	400*25	0.1on 0.9off	400*25	0.1on 1.9off	400	0.15on 4.85off
14	Indonesia	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
15	Iran	425	0.1on 0.9off	425	0.1on 1.9off	425	0.1on 2.9off
16	Iraq	400	0.1on 0.9off	400	0.1on 1.9off	400	0.1on 2.9off
17	Israel	400	0.1on 0.9off	400	0.1on 1.9off	400	0.1on 2.9off
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

CPTG Type	Region	Feature/Programming/ Prompt Tone		Routing Tone		IntrusionTone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
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	UCP/VFX TH 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	UCP/VFX TH Type5	350+440	0.1on 0.9off	350+440	0.1on 1.9off	350+440	0.5on 0.5off 1.0on 5.0off

Class of Service

SETU VFXTH supports variety of features. In an organization, requirement of all the users vary with respect to their position and type of work. All features are not required by all FXS Port users. SETU VFXTH gives you the flexibility to allow or deny a feature to the FXS Port user using Class of Service feature. Class of Service defines accessibility rights of the user relating to various features.

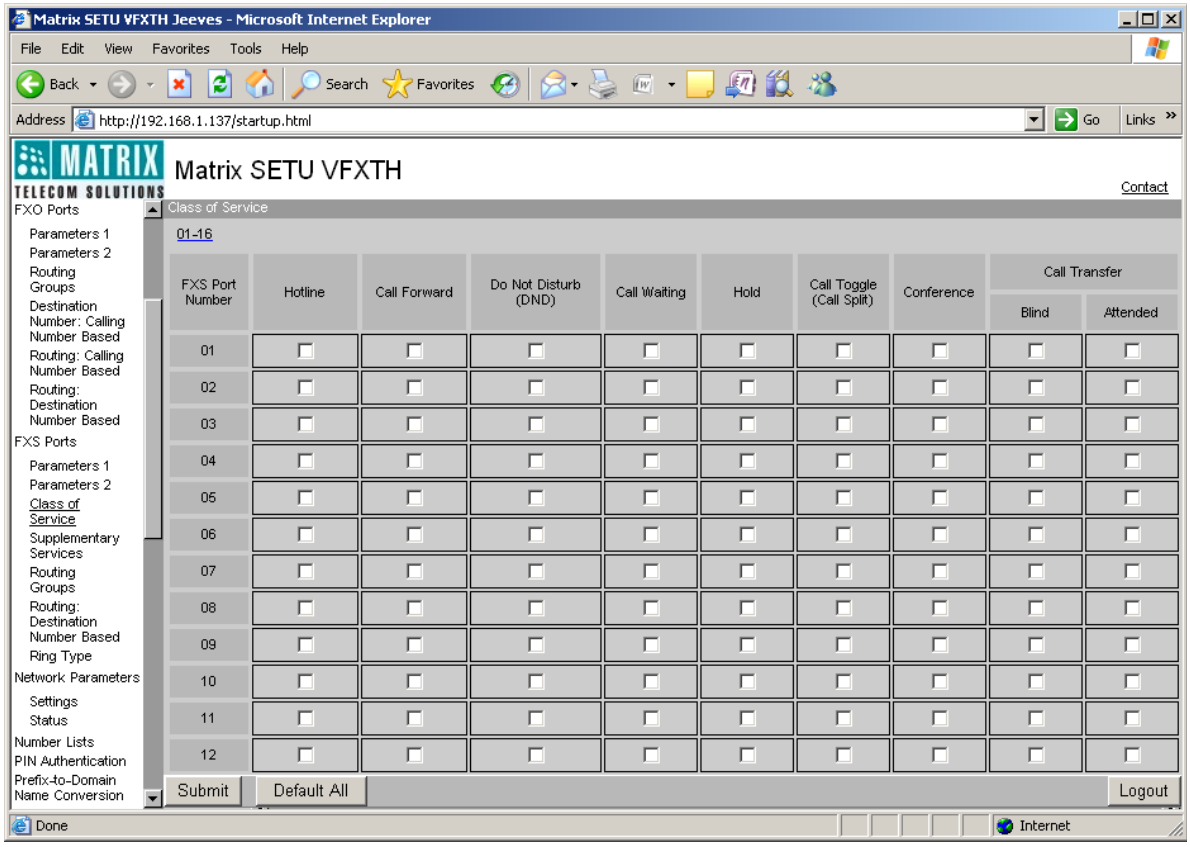
Following features are included in Class of Service:

- Hotline
- Call Forward
- Do Not Disturb (DND)
- Call Waiting
- Hold
- Call Toggle
- Conference
- Blind Transfer
- Attended Transfer

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Class of Service'** link. Enable required features for each FXS Port. By default, all the features are disabled for all FXS Ports.



- Click on **'Submit'** link at the bottom of the page to save the changes.

a. *When in Class of Service, if Hotline, Call Forward, Call Waiting or DND feature is disabled for any FXS Port then the same will be disabled automatically in Supplementary Services too.*

b. *When user tries to set or cancel a feature using access code, he will get error tone if the feature is disabled in the Class of Service.*

Relevant Topics:

1. “Access Codes” 25
2. “Attended Call Transfer” 182
3. “Blind Call Transfer” 184
4. “Call Forward” 187
5. “Call Hold” 190
6. “Call Toggle (Call Split)” 192
7. “Call Waiting” 194
8. “Conference” 198
9. “Do Not Disturb (DND)” 201
10. “Hotline” 203

Date and Time Settings

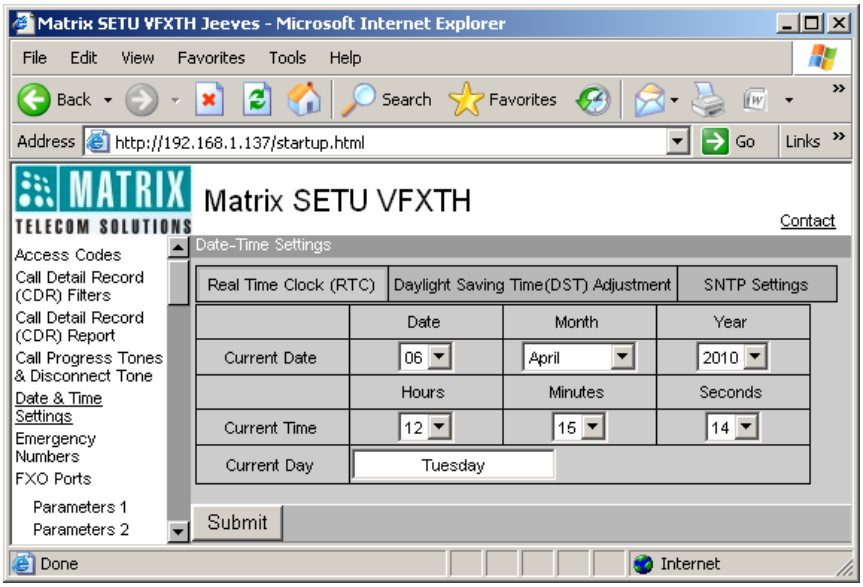
Various features like Call Detail Records, Daylight Savings Time etc. use date and time parameters for its proper functioning. Thus, programming 'Date and Time' parameters, is one of the most important step in programming SETU VFXTH.

SETU VFXTH has its own Real Time Clock (RTC) to store date and time. It uses Simple Network Time Protocol (SNTP) to get Greenwich Mean Time from the Time Server. Depending on the country of installation, SETU VFXTH adjusts its clock to match the time of the time server.

SETU VFXTH also provides an option to sync system's date and time with the SNTP server automatically at power on.

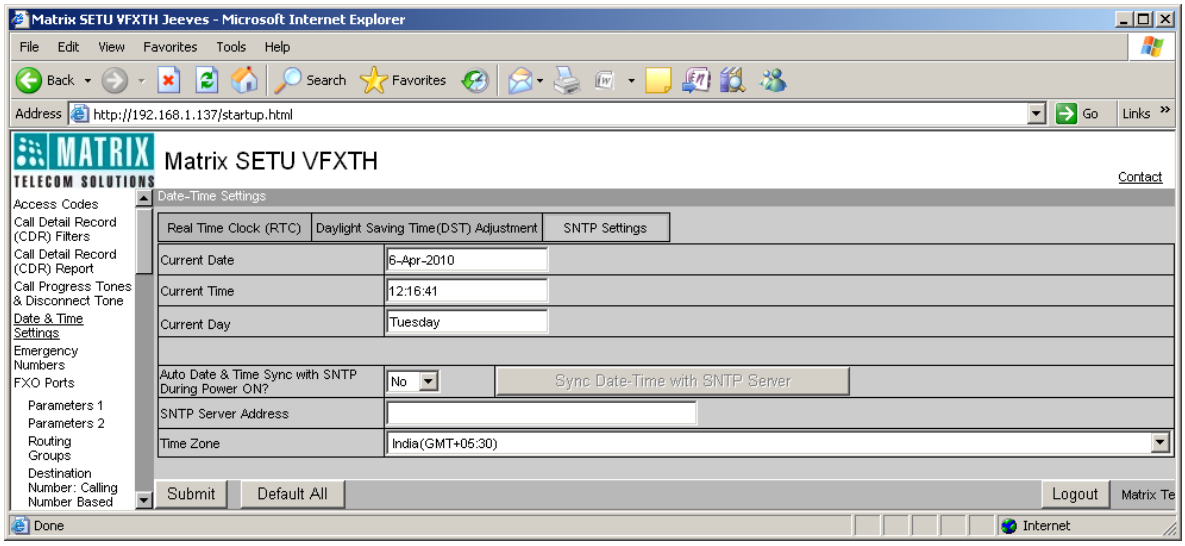
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on **'Date and Time Settings'** link.
- Click on ‘**Real time Clock (RTC)**’ link and program the following parameters.



- **Current Date:** Enter current Date in DD-MM-YYYY format. Valid range for Date is 01 to 31, Month is January to December and Year is 2009 to 2099.
- **Current Time:** Enter current Time in HH-MM-SS format. Valid range for Hour is 00 to 23, Minutes is 00 to 59 and Seconds is 00 to 59.
- **Current Day:** Depending on the current Date entered, SETU VFXTH automatically sets current Day in this field.
- Click on **'Submit'** link at the bottom of the page to save the changes.

- Click on **'SNTP Settings'** link and program the following parameters.



- **Auto Date and Time Sync with SNTP during Power ON?:** Select 'Yes' in this field if you want to sync date and time with SNTP server automatically at Power On. By default, it is set to 'No'.

If 'Auto Date & Time Sync with SNTP during power ON' option is set to 'Yes', server address field cannot be left blank. If done so, error message 'Error! SNTP server address can not be left blank' will appear on the screen.

- **SNTP Server Address:** Program SNTP server address in this field, if you have selected to sync date and time with SNTP server automatically at power on. SETU VFXTH will sync its date and time with the server whose address is programmed in this field. SNTP Server address can be of maximum 40 characters. All ASCII characters are allowed. By default, it is blank.

- **Time Zone:** Select the country/region in which SETU VFXTH is installed. By default, 'India (GMT+05:30)' is selected.

- **Sync SNTP with Date and Time Server:** Click this button to sync date and time of SETU VFXTH with SNTP server manually whenever required.

- Click on **'Submit'** link at the bottom of the page to save the changes.

Relevant Topic:

1. "Call Detail Record" 39

Daylight Saving Time (DST) Mode

Real Time Clock of SETU VFXTH moves forward or backward automatically in tune with the Day Light Savings requirement of the country in which it is installed. Typically clocks are adjusted forward by one hour near the start of spring and are adjusted backward in autumn. The start and end dates of DST vary with location and year.

SETU VFXTH provides option to enable or disable DST as per the requirement of the user. In most countries in Asia and Africa, and in certain countries of South America, DST is not observed. Thus, enable DST if SETU VFXTH is installed in the country where DST is observed. When DST is observed in a specific country, the time difference between countries in different time zone will also vary.

SETU VFXTH enables you to program DST manually or automatically:

- If Scheduled DST is enabled then DST parameters will be updated automatically as per the country selected.
- If Manual DST is enabled then program DST parameters manually as per the requirement of the user. There are two ways to adjust DST manually:

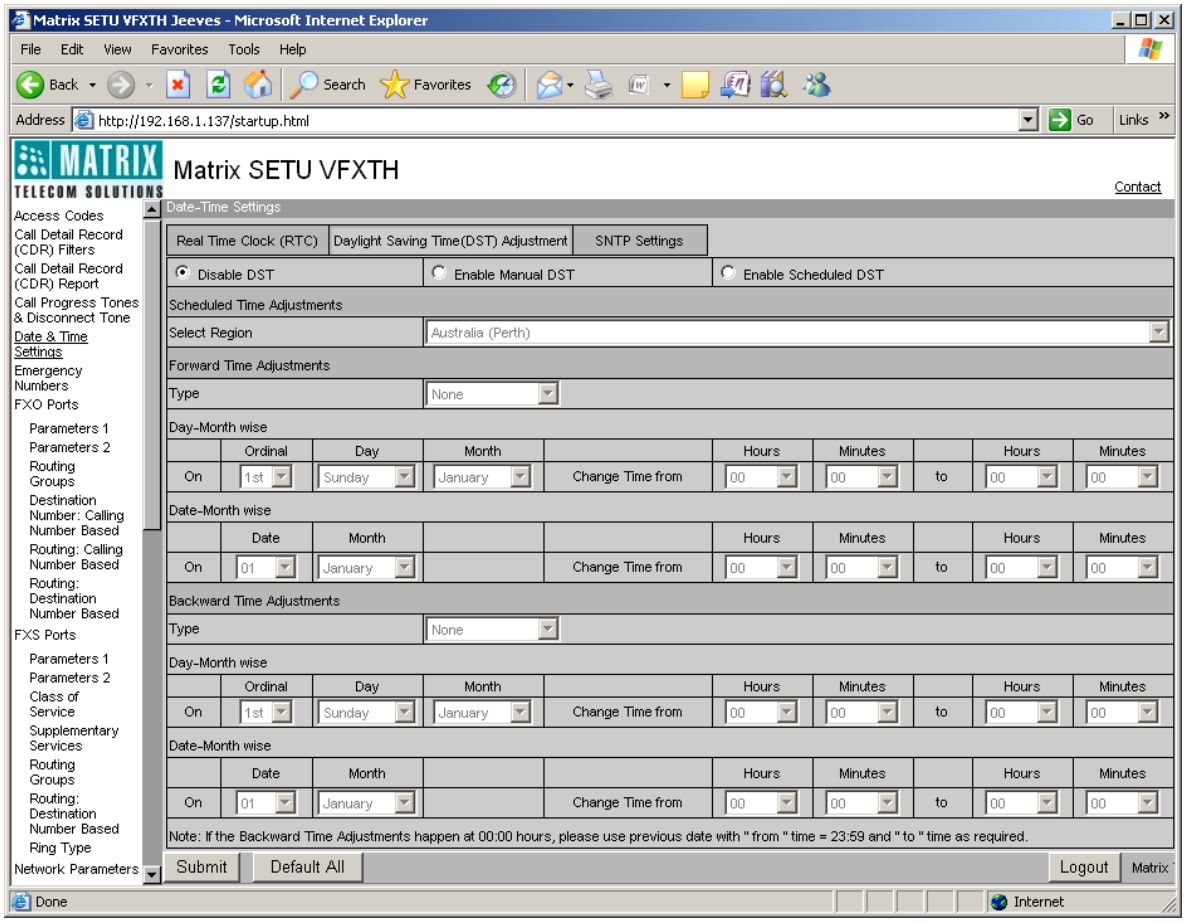
  a. **'Day-Month wise':** This method specifies a day of the month on which DST will start or end. For example, DST starting on 2nd Sunday of March and ending on 1st Sunday of November.

  b. **'Date-Month wise':** This method specifies a date of the month on which DST will start or end. For example, DST starting on 11th March and ending on 4th November.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on **'Date and Time Settings'** link.

- Click on '**Daylight Saving Time (DST) Adjustment**' and program the following parameters.



By default, Disable DST is selected. If SETU VFXTH is installed in the country in which DST is observed then enable DST by selecting either Manual DST or Scheduled DST option.

- If Scheduled DST is enabled then select Region where SETU VFXTH is installed from the list. All other parameters will be set automatically.

- If Manual DST is enabled then program following parameters:

  - **Forward Time Adjustments:** Go to **'Forward Time Adjustments'** and select the desired type of forward time adjustment to advance the time when DST will start. You have two options:

    - **Day-Month Wise:** If this option is selected then select suitable option in each of the following combo boxes:

      - *Ordinal:* Select 1st, 2nd, 3rd, 4th or 5th day of the month as ordinal.

      - *Day:* Select a day from Sunday to Saturday as the day on which DST shall start.
      - *Month:* Select a month from January to December in which DST shall start.

      - *Change Time form:* This is the time from when DST shall start changing. The time is in 24 hours format and valid range for hours is 00 to 23 and for minutes is 00 to 59.

      - *Change Time to:* The time to which DST is to be advanced. The time is in 24 hours format and valid range for hours is 00 to 23 and for minutes is 00 to 59.

- **Date-Month Wise:** If 'Date-Month wise' type of DST is selected then select the suitable option in each of the following combo boxes:

  - *Date:* Select a date on which DST shall start.

  - *Month:* Select a month from January to December in which DST shall start.

  - *Change Time form:* This is the time from when DST shall start changing. The time is in 24 hours format and valid range for hours is 00 to 23 and for minutes is 00 to 59.

  - *Change Time to:* The time to which DST is to be advanced. The time is in 24 hours format and valid range for hours is 00 to 23 and for minutes is 00 to 59.

  By default, Forward time adjustments type is 'None'. None is selected when SETU VFXTH is installed in the country where DST is not observed.

- **Backward Time Adjustments:** Go to **'Backward Time Adjustments'** and program its parameters in the same manner as mentioned above for Forward Time Adjustments.

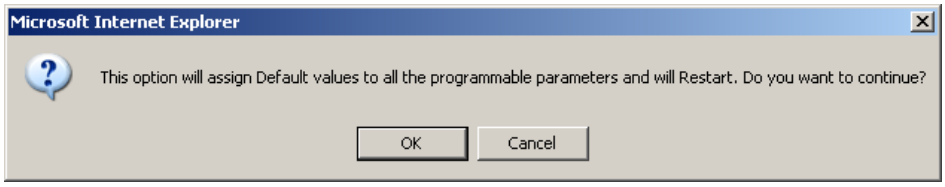
Default SETU VFXTH

SETU VFXTH enables user to default the system and assign default values to all programmable parameters by using 'System Default' feature. This feature will set all parameters to factory set values except the following features:

1. Call Detail Records
2. Call Progress Tones
3. Date and Time
4. Region/Country
5. Ring Type

Default SETU VFXTH using Web JEEVES

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on **'Default SETU VFXTH'** link. An alert message window will appear stating: "This option will assign default values to all the programmable parameters and will restart. Do you want to continue?"



- Click on **'OK'** button. The system will restart. All the programmable parameters except those mentioned above will get default.

Default SETU VFXTH using Commands

- Dial ***#19-1234***. You will get programming tone. (#19 is the default programming access code and 1234 is the default SE Password.)
- Dial ***51-Reverse SE Password-#****, Where Reverse SE Password is of four digits. By default, Reverse SE Password would be 4321.

  As soon as the command to default the system is issued, the system restarts. All the programmable parameters will be assigned default value except those mentioned above.

Relevant Topic:

1. “Software Upgrade” 171

Digest Authentication

SETU VFXTH supports challenge based authentication service on SIP Trunk called SIP Digest Authentication to authenticate the caller. Using digest authentication, INVITE message is challenged in form of SIP message and the caller is requested to prove his credentials. Credentials are User ID and Password of the calling party which is known to the SETU VFXTH.

Calling Party shall send his credentials (i.e. User ID and Password) when challenged by SETU VFXTH. If the credentials matches the entry stored in the database of SETU VFXTH, the call is allowed else it is rejected.

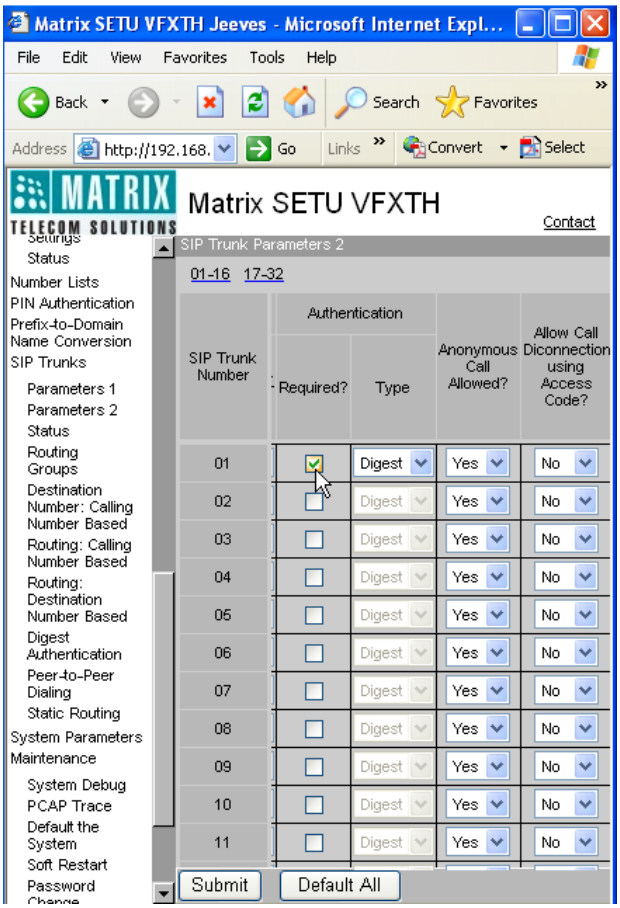
Let us understand Digest Authentication with the help of an example:

- Suppose a company has its branch offices in Mumbai, Kolkata, Chennai and New Delhi.
- Company wants to establish a voice communication over IP between all its offices. Therefore, it installs SETU VFXTH at all its offices. Static IP address is assigned to the WAN Port of all SETU VFXTH and peer to peer call functionality is enabled between all the offices.
- Company does not want to receive calls from outside callers and want to dedicate a SIP Trunk exclusively for inter-office calling only. Therefore, the company shall enable, Digest Authentication on the dedicated SIP Trunk, in all its branch offices and program numbers/contact addresses of all its branch offices in Digest Authentication table.
- Now, whenever an incoming call lands on SIP Trunk of SETU VFXTH, it will request credentials (User ID and Password) from the calling party.
- When called party send credentials, SETU VFXTH will compare it with its Digest Authentication table. If best match is found, SETU VFXTH will allow that incoming call else reject it.

How to Program?

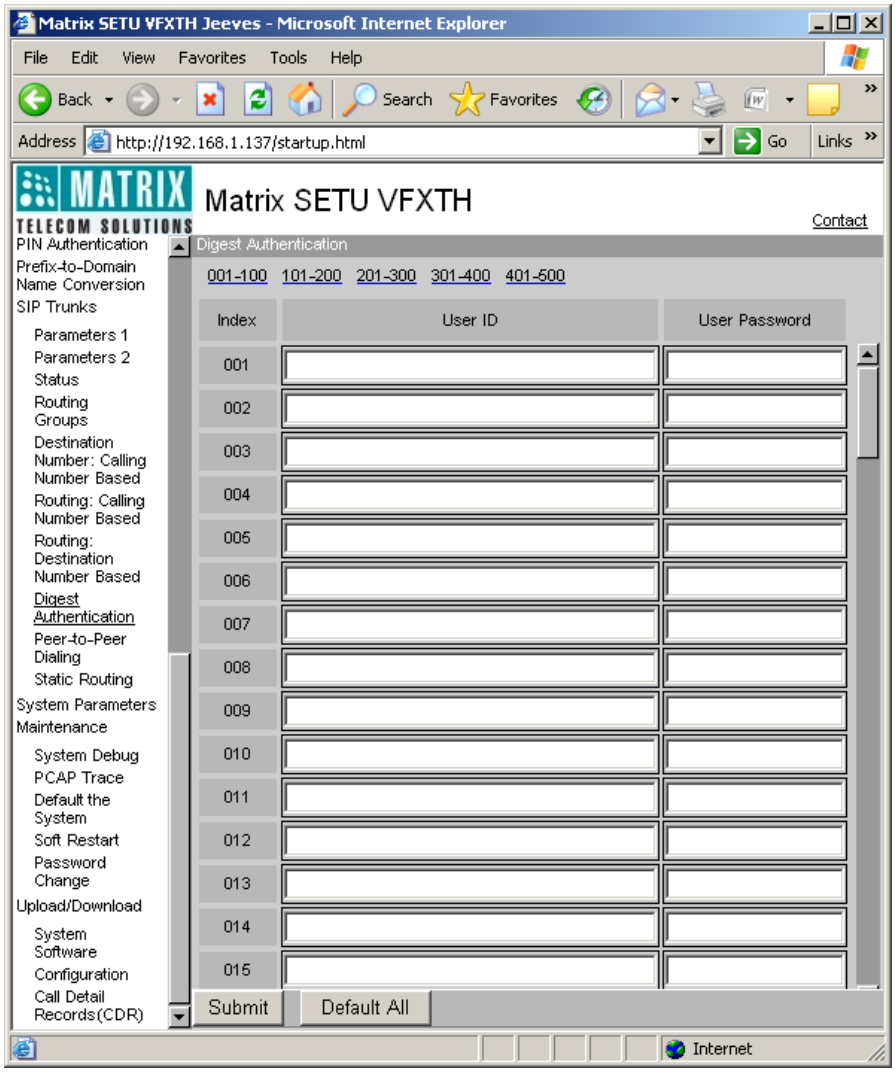
- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'SIP Trunk Parameters 2'** link and program the following parameters.



- **Digest Authentication?** Tick this Check box to enable Digest Authentication on the SIP Trunk. By default, it is un-ticked.

- Click on **'Digest Authentication'** link and program User ID and User Password for each caller in Digest Authentication table.



  - **User ID:** User ID can be of maximum 40 characters. All ASCII characters are allowed.

  - **User Password:** User Password can be of maximum 16 characters. All ASCII are characters allowed.

a. *Caller will be given only one attempt to send valid User ID and Password for Digest Authentication. If he sends invalid User ID and User Password, call will be rejected.*

b. *Digest Authentication table provides 500 entries. Thus maximum 500 callers can be authenticated using Digest Authentication.*

Relevant Topics:

1. “Call Processing” 47
2. “PIN Authentication” 101
3. “Port Parameters-SIP” 117

Emergency Number Dialing

Emergency Number Dialing feature enables user to call emergency services such as Ambulance, Fire Brigade, Police, etc. in adverse situations. SETU VFXTH supports dialing of Emergency Numbers from FXO Port and SIP Trunk. To use this feature, SE must program the Emergency Number Table. Emergency Number Table has total 10 entries and each entry has two parameters:

1. Emergency Number
2. Routing Group

Each Emergency Number is dialed using the Routing Group assigned to it. Emergency Number will be dialed out only if the status of that port is enabled.

Emergency number Dialing will not work if Mains power to SETU VFXTH fails.

How to Program?

- You can program numbers of your choice in index numbers 5 to 10 of the Emergency Number Table.

You are recommended to program the numbers prevalent as Emergency Numbers in your region, in the Emergency Number Table.

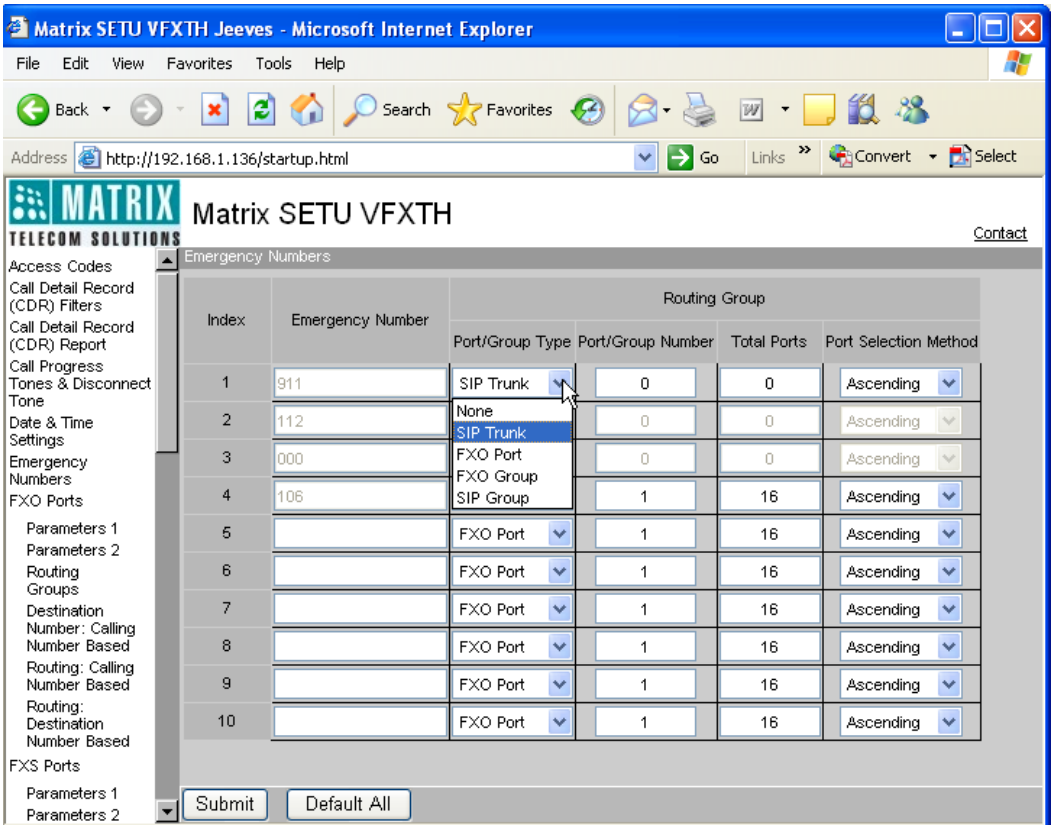
- Numbers available in index numbers 1 to 4 are fixed i.e. they are non-programmable.
- You can program Routing Group for all ten Emergency Numbers.

  Default configuration of Emergency number table is as shown below:

Index	Emergency Number	Routing Group			
		Port/Group Type	Port/Group Number	Total Ports	Port Selection Method
1	911	None	1	0	Ascending
2	112	None	1	0	Ascending
3	000	None	1	0	Ascending
4	106	FXO	1	16	Ascending
5		FXO	1	16	Ascending
6		FXO	1	16	Ascending
7		FXO	1	16	Ascending
8		FXO	1	16	Ascending
9		FXO	1	16	Ascending
10		FXO	1	16	Ascending

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Emergency Numbers'** link.



- Program emergency numbers in the 'Emergency Number' column of the emergency number table. You can program emergency numbers in the index numbers 5 to 10 whereas entries in the index numbers 1 to 4 are fixed i.e. non-programmable. Emergency Number can be of maximum 3 digits and digits 0 to 9 are allowed.

- Program Routing Group parameters for each Emergency Number as shown below.

  - **Port/Group Type:** Select Port/Group Type for routing each emergency number. Group can be of following types:

    i. None
    ii. FXO Port
    iii. SIP Trunk
    iv. FXO Group
    v. SIP Group

  - **Port/Group number:** Program Group Number for routing Emergency Number in this field. Valid Group Number options are:

    i. For FXO Port: 1 to 16
    ii. For SIP Trunk: 1 to 32
    iii. For FXO Group: 1 to 16
    iv. For SIP Group: 1 to 9

  - **Total Ports:** Program the number of ports that can be selected for routing the call.

i. For SIP Trunk: 1 to 32
ii. For FXO Port: 1 to 16

- **Port Selection Method:** Program the method for selecting Port for routing the call in this field. SETU VFXTH supports following methods for selecting the port or channel:
  i. Ascending
  ii. Descending

Refer Default Emergency Number Table shown above for default settings.

- *Avoid programming Access Codes which might conflict with the Emergency Numbers.*
- *Emergency Numbers have priority over Destination Number Table, PIN Number and Access Codes.*
- *End-of-Dialing is not applicable while dialing Emergency Number.*
- *Emergency Number can be dialed out even if outgoing calls are disabled for the FXS Port.*
- *Automatic Number Translation logic and Allowed-Denied Number logic is not applicable on Emergency Numbers.*

Relevant Topics:

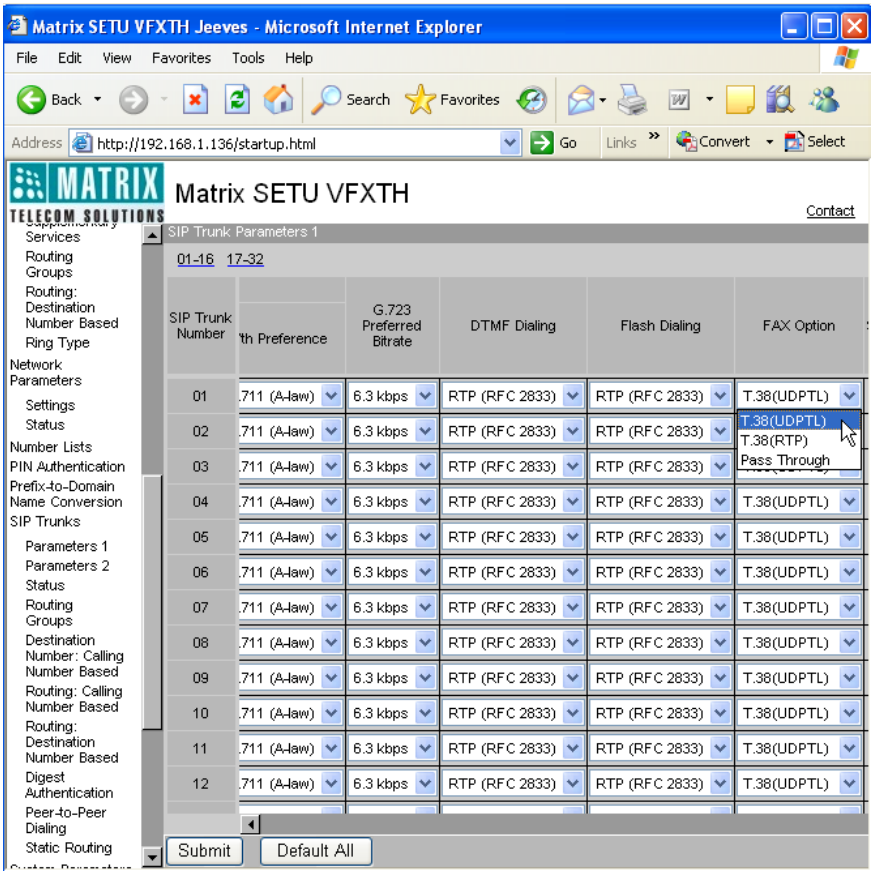
1. “Allowed-Denied Numbers” 28
2. “Automatic Number Translation” 33
3. “Call Processing” 47
4. “PIN Authentication” 101
5. “Routing Mechanism on FXS” 155
6. “Routing Mechanism on FXO” 143
7. “Routing Mechanism on SIP” 161

FoIP (Fax over IP) using T.38

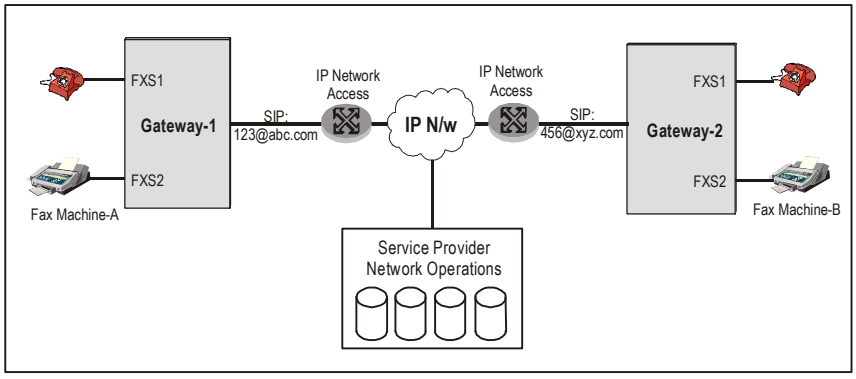
SETU VFXTH supports facility of Fax over IP network.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on **'SIP Trunk Parameters 1'** link and select the method for making FAX calls in 'FAX Option' field. SETU VFXTH supports T.38 (UDPTL), T.38 (RTP) and Pass Through. By default, T.38 (UDPTL) is selected.



Following example explains how fax calls shall be handled by SETU VFXTH.



Suppose SETU VFXTH is installed in Office 1 at Mumbai as well as in Office 2 at Chennai.

At Office 1

- SE shall program SETU VFXTH in such a way that, incoming calls on SIP Trunk 123@abc.com are routed on FXS2 of the Gateway-1, where the Fax machine-A is connected as shown in above figure.
- Thus Mumbai office (Office 1) fax number will be 123@abc.com and should be given to all the outsiders.

At Office 2

- SE shall program SETU VFXTH in such a way that, incoming calls on SIP Trunk 456@xyz.com are routed on FXS2 of the Gateway-2 where the Fax machine-B is connected as shown in above figure.

- Thus Chennai office (Office 2) fax number will be 456@xyz.com and should be given to all the outsiders.

For receiving voice calls both the Gateway owner should subscribe to another SIP Trunk and route those SIP Trunk calls to the FXS of the Gateway where SLTs are connected.

Case-1: Call from 123@abc.com to 456@xyz.com

- When user of Fax machine-A dials 456@xyz.com.
- Gateway shall make the call using SIP Trunk programmed for FXS Port with preferred Fax option. SETU VFXTH supports following Fax options:

  a. T.38 (UDPTL)
  b. T.38 (RTP)
  c. Pass Through

- Fax machine-A shall send 1100Hz signal when user presses start button of fax machine.

- Fax machine-B shall respond with 2100Hz signal.

- On detecting 2100Hz signal, Gateway-2 shall identify it as a fax call and shall re-negotiate with Gateway-1 for vocoder T.38.

- If Gateway-1 supports vocoder T.38, it shall entertain the request and fax call shall get completed using vocoder T.38.

- If Gateway-1 does not support vocoder T.38 then Gateway-2 shall again re-negotiate with Gateway-1 for another vocoder [Generally G.711] and get the fax call completed, which is called "Fax Pass Through".

Case-2: Similarly call from 456@xyz.com to 123@abc.com shall be handled.

When the call originates on the 456@xyz.com and if destination port is programmed as FXO or SIP or if the FAX machine is not connected to the FXS Port, this call will be handled as the normal SIP Trunk call instead of FAX call.

Related Topic:

1. “Port Parameters-SIP” 117

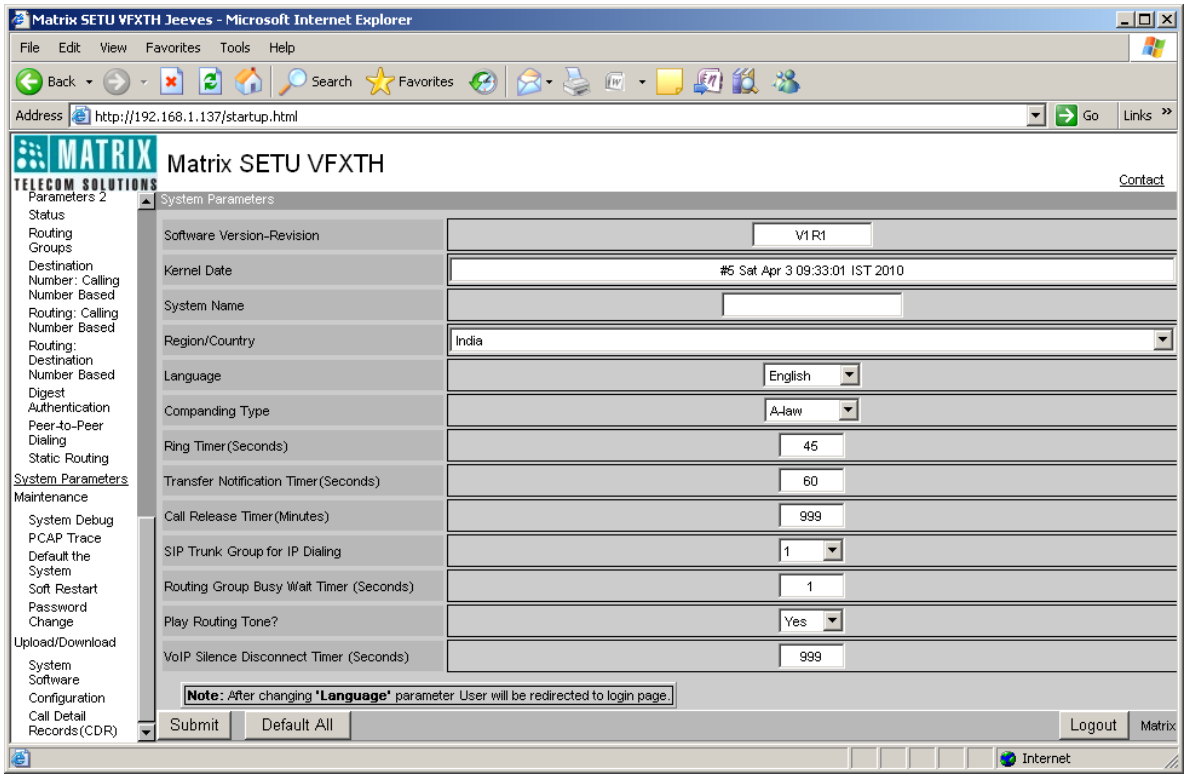
IP Dialing

SETU VFXTH supports multiple port types' viz. FXO, FXS and SIP. When a number is dialed out from any port, SETU VFXTH shall route that call to the desired destination as per the routing mechanism programmed for that port. When an IP Address is dialed from any port of SETU VFXTH, it is routed to the desired destination through the SIP Trunk group programmed for IP dialing. SIP Trunk group for IP dialing is applicable to all port types.

For dialing an IP Address, user should dial (*) in place of (.) i.e. if user wants to dial 192.167.100.001 then he should dial 192*167*100*001 from his telephone instrument. When IP Address is dialed, logic will not check Destination Port Determination Method.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")
- Click on **'System Parameters'** link.



- Program 'SIP Trunk Group for IP Dialing' parameter. Range of this parameter is 'None' and SIP Trunk Group 1 to 9. By default, SIP Trunk Group selected for IP Dialing is 1.

If SIP Trunk Group for IP Dialing is programmed as 'None' then SETU VFXTH will give error tone and reject the call.

Relevant Topics:

1. "Call Processing" 47
2. "Routing Group" 135
3. "Routing Mechanism on FXO" 143
4. "Routing Mechanism on FXS" 155
5. "Routing Mechanism on SIP" 161

Network Port Parameters

Network conditions vary for every customer. Configuring Network Port Parameters is an important step in configuring SETU VFXTH1616. When SETU VFXTH1616 is connected to LAN of an organization, the IP Address of SETU VFXTH1616 is to be changed according to the LAN Addressing Scheme. All the Network Port Parameters can be programmed using Web JEEVES while some parameters like IP Address, Subnet Mask, Connection Type etc., can be programmed using telephone instrument also.

Programming Network Port Parameters using Telephone Instrument:

To program Network Port Parameters using telephone instrument, dial ***#19-1234.*** (#19 is the programming access code and 1234 is the default SE password.)

Use following command to program the Connection Type:
10-Code-#*
Where,

Code	Connection Type
1	Static
2	DHCP
3	PPPoE

By default, Connection Type set for SETU VFXTH1616 is Static.

Use following command to program the IP Address:
11-IP Address-#*
Where,
IP Address is of 12 digits in XXX.XXX.XXX.XXX format
By default, IP Address of SETU VFXTH1616 is 192.168.001.136.

Use following command to program the Subnet Mask:
12-Subnet Mask-#*
Where,
Subnet Mask is of 12 digits in XXX.XXX.XXX.XXX format
By default, Subnet Mask of SETU VFXTH1616 is 255.255.255.000.

Use following command to enable/disable VLAN tag:
31-Code-#*
Where,

Code	Meaning
0	Disable
1	Enable

By default, VLAN tag is disabled.

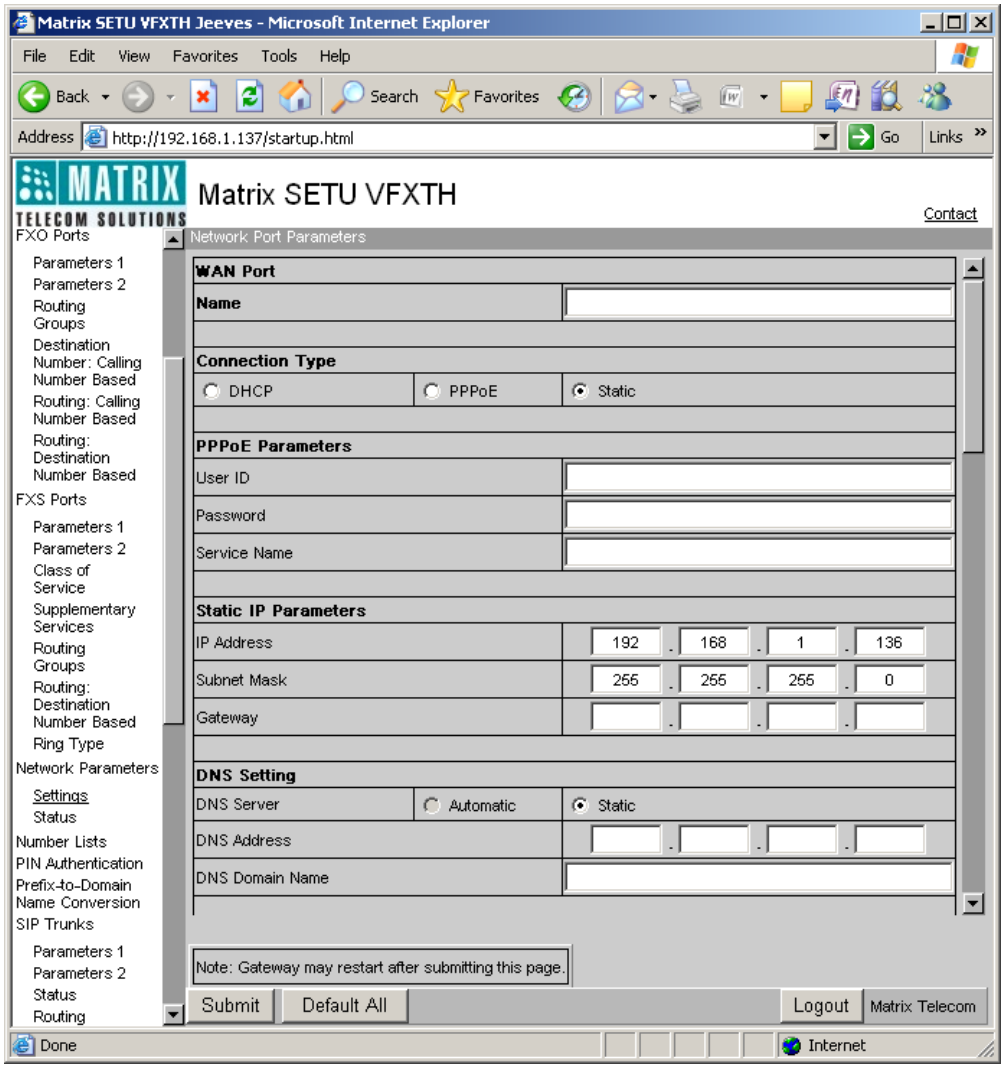
You can also display values of various Network Port Parameters on the LCD of your telephone instrument.

Follow the steps given below:

1. Pick up handset of the telephone instrument connected to your SETU VFXTH1616.
2. Dial ***'#19'*** to enter the programming mode.
3. Enter the following commands to display the value of desired Network Port Parameter and go On-Hook.
   a. To display the Connection Type, use command: ***20-#****
   b. To display the Network IP Address, use command: ***21-#****
   c. To display the Subnet Mask, use command: ***22-#****
   d. To display the Gateway Address, use command: ***23-#****
   e. To display the DNS Address, use command: ***24-#****
4. Value for the desired Network Port Parameter will be displayed on the LCD of your telephone instrument.
5. Exit programming mode by dialing ***00-#****, when you are finish.

Programming Network Port Parameters using Web JEEVES:

- Open Web JEEVES of SETU VFXTH1616. (Refer "Accessing Web JEEVES")
- Click on **'Network Port Parameters'** link and program the following parameters.



- **Name:** In this filed, a Name can be assigned to WAN port of SETU VFXTH1616. Name can be of maximum 12 characters and all ASCII characters are allowed. By default, it is blank. This name is just a tag to the Network Port and it is not used anywhere in the system.

- **Connection Type:** Select a Connection Type depending on the IP Addressing Scheme of the network. SETU VFXTH1616 supports following connection types:

  - ***Static:*** Select **'Static'** if you want to assign IP Address, Subnet Mask and Gateway Address manually. Obtain necessary information from your LAN administrator and enter it manually. By default, 'Static' is selected.

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

  - ***PPPoE:*** Select **'PPPoE'**, if you want PPPoE server to assign IP Address, Subnet Mask, DNS Address and Gateway Address automatically. If you select this option, you must enter User ID and Password provided by your service provider in PPPoE Parameters.

When connection type is changed from DHCP or PPPoE to Static then DNS connection type should also be changed to Static.

- **PPPoE Parameters:** If you have selected 'PPPoE' connection type then program following parameters:

  - ***User ID:*** Enter PPPoE User ID provided by your service provider manually in this field. User ID can be of maximum 16 characters. All ASCII characters are allowed in this field. By default, it is blank.

  - ***Password:*** Enter PPPoE User Password provided by your service provider manually in this field. Password can be of maximum 16 characters. All ASCII characters are allowed in this field. By default, it is blank.

  - ***PPPoE Service Name:*** Configure 'PPPoE Service Name' if it is provided by your ISP else keep it blank. PPPoE Service Name can be of maximum 16 characters. All ASCII characters are allowed. By default, it is blank.

- **Static IP Parameters:** If you have selected 'Static' connection type then program following parameters:

  - ***IP Address:*** Enter 'Static IP Address' provided by your service provider in this field. Default Static IP Address of SETU VFXTH1616 is 192.168.001.136. IP Address can be of maximum 15 characters. Valid range for first three octets is from 000 to 255 and for other octets is from 000 to 254.

  - ***Subnet Mask:*** Enter 'Subnet Mask' provided by your service provider in this field. Default subnet mask of SETU VFXTH1616 is 255.255.255.000. Subnet Mask can be of 15 characters. Valid octets for subnet mask are 000, 128, 192, 224, 240, 248, 252, 254 and 255.

  - ***Gateway:*** Enter Gateway Address (if provided by your ISP) in this field. Gateway Address can be of maximum 15 characters. Valid octet for Gateway Address is 000 to 255. By default, Gateway Address is blank.

- **DNS Setting:** DNS stands for Domain Name Server which is used to resolve domain name in to IP address. Select either Static DNS or Automatic DNS. By default, Static DNS is selected.

- ***Automatic DNS:*** Select 'Automatic DNS' if your ISP provides DHCP or PPPoE type of connection.
- ***Static DNS:*** If Static DNS is selected then program the following parameters manually:

    1. *DNS Address:* Enter DNS address provided by the service provided manually in this field. DNS Address can be of maximum 15 characters. Valid range is 000 to 255 for all octets. By default, it is blank.

    2. *DNS Domain Name:* Enter DNS Domain Name in this field. It can be of maximum 40 characters. All ASCII characters are allowed. By default, it is blank.

- **Dynamic DNS (DDNS):** Dynamic DNS (DDNS) is a service that maps internet domain names to IP addresses. DDNS Service Provider provides the hostname/domain name to the internet devices and also embed DDNS client in the internet device. Doing so, whenever new IP Address is assigned to internet host, DDNS client running in the internet host update 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 name of the system.

    - ***Enable Dynamic DNS?:*** Select 'Yes' in this field to enable dynamic DNS service. By default, it is set to 'No'. If Dynamic DNS service is enabled, program following parameters.

    - ***User Name:*** Program User Name provided by DynDNS. Org in this field. User Name can be of maximum 40 characters. All ASCII characters are allowed. By default, 'User Name' is blank.

    - ***Password:*** Program Password provided by DynDNS. Org in this field. Password can be of maximum 24 characters. All ASCII characters are allowed. By default, 'Password' is blank.

    - ***Host Name:*** Program Host Name provided by DynDNS.Org in this field. Host Name can be of maximum 40 characters. All ASCII characters are allowed. By default, 'Host Name' is blank.

- **UDP NAT Keep Alive:** When SETU VFXTH1616 is connected behind the NAT router and SIP messages are transported over UDP then UDP NAT Keep Alive messages are to be sent to refresh the UDP binding in the NAT router. UDP NAT Keep Alive settings can be done as follows:

    - ***Enable UDP NAT Keep Alive?:*** Select 'Yes' to enable SETU VFXTH1616 to send Keep Alive messages periodically to refresh UDP binding in the NAT router. By default, 'No' is selected.

    - ***Interval (Seconds):*** Select time period after which the SETU VFXTH1616 should send Keep Alive messages. This time period should be less than the binding timer of the router. The valid range is 001 to 999 seconds. By default, it is 120 seconds.

    - ***Message Type:*** Select the type of message to be sent to refresh UDP binding in the NAT router. Select either REGISTER or NOTIFY. By default, NOTIFY is selected.

- **TCP NAT Keep Alive:** When SETU VFXTH1616 is connected behind the NAT router and SIP messages are transported over TCP then TCP NAT Keep Alive messages are to be sent to refresh the TCP binding in the NAT router. SETU VFXTH1616 sends CR-LF (Carriage Return - Line Feed) message as TCP NAT Keep Alive message to refresh the TCP binding. TCP NAT Keep Alive settings can be done as follows:

    - ***Enable TCP NAT Keep Alive?:*** Select 'Yes' to enable SETU VFXTH1616 to send Keep Alive messages periodically to refresh the binding in the NAT router. By default, 'No' is selected.

- ***Interval:*** Select the time period after which the SETU VFXTH1616 should send keep alive messages. This time period should be less than the TCP binding timer of the router. The valid range is 0001 to 9999 seconds. By default, it is 120 seconds.

- **Router's Public IP Address:** Router's Public IP Address is the IP address of the router behind which SETU VFXTH1616 is located.

  Enter public IP address of the router if 'Source Port IP Address' is set to 'Use Router's Public IP Address' in SIP parameters. Maximum 15 characters are allowed. Valid range of IP address is 000 to 255. By default, it is blank.

  (Refer "Port Parameters-SIP" for more details)

- **STUN:** STUN is required if SETU VFXTH1616 is located behind the NAT router. STUN server facilitates traversing through most NATs except symmetric NATs. If your router has symmetric NAT, do not program STUN.

  - ***Server Address:*** Enter STUN server address in this field. Server Address can be of maximum 40 characters. All ASCII characters are allowed. By default, STUN server address is blank.
  - ***Server Port:*** Enter STUN Server's listening port in this field. Valid range of Server Port is from 1024 to 65535. By default, the value is 3478.

  - ***Use SIP Port fetched using STUN?:*** This parameter is applicable only when 'Source Port IP Address' is set to 'Use IP Address fetched using STUN'. If the SETU VFXTH1616 is located behind the NAT router and SIP Listening Port is forwarded to the NAT router, you do not need to use the port provided by STUN server. Hence, select 'No' in this field. Select 'Yes' in this field, if you have not forwarded the SIP Port to the NAT router. By default, this field is set to 'No'.

- **Quality of Service (Layer 3):**

  - ***SIP Diffserve/ToS:*** SETU VFXTH1616 will send all SIP messages using SIP QoS setting. Program SIP Diffserve/ ToS for SIP signaling. Valid range for SIP Diffserve /ToS is from 00 to 63. By default, '26' is selected.

  - ***RTP Diffserve/ToS:*** SETU VFXTH1616 will send all the RTP packets using RTP QoS setting. Program RTP Diffserve/ ToS for RTP traffic. Valid range for RTP Diffserve/ ToS is from 00 to 63. By default, '46' is selected.

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

  - ***Enable VLAN/ CoS?:*** Select 'Yes' if you want SETU VFXTH1616 to add L2 VLAN header to all the packets leaving the Network Port. L2 VLAN header include VLAN ID and Class of Service (CoS) bits. By default, it is set to 'No'.

  - ***VLAN ID:*** This is the Virtual LAN Identifier. Program VLAN ID in this field. This ID will be added to the VLAN header if 'Enable VLAN/ CoS?' is set to Yes. Range of VLAN ID is 0-4094. By default, it is 1.

  - ***SIP CoS:*** Program SIP CoS in this field. SETU VFXTH1616 will add CoS bits to the SIP packets leaving the Network Port, if 'Enable VLAN/ CoS?' is set to Yes. Range of SIP CoS is 0-7. By default, it is 3.

- ***RTP CoS:*** Program RTP CoS in this field. SETU VFXTH1616 will add CoS bits to the RTP packets leaving the Network port, if 'Enable VLAN/ CoS?' is set to Yes. Range of RTP CoS is 0-7. By default, it is 6.

- **SIP/ RTP Port:** Program SIP/RTP Port parameters as shown below.

  - ***SIP UDP Port:*** SIP UDP port defines the port on which SETU VFXTH1616 listens for SIP messages transported over UDP. Range of SIP UDP Port is 1024-65535. By default, SIP UDP Port is 5060.

  - ***SIP TCP Port:*** SIP TCP port defines the port on which SETU VFXTH1616 listens for SIP messages transported over TCP. Range of SIP TCP Port is 1024-65535. By default, SIP TCP Port is 5060.

  - ***RTP Listening Port:*** RTP Listening port defines the port on which SETU VFXTH1616 listens for RTP packets. System also uses this port as source port in RTP packets while sending RTP to remote peer. Range of RTP Listening Port is 1024-65525. By default, RTP Listening port is 8000.

- **Enable SIP over TCP?:** By default, Enable SIP over TCP is set to 'Yes' i.e. SETU VFXTH1616 listens for SIP messages over TCP. Select 'No' if you do not want SETU VFXTH1616 to send and receive SIP messages over TCP.

- **Send ICMP Message?:** By default, this field is set to 'Yes' to send ICMP error message whenever packets are received on closed TCP/ UDP ports. Select 'No' if you do not want ICMP error message to be sent on receipt of packets on closed TCP/ UDP ports.

- **Enable 100rel?:** Select 'Yes' if you want SETU VFXTH1616 to use 100rel extension for reliable transmission of SIP (provisional) responses and use of PRACK. By default, it is set to 'No'.

- **Timer:** SETU VFXTH1616 support three programmable SIP timers viz. SIP INVITE Timer, SIP Provisional Timer and General Request Timer.

  - ***SIP INVITE Timer:*** SETU VFXTH1616 waits for a response from the called party after sending INVITE message for this timer. Range of SIP INVITE Timer is 010 to 180 seconds. By default, it is 30 seconds.

  - ***SIP Provisional Timer:*** SETU VFXTH1616 waits for final response after receiving provisional response from the called party for this timer. Range of SIP Provisional Timer is 010 to 180 seconds. By default, it is 60 seconds.

  - ***SIP General Request Timer:*** SETU VFXTH1616 waits for the response of a transaction request for this timer. Range of General Request Timer is 10 to 60 seconds. By default, it is 20 seconds.

- **MAC Address:** You can use either Unique or Clone MAC Address as per your requirement. Select the type of MAC Address you want to use in this field. By default, Unique is selected.

  - ***Unique MAC Address:*** If 'Unique' is selected, SETU VFXTH1616 will use unique MAC address assigned to it as source MAC address in all Ethernet frames.

  - ***Clone MAC Address:*** Select 'Clone' if you want to use MAC Address other than Unique MAC Address. Enter desired MAC Address manually in this field. Only Hexadecimal characters are allowed and by default, it is blank.

SETU Vfxth1616 will reboot as soon as you submit the page after changing any of the parameters of Network Port.

Number Collection Method

The method of collecting digits for processing a call is called Number Collection Method. Number collection includes gathering of valid digits and detecting end-of-dialing to initiate processing of the call.

Following steps are involved in collection of number by SETU VFXTH:

1. First Digit Wait Timer is loaded as soon as the user goes Off-hook or the system answers the incoming call on FXO Port.
2. As soon as the user dials the first digit, First Digit Wait Timer is stopped and Inter Digit Wait Timer is loaded.
3. Inter Digit Wait Timer is loaded each time the user dials a new digit until it detects termination digit.
4. After detecting termination digit, SETU VFXTH will process the call as per the programming done for routing the call.

Therefore, number collection method involves two things:

1. First Digit Wait Timer
2. End-of-Dialing

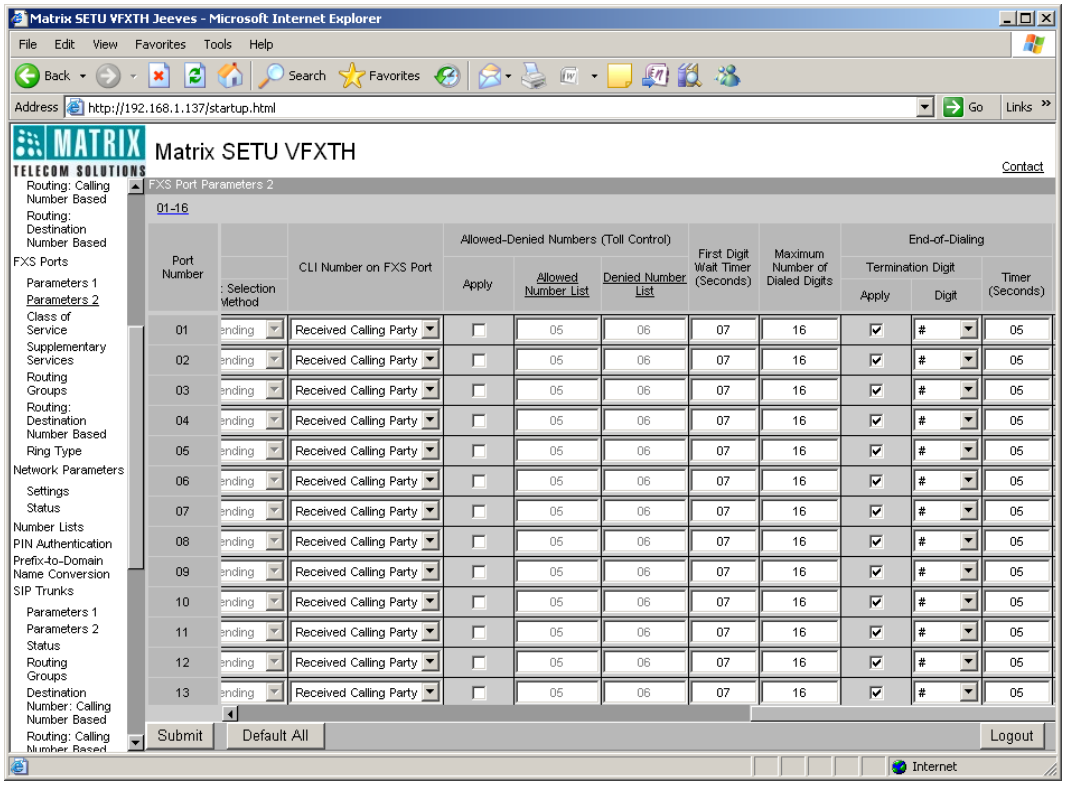
First Digit Wait Timer

First Digit Wait Timer is the time period for which SETU VFXTH waits for the user to dial first digit when FXS Port goes Off-hook or when the system answers the call on FXO Port and the Destination Number Determination Method is set to 'Manual Dial'. First Digit Wait Timer is not applicable on SIP Trunks.

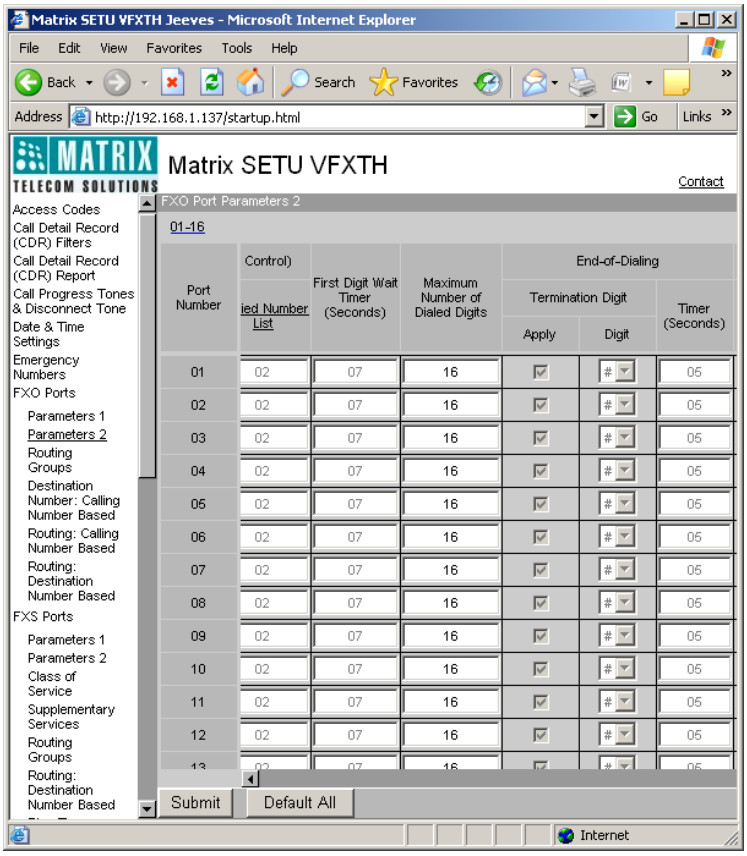
Program ‘First Digit Wait Timer’ using JEEVES

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'FXS Port Parameters 2'** link. Program First Digit Wait Timer for all FXS Ports. Range of First Digit Wait Timer is 01 to 99 seconds. By default, it is 07 seconds for all ports.



- Click on **'FXO Port Parameters 2'** link. Program First Digit Wait Timer for all FXO Ports. Range of First Digit Wait Timer is 01 to 99 seconds. By default, it is 07 seconds for all ports.



First Digit Wait Timer is required to decide that if no digit is dialed by the user before the expiry of this timer, SETU VFXTH should stop the dial tone and give error tone to the user or proceed as per the programming of the system.

End-of-Dialing

End-of-Dialing is a mechanism by which SETU VFXTH detects end of dialing of a number string and further processes the call. It enables faster out dialing of a number. End-of-Dialing is applicable only when Destination Number Determination Method is set to 'Manual Dial'. Hence, it is not applicable to the SIP Trunks.

SETU VFXTH supports following methods for End-of-Dialing:

1. Dialing Termination Digit

    - It is applicable when the call is answered on source port and the system is waiting for number collection which is to be out dialed on the destination port.
    - If Termination Digit flag is enabled and user dials digit programmed as Termination Digit then SETU VFXTH shall consider it as end-of-dialing of a number and shall start further processing of the call.
    - End-of-Dialing is applicable only if Destination Number Determination Method is set to Manual Dial.

2. Timer (Seconds)

    - Timer (seconds) refers to 'Inter Digit Wait Timer'. This Timer is loaded as soon as any digit is dialed during First Digit Wait Timer.
    - If the user does not dial the next digit before expiry of Inter Digit Wait Timer then SETU VFXTH shall consider it as end-of-dialing and shall process the call further as per the programming done for processing the call.
    - It is applicable only when Destination Number Determination Method is set to 'Manual Dial'.
    - This timer is applicable only on FXO and FXS Ports.

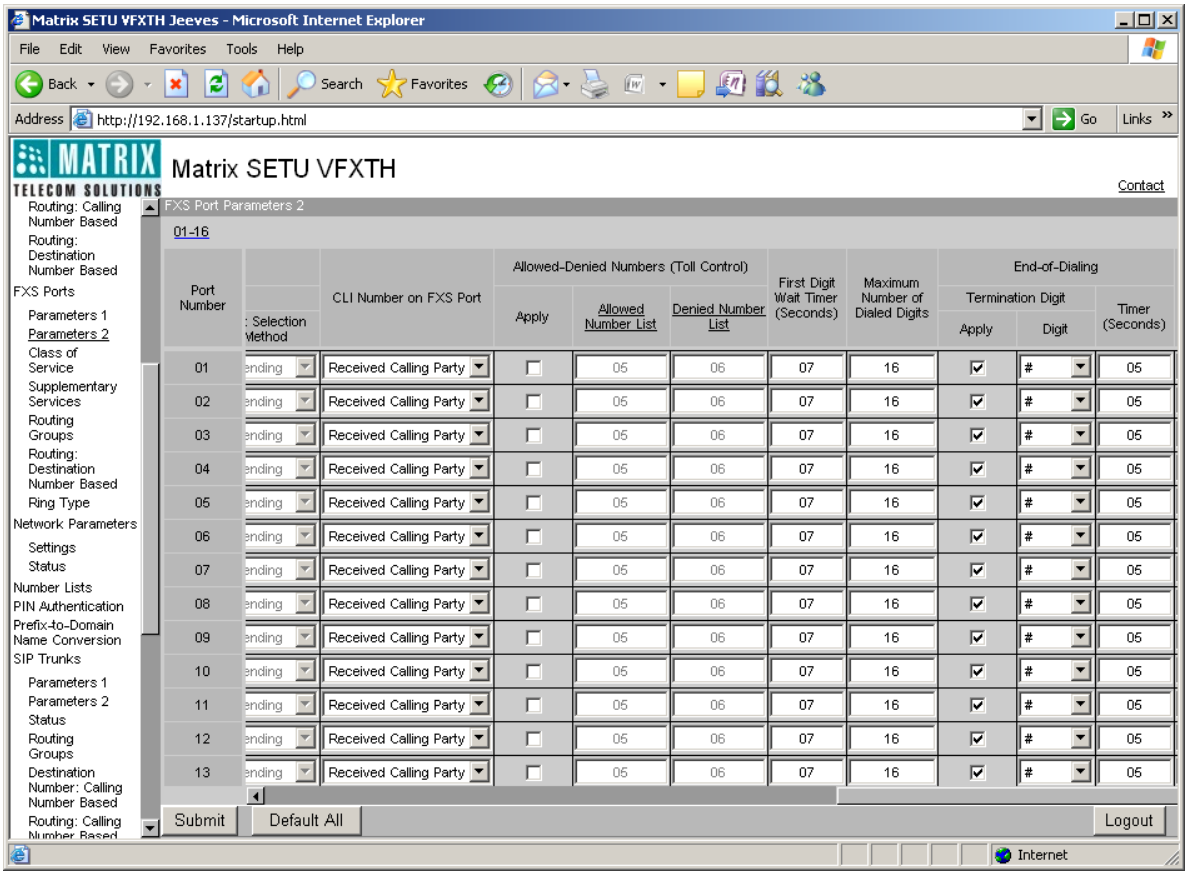
3. Maximum Number of Dialed Digits

    - SETU VFXTH shall consider it as end-of-dialing when maximum number of digits allowed to be dialed from the port is dialed out.
    - This method is applicable only when the user dials number equal to the maximum number of digits allowed before the expiry of Inter Digit Wait Timer.
    - It is applicable when Destination Number Determination Method is set to Manual Dial and Destination Port Determination Method is set to 'Fixed' or 'As per Calling Number Based'. It is also applicable for Fixed Destination Number, Hotline number and Call Forward number.

Program ‘End of Dialing’ using JEEVES

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'FXS Port Parameters 2'** link and program the following parameters:



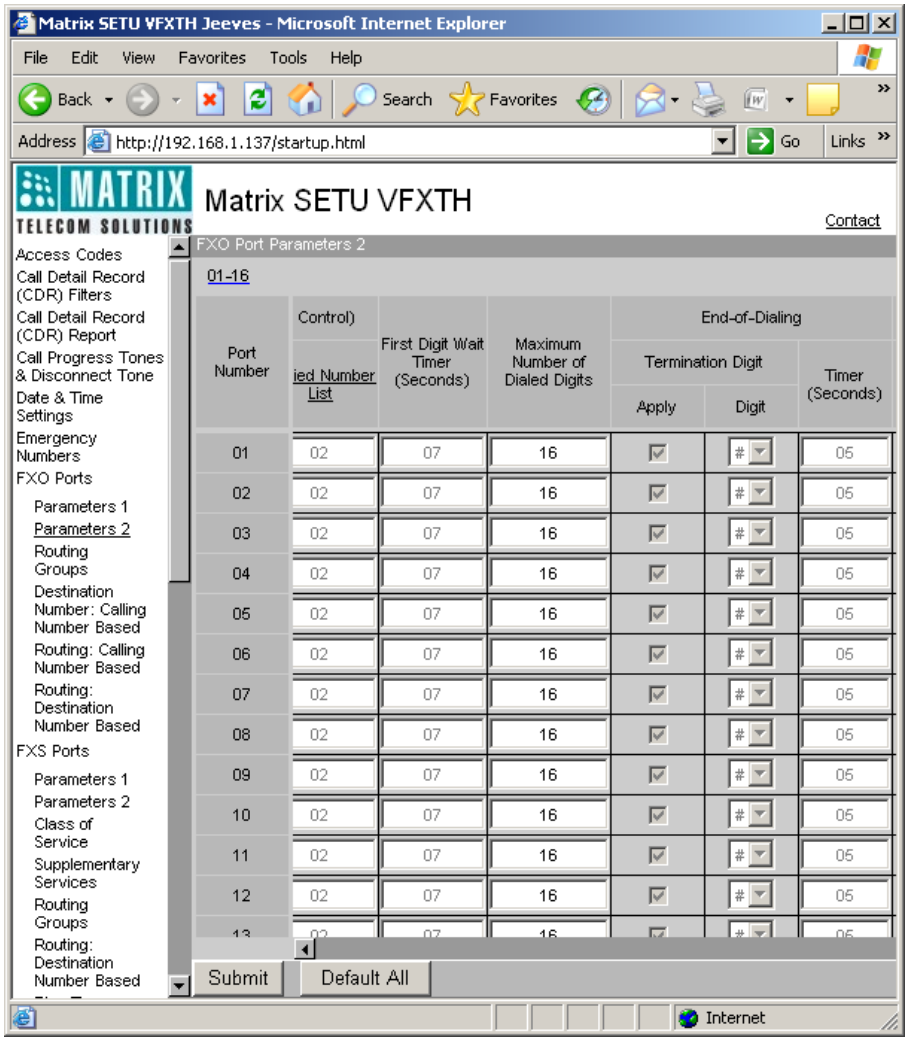
- **Maximum Number of Dialed Digits:** Program maximum number of digits allowed to be dialed by the user. Valid range of this parameter is from 01 to 24. By default, it is 16.

- **Termination Digit:**
  - ***Apply:*** Tick this flag to enable End-of-Dialing using termination digit. By default, it is enabled for all ports.

  - ***Digit:*** Select a digit you wish to dial to indicate End-of-Dialing in this field. You can select either '*' or '#' as termination digit. By default, '#' is selected for all ports.

- **Timer (seconds):** Select Inter Digit Wait Timer in this field. Range of this Timer is from 01 to 99 seconds. By default, it is programmed as 05 seconds. By default, SETU VFXTH use this 'Timer' to detect End-of-Dialing.

- Similarly click on **'FXO Port Parameters 2'** link and program 'Maximum Number of Dialed Digits' and 'End-of-Dialing' parameters as described for FXS port.



Relevant Topics:

1. “Call Processing” 47
2. “Port Parameters-FXS” 112
3. “Port Parameters-FXO” 104
4. “Routing Mechanism on FXO” 143
5. “Routing Mechanism on FXS” 155

Number Lists

Number List is the data structure in which specific number strings consisting of digits and/or characters are programmed for functioning of features such as Automatic Number Translation, Black Listed Callers and Allowed-Denied Numbers.

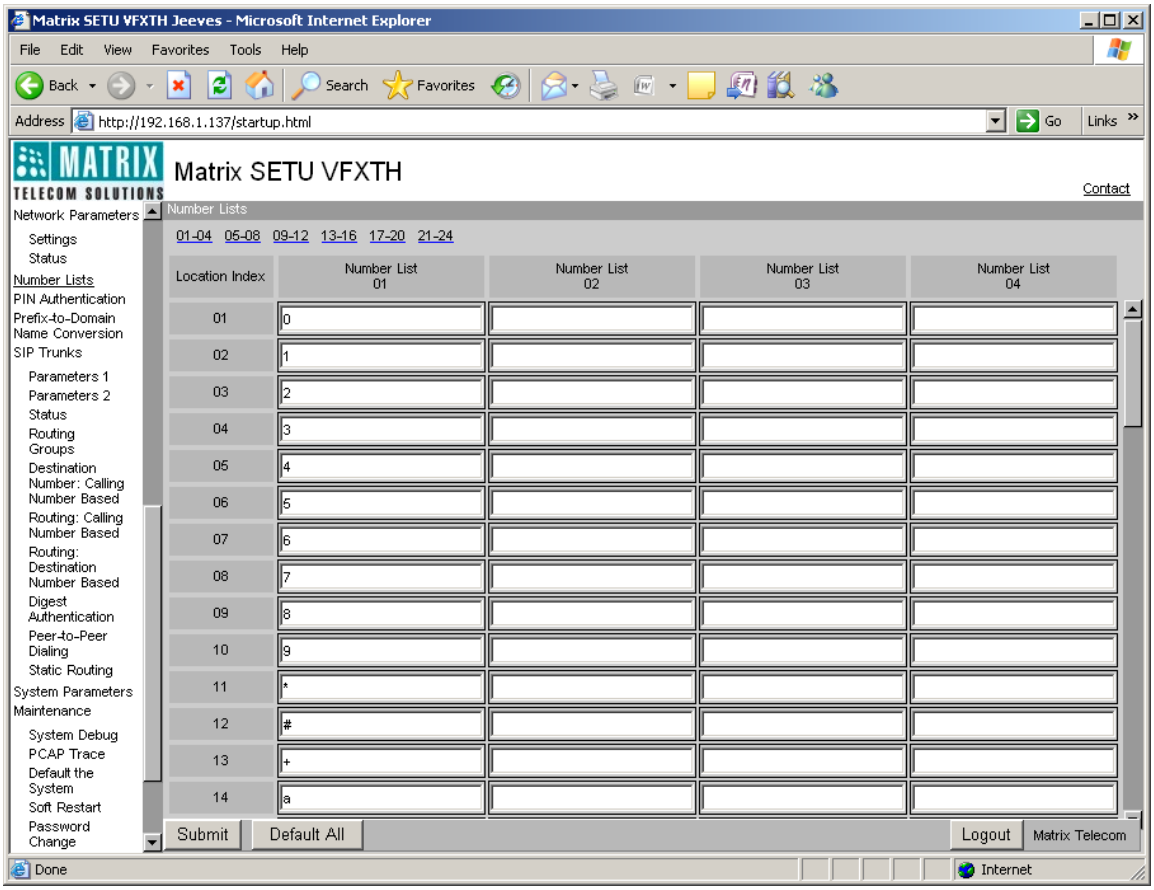
Number Lists are programmed for:

- Allowed-Denied Numbers on a source port. (FXO, FXS and SIP)
- Automatic Number Translation on a destination port. (FXO and SIP)
- Black Listed Callers on a SIP source port.
- Call Detail Records Filters for called party and calling party filter settings.

SETU VFXTH supports 24 number lists, each number list having total 64 entries. Each entry can be of maximum 24 characters and all ASCII characters are allowed. By default, in Number list 01, following values are programmed: 0 to 9, a to z, *, # and +. All other number lists are blank.

How to Program?

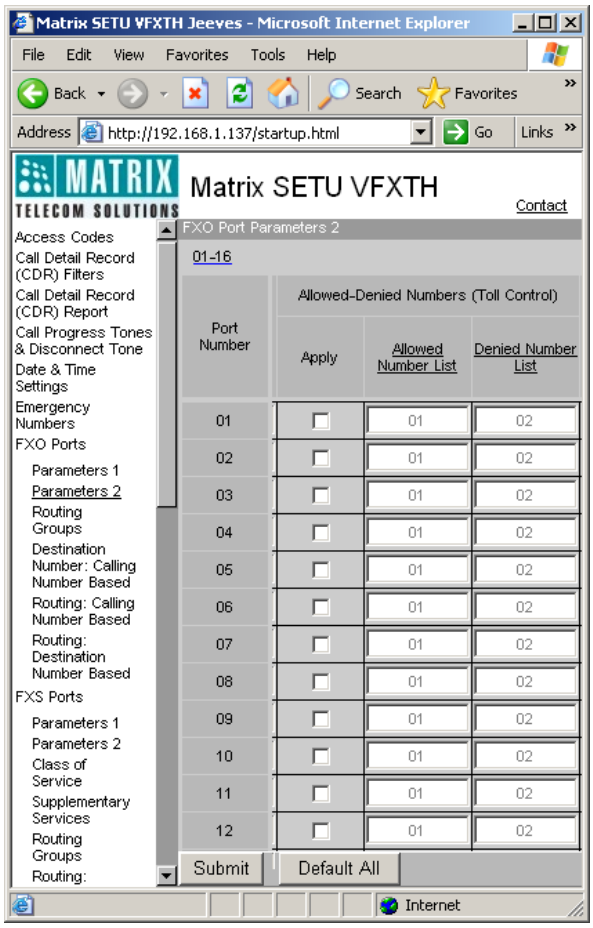
- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on **'Number Lists'** link. Program numbers for different purposes in different Number Lists such as numbers to be allowed in allowed number list, numbers to be denied in denied number list, etc.



- Program the number of Number List in relevant column of Port Parameters page.

Let us take an example to understand this feature:

Suppose SE has programmed numbers allowed on FXO Port 1 in list number 01 and numbers denied on FXO Port 1 in list number 02 in the Number List table.



- Click on **'FXO Port Parameters 2'** link.
- Go to **'Allowed-Denied Numbers (Toll Control)'** column and tick the 'Apply' flag for FXO Port 1.
- Enter allowed list number i.e. 01 in 'Allowed Number List' column and denied list number i.e. 02 in 'Denied Number List' column.
- Now whenever the port user dials any number, SETU VFXTH will check allowed and denied number list before dialing the number.
- The number will be dialed out only if it is allowed for that port.
- If the dialed number is programmed in both allowed and denied number list or it is not programmed in any of the number list then the system will dial out that number.

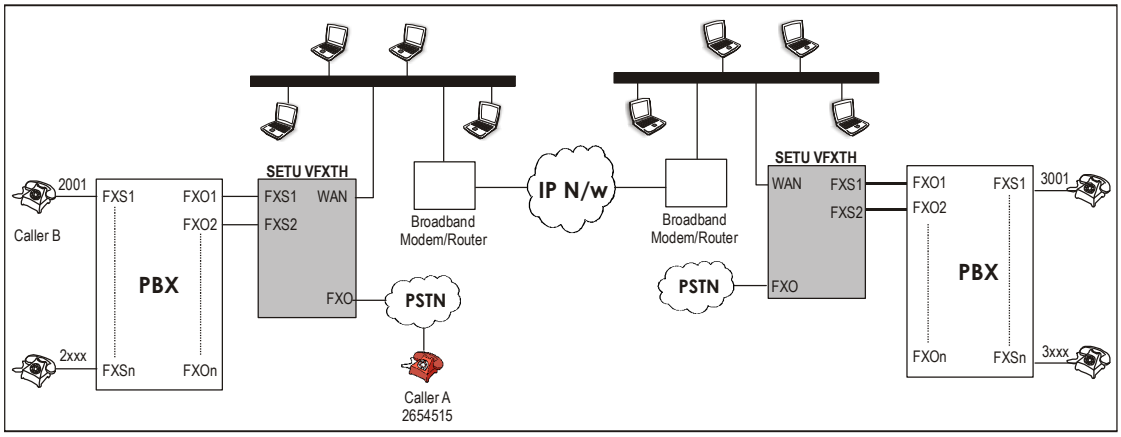
This example explains one of the applications of Number List feature. Similarly, number list is also useful for features such as Automatic Number Translation, Black Listed Callers etc.

Relevant Topics:

1. “Allowed-Denied Numbers” 28
2. “Automatic Number Translation” 33
3. “Black Listed Callers” 37
4. “Call Detail Record” 39

Number Presentation on FXS Port

SETU VFXTH supports Calling Party Number or Called Party Number Presentation on its FXS Port. Most of the applications demand Calling Party Number presentation on FXS Port. However, there are certain applications in which Called Party Number presentation is desirable on FXS Port. One such application is explained below.



As shown in the figure,

- SETU VFXTH is used to make P2P calls with its branch office.
- FXO Ports are connected to the PSTN.
- FXS Ports are connected to the FXO Ports of the PBX.
- Destination Number Determination Method on FXO Port is programmed as 'Manual Dial'.
- Destination Number Determination Method on SIP Port is programmed as 'Called Party Number'.
- The PBX is programmed for CLI based routing on its FXO Ports.
- Following is desirable:

  a. When Caller A (2654515) calls SETU VFXTH and dials extension 3001, the call should be placed on extension 3001 of the PBX.

  b. When Caller B (2001) dials 3001, the call is placed on extension 3001 of the PBX.

Let us understand how it works

Case 1:

- Suppose 'CLI Number presentation on FXS port' on FXO Port is programmed as 'Received Called Party' in SETU VFXTH installed at both the offices (Refer above figure).
- Now when Caller A calls SETU VFXTH installed at Baroda office, the call lands on the FXO Port and as per the routing mechanism programmed the call will be routed to the WAN Port.

- Through IP network, the call routes to the WAN Port of SETU VFXTH installed at Mumbai office. SETU VFXTH places the call on one of the FXS Port and sends 3001 in the CLI using CLI Number Presentation protocol.

- From FXS Port of SETU VFXTH, the call routes to the FXO Port of the PBX B. The FXO Port of PBX B receives 3001 in the CLI and routes the call to the desired extension using CLI Based Routing table.

Case 2:

- Suppose 'CLI Number presentation on FXS port' on SIP Trunk is programmed as 'Received Called Party' in SETU VFXTH installed at both the offices.

- When Caller B (2001) dials 3001, the call is routed to the FXS Port of SETU VFXTH installed at Baroda office as per the routing mechanism programmed in PBX A.

- SETU VFXTH routes the call to the WAN Port and through IP network it routes to the WAN Port of SETU VFXTH installed at Mumbai office.

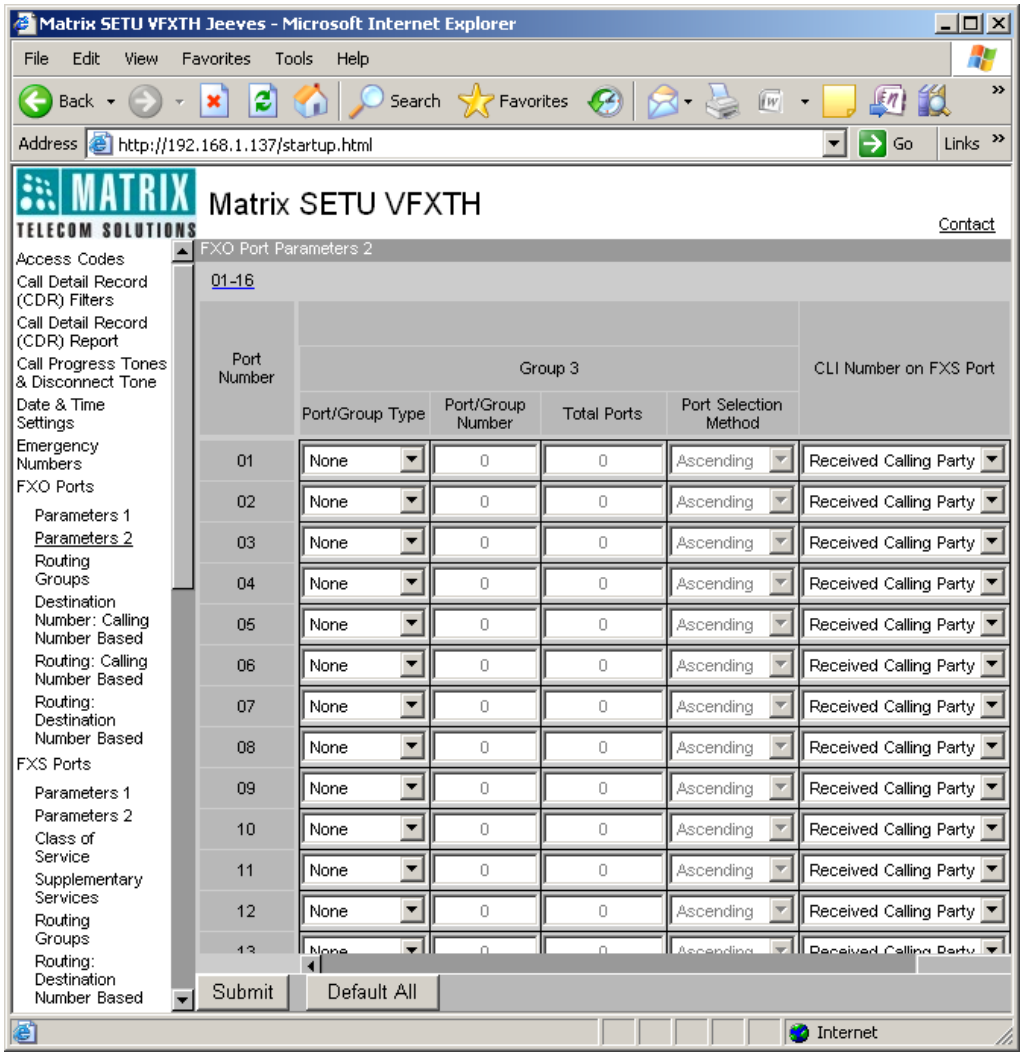
- At Mumbai office, CLI '3001' is received in the 'To' field of the Incoming call. As per the programming of the SETU VFXTH of Mumbai office, the call is routed to one of the FXS Ports of the PBX and send '3001' on the FXO port of PBX B.

- The PBX B on receiving '3001' on its FXO Port routes the call to the desired destination as per the CLI based routing table programmed in the PBX.

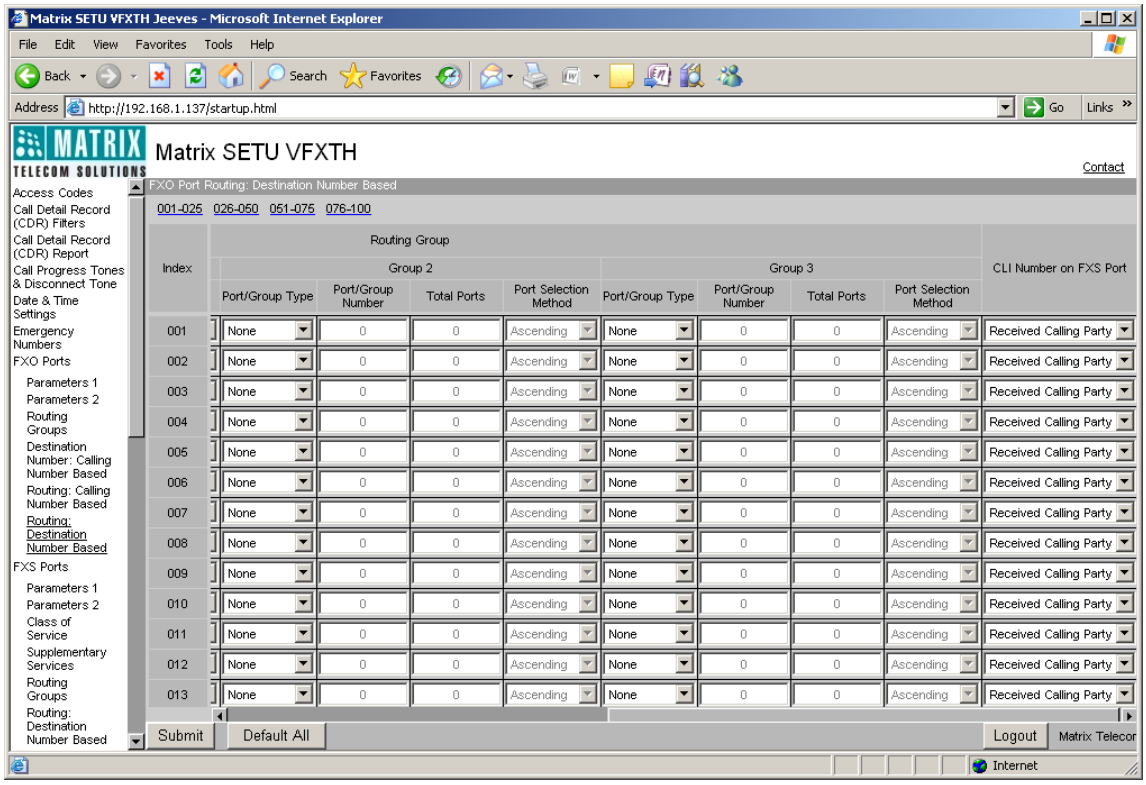
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

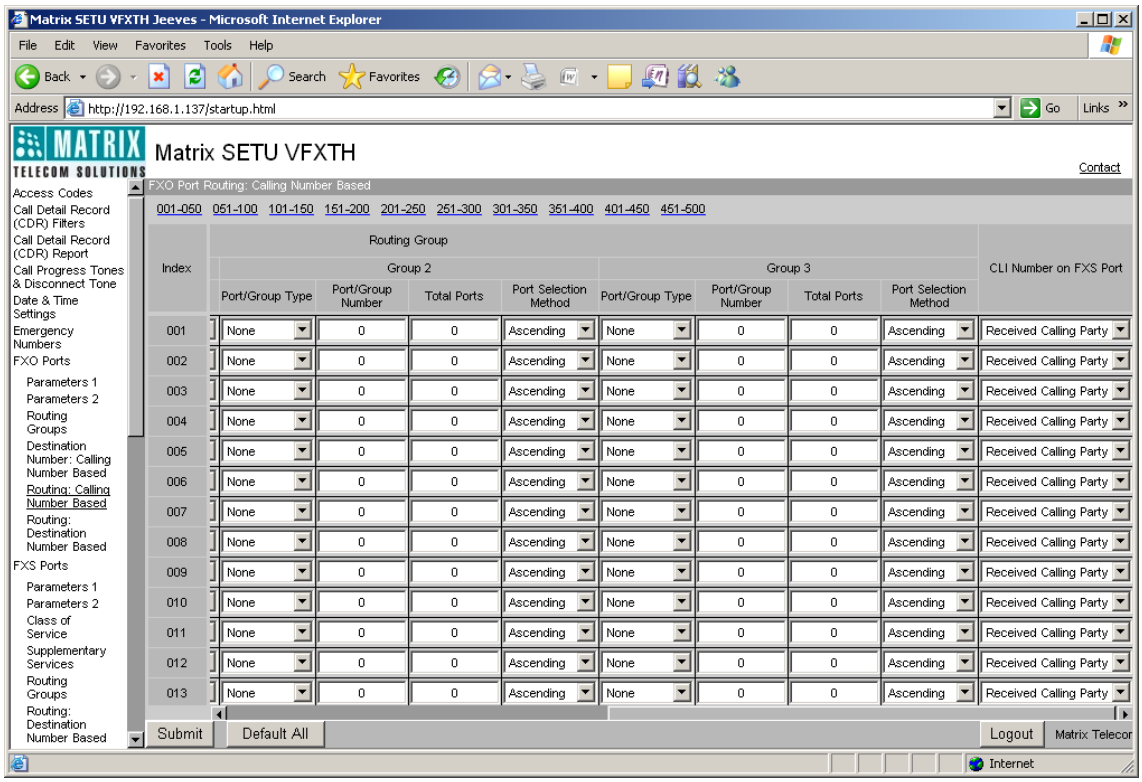
- Click on **'FXO Port Parameters 2'** link. Go to 'CLI Number on FXS Port' column. SETU VFXTH supports two types of CLI number presentation on FXS Port viz., 'Received Calling Party' and 'Received Called Party'. Select the desired type of number presentation. By default, 'Received Calling Party' is selected.



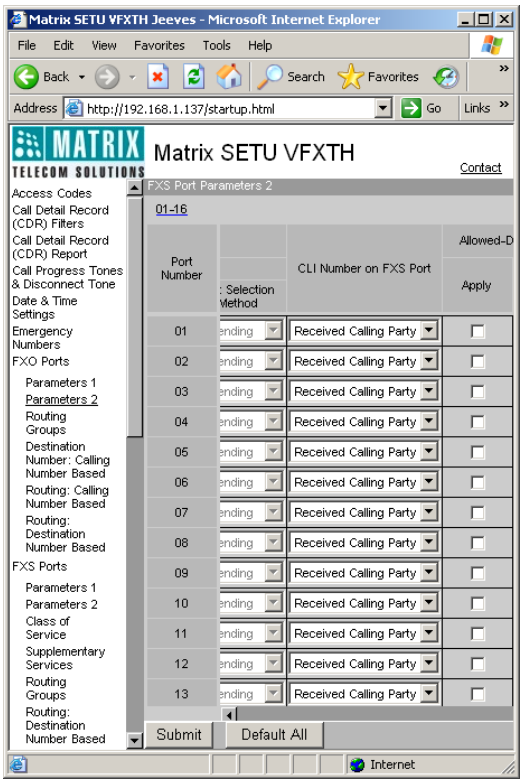
- Click on **'FXO Port Routing: Destination Number Based'** link. Select the desired type of CLI number presentation in 'CLI Number on FXS Port' column.



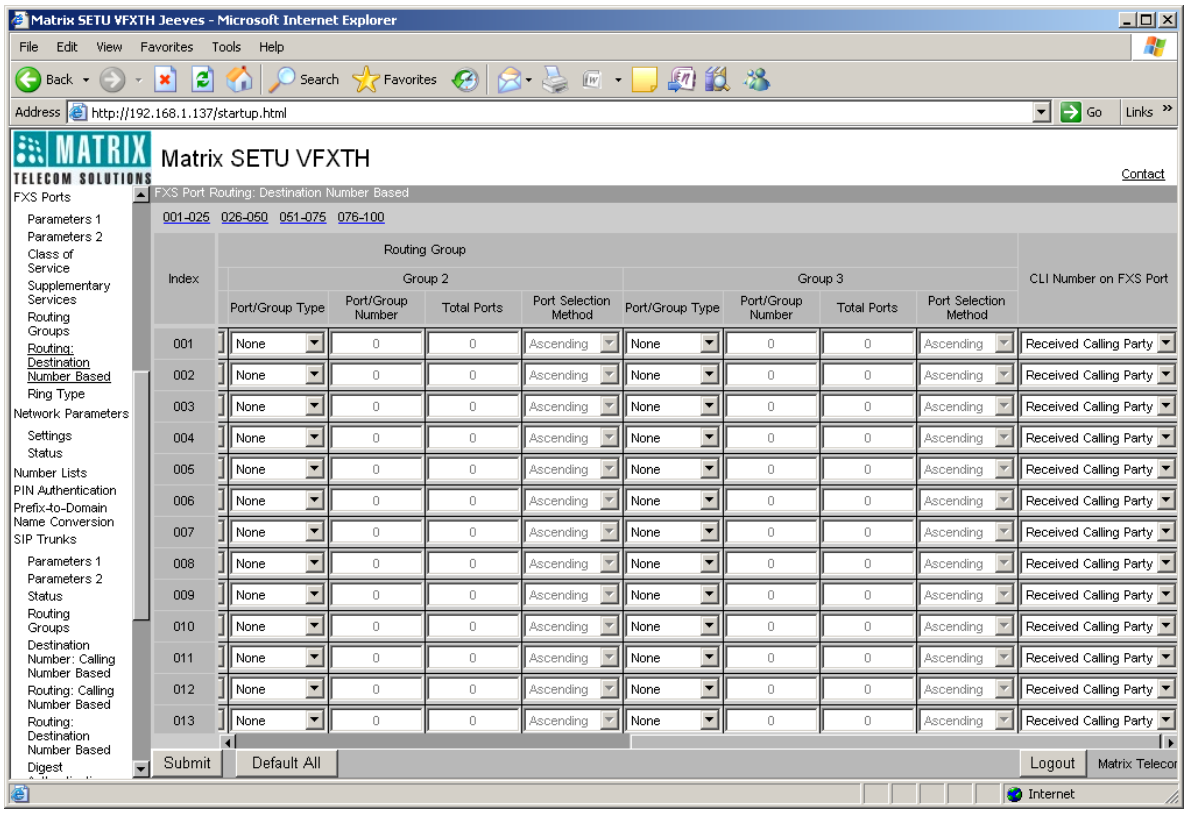
- Click on **'FXO Port Routing: Calling Number Based'** link. Select the desired type of CLI number presentation in the 'CLI Number on FXS Port' column.



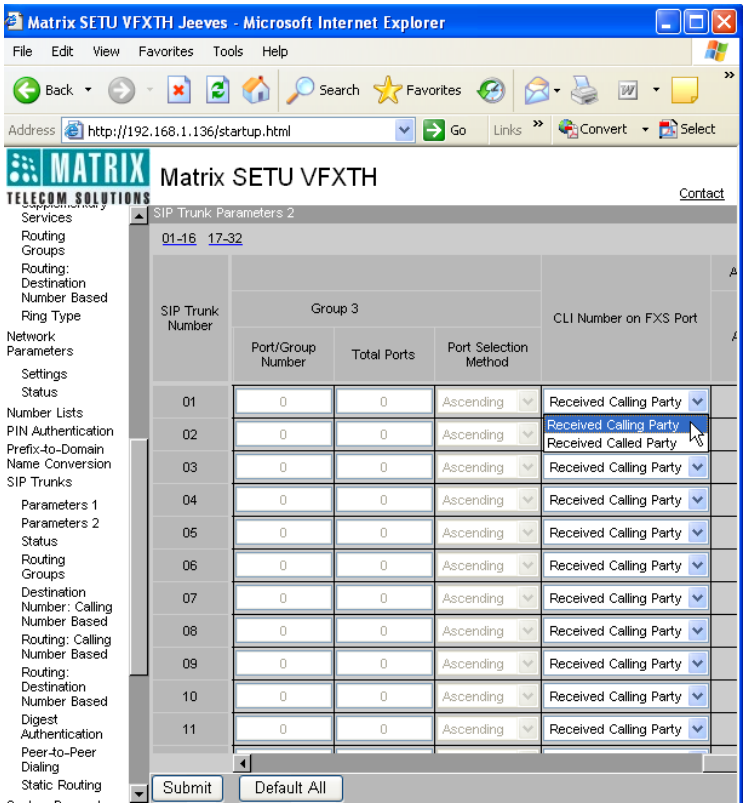
- Click on **'FXS Port Parameters 2'** link. Select the desired type of CLI number presentation in the 'CLI Number on FXS Port' column.



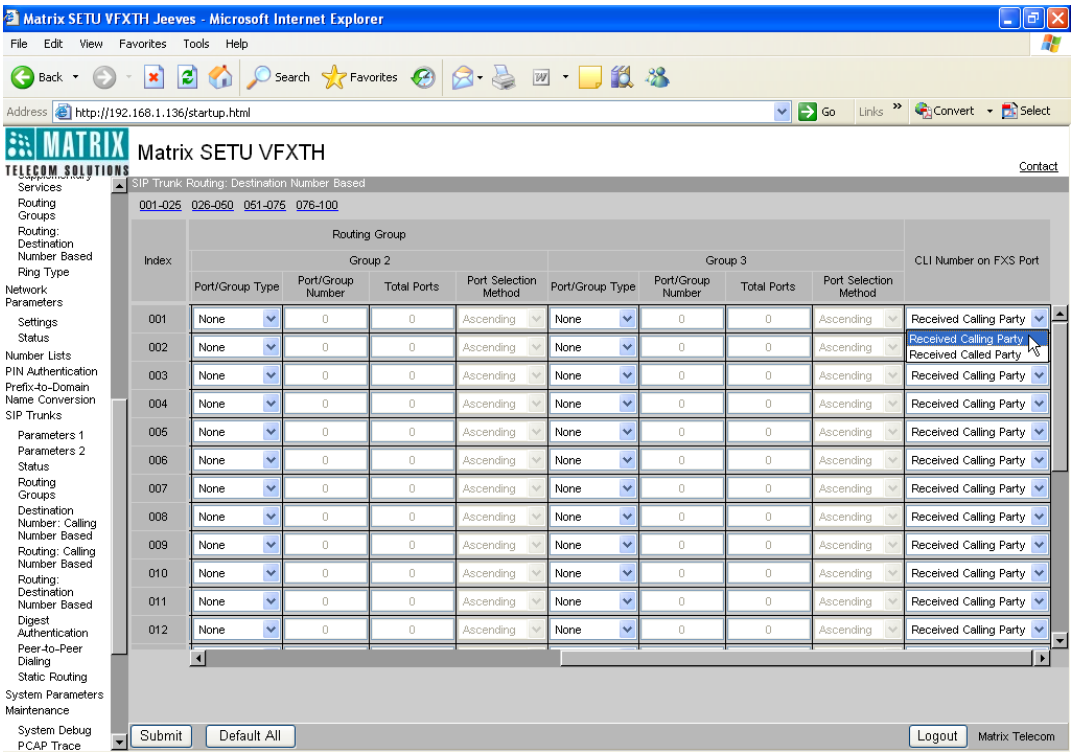
- Click on **'FXS Port Routing: Destination Number Based'** link. Select the desired type of CLI number presentation in the 'CLI Number on FXS Port' column.



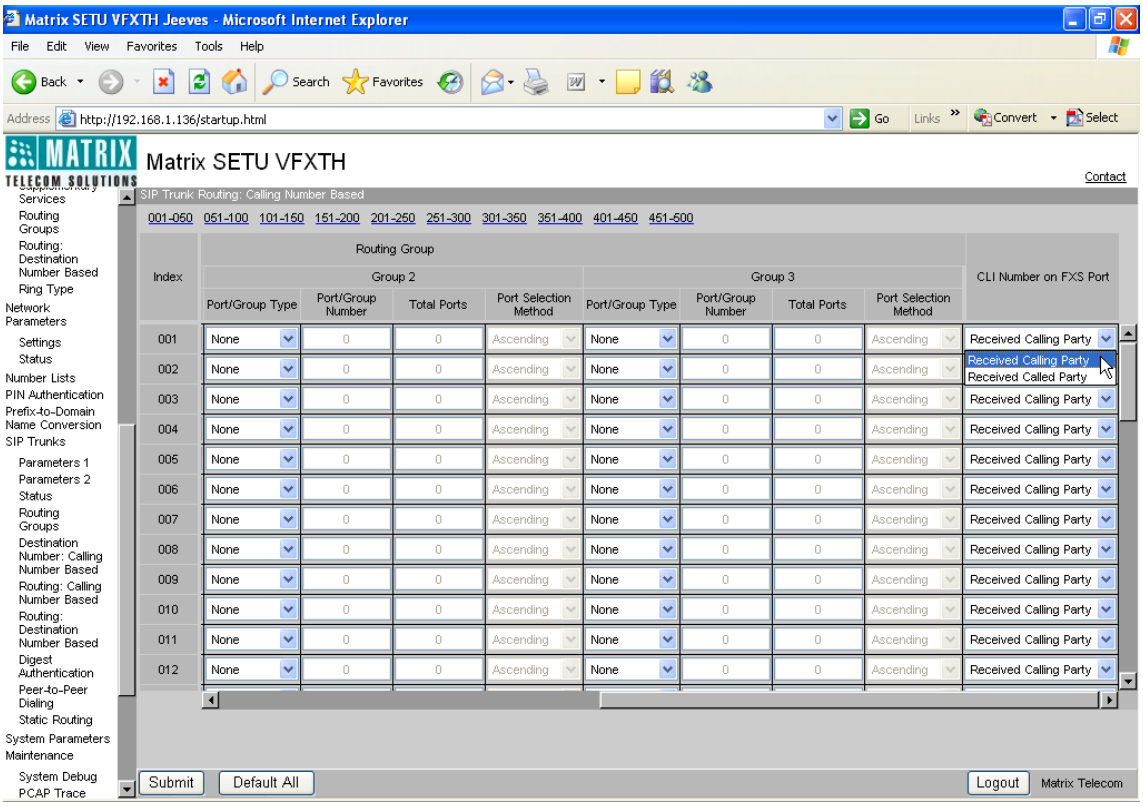
- Click on **'SIP Trunk Parameters 2'** link. Select the desired type of CLI number presentation in the 'CLI Number on FXS Port' column.



- Click on **'SIP Trunk Routing: Destination Number Based'** link. Select the desired type of CLI number presentation in the 'CLI Number on FXS Port' column.



- Click on **'SIP Trunk Routing: Calling Number Based'** link. Select the desired type of CLI number presentation in the 'CLI Number on FXS Port' column.



Relevant Topics:

1. “Call Processing” 47
2. “Port Parameters-FXO” 104
3. “Port Parameters-FXS” 112
4. “Port Parameters-SIP” 117
5. “Routing Mechanism on FXO” 143
6. “Routing Mechanism on FXS” 155
7. “Routing Mechanism on SIP” 161

PCAP Trace

PCAP or packet capture consists of intercepting and logging the traffic passing over a digital network or a part of a network. PCAP intercepts each packet in the data streams that flow across the network, and can decode and analyze its contents.

PCAP is used to monitor the network, analyze network problems, debug client/server communications, debug network protocol implementations, etc.

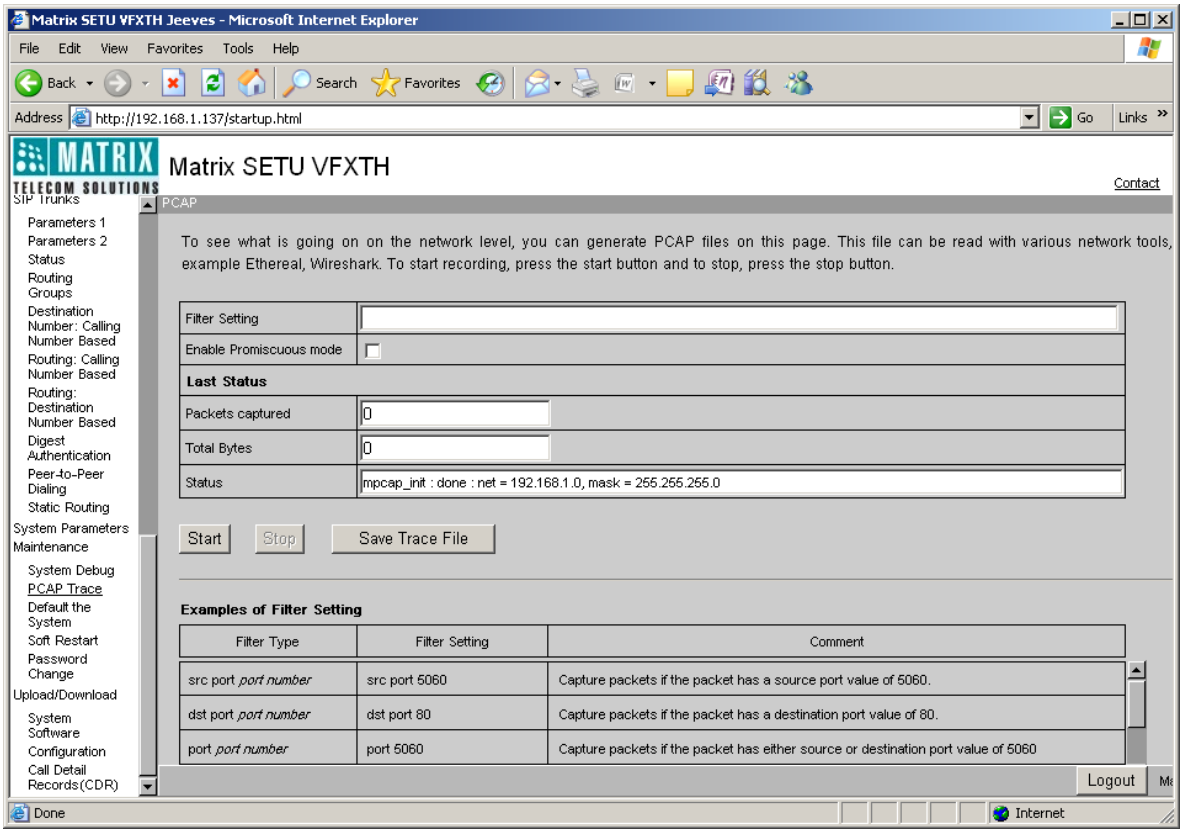
SETU VFXTH supports PCAP Trace, which you can use to detect and diagnose network related problems, for example, when the SIP account is not getting registered, or any SIP related feature is not functioning.

Packets traveling over a network are captured and saved in the SETU VFXTH. You can save these trace files (packets captured by the system) on a PC and open these trace files using a graphical packet capture and protocol analysis tool such as Wireshark or Ethereal.

A maximum of 1 MB of packets can be captured and stored in the SETU VFXTH. It also supports Filters and 'Promiscuous' mode for capturing packets, which you can use to specify the types of data packets to be captured.

How to use?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”).
- Click on the link **'PCAP Trace'** to open the page.



- **Filter Setting:** Decide the type of packets to be captured and set the Filter accordingly. The Filter Settings parameter should be maximum 60 characters in length; all ASCII characters are allowed. By default, this field is blank. So all packets will be captured.

Refer to the following examples to know how to set the Filters.

- To capture only SIP traces:
  - Filter Settings = port 5060, where, 5060 is the SIP Port number for which the traces are to be captured.
- To capture packets which are transmitted from the system i.e. from IP address 192.168.1.191:
  - Filter Settings = src 192.168.1.191
- To capture packets which are received for the system i.e. to IP address 92.168.1.191:
  - Filter Settings = dst 192.168.1.191
- To capture only packets which are transmitted from the system and received to the system i.e. IP address 192.168.1.191:
  - Filter Settings = src 192.168.1.191 or dst 192.168.1.191

- *Jeeves will throw up an alert message 'Invalid filter! Please enter valid filter' if you do not enter a valid filter.*
- *It is not mandatory to set Filters. When the Filter Settings field is left blank, the system will capture all packets.*

- **Enable Promiscuous Mode:** You can set the flag 'Enable Promiscuous Mode?' to Yes, if you want.

  When you enable Promiscuous mode, the SETU VFXTH will capture all network traffic. However, this will work only in a non-switched environment.

  When Promiscuous Mode flag is disabled, the system will capture only traffic that is directly related to it. Only traffic to, from or routed through the SETU VFXTH will be picked up by the PCAP Trace.

'Filter Settings' and 'Promiscuous Mode' (enabled) will not be cleared during power down.

- Press **'Start'** button to begin the capturing of the packets.
- Press **'Stop'** button to stop the packet capturing.

  OR

  Wait for the system to stop packet capturing. The SETU VFXTH stops packet capturing once the maximum allotted memory of 1 MB (RAM) is utilized.

  Number of Packets and bytes captured as per the filter setting will be displayed as 'Packets Captured' and 'Total Bytes'.

Capturing of packets will not stop if you open any other page of Jeeves. So, you may continue using Jeeves for any other purpose while PCAP Trace is being used.

- When the packet capturing is stopped (by you or the system), click on the 'Save Trace File' button to save the files on your PC or another PC.

  A dialog box will open. You can select the path for saving the trace file.

The current packets captured will not be deleted after you have saved the trace file. The current packets will be deleted when you start the PCAP capture again.

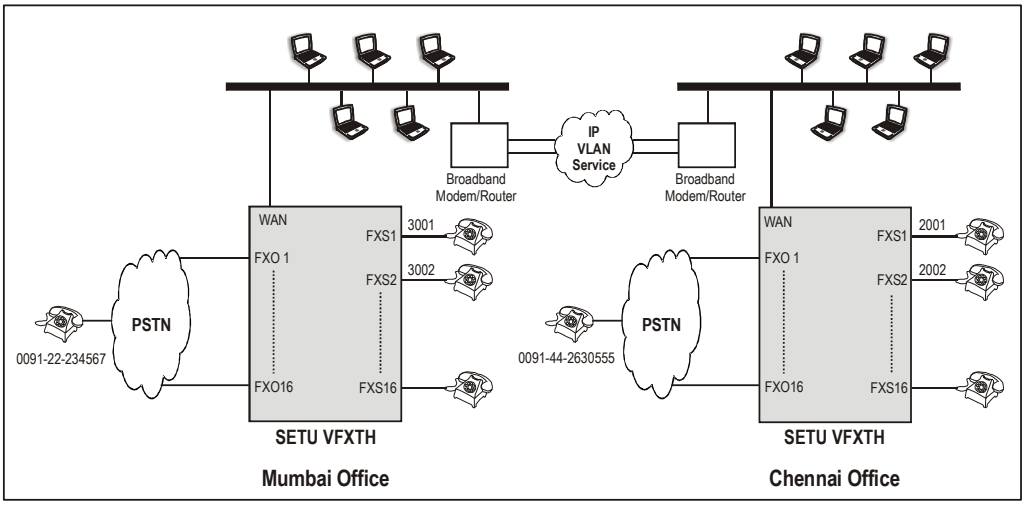
- You may log out of Jeeves.
- Now, you can open the trace files using Wireshark/Ethereal or any other similar software which supports opening of trace files.

Peer to Peer Calling

Making and receiving calls over IP network without intervention of Proxy (SIP) Server is called Peer to Peer Calling. As Peer to Peer call application does not require a SIP server, voice communication using this application is done virtually free of cost. The major cost saving offered by this application makes it a very attractive mode of inter-branch or intra-office voice communication.

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

- Suppose a Company has its Head-Office in Mumbai and Branch-Office in Chennai.
- Both the Offices are connected using VLAN services offered by ISP/ITSP.
- SE configures the network settings at both offices in such a way that peer to peer calls can be made between both the Offices.
- Suppose IP Address of SETU VFXTH installed at the Mumbai office is 192.16x.1.xx and of SETU VFXTH installed at the Chennai office is 192.16y.1.yy
- The extension number programmed at Mumbai office is 3001, 3002, 3003 and so on and the extension number programmed at Chennai office is 2001, 2002, 2003 and so on.



- Now if extension user 3001 at Mumbai office wants to call extension user 2002 at Chennai office, he should dial 2002@192.16y.1.yy or 2002@domain name and talk. This is how Peer to Peer calls are made using SETU VFXTH.

  Peer to Peer table has total 500 entries therefore 500 numbers can be configured for peer to peer call application. First entry of Peer to Peer table is used for no match found case.

  Each entry in peer to peer table consists of Destination Number, Destination Address, Minimum Digit, Maximum Digit and Name.

- **Destination Number:** Destination Number is the extension number of the remote called party.
- **Destination Address:** Destination Address corresponds to the IP Address or the domain name of SETU VFXTH to which the call has to be made for the number programmed.

- **Minimum Digit:** Minimum Digit defines the minimum number of digits required to be dialed for routing the call to the desired destination. Minimum length of number dialed should be less than or equal to Maximum Digit programmed.

- **Maximum Digit:** Maximum Digit defines the maximum number of digits accepted by SETU VFXTH to route the call to the desired destination. If the destination number dialed by the user is greater than the maximum number of digits allowed to be dialed through that port then SETU VFXTH shall strip off the extra digits dialed.

- **Name:** Name field is used to program the name of the contact person to whom the call is made. This name is just for information purpose and is not used anywhere in the system.

SETU VFXTH uses Best Match Found logic to compare the destination number to be out dialed with the entries programmed in Peer to Peer table. Number string dialed by the user should fulfill the Minimum and Maximum Digit criteria programmed for that destination number.

- If destination number dialed out does not match with the entries programmed in Peer to Peer table then no match found entry criteria for Minimum and Maximum Digit should be verified.

- If best match entry is found for the destination number dialed but it does not fulfill the criteria of Minimum Digit programmed for that entry then the system will give error tone to the user.

- If best match entry is found for the destination number dialed but the destination number string is greater than Maximum Digit programmed for that entry then the system will strip off the extra digits dialed and use the destination address for routing the call.

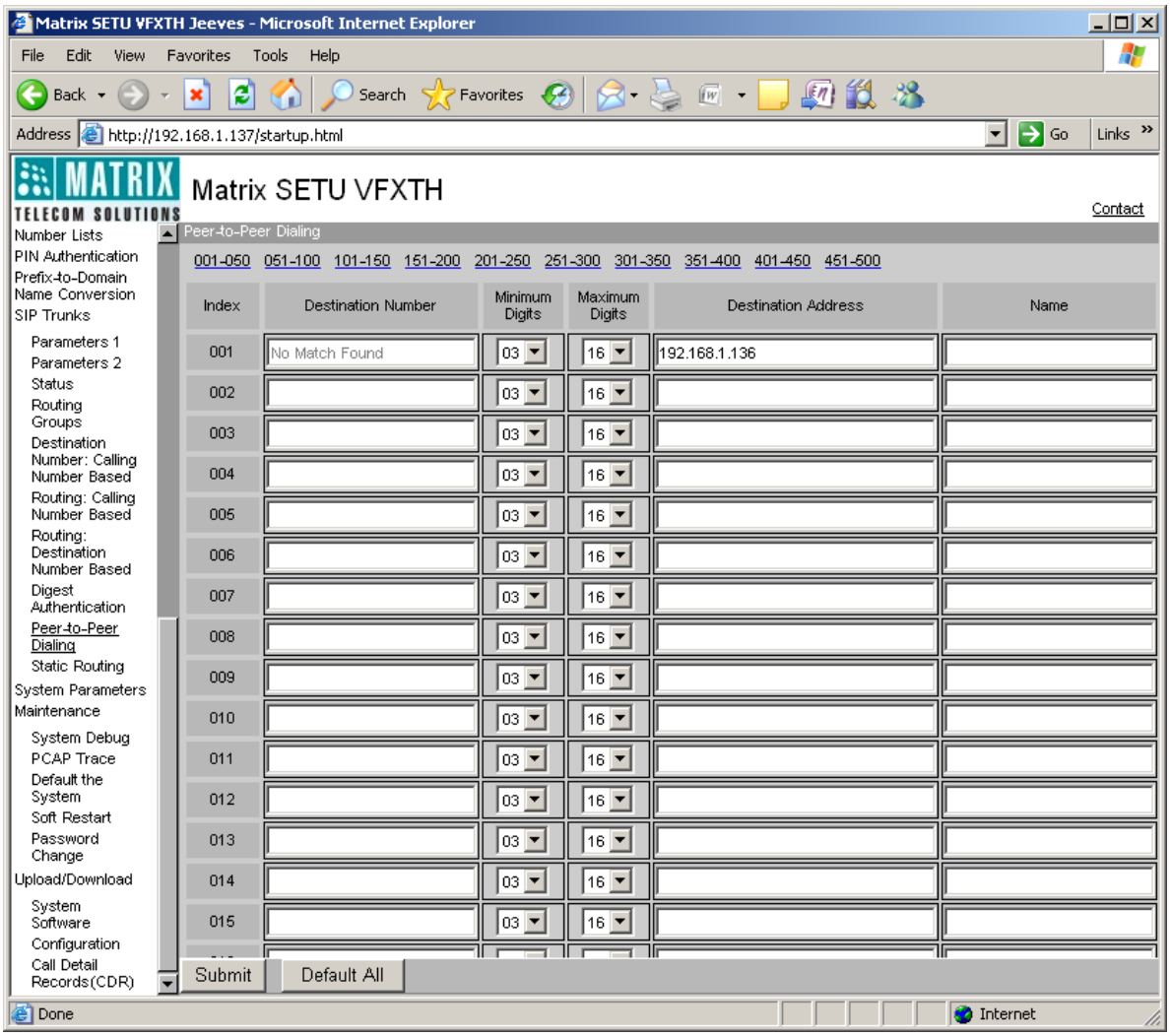
- If destination number that is to be out-dialed matches with the multiple entries in the peer to peer table then the system should dial out the number for which the Maximum and Minimum Digit programmed best matches the dialed number.

- Thus Minimum and Maximum Digit programmed for each entry removes the possibility of conflict numbers to a great extent in Peer to Peer calls application.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Peer to Peer Dialing'** link. Program the following parameters:



- **Index:** Index number shows the number of entry at which destination number and destination address for making peer to peer calls is stored. Index number 001 is used for no match found case.
- **Destination Number:** Destination number can be of maximum 24 digits. Digits 0 to 9, # and * is allowed in destination number field. By default all entries are blank.
- **Minimum Digit:** Program Minimum Digits to be dialed in this field. Minimum length of dialed digits should be in the range of 01 to 24. By default, it is 3 for all entries.
- **Maximum Length:** Program Maximum Digits to be dialed in this field. Maximum length of digits to be dialed should be in the range of 01 to 24. By default, it is 16 for all entries.
- **Destination Address:** Destination Address can be of maximum 40 characters. All ASCII characters are allowed. By default, it is blank for all entries.
- **Name:** Name of the called person is stored in this field. Name can be of maximum 12 characters. All ASCII characters are allowed.

a. *Peer to Peer table is used only when destination port is SIP Trunk.*

b. *When Peer to Peer call lands on SETU VFXTH, it will route that incoming call using the first SIP Trunk whose SIP ID is programmed as '*'.*

c. *Using SETU VFXTH, Peer-to-Peer calls can be made between any two clients when:*
   - *Both the clients are in Public Network.*
   - *Both the clients are in same Private Network and make calls only in private network (STUN is disabled).*
   - *Both the clients are in different Private Networks and both the private networks do not use symmetric NAT (STUN is compulsory to use by both the clients).*
   - *One client is in Public network (Using Symmetric RTP is compulsory) and other client is in Private network.*

d. *Using SETU VFXTH, Peer-to-Peer calls cannot be made:*

   - *Between the clients in different Private Networks with both the private networks using routers with Symmetric NAT type.*
   - *Between clients in different Private Network and public network simultaneously*
   - *Between clients in same Private Network and public network simultaneously*
   - *Between clients in same private networks, different private network and public network simultaneously.*

Relevant Topics:

1. “Call Processing” 47
2. “Network Port Parameters” 72
3. “Port Parameters-SIP” 117
4. “Routing Mechanism on SIP” 161

PIN Authentication

PIN Authentication is used to authenticate the caller to prove his identity before the call is being processed by SETU VFXTH to avoid the possibility of malicious calls and to avoid misuse of its services.

PIN Authentication is applicable only on source port. It is applicable on FXO Ports only. This feature is useful only when Destination Number Determination Method is Manual Dial. To use this feature, PIN Number and PIN Password should be programmed in PIN Authentication table. PIN Authentication table supports 500 entries.

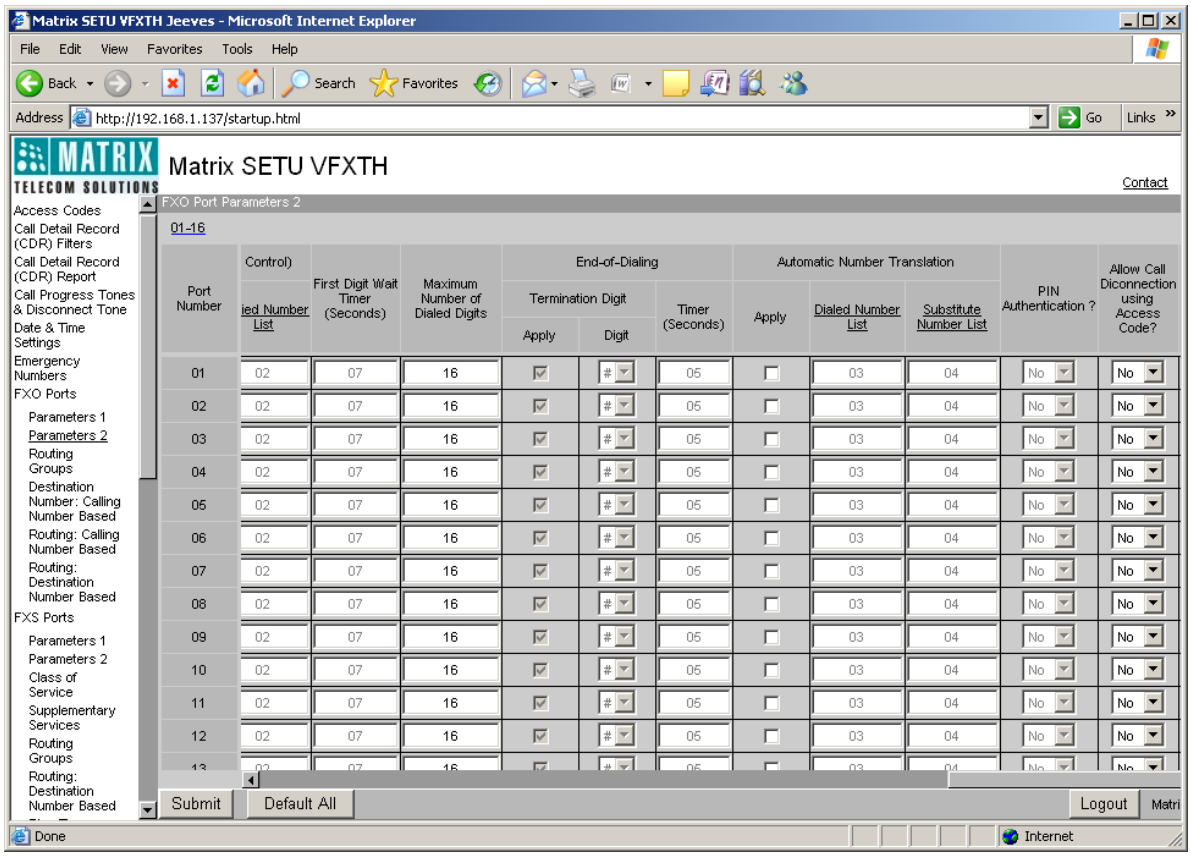
Let us understand PIN Authentication with the help of an example:

- Suppose a company has its head quarter in New York and branch office in Mumbai. Employees in Mumbai frequently calls headquarter in New York.
- To minimize the communication cost between head office and branch office, company installed SETU VFXTH at both the places.
- SETU VFXTH allows everyone to call its FXO Port and make an IP call to New York. Company employees also make calls to New York office using this service. However, a possibility of misusing this service arises.
- To avoid this problem, some type of authentication is required to prove identity of the user before services being provided by the system.
- This requirement is met using PIN Authentication feature. PIN Authentication process involves following steps:
    a. Employees of the company calls FXO Port of SETU VFXTH to make calls to its head office in New York.
    b. As soon as call lands on FXO Port, it will check for Destination Number Determination Method.
    c. If Destination Number Determination Method selected for FXO Port is 'Manual Dial' then SETU VFXTH will play prompt tone for PIN Number and PIN Password.
    d. SETU VFXTH will allow user to dial a number only if valid PIN Number and PIN Password is entered by the user.
- Whenever a call is made using SETU VFXTH, it processes the call only after the caller is being authenticated.

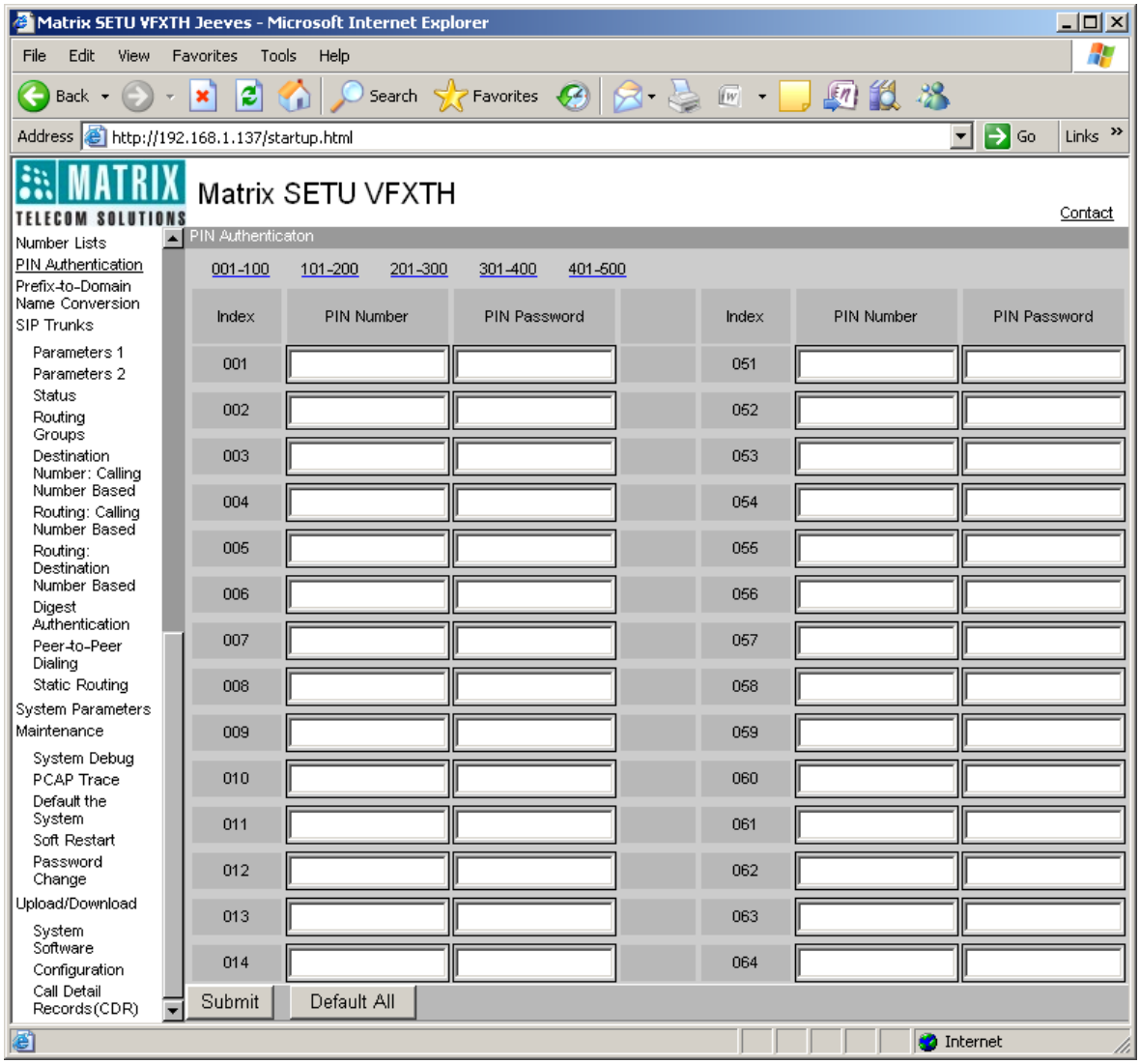
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'FXO Port Parameters 2'** link. Select 'Yes' to enable PIN Authentication on the desired FXO Ports.



- Click on **'PIN Authentication'** link and program PIN Number and PIN Password for each caller who is authorized to use services of SETU VFXTH in PIN Authentication table.



- **PIN Number:** PIN Number can be of maximum 4 digits. Only digits 0 to 9 allowed. By default, all entries are blank.

- **PIN Password:** PIN Password can be of maximum 4 digits. Only digits 0 to 9 allowed. By default, all entries are blank.

a. *If PIN Authentication is enabled, user will get prompt tone instead of dial tone or feature tone while going Off-hook to make a call.*

b. *User is given maximum three attempts to enter valid PIN Number and PIN Password. If invalid PIN Number and Password is entered for three times then the call gets disconnected.*

Relevant Topics:

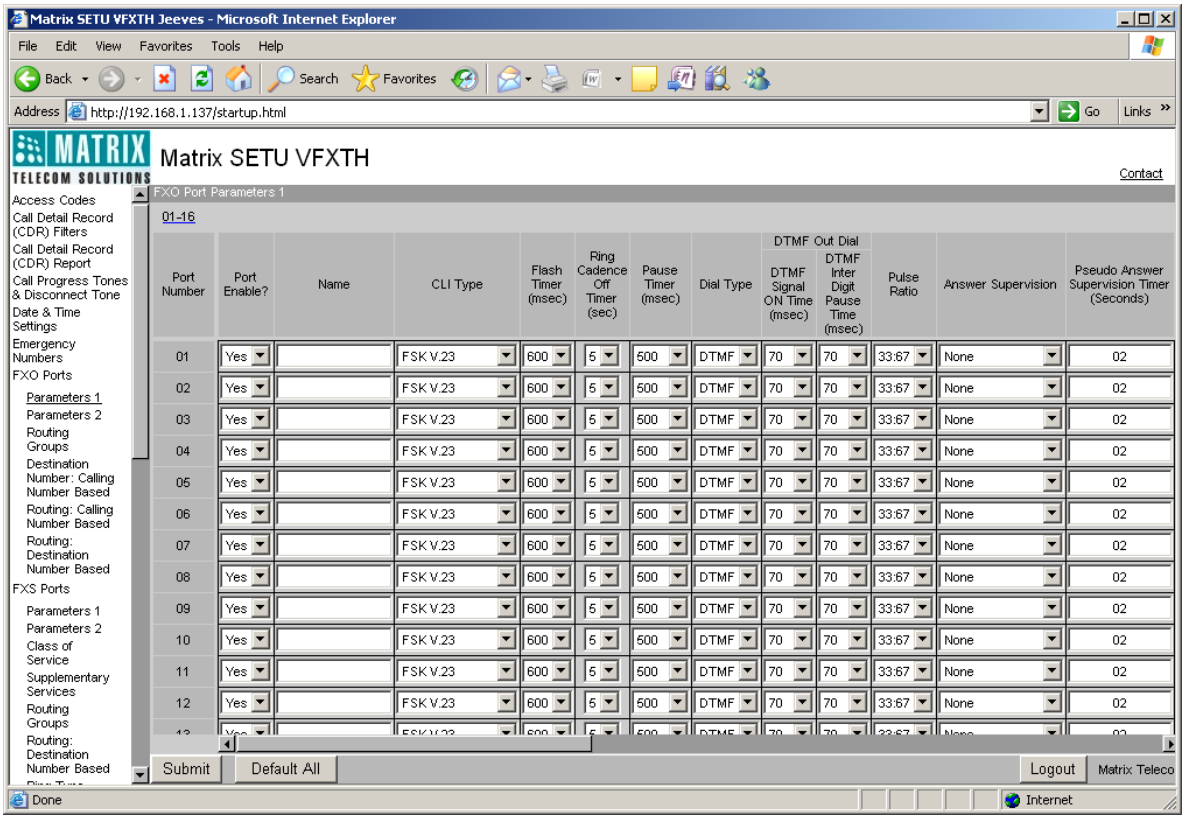
1. “Call Processing” 47
2. “Call Progress Tones” 48
3. “Digest Authentication” 63
4. “Port Parameters-FXO” 104
5. “Routing Mechanism on FXO” 143

Port Parameters-FXO

SETU VFXTH1616 supports maximum 16 FXO Ports. SETU VFXTH1616 shall route the call to and from FXO Port only if that port is enabled.

To program various FXO Port Parameters, open Web Jeeves of SETU VFXTH1616. (Refer “Accessing Web JEEVES”)

- Click on **'FXO Port Parameters 1'** link and program all the parameters on the page as shown below:



- **Port Enable?:** You can enable or disable FXO Ports as per your requirement. Select 'No' to disable FXO Port if you do not want to use it for routing the calls or there is a problem in exchange, hardware failure, etc.

  By default, 'Yes' is selected for all FXO Ports i.e. all FXO Ports are enabled.

- **Name:** You can assign name to each FXO Port. Name assigned to FXO Port is relevant only for the identification of that Port. However, if during an incoming call, name is not received in the CLI of FXO Port then the name programmed in this field is used:

  - As display name in the 'From' field of the INVITE message when the call is routed through the SIP Trunk.

  - For displaying the name on CLI of the FXS Port.

  Name can be of maximum 12 characters. In name field, following ASCII characters are supported:

- 0 to 9
- A to Z
- a to z
- ! (exclamation mark)
- % (percentage)
- ' (single quote)
- * (asterisk)
- - (minus or dash)
- + (plus)
- . (dot)

By default, name field is blank for all FXO Ports.

- **Calling Line Identification (CLI) Type:** Calling Line Identification means detecting calling party's number, sent by exchange, on the FXO Port of SETU VFXTH1616.

  Different type of signaling is used, by different exchanges of different countries, for sending the calling party number. Select CLI type supported by your exchange in this field.

  SETU VFXTH1616 is capable of detecting following type of signaling on its FXO Port:

  - None
  - DTMF-India
  - DTMF-ETSI
  - DTMF-Denmark
  - DTMF-Brazil
  - FSK V.23
  - FSK Bellcore

  By default, 'FSK V.23' type of CLI is selected for all FXO Ports.

  CLI type 'None' is selected when the exchange does not support any type of CLI or user does not want to detect the CLI. Detecting CLI takes time and thus the routing of call gets delayed by 2-3 seconds, this delay can be avoided by using CLI type = 'None'.

- **Flash Timer (msec):** SETU VFXTH1616 uses flash timer to generate Flash on FXO Port for using various supplementary features viz. Call Hold, Call Transfer etc. Flash Timer signifies the time period for which the loop current breaks.

  Select Flash Timer as per you requirement in this field. Valid options are:

  - 83
  - 100
  - 200
  - 300
  - 400
  - 500
  - 600
  - 700
  - 800
  - 900

  By default, Flash Timer is 600 msec for all FXO Ports.

- **Ring Cadence Off Timer:** Program Ring Cadence Off Timer in this field. Range of this timer is 1 to 9 seconds. By default, it is set to 5 seconds.

- **Pause Timer :** Pause Timer is the time period for which SETU VFXTH1616 waits before dialing out a number on the FXO Port. Some exchange takes time to detect Off-hook event on the FXO Port but SETU VFXTH1616 dials the number immediately after going Off-hook. This may result in dialing of a wrong number, as the exchange might miss the initial digits of a number dialed by the system. To avoid this situation, SETU VFXTH1616 supports programming of Pause Timer.

  Select Pause Timer for each FXO Port in this field. Valid options are:

  - 500
  - 1000
  - 1500
  - 2000
  - 2500

  By default, Pause Timer is 500 msec for all FXO Ports.

- **Dial Type:** Select a method to out-dial a number on FXO Port. SETU VFXTH1616 supports following methods:

  - ***Pulse:*** If Pulse is selected then ratio of make/break depends on the pulse ratio programmed for the FXO Port.

  - ***DTMF:*** If DTMF is selected then DTMF On/Off time depends on the DTMF dial timer programmed for the FXO Port.

  By default, 'DTMF' Dial Type is selected for each FXO Port.

- **DTMF Outdial Timer:** Select DTMF Outdial Timer if dial type selected for FXO Port is 'DTMF'.

  - ***DTMF Signal ON Time:*** This timer signifies the time period for which the DTMF digit will remain ON while dialing the DTMF number on the FXO Port. DTMF Signal ON Timer range is from 50 msec to 200 msec in steps of 10 upto 200. By default, it is set to 70 msec.

  - ***DTMF Inter-Digit Pause Time:*** This timer signifies the pause time between two subsequent DTMF digits. DTMF Inter-Digit Pause Timer range is from 50 msec to 200 msec in steps of 10 upto 200. By default, it is set to 70 msec.

- **Pulse Ratio:** Select Pulse Ratio if Dial type selected for FXO Port is 'Pulse'. Different country has different pulse ratio. SETU VFXTH1616 supports following types of Pulse Ratios:

  - 40:60
  - 50:50
  - 33:67

  By default, Pulse Ratio set for each FXO Port is 33:67.

- **Answer Supervision:** Answer Supervision is a signal which is provided from the CO to the calling party, which indicates that the called party has answered and the call is matured.

Answer Supervision on FXO Port can be done by using following methods:

- ***None:*** This option is selected when no signaling is available from the CO. Call gets matured on expiry of 'Pseudo Answer Supervision Timer' which is programmed as 20 seconds by default irrespective of whether the call actually gets matured or not. After expiry of this Timer, call duration timer will start and SETU VFXTH1616 will start detecting Disconnect Supervision signal programmed for that port.

- ***Battery Reversal:*** Battery Reversal is a maturity signal generated by the PSTN by reversing the battery polarity. For example, in dialing state if battery polarity is negative for TIP and positive for RING then after maturity of call TIP will become positive and RING will become negative. After maturity of a call, call duration timer will start and SETU VFXTH1616 will start detecting Disconnect Supervision signal programmed for that port.

By default, Answer Supervision on FXO Port is set to 'None'.

- **Pseudo Answer Supervision Timer (seconds):** Pseudo Answer Supervision Timer (seconds) is applicable when Answer Supervision on FXO Port is set to 'None'. In this case, call matures on expiry of 'Pseudo Answer Supervision Timer' irrespective of whether the call actually gets matured or not. Select Pseudo Answer Supervision Timer in this field. Range of this Timer is 01 to 99 seconds.

  By default, Pseudo Answer Supervision Timer is programmed as 20 seconds for all FXO Ports.
- **Disconnect Supervision:** Disconnect supervision is the signal given by the CO to detect far end disconnection in case of outgoing as well as incoming call. Disconnect Supervision on FXO Port can be provided by CO in one of the following forms:

  - ***None:*** This option is selected when no signaling is available from the CO. If this option is selected:

    - In case of outgoing calls, FXO Port will be released only if the calling party disconnects or the disconnect tone is detected on FXO Port when the called party disconnects the call.

    - In case of incoming calls, FXO Port will be released only if the called party disconnects or if the disconnect tone is detected on the FXO Port when the calling party disconnects the call.

  - ***Battery Reversal:*** Battery Reversal is a signal generated by the PSTN by reversing the battery polarity to indicate call disconnection in case of incoming as well as outgoing calls. For example, if in speech condition the battery polarity is negative for TIP and positive for RING then after call disconnection battery polarity for TIP becomes positive and RING becomes negative. When FXO Port goes on-hook, idle state polarity is restored.

  - ***Open Loop Disconnect:*** Open Loop Disconnect is a pulse which is generated by the CO on removing the battery for open loop disconnect timer and again connects it. The duration for Open Loop Disconnect pulse varies from one service provider to other and countrywise. If Open Loop signal is detected for the time less than the Open Loop Disconnect timer programmed then it will not be considered as valid open loop signal for releasing the port.

  By default, Disconnect Supervision on FXO Port is set to 'Open Loop Disconnect'.

- **Open Loop Disconnect Timer:** Open Loop Disconnect Timer is applicable when Disconnect Supervision method selected is 'Open Loop Disconnect'. System breaks the loop current for the time period programmed as Open Loop Disconnect Timer. The range of Open Loop Disconnect timer is 001 to 999 misc.

  By default, Open Loop Disconnect Timer is 200 msec.

- **Disconnect Tone Detection**
  (Refer "Call Disconnect Tone" for details)

- **Disconnect Tone to be detected**
  (Refer "Call Disconnect Tone" for details)

- **AC Termination Impedance:** Select AC Termination Impedance for FXO Port in this field. SETU VFXTH1616 supports following options:

  - 600 Ω
  - 900 Ω
  - 270 Ω + (750 Ω || 150 nF) and 275 Ω + (780 Ω || 150 nF)
  - 220 Ω + (820 Ω || 120 nF) and 220 Ω + (820 Ω || 115 nF)
  - 370 Ω + (620 Ω || 310 nF)
  - 320 Ω + (1050 Ω || 230 nF)
  - 370 Ω + (820 Ω || 110 nF)
  - 275 Ω + (780 Ω || 115 nF)
  - 120 Ω + (820 Ω || 110 nF)
  - 350 Ω + (1000 Ω || 210 nF)
  - 200 Ω + (680 Ω || 100 nF)
  - 600 Ω + 2.16 µF
  - 900 Ω + 1 µF
  - 900 Ω + 2.16 µF
  - 600 Ω + 1 µF
  - Global complex impedance

  By default, AC Termination Impedance is 600Ω.

- **Rx Gain (db):** Select Rx Gain for FXO Port in this field. SETU VFXTH1616 enables you to adjust receive gain of the FXO Port to increase the audibility of incoming speech. Range of Rx Gain is from +10dB to -15dB.

  By default, Rx Gain for FXO Port is 0.

- **Tx Gain (db):** Select Tx Gain for FXO Port in this field. SETU VFXTH1616 enables SE to adjust the transmit gain of the FXO Port to adjust the right intensity for the transmitted signal. Range of Tx Gain is from +10dB to -15dB.

  By default, Tx Gain for FXO Port is 0.

- **On-Hook Speed (msec):** Select On-Hook Speed in this field. On-Hook Speed sets the time for the line-side device (DAA) to go on-hook. The On-Hook Speed is measured, from the time the on-hook bit is cleared, until the loop current equals zero. SETU VFXTH1616 supports following options:

  - <0.5 msec
  - 3 msec (ETSI)
  - 26 msec (Australia)

  By default, On-Hook speed is <0.5msec.

- **Off-Hook Speed (msec):** Select Off-Hook Speed in this field. Off-Hook Speed decides the time to settle the line transients after which transmission or reception can occur. Total Off -Hook speed

includes off hook Counter + 17 msec (resistor calibration) + 256 msec (ADC calibration) + 1.5 msec (FIR filter delay). All other parameters are fixed for determination of Off-Hook speed except Off-Hook counter. SETU VFXTH1616 supports following options:

- 512 msec
- 128 msec
- 64 msec
- 8 msec

By default, Off-Hook Speed is 8 msec.

- **Current Limiting Mode:** Current Limiting mode is used to limit the loop current. Enable current limiting mode to limit the loop current in SETU VFXTH1616. It will limit the loop current to maximum 60mA.

  By default, it is disabled.

- **Minimum Loop Current (mA):** Select Minimum Loop Current in this field. Minimum Loop Current helps to set the minimum loop current at which DAA operates. SETU VFXTH1616 supports following options:

  - 10 mA
  - 12 mA
  - 14 mA
  - 16 mA

  By default, minimum loop current is 10 mA.

- **TIP/ Ring Voltage (Volts) Adjustment:** Select TIP/Ring Voltage in this field. It is used to adjust TIP/ RING Voltage on the line side. SETU VFXTH1616 supports following options for TIP/Ring Voltage Adjustment:

  - 3.1 V
  - 3.2 V
  - 3.35 V
  - 3.5 V

  By default, TIP RING Voltage is 3.5 V.

- **Ringer Impedance:** Select Ringer Impedance in this field. You may select either 'High' or 'Synthesized'. Some countries like Poland, South Africa and Slovenia requires high ring impedance which is achieved by the DAA module by setting ringer impedance to 'Synthesized'.

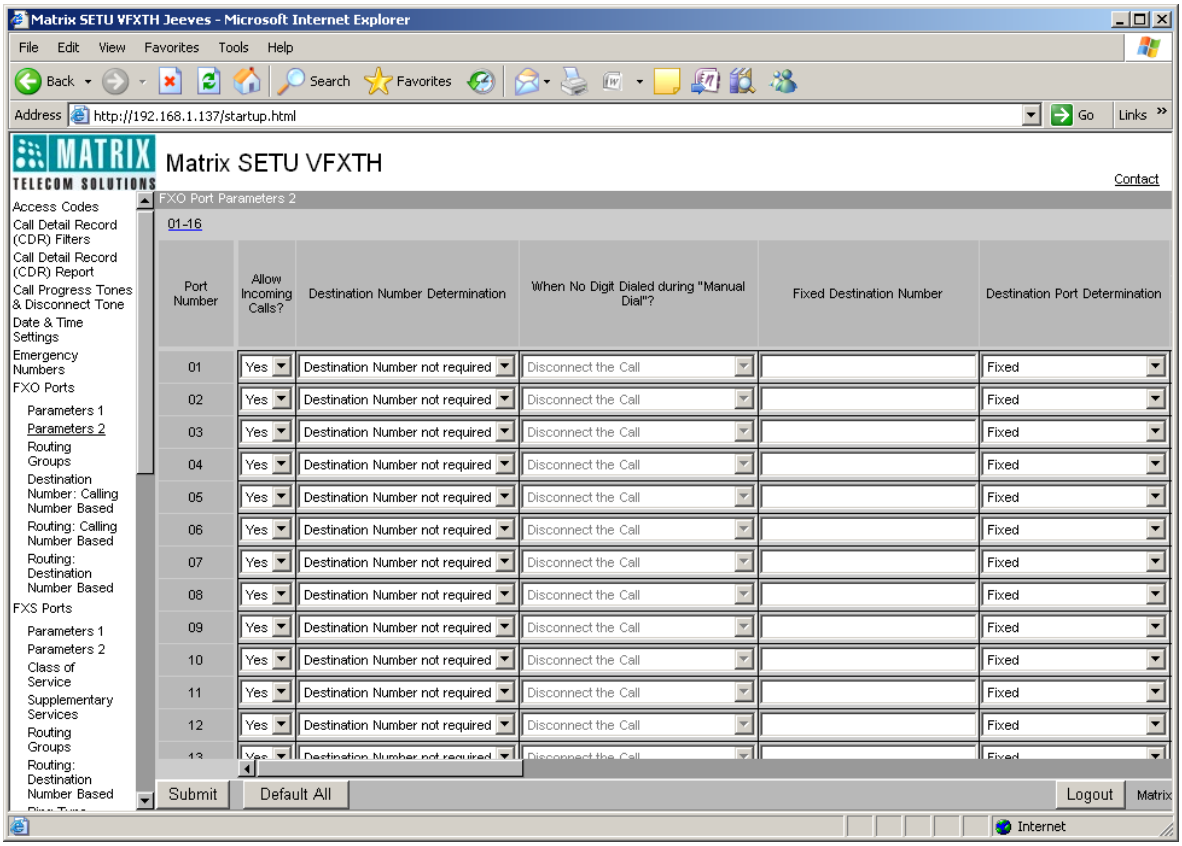
  By default, Ringer Impedance is set to 'High'.

- **Ringer Threshold (Vrms):** Ringer Threshold is used to decide the level at which SETU VFXTH1616 will generate the ring event. Select the range for Ringer Threshold (Vrms) in this field. SETU VFXTH1616 supports following range for ring generation:

  - 13.5 - 16.5 Vrms
  - 19.35 - 23.65 Vrms
  - 40.5 - 49.5 Vrms

  By default, Ringer Threshold is 13.5-16.5 Vrms.

- Click on **'FXO Port Parameters 2'** link and program following parameters:



- **Allow Incoming Calls?:** This field is applicable only for incoming calls on FXO Port. Select 'No' in this field to prohibit incoming calls on FXO Port. If this field is set to 'Yes', the call originated on FXO Port is routed to the destination port as per the routing mechanism programmed for that FXO Port.

  By default, this field is set to 'Yes'.

- **Destination Number Determination**
  (Please refer "Routing Mechanism on FXO" for more details)

- **When No Digit Dialed during "Manual Dial"?**
  (Please refer "Routing Mechanism on FXO" for more details)

- **Fixed Destination Number**
  (Please refer "Routing Mechanism on FXO" for more details)

- **Destination Port Determination**
  (Please refer "Routing Mechanism on FXO" for more details)

- **Routing Groups**
  (Please refer "Routing Group" for more details)

- **CLI number on FXS Port**
  (Please refer "Number Presentation on FXS Port" for more details)

- **Allowed- Denied Numbers (Toll Control)**
  (Please refer “Allowed-Denied Numbers” for more details)

- **First Digit Wait Timer (seconds):** Program First Digit Wait Timer for FXO Port in this field. Range for First Digit Wait Timer is 01 to 99 seconds. By default, it is 07 seconds.

  (Please refer “Number Collection Method” for more details)

- **Maximum Number of Dialed Digits:** Enter the maximum number of digits allowed to be dialed out on FXO Port in this field. Maximum number of dialed digits is used to detect end-of-dialing of a number on FXO Port. Range of this field is 01 to 24 characters. By default, it is 16 digits.

  (Please refer “Number Collection Method” for more details)

- **End-of-Dialing:** Untick 'Apply' flag to dis-allow End-of-Dialing using termination digit in End-of-Dialing table. By default, it is ticked. Valid Termination Digits are # and *. By default, # is selected.

  Also program Inter Digit Wait Timer in 'Timer (seconds)' field. Range of Inter Digit Wait Timer is 01 to 99 seconds. By default, it is 05 seconds.

  (Please refer “Number Collection Method” for more details)

- **Automatic Number Translation**
  (Please refer “Automatic Number Translation” for more details)

- **PIN Authentication:** Enable this field if PIN Authentication is required on FXO Port. By default, it is disabled.

  (Please refer “PIN Authentication” for more details)

- **Allow Call Disconnection using Access Code?:** Enable this field if disconnection of the call using access code is to be allowed. By default, it is disabled.

  (Please refer “Disconnect call using Access Code” for more details)

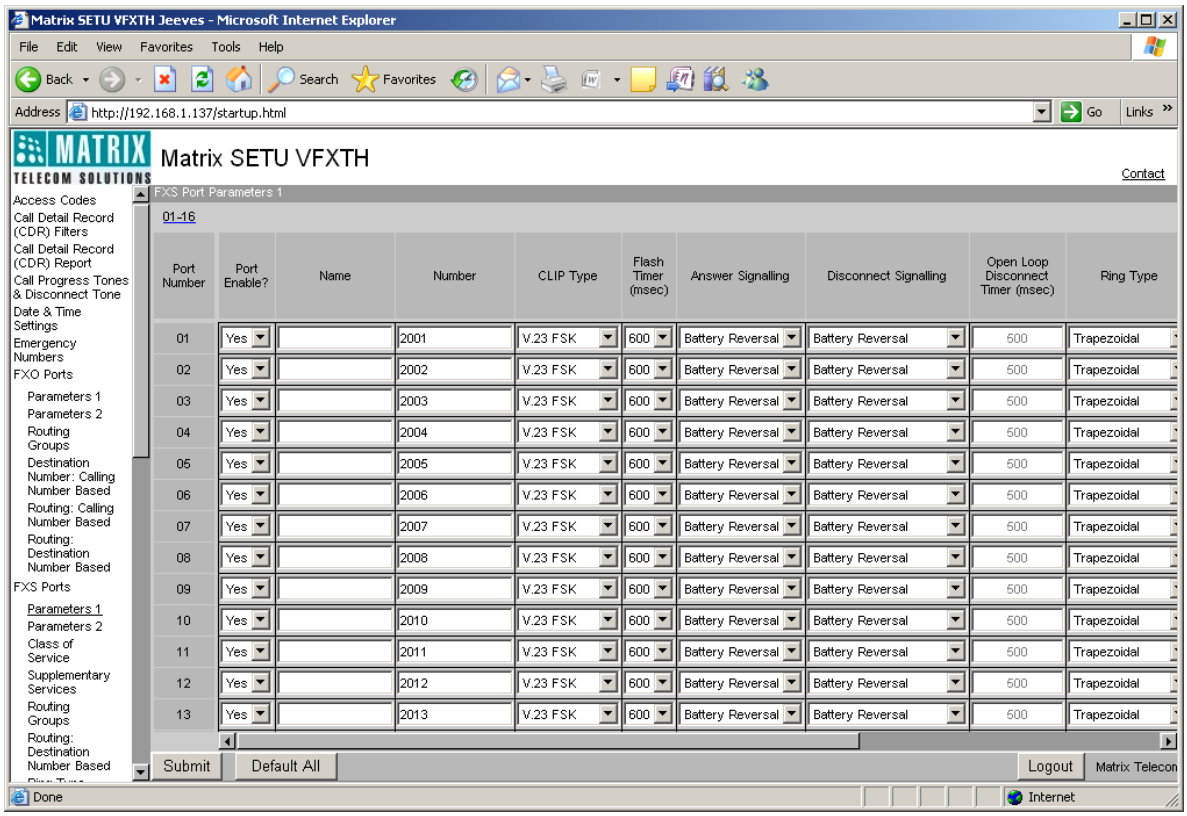
- **Allow New Call using Access Code?:** Enable this field if making of new call using access code is to be allowed. By default, it is disabled.

  (Please refer “Making New Call” for more details)

Port Parameters-FXS

SETU VFXTH1616 supports maximum 16 FXS Ports. SETU VFXTH1616 shall route the call to and from FXS Port only if that port is enabled. To program various FXS Port Parameters, open Web Jeeves of SETU VFXTH1616.

- Click on **'FXS Port Parameters 1'** link and program all the parameters on the page as shown below:



- **Port Enable?:** You can enable or disable FXS Ports as per your requirement. Disabling of FXS Port is required only in case of hardware failure. If FXS Port is disabled, it will not be used for routing the calls.

  By default, all FXS Ports are enabled.

- **Name:** You can assign name to the FXS Port. This name is displayed on the LCD of the called party if the instrument of the called party supports display name functionality. Name can be of maximum 12 characters and following ASCII characters are supported:

  - 0 to 9
  - A to Z
  - a to z
  - ! (exclamation mark)
  - % (percentage)
  - ' (single quote)
  - * (asterisk)
  - - (minus or dash)
  - + (plus)
  - . (dot)

  By default, name field is blank for all FXS Ports.

- **Number:** A four digit number can be programmed for each FXS Port. This number is used as calling party number for the remote party.

  By default, number 2001 to 2032 is assigned to FXS Port 01 to 32 respectively.

- **Calling Line Identification Presentation (CLIP) Type:** When a call is placed on the FXS Port, calling party number received on the source port is to be displayed on the LCD of the telephone instrument connected to FXS Port. Different type of signaling is used to present the calling party number on the FXS Port. SETU VFXTH1616 supports following types of CLIP signaling on the FXS Port:

  - None
  - DTMF
  - V.23 FSK
  - Bellcore FSK

  By default, CLIP Type V.23 FSK is selected for FXS Ports.

  CLIP type 'None' is selected when the telephone instrument connected to the FXS Port does not have the LCD to display the number or user does not want to display the CLIP.

- **Flash Timer (msec):** SETU VFXTH1616 uses flash timer to detect Flash on FXS Port for using various supplementary features viz. Call Hold, Call Transfer etc. Flash Timer signifies the time period for which the loop current breaks.

  Select Flash Timer as per you requirement in this field. Valid options are:

  - 83
  - 100
  - 200
  - 300
  - 400
  - 500
  - 600
  - 700
  - 800
  - 900

  By default, Flash Timer is 600 msec for all FXS Ports.

- **Answer Signaling (H2):** Answer Signaling is a signal generated on FXS Port, which indicates that the called party has answered and the call is matured. Answer Signaling on FXS Port can be in one of the following forms:

  - ***None:*** This option is selected when no answer signaling is to be generated on the FXS Port.

  - ***Battery Reversal:*** This option is used when answer signaling is to be generated in the form of Battery Reversal on the FXS Port. In Battery Reversal, battery polarity of FXS Port is reversed. For example, suppose the battery polarity of the FXS Port is negative for TIP and positive for RING in dialing state. After call maturity, TIP will become positive and RING will become negative.

  By default, 'Battery Reversal' is used for Answer Signaling on FXS Port.

- **Disconnect Signaling:** Disconnect signaling is the signal generated on FXS Port to indicate the call disconnection by the other party. Disconnect Signaling on FXS Port can be generated in one of the following forms:

  - ***None:*** This option is selected when no disconnect signaling is to be generated on the FXS Port.

  - ***Battery Reversal:*** This option is used when disconnect signaling is to be generated in the form of Battery Reversal on the FXS Port. In Battery Reversal, battery polarity of FXS Port is reversed. For example, suppose the battery polarity of the FXS Port is positive for TIP and negative for RING in speech condition. After call disconnection, TIP will become negative and RING will become positive.

  - ***Open Loop Disconnect:*** This option is used when call disconnection is to be signaled in the form of Open Loop Disconnect pulse. In Open Loop Disconnect, the battery voltage on FXS Port is removed for Open Loop Disconnect timer and then it is restored again.

  By default, Disconnect Signaling is set to Battery Reversal on FXS Port.

- **Open Loop Disconnect Timer:** When the Disconnect signaling is generated in the form of 'Open Loop Disconnect' then the system breaks the loop current for the time period programmed as Open Loop Disconnect Timer and thus releases the call.
  Program Open Loop Disconnect Timer in this field. The range of Open Loop Disconnect timer is 001 to 999 msec.

  By default, Open Loop Disconnect Timer is 500 msec.

- **Ring Type:** Select Ring Type in this field. Ring Type parameter is used to select the type of ring to be generated on FXS Port. SETU VFXTH1616 supports following Ring Types:

  - Low Sinusoidal
  - Low Trapezoidal
  - Sinusoidal
  - Trapezoidal

  By default, 'Trapezoidal' type of ring is generated on each FXS Port.

- **Rx Gain (db):** Select Receive Gain for FXS Port in this field. SETU VFXTH1616 enables you to adjust receive gain of the FXS Port to increase the audibility of incoming speech. Valid range of Rx Gain is:

  - -3
  - 0
  - +3
  - +6

  By default, Rx Gain for FXS Port is set to '0'.

- **Tx Gain (db):** Select Transmit Gain for FXS Port in this field. SETU VFXTH1616 enables you to adjust the transmit gain of the FXS Port to adjust the right intensity for the transmitted signal. Valid range of Tx Gain is:

  - -3
  - 0
  - +3

- +6

By default, Tx Gain for FXO Port is set to '0'.

- **AC Impedance:** Select AC Impedance for FXS Port. SETU VFXTH1616 supports following AC Impedances for FXS Port:

  - 600Ω
  - 900Ω
  - 350Ω + (1000Ω || 0.21 F)
  - 220Ω + (820Ω || 120 nF)

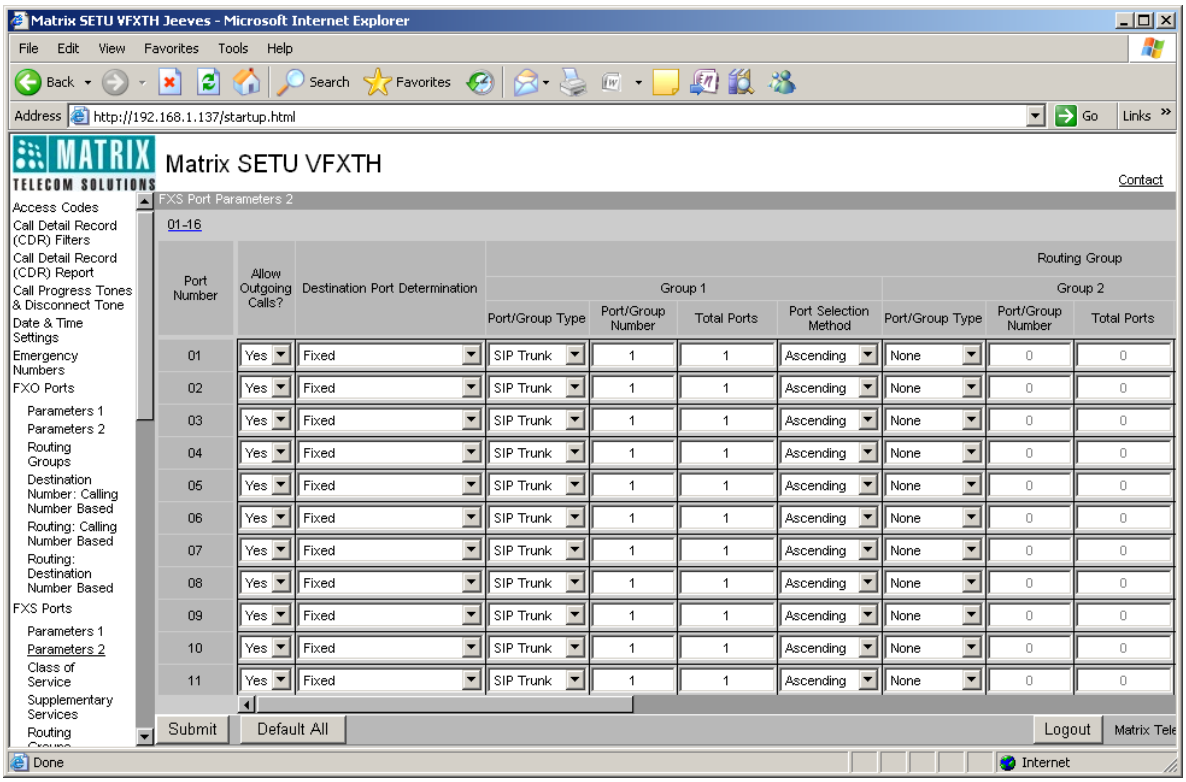
  By default, AC Termination Impedance is set to 600 ?.

- **Loop Current:** SETU VFXTH1616SETU VFXTH1616 supports configurable loop current as per the loop length programmed for the FXS Port. You can select the appropriate loop current from the following options:

  - 25 mA
  - 30 mA
  - 35 mA
  - 40 mA

  By default, Loop Current is set to 35 mA.

- Click on **'FXS Port Parameters 2'** link and program the necessary parameters as shown below:



- **Allow Outgoing Calls?:** You can allow or dis-allow making of outgoing calls from FXS Port. Select 'No' to restrict outgoing calls from the FXS Port.

  By default, 'Allow Outgoing Calls?' is set to 'Yes' for all FXS Ports.

- **Destination Port Determination**
  (Refer “Routing Mechanism on FXS” for details)

- **Routing Groups**
  (Refer “Routing Group” for details)

- **CLI Number on FXS Port**
  (Refer “Number Presentation on FXS Port” for details)

- **Allowed Denied Numbers (Toll Control)**
  (Refer “Allowed-Denied Numbers” for details)

- **First Digit Wait Timer (seconds):** Program First Digit Wait Timer for FXS Port in this field. Range for First Digit Wait Timer is 01 to 99 seconds.

  By default, it is 07 seconds.

  (Refer “Number Collection Method” for more details)

- **Maximum Number of Dialed Digits:** Enter the maximum number of digits allowed to be dialed out on FXS Port in this field. Maximum number of dialed digits is used to detect end-of-dialing of a number on FXS Port. Range of this field is 01 to 24 characters.

  By default, it is 16 digits.

  (Please refer “Number Collection Method” for more details)

- **End of Dialing:** Untick 'Apply' flag to dis-allow End-of-Dialing using termination digit in End-of-Dialing table. By default, it is ticked. Valid Termination Digits are # and *. By default, # is selected.

  Also program Inter Digit Wait Timer in 'Timer (seconds)' field. Range of Inter Digit Wait Timer is 01 to 99 seconds. By default, it is 05 seconds.

  (Please refer “Number Collection Method” for more details)

- **Subscriber Type:** Subscriber Type signifies the type of services FXS Port user can access. Subscriber Type can be either Gateway or Network.

  - In ***Gateway Type***, the FXS Port user can use features of SETU VFXTH1616 like call hold, call transfer etc. along with the Supplementary Services of Service Provider.

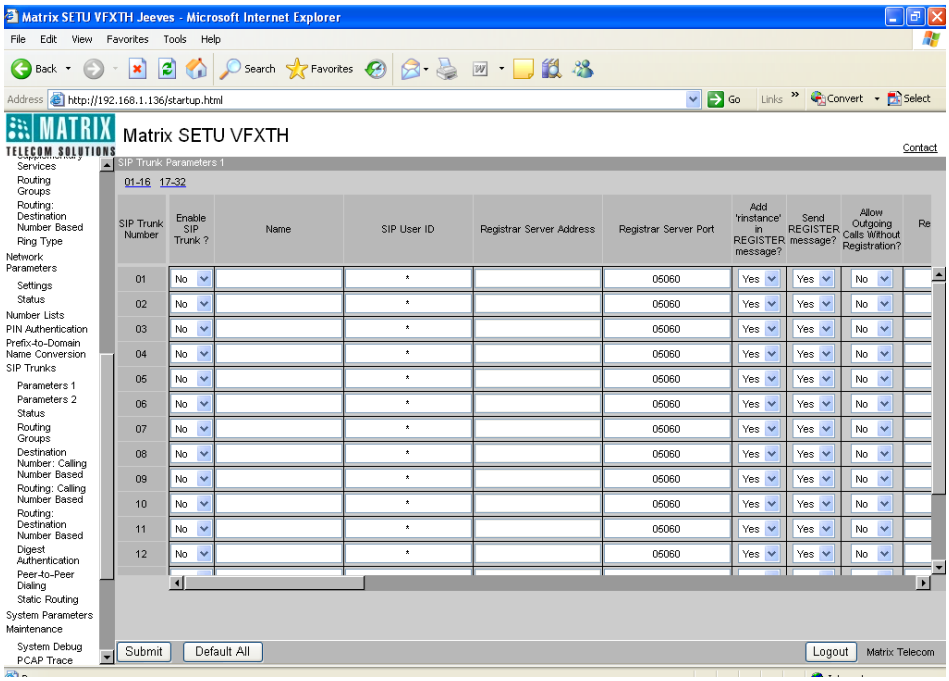
  - In ***Network Type***, the FXS Port user can use all the remote end features i.e. the features of the PBX connected to SETU VFXTH1616. However, user cannot use any feature of SETU VFXTH1616 that requires Flash i.e. Call hold, Call waiting etc.

  By default, Subscriber Type is set to 'Gateway' for all FXS Ports.

Port Parameters-SIP

SETU VFXTH1616 supports 32 SIP Trunks. You may register all SIP Trunks with one ITSP or with different ITSPs. To program various SIP Trunk Parameters, open Web Jeeves of SETU VFXTH1616.

- Click on **'SIP Trunk Parameters 1'** link and program all the parameters on the page as shown below:



- **Enable SIP Trunk?:** You can enable or disable the SIP Trunks as per your requirement. SIP Trunk is disabled if not registered or if there is some problem on the service provider side. SETU VFXTH1616 shall route the call from SIP Trunk only if it is enabled.

  By default, all SIP Trunks are disabled.

- **Name:** You can assign a name to the SIP Trunk. Name assigned to the SIP Trunk is relevant only for identification of that trunk. It is used to remember the ITSP with which it is registered or to remember the name of the person using it.

  Name can be of maximum 12 characters. All alpha-numeric characters are allowed.

  By default, name field is blank for all SIP Trunks.

- **SIP User ID:** SIP ID is the user part of the full SIP URI provided by the ITSP. For e.g. If SIP URI provided by ITSP is 12345@abc.com then 12345 is the SIP ID. SIP ID can be of maximum 40 characters. All ASCII characters are allowed. By default, SIP ID programmed for each SIP Trunk is *.

- **Registrar Server Address:** SIP Registrar Server Address is the address of the SIP Registrar. It can be an IP address or domain. SIP registrar server address can be of maximum 40 characters. All ASCII characters are allowed.

  By default Register Server Address is Blank.

- **Registrar Server Port:** SIP Registrar Servers' Port is the Registrar Server's listening port for SIP. It may be same as SIP Server's Port Address. Valid range of registrar server port is from 1025 to 65535.

  By default register server port is 5060.

- **Add 'rinstance' in REGISTER message?:** 'rinstance' is any random value used by SETU VFXTH1616 to fetch its own contact binding. SETU VFXTH1616 will include 'rinstance' in REGISTER message, if this parameter is set to 'Yes'. Select 'No' if you do not want system to include rinstance in REGISTER message.

  By default, this parameter is set to 'Yes' for all the SIP Trunks.

- **Send REGISTER message?:** You can select whether or not to send REGISTER message from the SIP Trunk. Select 'No' if you do not want SETU VFXTH1616 to send REGISTER message to the Registrar/Outbound Proxy Server.

  By default, this parameter is set to 'Yes' for all the SIP Trunks.

- **Allow Outgoing Calls without Registration?:** Select 'Yes' to enable SETU VFXTH1616 to make outgoing calls from the SIP Trunk even when it is not registered with the Registrar Server.

  By default, this parameter is set to 'No' for all the SIP Trunks. Therefore, SETU VFXTH1616 will not allow making of Outgoing calls using SIP Trunk, if it is not registered with the SIP Server.

- **Re-Registration Timer:** The registrar server deletes an entry of its client from its database on expiry of a fixed timer which is set by the registrar server. SETU VFXTH1616 will send a registration request before this timer expires to remain registered on the server. Re-Registration timer should be less than or equal to the timer set by registrar server for deleting the client's entry from its database. Before expiry of this timer, SETU VFXTH1616 should send registration request to the registrar server to get registered again.

  Program Re-Registration Timer (seconds) in this field. Valid range of this timer is from 00001 to 65535.

  By default Re-Registration timer is 3600 seconds.

- **Registration Retry Timer:** Registration Retry Timer indicates the period between retries for registration. If the registration attempt fails, SETU VFXTH1616 shall send the registration request on expiry of this timer again. SETU VFXTH1616 keeps sending the registration request till SIP Trunk gets registered with SIP server. Valid range of this timer is from 00001 seconds to 65535 seconds.

  By default, Registration Retry Timer is 10 seconds.

- **Authentication User ID:** Authentication User ID is the user name/ number for registering the SIP Trunk with the SIP register server. It is provided by an ITSP and is relevant when SIP user ID and Authentication user ID are not same. User ID can be of maximum 40 characters. All ASCII characters are allowed.

  By default Authentication User ID is Blank.

- **Authentication User Password:** Authentication User Password is the password associated with the Authentication User ID. Password can be of maximum 24 characters.

  By default Authentication password is Blank.

- **Outbound Proxy:**

  - ***Enable Outbound Proxy?*:** Outbound Proxy Server can be enabled or disabled as per requirement. It can be enabled only if the ITSP service provider has a SIP outbound server to handle voice calls.

    By default, Outbound Proxy is set to 'No'.

  - ***Server Address:*** Outbound Proxy server address is the address of the outbound proxy server. This may be same as SIP server address. It can be IP address or domain. Outbound proxy server address can be of maximum 48 characters and all ASCII characters are allowed.

    By default, Outbound Proxy Server Address is Blank.

  - ***Server's Port:*** Outbound Proxy Servers' Port is the outbound proxy server's listening port for SIP. This may be same SIP server's port. Valid range of server port is from 1025 to 65535.

    By default, Outbound Proxy Server Port is 5060.

- **Source Port IP Address:** Source Port IP Address means IP Address of the source computer from which the data packets to be sent over IP Address has originated. SETU VFXTH1616 provides three options for selecting Source Port IP Address:

  - ***Use Network Port IP Address:*** Select this option if SETU VFXTH1616 is connected to the public internet.

  - ***Use IP Address fetched using STUN:*** STUN is the most widely used protocol by SIP clients when located behind the NAT router. STUN is used to map the public IP address and port of the NAT router behind which the SIP client is located. Select this option, if SETU VFXTH1616 is located behind the NAT router other than Symmetric. This option will work only if outbound is disabled on the SIP Trunk. If you have selected this option, program the STUN server's address and STUN server's port in 'Network Port Parameters'.

  - ***Use Router's Public IP Address:*** Select this option if SETU VFXTH1616 is located behind the NAT router. This option will work only if outbound is disabled on SIP Trunk. If you have selected this option, program router's public IP address in 'Network Port Parameters'.

  By default, 'Use Network Port IP Address' is selected as Source Port IP Address.

- **Vocoder:** Vocoders are the various voice codecs used to compress the data in RTP packets for optimum use of bandwidth and for ensuring voice quality. SETU VFXTH1616 supports total 6 types of Vocoders. The Preferred Vocoders are:

  - G.729ab (1st Preference)
  - G.723-L (2nd Preference)
  - G.723-H (3rd Preference)
  - GSM FR (4th Preference)
  - iLBC-30ms (5th Preference)
  - iLBC-20ms (6th Preference)
  - G.711 (u-law) (7th Preference)
  - G.711 (A-law) (8th Preference)

  By default, Preferred Vocoders shall follow the same sequence as shown above.

i. *The Vocoders for the OG calls shall be selected as per the preferences programmed.*

ii. *This parameter will be checked in case of outgoing calls only.*

iii. *The Vocoder requested by the remote end shall be entertained only if supported by SETU VFXTH1616, else the call shall be rejected.*

- **DTMF Dialing:** DTMF dialing option decides the method for sending the DTMF digits over IP network, when a DTMF digit is pressed. SETU VFXTH1616 supports following options in SIP for DTMF Dialing:

  - ***RTP (RFC 2833):*** RTP (RFC 2833) is also known as Out-band. Out-band means digits are to be sent using RTP (RFC 2833) packets.

  - ***SIP INFO:*** SIP INFO means digits are to be sent in SIP INFO message.

  - ***In-Band:*** In-band means DTMF is combined in audio signal.

  By default, DTMF Dialing is set to RTP (RFC 2833).

- **Flash Dialing:** Flash Dialing is used for sending flash digit to the remote end. Flash Dialing options are not applicable while receiving flash digit from remote end. Following options are supported in SIP for Flash Dialing:

  - RTP (RFC 2833)

  - SIP INFO

  By default, flash dialing is set to RTP (RFC 2833).

- **FAX Option:** This parameter signifies how FAX messages are handled by the SETU VFXTH1616. SETU VFXTH1616 supports following FAX options:

  - T.38 (UDPTL)
  - T.38 (RTP)
  - Pass Through.

  By default, T.38 (UDPTL) is selected.

  Refer “FoIP (Fax over IP) using T.38” for details)

- **Use Symmetric RTP?:** Select 'Yes' to enable symmetric RTP if SETU VFXTH1616 is located on the public IP and outgoing calls are made to the SIP client located behind the NAT router or incoming calls are received from the SIP client located behind the NAT router. This parameter is applicable only for peer to peer calls.

  By default, 'No' is selected for all SIP Trunks.

- **Use 183 (Session Progress) to connect Media:** To connect SETU VFXTH1616 to the media after answering the call, select 'Yes' in this field. If this option is enabled, SETU VFXTH1616 will send either 183 Session Progress or 200 OK to acknowledge answering of the call by the system further enabling it to connect to the media.

By default, this option is set to 'No' for all SIP Trunks.

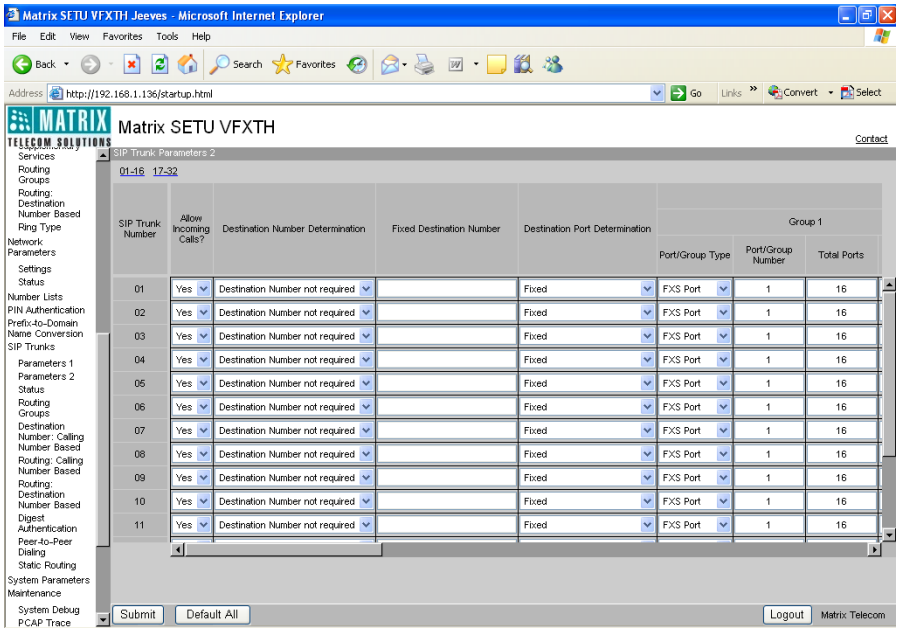
- **Default Transport for outgoing messages:** SETU VFXTH1616 supports following three options for outgoing SIP messages. These options are checked only if you have enabled SIP over TCP in Network Port Parameters else all outgoing messages are transported over UDP only.
  - ***UDP:*** If this option is selected, UDP will be used as default transport protocol for outgoing messages.
  - ***TCP:*** If this option is selected, TCP will be used as default transport protocol for outgoing messages.
  - ***TCP (with fallback to UDP):*** If this option is selected, TCP will be used as default transport protocol for outgoing messages. However, if TCP connection fails, SETU VFXTH1616 will try sending messages again over UDP.

  By default, UDP is programmed as default transport for outgoing messages.

- **Call Hold Method:** SETU VFXTH1616 enables user to hold a SIP call from FXS Port. Select the method supported by your ITSP. Following two methods are supported:
  - ***RFC 2543:*** If you select this option, SETU VFXTH1616 will send connection information: 0.0.0.0 to the ITSP.
  - ***RFC 3261:*** If you select this option, SETU VFXTH1616 will send connection information: IP Address as used in Contact to the ITSP.

  By default, Call Hold Method is set to 'RFC 3261' for each SIP Trunk.

- Click on **'SIP Trunk Parameters 2'** link and program all the parameters on the page as shown below:



- **Allow Incoming Calls?:** This field is applicable only for incoming calls on SIP Trunk. Select 'No' in this field to prohibit incoming calls on a SIP Trunk. If this field is set to 'Yes', the call will be routed to the destination port as per the routing mechanism programmed for that SIP Trunk.

  By default, this field is set to 'Yes'.

- **Destination Number Determination**
  (Refer “Routing Mechanism on SIP” for details)

- **Fixed Destination Number**
  (Refer “Routing Mechanism on SIP” for details)

- **Destination Port Determination**
  (Refer “Routing Mechanism on SIP” for details)

- **Routing Groups**
  (Refer “Routing Group” for details)

- **CLI number on FXS Port**
  (Refer “Number Presentation on FXS Port” for details)

- **Allowed-Denied Numbers (Toll Control)**
  (Refer “Allowed-Denied Numbers” for details)

- **Digest Authentication:** Tick to enable Digest authentication on the desired SIP Trunk.

  By default, this flag is Untick i.e. Digest authentication is disabled on all SIP Trunks.

  (Refer “Digest Authentication” for more details)

- **Anonymous Call Allowed?:** Select 'No' in this field to restrict anonymous calls on a SIP Trunk. By default, anonymous calls are allowed on all SIP Trunks.

- **Allow Call Disconnection using Access Code?:** Select 'Yes', to enable disconnection of calls using Access Code.

  By default, 'No' is selected in this field i.e. disconnecting call using access code is not allowed.

- **Maximum Calls:** Program the maximum number of calls that is to be allowed on each SIP Trunk. Valid range for maximum calls allowed is 01 to 32.

  By default, maximum calls allowed on all SIP Trunks are 32.

- **Black Listed Callers**
  (Refer “Black Listed Callers” for details.)

- **Send Caller ID?:** Select 'No' if you do not want to send CLI while making outgoing calls using SIP Trunk. SETU VFXTH1616 will send SIP ID configured on the SIP Trunk in the CLI of outgoing calls when this parameter is set to 'Yes'.

  By default, 'Send Caller ID' is set to 'Yes'.

- **Send Caller-ID received on Source Port?:** If this parameter is set to 'Yes', SETU VFXTH1616 will send the CLI received on the Source port in the 'From' field of INVITE message when an outgoing call is made using SIP Trunk else it will check 'Send Caller ID?' option programmed for the SIP Trunk.

  By default, 'Send Caller-ID received on Source Port' is set to 'No'.

SETU Vfxth1616 will send Anonymous in the From field if 'Send Caller ID?' option is set to 'No', whether 'Send Caller-ID received on Source Port' is programmed as 'Yes' or 'No'.

- **Automatic Number Translation**
  (Refer “Automatic Number Translation” for details)

Prefix to Domain Name Conversion

SETU VFXTH supports multiple SIP Trunks and FXS ports. When the FXS Port user dials a SIP number, system routes this call to IP network using the SIP Trunk determined by the routing mechanism.

Now, it is necessary that the number string dialed by SETU VFXTH is understood by the ITSP through which the call is routed. Thus, for this purpose 'Prefix to Domain Name Conversion' feature is useful. This feature is applicable only when the destination port is SIP.

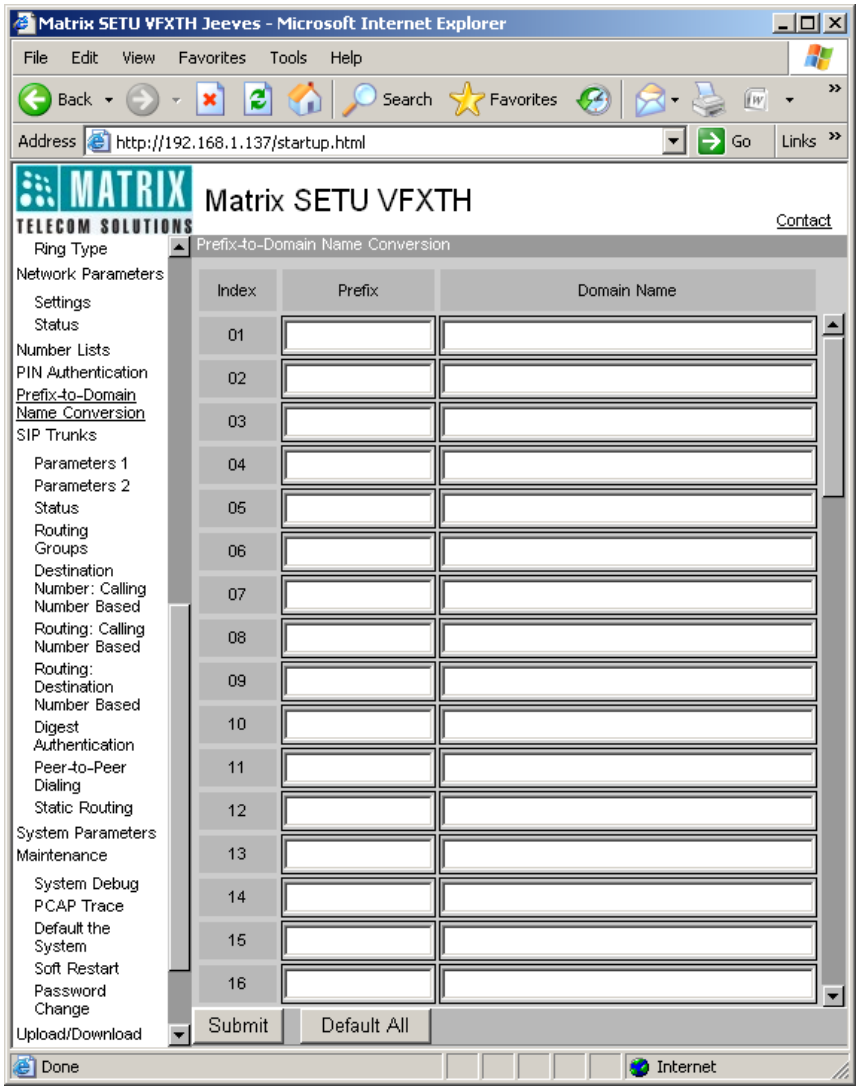
Let us understand this feature with the help of an example:

> Suppose, SETU VFXTH is so programmed that all the calls made to 'abc.com' from FXS Port are routed through the SIP Trunk registered with 'Pulver.com'. SE assigns a Prefix code, *234 to the domain 'abc.com'. Now, when FXS user wants to dial the SIP ID 9874@abc.com, he should dial *234 followed by 9874. The SETU VFXTH 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 in above scenario, assume that the FXS user forwards his calls to *2349874 (i.e. 9874@abc.com). If any external caller calls the FXS user, he would not be able to reach the desired number because FXS user has forwarded his calls to *2349874. The external caller would not recognize *234, the prefix code assigned to abc.com. In this situation, SETU VFXTH should convert this prefix code to the domain name abc.com, using 'Prefix-to-Domain name conversion' table and send it in the redirect message to the external caller to inform him of the new contact address. On receiving this information, external caller can call the new contact number and talk to the desired person.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'Prefix to Domain Name Conversion'** link and program the following parameters of 'Prefix to Domain Name Conversion' table.



- **Index:** Index number shows the serial number at which the entry is stored.
- **Prefix:** Enter Prefix Code for the Domain Name in this field. Prefix code is the code that is assigned to the Domain (Server) by the SE. Prefix code can be of maximum four digits. Digits 0 to 9, * and # are allowed. By default, all entries are blank.
- **Domain Name:** Enter the Domain Name whose prefix code is entered in corresponding 'Prefix' field. Domain name can be of maximum 40 characters. All ASCII characters are allowed. By default, all entries are blank.

i. *Prefix to Domain Name Conversion table has 64 entries.*

ii. *This feature is used when user has set Call Forward or user is using Blind Transfer on SIP.*

Relevant Topics:

1. “Blind Call Transfer” 184
2. “Call Forward” 187
3. “Call Processing” 47

Region Selection

SETU VFXTH is a versatile system and it is designed to operate world wide. Several important parameters in it are country specific such as:

- Language
- Time Zone
- Schedule DST
- CPTG
- Ring Type

Therefore, when SETU VFXTH is installed in the country other than India, the values of these parameters should be changed according to the region in which the system is installed. By default, values of these parameters are set for India.

Region selection enables you to set the values of all the region/country specific parameters as per the country or region of installation. After changing the country or region, SE must default the system to change the values of these parameters, for proper functioning of the system.

Once the values of these parameters are changed as per the region programmed, it is possible to change the values of these parameters separately also if required. For e.g. when the region/country is changed from India to Portugal, the values assigned to region/country specific parameters would be as shown below:

- Language - Portuguese
- Time Zone - GMT
- Schedule DST - Type 07 (Schedule DST)
- CPTG - 27 (Portugal)
- Ring Type - 12 (Portugal)

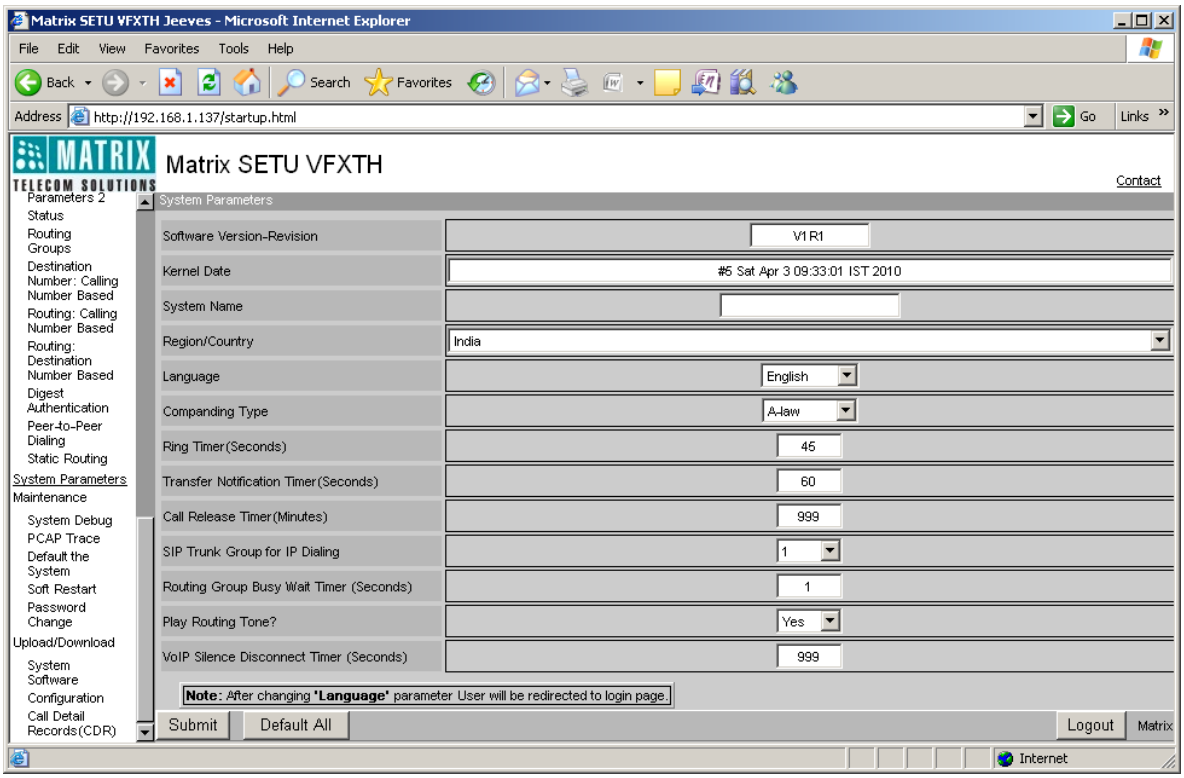
Now SE can change language from Portuguese to English if he wishes to do so.

Refer table given at the end of this topic for the values assigned to above mentioned parameters in region/country other than India.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on **'System Parameters'** link and select 'Region/Country' from the list.



- After selecting Region/Country when SE submits the page, an alert message "Changing Region shall assign default values to all parameters of System. Do you want to continue?" OK/Cancel will appear.

- Click **'OK'** button. All parameters will be assigned default values followed by system restart. Check 'Country/Region' programmed after the system restarts.

'Region/ Country' does not default when the system is defaulted.

Following table shows the values used for various countries for the parameters stated above. (Values which are blank are as per INDIA except that of Default DST Type. DST Mode is disabled wherever not specified.)

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type
001	Afghanistan	English	GMT+04:30			
002	Algeria	English	GMT+01:00			
003	Antigua and Barbuda	English	GMT-04:00			
004	Argentina	Spanish	GMT-03:00		04	
005	Australia (Perth)	English	GMT+08:00	02	05	08
006	Australia (Adelaide)	English	GMT+09:30	02	05	08
007	Australia (Brisbane, Canberra, Melbourne, Sydney)	English	GMT+10:00		05	08
008	Austria	German	GMT+01:00	01		

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type
009	Bahamas	English	GMT-05:00			
010	Bahrain	English	GMT+03:00	03		
011	Bangladesh	English	GMT+06:00			
012	Belarus	English	GMT+02:00			
013	Belgium	French	GMT+01:00	02	39	11
014	Bhutan	English	GMT+06:00			
015	Bolivia	Spanish	GMT-04:00			
016	Bosnia and Herzegovina	English	GMT+01:00			
017	Botswana	English	GMT+02:00			
018	Brunei	English	GMT+08:00			
019	Brazil (Fernando De Noronha)	Portuguese	GMT-02:00		06	06
020	Brazil (Brasilia, Rio de Janeiro, Sao Paulo)	Portuguese	GMT-03:00	04	06	06
021	Brazil (Manaus)	Portuguese	GMT-04:00		06	06
023	Brazil (Acre)	Portuguese	GMT-05:00		06	06
024	Bulgaria	English	GMT+02:00			
025	Cambodia	English	GMT+07:00			
026	Cameroon	English	GMT+01:00			
027	Canada (St. John's)	English	GMT-03:30	05	07	07
028	Canada (Halifax)	English	GMT-04:00	05	07	07
029	Canada (Montreal, Ottawa, Toronto)	English	GMT-05:00	05	07	07
030	Canada (Winnipeg)	English	GMT-06:00	05	07	07
031	Canada (Calgary)	English	GMT-07:00	05	07	07
032	Canada (Vancouver)	English	GMT-08:00	05	07	07
033	Chile	Spanish	GMT-04:00	06		
034	China	English	GMT+08:00		08	11
035	Colombia	Spanish	GMT-05:00			
036	Costa Rica	Spanish	GMT-06:00			
037	Croatia	English	GMT+01:00			
039	Cuba	Spanish	GMT-05:00	18		
040	Cyprus	English	GMT+02:00			
041	Czech Republic	English	GMT+01:00			
042	Denmark	English	GMT+01:00	07		

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type
043	Egypt	English	GMT+02:00	11	09	07
044	Fiji	English	GMT+12:00			
045	Finland	English	GMT+02:00	08		
046	France	French	GMT+01:00	02	10	14
047	Germany	German	GMT+01:00	02	11	06
048	Greece	English	GMT+02:00	02	12	06
049	Guyana	English	GMT-04:00			
050	Hong Kong	English	GMT+08:00			
051	Hungary	English	GMT+02:00	02		
052	India	English	GMT+05:30		13	08
053	Indonesia	English	GMT+07:00		14	
054	Iran	English	GMT+03:30		15	
055	Iraq	English	GMT+03:00	09	16	
056	Ireland	English	GMT	07		
057	Israel	English	GMT+02:00		17	15
058	Italy	Italian	GMT+01:00	02	18	06
059	Japan	English	GMT+09:00		19	10
060	Jordan	English	GMT+02:00			
061	Kazakhstan	English	GMT+06:00			
062	Kenya	English	GMT+03:00		20	
063	Korea - North	English	GMT+09:00		21	11
064	Korea - South	English	GMT+09:00		21	11
065	Kuwait	English	GMT+03:00			
066	Kyrgyzstan	English	GMT+06:00	10		
067	Lebanon	English	GMT+02:00	12		
068	Libya	English	GMT+02:00			
069	Malaysia	English	GMT+08:00		22	15
070	Maldives	English	GMT+05:00			
071	Mauritius	English	GMT+04:00			
072	Mexico (Mexico City)	Spanish	GMT-06:00	03	23	
073	Mexico (Chihuahua)	Spanish	GMT-07:00	03	23	
074	Mexico (Tijuana)	Spanish	GMT-08:00	03	23	
075	Mongolia	English	GMT+08:00			
076	Mozambique	Portuguese	GMT+02:00			

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type
077	Myanmar	English	GMT+06:30			
078	Namibia	English	GMT+01:00	13		
079	Nepal	English	GMT+05:45			
80	Netherlands	English	GMT+01:00			
81	New Zealand	English	GMT+12:00	14	24	15
82	Nigeria	English	GMT+01:00			
83	Norway	English	GMT+01:00	15		
84	Oman	English	GMT+04:00			
85	Pakistan	English	GMT+05:00			
86	Paraguay	Spanish	GMT-04:00	16		
87	Peru	Spanish	GMT-05:00			
88	Philippines	English	GMT+08:00		25	
89	Poland	English	GMT+01:00	01	26	15
90	Portugal	Portuguese	GMT	07	27	12
91	Qatar	English	GMT+03:00			
92	Romania	English	GMT+02:00			
93	Russia (Moscow, St. Petersburg)	English	GMT+03:00	01	28	11
94	Russia (Novosibirsk)	English	GMT+06:00	01	28	11
95	Russia (Vladivostok)	English	GMT+10:00	01	28	11
96	Singapore	English	GMT+08:00		30	08
97	Slovakia	English	GMT+01:00			
98	South Africa	English	GMT+02:00		31	08
99	Spain	Spanish	GMT+01:00	01	32	13
100	Sri Lanka	English	GMT+05:30			
101	Sudan	English	GMT+03:00			
102	Sweden	English	GMT+01:00	02		
103	Switzerland	German	GMT+01:00	02		
104	Syria	English	GMT+02:00	17		
105	Taiwan	English	GMT+08:00			
106	Tajikistan	English	GMT+05:00			
107	Thailand	English	GMT+07:00		33	15
108	Turkey	English	GMT+02:00		34	
109	Uganda	English	GMT+03:00			
110	Ukraine	English	GMT+02:00			

Region Code	Country/ Region	Default Language	Default Time Zone	Default DST Type	Default CPTG	Default Ring Type
111	United Arab Emirates	English	GMT+04:00		35	15
112	United Kingdom	English	GMT	07	36	08
113	United States (Atlanta, Augusta, Boston, Charlotte, Columbus, Detroit, Indiapolis, Miami, NY, Philadelphia, Washington)	English	GMT-05:00	03	37	07
114	United States (Chicago, Dallas, Des Moines, Memphis, Minneapolis, New Orleans, Oklahoma, Omaha, St. Louis)	English	GMT-06:00	03	37	07
115	United States (Albuquerque, Boise, Cheyenne, Denver, Salt Lake City)	English	GMT-07:00	03	37	07
116	United States (Las Vegas, Los Angeles, Phoenix, San Francisco, Seattle)	English	GMT-08:00	03	37	07
117	United States (Juneau)	English	GMT-09:00	03	37	07
118	United States (Hawaii)	English	GMT-10:00		37	07
119	Uzbekistan	English	GMT+05:00			
120	Venezuela	Spanish	GMT-04:30			
121	Vietnam	English	GMT+07:00			
122	Yemen	English	GMT+03:00			
123	Yugoslavia	English	GMT+02:00			
124	Zambia	English	GMT+02:00			
125	Zimbabwe	English	GMT+02:00			

Relevant Topics:

1. “Call Progress Tones” 48
2. “Date and Time Settings” 57
3. “Daylight Saving Time (DST) Mode” 59

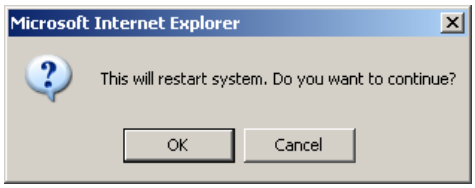
Restart SETU VFXTH

Sometimes it becomes necessary for the user to restart the system. Many a times it is difficult to walk to the system and restart it. Thus, SETU VFXTH gives facility to restart it through Web Jeeves. Restarting SETU VFXTH gives the same effect as switching OFF the system and switching it ON again.

How to Restart the System using Web Jeeves?

- Open Web Jeeves of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on **'Soft Restart'** link. An alert message will appear on the screen:

  "This will Restart System. Do you want to continue?"



  - Click **'OK'** to restart SETU VFXTH else click 'Cancel'.

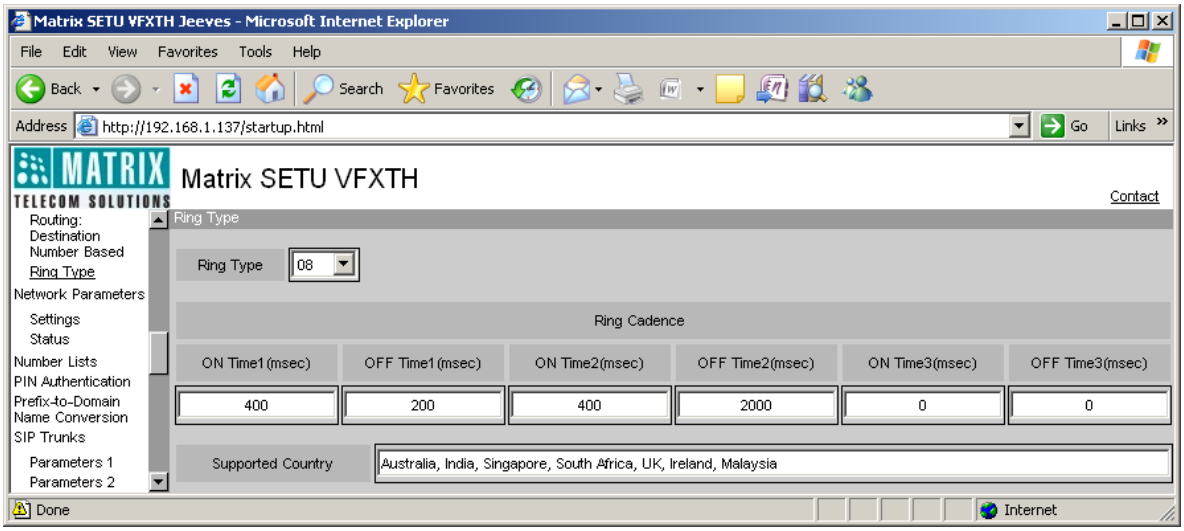
Restarting SETU VFXTH will not alter any programming parameters. But all active calls will be disconnected and the ports in use will be released.

Ring Type

SETU VFXTH has inbuilt ring cadences of various countries. It gives you the flexibility to match ring cadences of SETU VFXTH to that of the country standard in which it is installed and stimulate the same Ring as given by PSTN.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")
- Click on **'Ring Type'** link.



- In Ring Type field, select the country in which SETU VFXTH is installed. By default, Ring Type selected is 08 (India).
- Ring cadence value will change in Ring Cadence table automatically as per the country programmed in Ring Type field. Ring Cadence table is un-editable.

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, Netherlands, Switzerland, Finland, Germany
07	2.0	4.0					Egypt, USA, Canada, Namibia

Ring Type Number	CADENCE (In Seconds)						Countries
	TON1	TOFF1	TON2	TOFF2	TON3	TOFF3	
08	0.4	0.2	0.4	2.0			Australia, India, Singapore, South Africa, UK, Ireland
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, Malaysia, New Zealand, Poland, Thailand, UAE, Czechia, Norway, Hongkong, Austria, Hungary, Slovakia
16	3.5	5.5	0.79	1.1			

Ring Type does not get default when SETU VFXTH is defaulted.

Routing Group

SETU VFXTH1616 supports multiple port types' viz. FXS, FXO and SIP. A call can originate from and terminate on any port type. When the call originates on the source port, it is required to be routed to the destination port as per the routing mechanism programmed for that source port.

If only one destination port is programmed for a source port and if that port is found busy, the call cannot be routed. SETU VFXTH1616 facilitates formation of a Routing Group. You can program more than one destination port in the same group for routing the call originated on the source port, so that if one port is busy the call can be routed to another port.

You can form following types of Routing Groups:

FXS Port Group: SETU VFXTH1616 supports 16 FXS Ports. You can create maximum 16 FXS Port routing groups. Each group can have up to 8 members and you can program only FXS Ports as members in FXS Port Groups.

FXO Port Group: SETU VFXTH1616 supports 16 FXO Ports. You can create maximum 16 FXO Port routing groups. Each group can have up to 8 members and you can program only FXO Ports as members in FXO Port Groups.

SIP Trunk Group: SETU VFXTH1616 supports 32 SIP Trunks. You can create maximum 9 SIP Trunk routing groups. Each group can have up to 9 members and you can program only SIP Trunks as members in SIP Trunk Groups.

SETU VFXTH1616 selects the routing group for routing the call as per routing mechanism programmed for the source port. In a group, selection of a member for routing the call can be done in one of the following two methods:

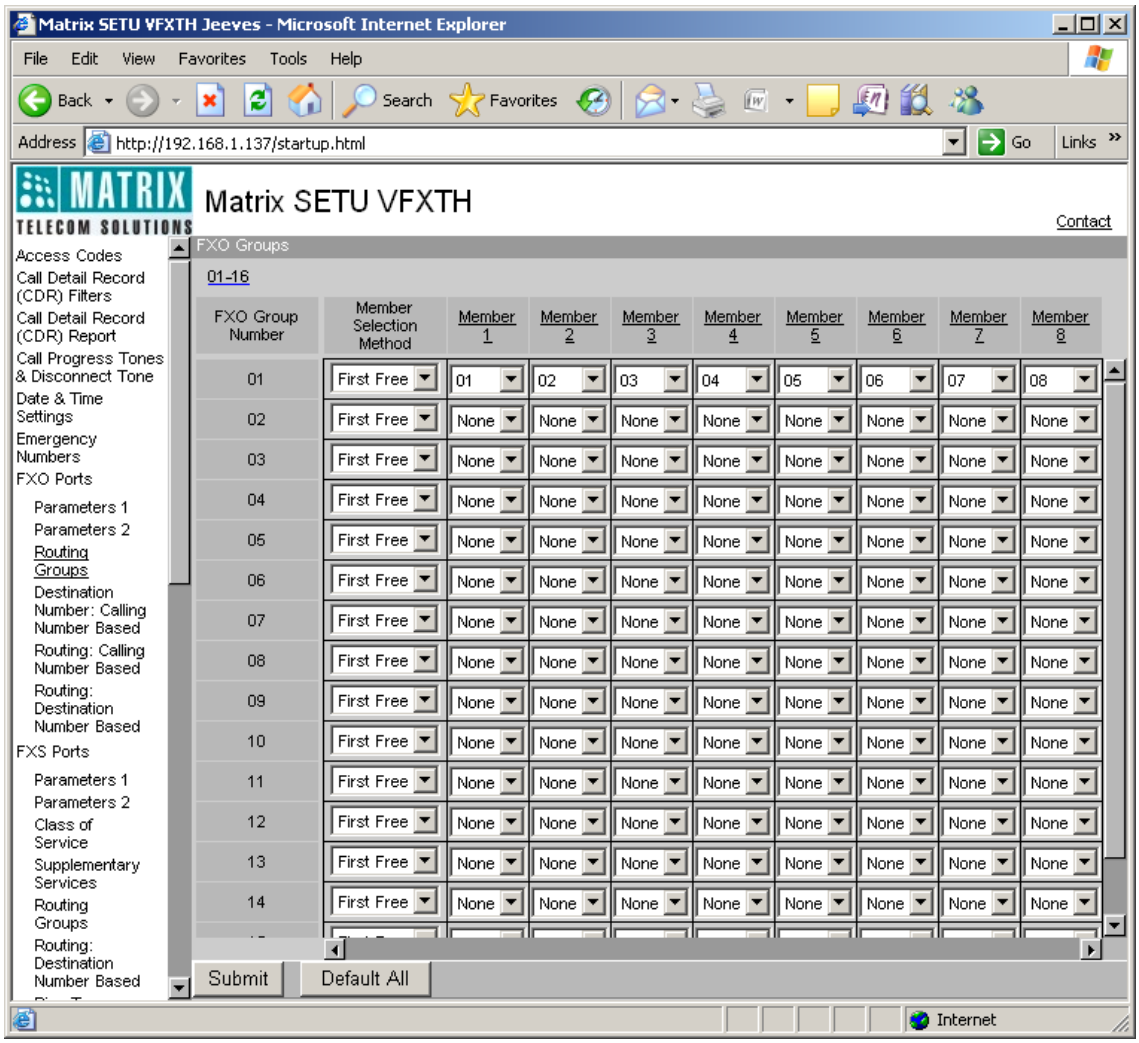
1. **First Free:** In this method, system selects first free member of the group to route the call.
2. **Rotation:** In this method, system selects (N+1) th member of the routing group to route the call, where N is the last member selected for routing the call.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer"Accessing Web JEEVES")

To program Routing Group for FXO Port, follow the steps given below:

- Click on '**FXO Groups**' link and form FXO Groups as shown below:

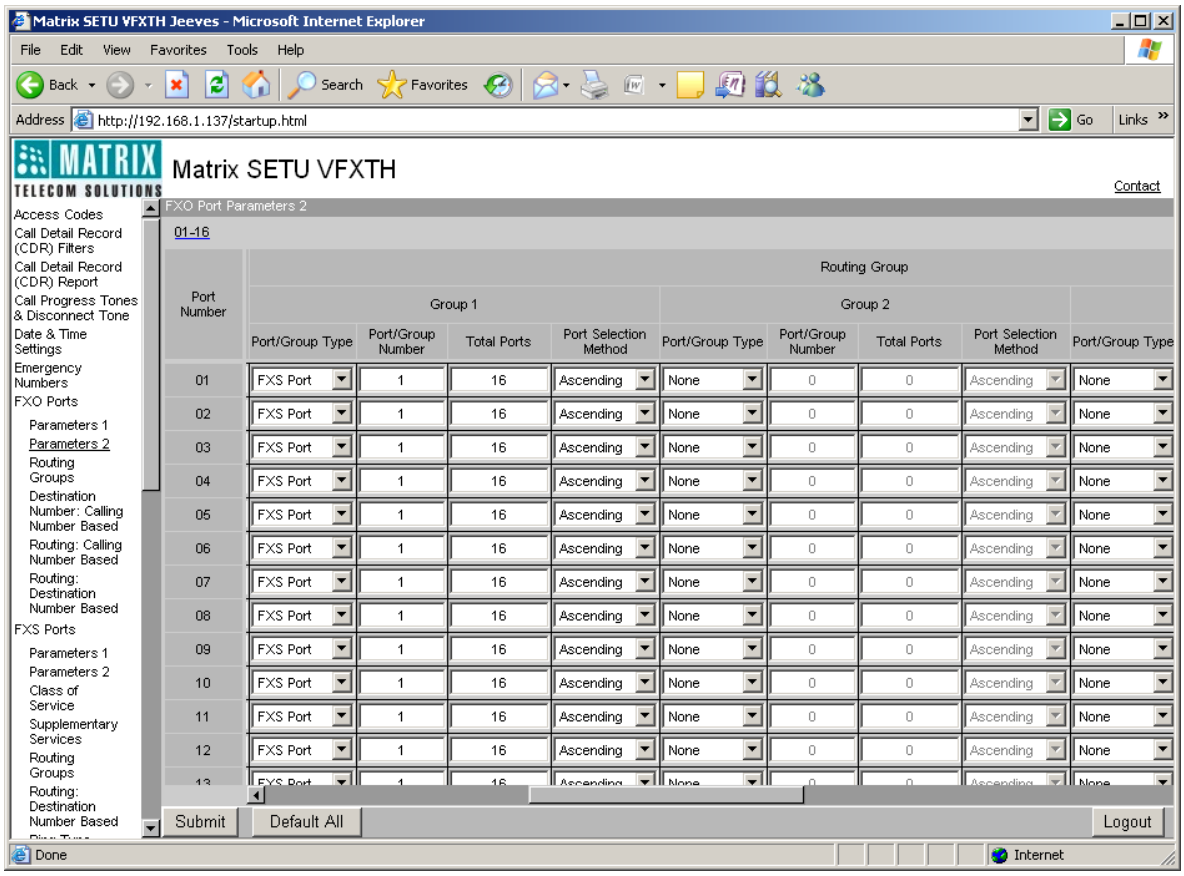


- **Member Selection Method:** Select a method for selecting a member for routing the call on the FXO port in this field. Member selection method can be either 'First Free' or 'Rotation'. By default, 'First Free' is selected.

- **Member (1 to 8):** Select desired FXO Port in each member field. When a call lands on FXO Port, SETU VFXTH1616 will route the call on any of these members as per member selection method programmed.

When Destination Port Determination Method selected for FXO Port is 'Fixed', program following parameters on the FXO Port parameter page.

- Click on '**FXO Port Parameters-2**'. In 'Routing Groups' table, program the following parameters.



For Group 1:

- **Port/Group Type:** Select desired port or group for routing the call originated on FXO Port. You can choose one of the following options:

  - None
  - FXS Port
  - FXO Port
  - SIP Trunk
  - FXS Group
  - FXO Group
  - SIP Group

  By default, 'FXS Port' is selected in Group 1 for all FXO Ports.

- **Port/Group Number:** Program Port Number here, if Port is selected in Port/Group Type field. Program Group Number, if Group is selected in Port/Group Type field. You may select from the following options:

  - 1-16 for FXS Port
  - 1-16 for FXO Port
  - 1-32 for SIP Trunk
  - 1-16 for FXS Group
  - 1-16 for FXO Group
  - 1-9 for SIP Group

  By default, Port number '01' is selected in Group 1 for all FXO Ports.

- **Total Ports:** Program total Ports to be checked while routing the call, if first port is found busy. You have following options:

  - 1-16 for FXO Port
  - 1-16 for FXS Port
  - 1-32 for SIP Trunk

  By default, it is set to '16' for all FXO Ports.

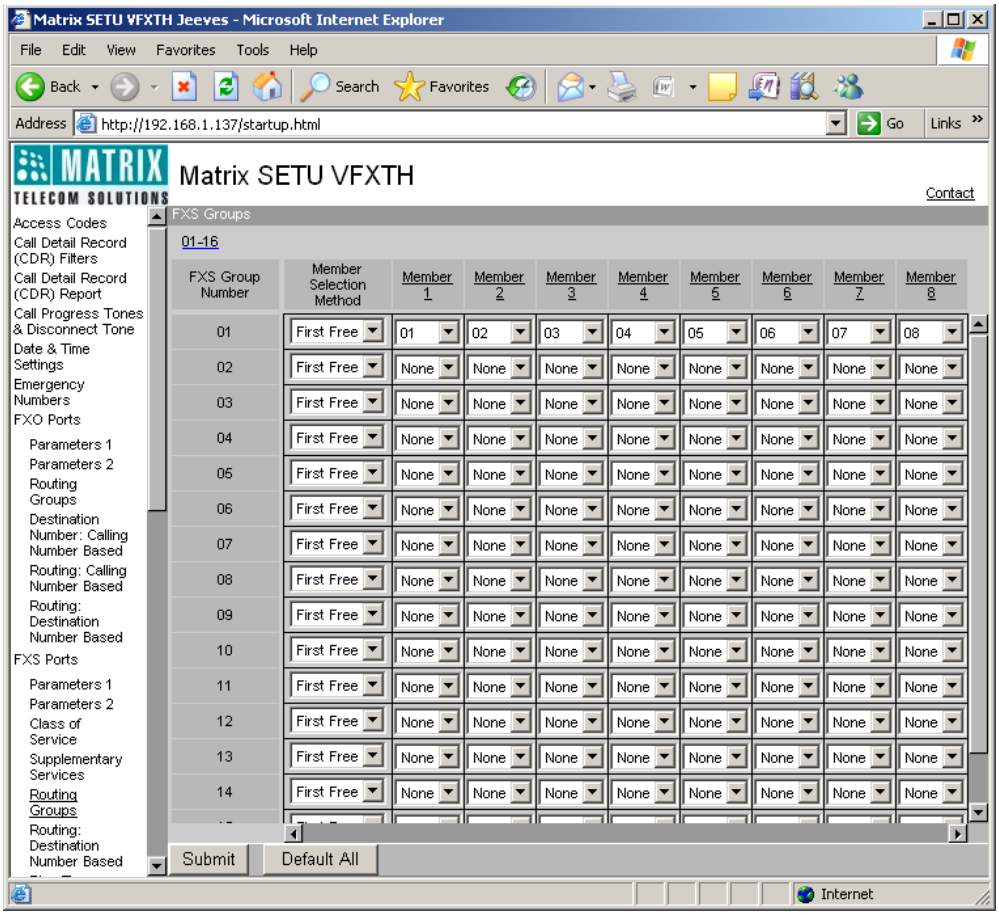
- **Port Selection Method:** Select the Port Selection method for routing the call to the destination port. SETU VFXTH1616 supports following port selection methods: Ascending and Descending. By default, it is 'Ascending' for all FXO Ports.

  When Destination Port Determination Method selected for FXO Port is 'Destination Number Based'; program Routing Groups parameters as shown above in FXO Port Routing: Destination Number Based table.

  When Destination Port Determination Method selected for FXO Port is 'Calling Number Based'; program Routing Groups parameters as shown above in FXO Port Routing: Calling Number Based table.

To program Routing Group for FXS Port, follow the steps given below:

- Click on '**FXS Groups**' link and form FXS Groups as shown below:

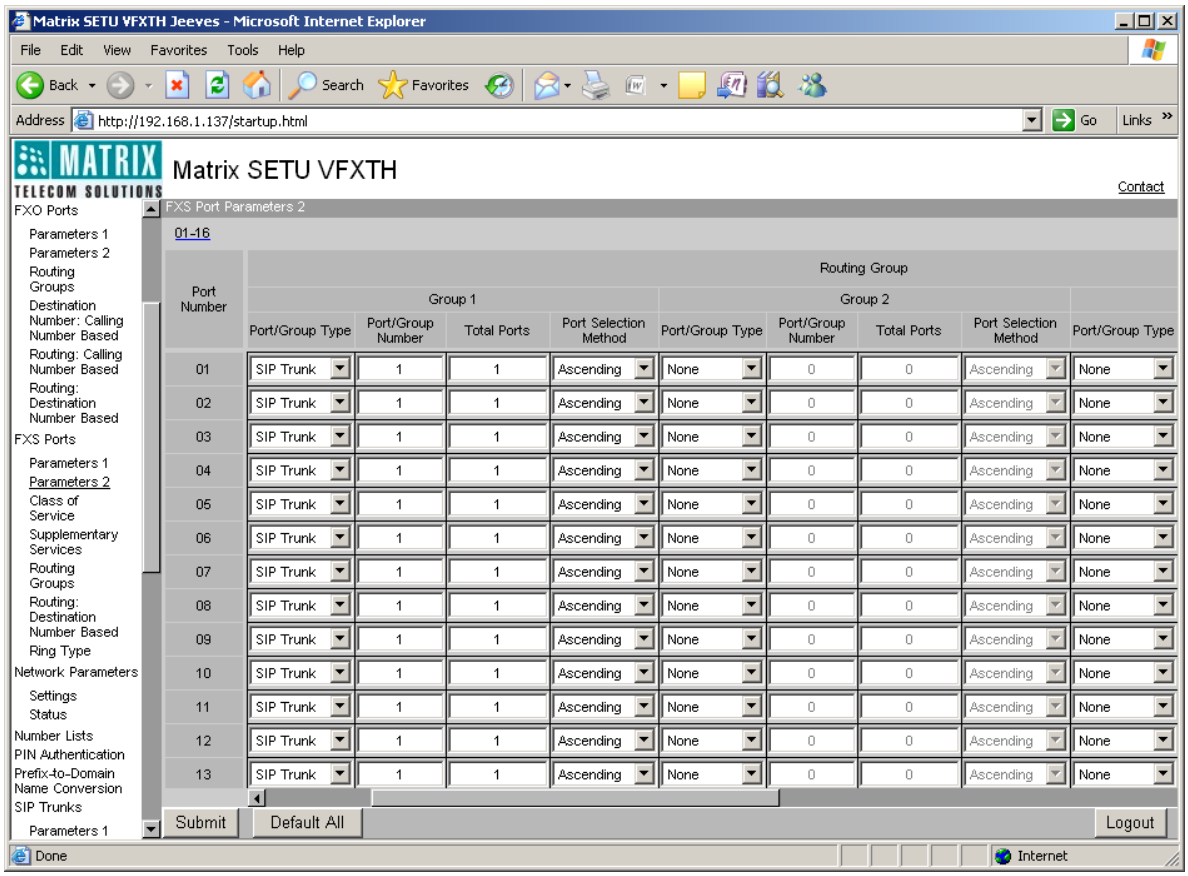


- **Member Selection Method:** In this field, select the method for selecting a member from a group for routing the call on FXS port. Member selection method can be either 'First Free' or 'Rotation'. By default, 'First Free' is selected.

- **Member (1 to 8)**: Select desired FXS Port in each member field. When a call lands on FXS Port, SETU VFXTH1616 will route the call on any of these members as per member selection method programmed.

When Destination Port Determination Method selected for FXS Port is 'Fixed', program following parameters on the FXS Port parameter page.

- Click on '**FXS Port Parameters-2**'. In 'Routing Groups' table, program the following parameters.



For Group 1:

- **Port/Group Type:** Select desired port or group for routing the call originated on FXS Port. You can choose one of the following options:

  - None
  - FXS Port
  - FXO Port
  - SIP Trunk
  - FXS Group
  - FXO Group
  - SIP Group

  By default, 'SIP Trunk' is selected in Group 1 for all FXS Ports.

- **Port/Group Number:** Program Port Number here, if Port is selected in Port/Group Type field. Program Group Number, if Group is selected in Port/Group Type field. You may select from the following options:

- 1-16 for FXS Port
- 1-16 for FXO Port
- 1-32 for SIP Trunk
- 1-16 for FXS Group
- 1-16 for FXO Group
- 1-9 for SIP Group

By default, Port number '1' is selected in Group 1 for all FXS Ports.

- **Total Ports:** Program total Ports to be checked while routing the call, if first port selected is found busy. You have following options:

  - 1-16 for FXO Port
  - 1-16 for FXS Port
  - 1-32 for SIP Trunk

  By default, it is set to '1' in Group 1 for all FXS Ports.

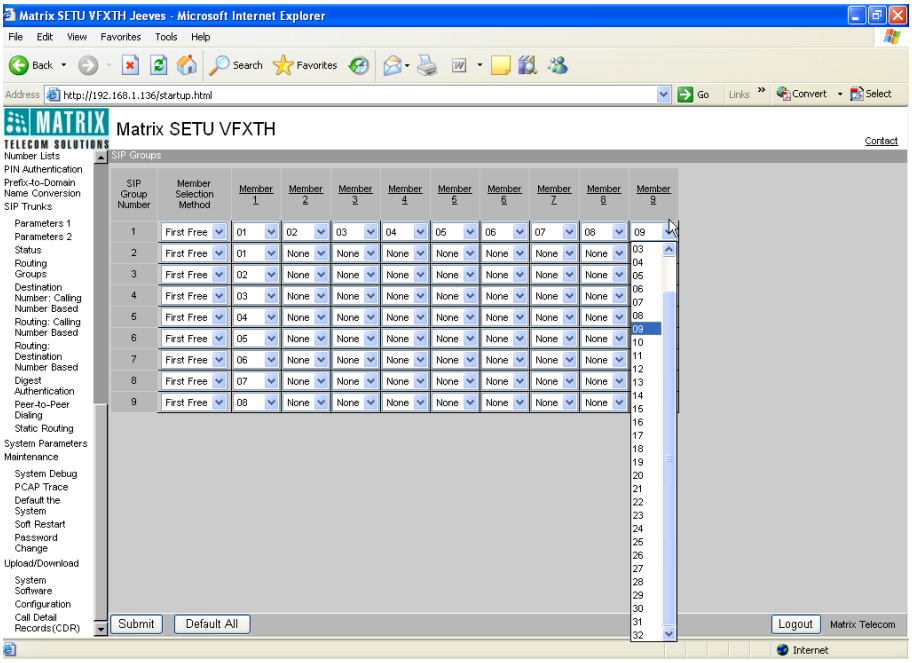
- **Port Selection Method:** Select the Port Selection method for routing the call to the destination port. SETU VFXTH1616 supports following port selection methods: Ascending and Descending. By default, it is 'Ascending' for all FXS Ports.

- Similarly program Routing Group parameters for Group 2 and Group 3 as shown above.

  When Destination Port Determination Method selected for FXS Port is 'Destination Number Based'; program 'Routing Groups' parameters as shown above in FXS Port Routing: Destination Number Based table.

To program Routing Group for SIP Trunk, follow the steps given below:

- Click on '**SIP Groups**' link and form SIP Groups as shown below:

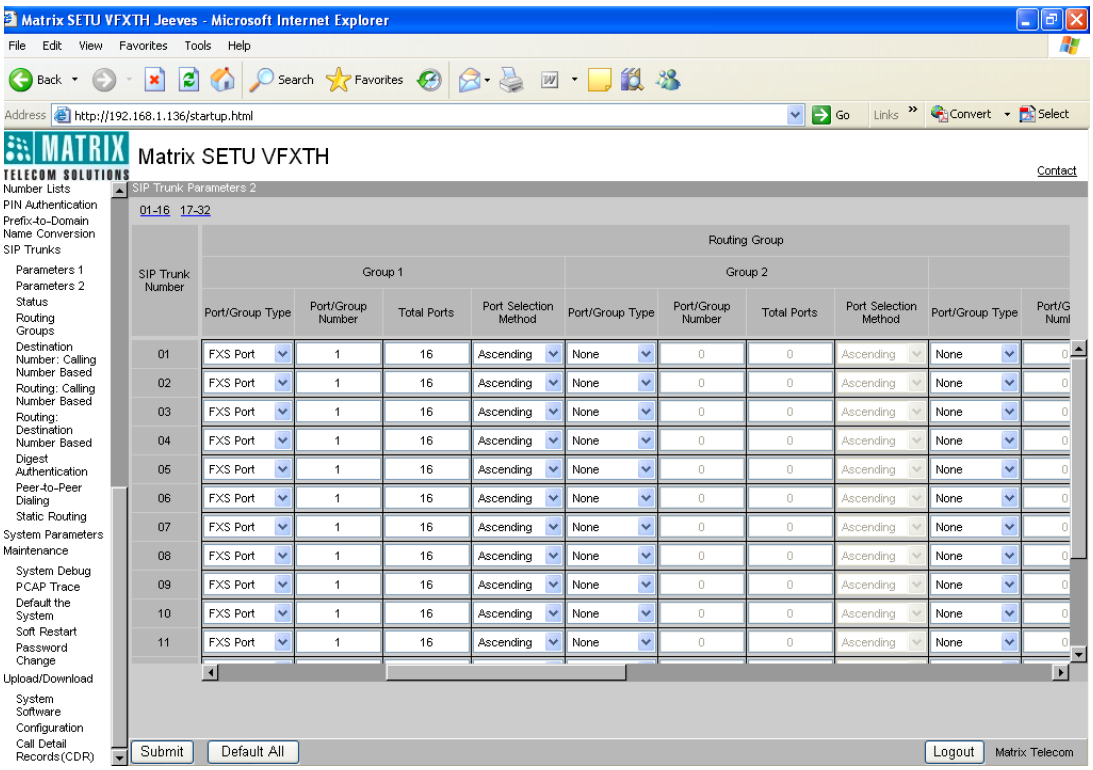


- **Member Selection Method:** Select a method for selecting a member for routing the call on the SIP Trunk in this field. Member selection method can be either 'First Free' or 'Rotation'. By default, 'First Free' is selected.

- **Member (1 to 9):** Select desired SIP Trunk in each member field. When a call lands on SIP Trunk, SETU VFXTH1616 will route the call on any of these members as per member selection method programmed.

  When Destination Port Determination Method selected for SIP Trunk is 'Fixed', program following parameters on the SIP Trunk parameter page.

- Click on '**SIP Trunk Parameters-2**'. In 'Routing Groups' table, program the following parameters.



For Group 1:

- **Port/Group Type:** Select desired port or group for routing the call originated on SIP Trunk. You can choose one of the following options:

  - None
  - FXS Port
  - FXO Port
  - SIP Trunk
  - FXS Group
  - FXO Group
  - SIP Group

  By default, 'FXS Port' is selected in Group 1 for all SIP Trunks.

- **Port/Group Number:** Program Port Number here, if Port is selected in Port/Group Type field. Program Group Number, if Group is selected in Port/Group Type field. You may select from the following options:

  - 1-16 for FXS Port
  - 1-16 for FXO Port
  - 1-32 for SIP Trunk

- 1-16 for FXS Group
- 1-16 for FXO Group
- 1-9 for SIP Group

By default, Port number '1' is selected in Group 1 for all SIP Trunks.

- **Total Ports:** Program total Ports to be checked while routing the call, if the first port selected is found busy. You have following options:

  - 1-16 for FXO Port
  - 1-16 for FXS Port
  - 1-32 for SIP Trunk

  By default, it is set to '2' for all SIP Trunks.

- **Port Selection Method:** Select the Port Selection method for routing the call to the destination port in this field. SETU VFXTH1616 supports following port selection methods: Ascending and Descending. By default, it is 'Ascending' for all SIP Trunks.

- Similarly program Routing Group parameters for Group 2 and Group 3 as shown above.

  When Destination Port Determination Method selected for SIP Trunk is 'Destination Number Based'; program Routing Groups parameters as shown above in SIP Trunk Routing: Destination Number Based table.

  When Destination Port Determination Method selected for SIP Trunk is 'Calling Number Based'; program Routing Groups parameters as shown above in SIP Trunk Routing: Calling Number Based table.

a. *Destination port (Member) from a routing group is selected as per member selection method assigned to that port group.*

b. *If only one member is programmed in a selected routing group and that member is found busy then SETU VFXTH1616 selects another routing group to route the call.*

c. *If all members in all the groups are found busy then SETU VFXTH1616 starts 'Routing Group Busy Wait Timer' and again checks for the first free member in all the groups for routing the call.*

d. *In case the free member is not found before the expiry of 'Routing Group Busy Wait Timer' then SETU VFXTH1616 will disconnect the call after giving busy tone.*

e. *In case of 'Blind Transfer', if free member is not found in a routing group then SETU VFXTH1616 will give error tone to the user.*

Relevant Topics:

1. “Emergency Number Dialing” 66
2. “Port Parameters-FXO” 104
3. “Port Parameters-FXS” 112
4. “Port Parameters-SIP” 117
5. “Routing Mechanism on FXO” 143
6. “Routing Mechanism on FXS” 155
7. “Routing Mechanism on SIP” 161

Routing Mechanism on FXO

From SETU VFXTH's perspective, a call is originated on a port and terminated on a port. A call received on FXO Port can be routed through any of the other ports of the system, to any number.

When a call is received on a FXO Port (originates on the FXO Port), two decisions are taken related to routing the call. Where to route this call and through which port to route this call? Two parameters are defined in SETU VFXTH for this purpose viz. 'Destination Number Determination Method' and 'Destination Port Determination Method'

Destination Number Determination Method:

SETU VFXTH supports four different methods of determining the destination number:

1. **Destination Number Not Required** - In this method, destination number is not required. The call initiated on FXO Port is simply placed on a fixed port determined as per Destination Port Determination Method for the FXO Port. No specific number is to be dialed in this method.

2. **Fixed Destination Number** - This method is applicable when a call received on the FXO Port is to be routed to a fix number, say a Global Technical Support number. In this case, a fixed destination number is programmed for the FXO Port, to which the call is placed.

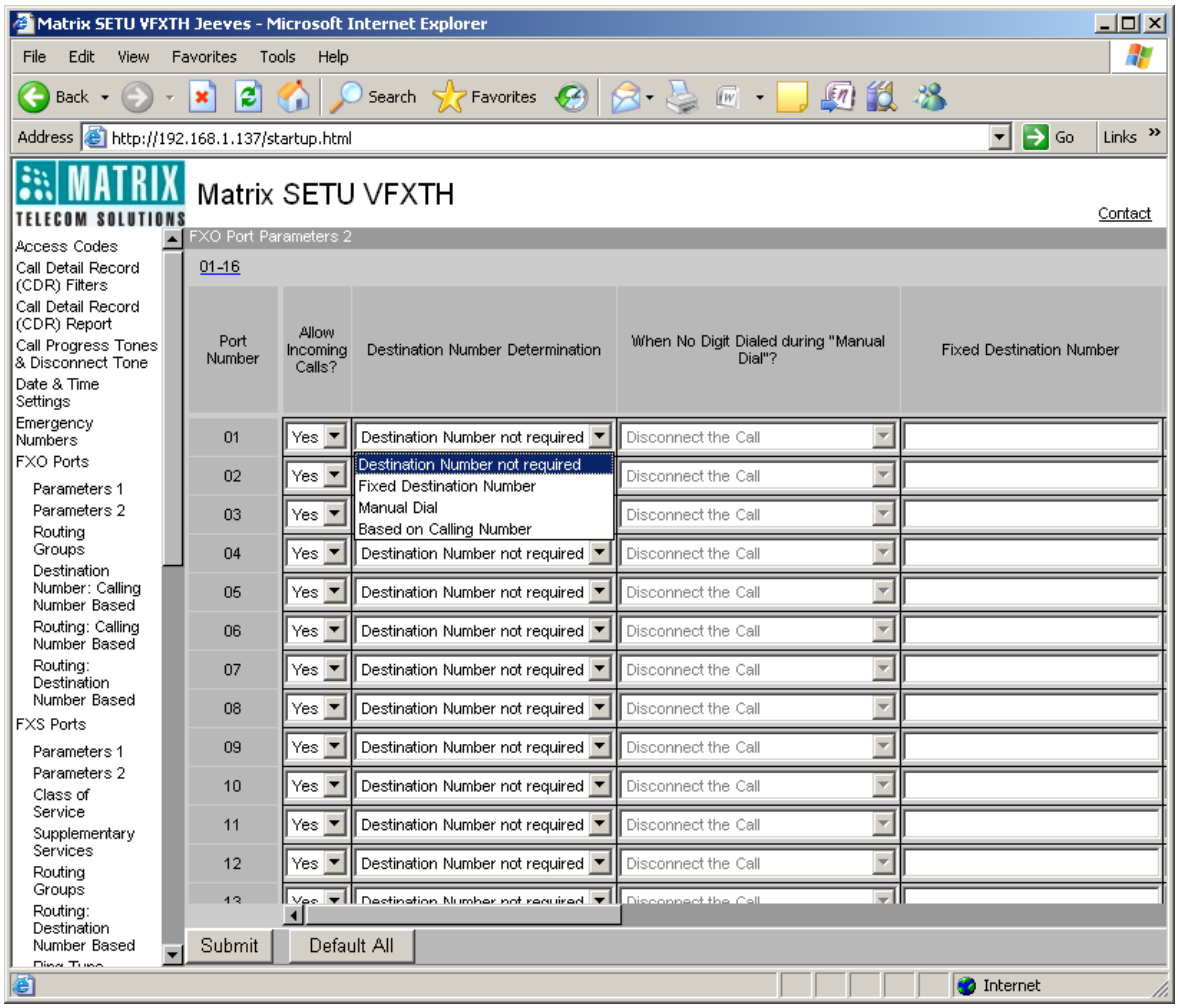
3. **Based on Calling Number** - This method is applicable when a call is to be routed to a specific number depending upon the calling party's number. For e.g. a call received from Chairman's wife should be routed to the chairman's international mobile number through SETU VFXTH over SIP.

4. **Manual Dial** - This method is applicable when the caller is given freedom to dial the number of his choice. (PIN authentication can be enabled/disabled in this case on the FXO Port)

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**FXO Port Parameters 2**' and select Destination Number Determination Method for each FXO Ports. By default, 'Destination Number Not Required' is selected.



- If Destination Number Determination Method selected is "Fixed Destination Number" then program destination number for the FXO Port in 'Fixed Destination Number' field. Destination number can be of maximum 24 digits. Valid digits are 0 to 9, *, #, (.) dot. By default, it is blank.

Fixed destination number should be less than or equal to the maximum number of digits allowed to be dialed on the FXO Port.

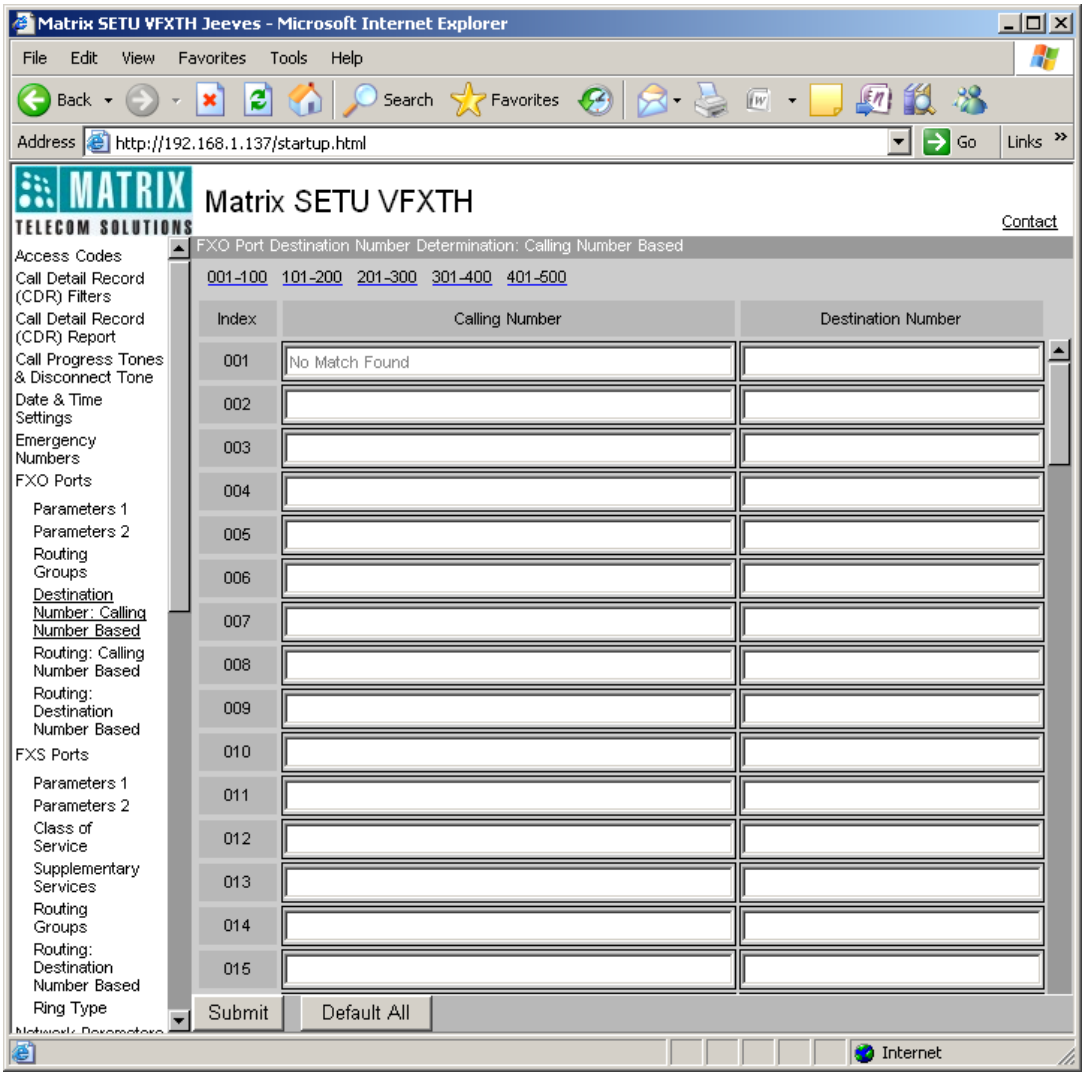
- If Destination Number Determination Method selected for FXO Port is "Manual Dial" then program whether to 'Disconnect the Call' or 'Route the call to Fixed Destination number' in 'When No Digit Dialed during Manual Dial' field, in case no number is dialed within First Digit Wait Timer.

- If Destination Number Determination Method selected for FXO Port is "Based on Calling Number" then click on 'FXO Port Destination Number Determination: Calling Number Based' link and program following parameters:

  - **Index:** In this field index number is given. Total 500 entries can be made in this table starting from 001 to 500.

- **Calling Number:** Program calling party number in this field. It can be of maximum 24 characters. All ASCII characters are allowed. By default, first entry is used for 'No Match Found' case and other entries are blank.

- **Destination Number:** Program destination number in this field. Destination number can be of maximum 24 characters. Digits 0 to 9, #, * and (.) dot are allowed. By default, it is blank.



If destination number to be dialed out is an IP Address then SETU VFXTH will not check Destination Port Determination Method. Instead it will route the call using the SIP Trunk Group programmed for IP Dialing. (Refer "IP Dialing" feature for more details)

Destination Port Determination Method:

SETU VFXTH supports three different methods of determining the destination port:

1. **Fixed** - This method is applicable when calls received on a FXO Port (originating on a FXO Port) are to be routed through fixed port. This fixed port can be another FXO Port or FXS Port or SIP Trunk.

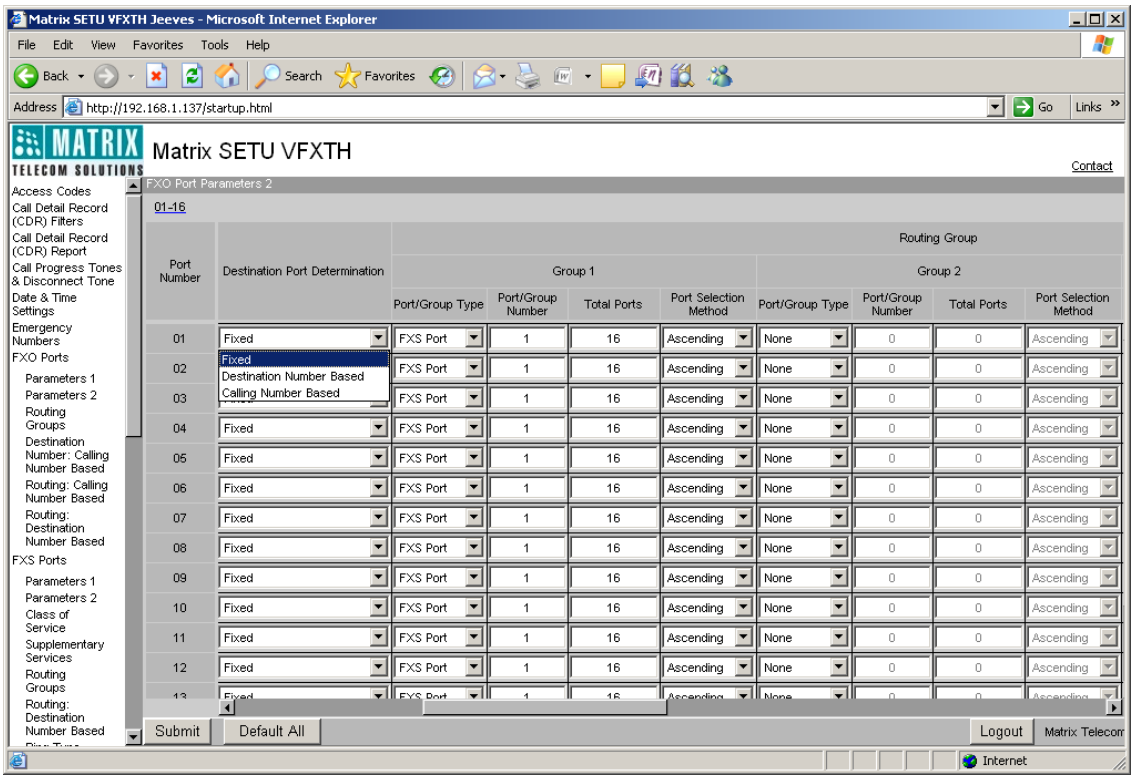
2. **Destination Number Based** - This method is applicable when the calls received on FXO Port are to be routed on the basis of the number dialed by the caller. To use this method, 'FXO Port Routing: Destination

Number Based' table should be programmed. (Please refer "Logic of FXO Port Routing: Destination Number Based" at the end of this topic.)

3. **Calling Number Based** - This method is applicable when the calls received on the FXO Port are to be routed through a specific port (another FXO Port, SIP Trunk or FXS Port) based on the Calling Party's Number. (Please refer "Logic of FXO Port Routing: Calling Number Based" at the end of this topic.)

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")
- Click on '**FXO Port Parameters 2**' and select destination port determination method for each FXO Ports. By default, 'Fixed' is selected.
- If Destination Port Determination Method selected is "Fixed" then program following parameters of Routing Groups (1, 2 and 3):



- **Port/Group Type:** Select desired port or group for routing the call originated on FXO Port in this field for each FXO Ports. You can choose one of the following options:
  - None
  - FXS Port
  - SIP Trunk
  - FXO Port
  - FXS Group
  - FXO Group
  - SIP Group

  By default, 'FXS Port' is selected for all FXO Ports.

- **Port/Group Number:** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

  - For FXS and FXO Port: 1-16
  - For SIP Trunk: 1-32
  - For FXS and FXO Group: 1-16
  - For SIP Group: 1-9

  By default, Port number '1' is selected for all FXO Ports.

- ***Total Ports:*** Program total ports to be checked while routing the call, if first selected port is busy. You have following options:

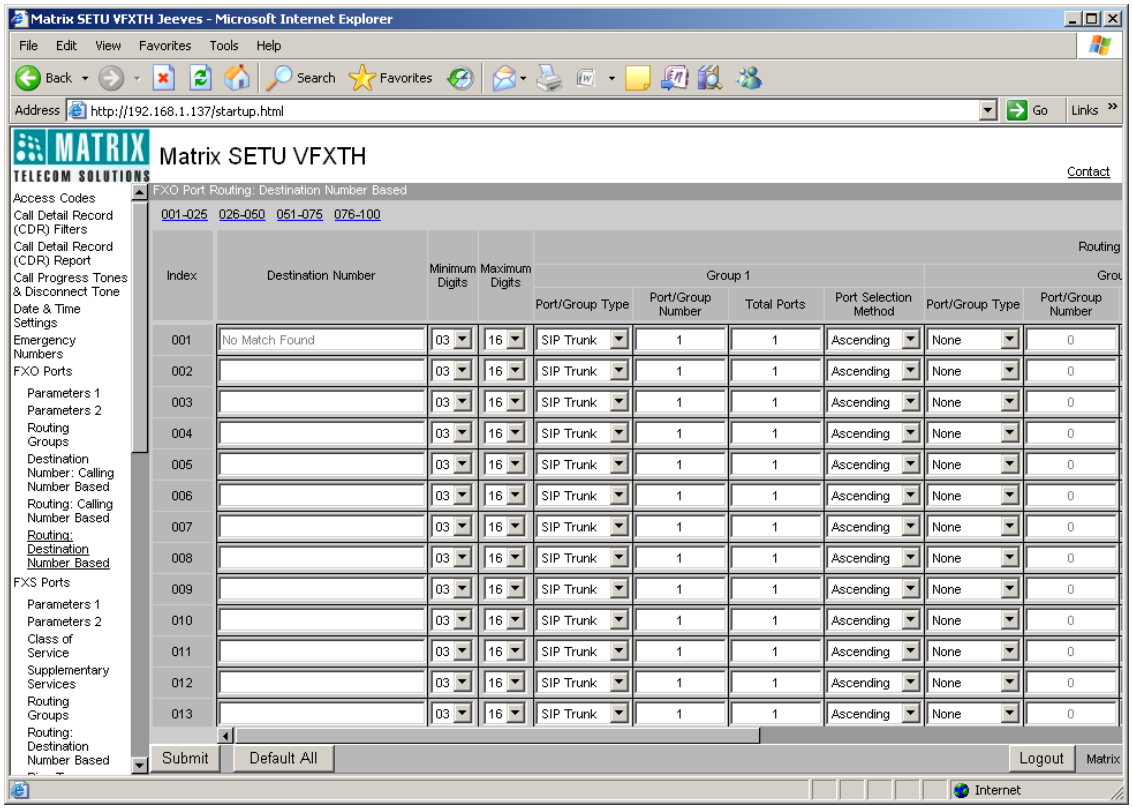
  - For FXO and FXS Port: 1-16
  - For SIP Trunk: 1-32

  By default, it is '16' for all FXO Ports.

- ***Port Selection Method:*** Select desired port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

- **CLI number on FXS Port:** This field is applicable only when group type 'FXS' is used for routing the call. (Please refer "Number Presentation on FXS Port" for more details)

- If Destination Port Determination Method selected is "Destination Number Based" then click on '**FXO Port Routing: Destination Number Based**' link and program the following parameters:



- **Index:** In this field index number is given. Total 500 entries can be made in this table starting from 001 to 500.

- **Destination Number:** Program frequently called numbers in this field. Destination number can be of maximum 24 digits. Digits 0 to 9, #, *, (.) dot are allowed. By default, it is blank.

- **Minimum Length:** Program minimum length of destination number required to be dialed by the user in this field. If the destination number dialed by the user is less than the minimum length programmed for that FXO Port then SETU VFXTH will reject the call. Range of minimum length of destination number is 01 to 24. By default, it is 03.

- **Maximum Length:** Program maximum length of destination number that can be dialed by the user in this field. If user dials a number greater than the number programmed for that FXO Port then SETU VFXTH shall strip off the extra digits and route the call. Range of maximum length of destination number is 01 to 24. By default, it is 16.

- **Routing Group:** Program Routing Groups (1, 2 and 3) parameters for Destination Port Determination Method: Destination Number Based.

  - ***Port/Group Type:*** Select desired port or group for routing the call originated on FXO Port in this field for each FXO Ports. You can choose one of the following options:

    - None
    - FXS Port
    - SIP Trunk
    - FXO Port
    - FXS Group
    - FXO Group
    - SIP Group

    By default, 'Port/Group Type' is set to 'SIP Trunk'.

  - ***Port/Group Number:*** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

    - For FXS and FXO Port: 1-16
    - For SIP Trunk: 1-32
    - For FXS and FXO Group: 1-16
    - For SIP Group: 1-9

    By default, Port number '1' is selected.

  - ***Total Ports:*** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

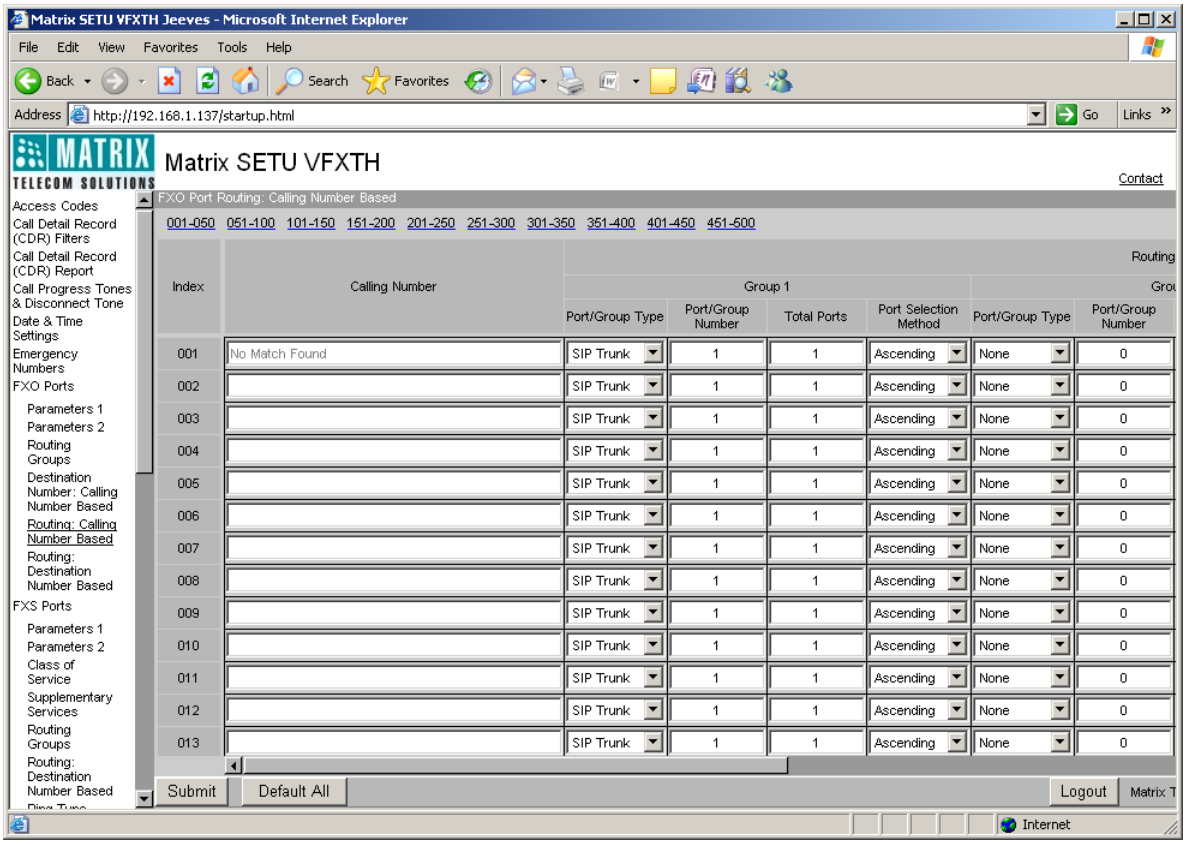
    - For FXO and FXS Port: 1-16
    - For SIP Trunk: 1-9

    By default, it is set to '1'.

  - ***Port Selection Method:*** Select desired port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

- **CLI number on FXS Port:** This field is applicable only when group type 'FXS' is used for routing the call. (Please refer "Number Presentation on FXS Port" for more details)

- If Destination Port Determination Method selected is "Calling Number Based" then click on '**FXO Port Routing: Calling Number Based**' link and program the following parameters:



- **Index:** In this field index number is given. Total 500 entries can be made in this table starting from 001 to 500.

- **Calling Number:** Program number of the calling party, who often calls SETU VFXTH, in this field. Calling party's number can be of maximum 24 digits. All ASCII characters are allowed. By default, it is blank.

- **Routing Group:** Program Routing Groups (1, 2 and 3) parameters for Destination Port Determination Method: Calling Number Based.

  - ***Port/Group Type:*** Select desired port or group for routing the call originated on FXO Port in this field. You can choose one of the following options:

    - None
    - FXS Port
    - SIP Trunk
    - FXO Port
    - FXS Group
    - FXO Group
    - SIP Group

    By default, 'Port/Group Type' is set to 'SIP Trunk'.

- ***Port/Group Number:*** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

  - For FXS and FXO Port: 01-16
  - For SIP Trunk: 1-32
  - For FXS and FXO Group: 01-16
  - For SIP Group: 1-9

  By default, Port number '1' is selected.

- ***Total Ports:*** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

  - For FXO and FXS Port: 01-16
  - For SIP Trunk: 1-9

  By default, it is set to '1'.

- ***Port Selection Method:*** Select desired Port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

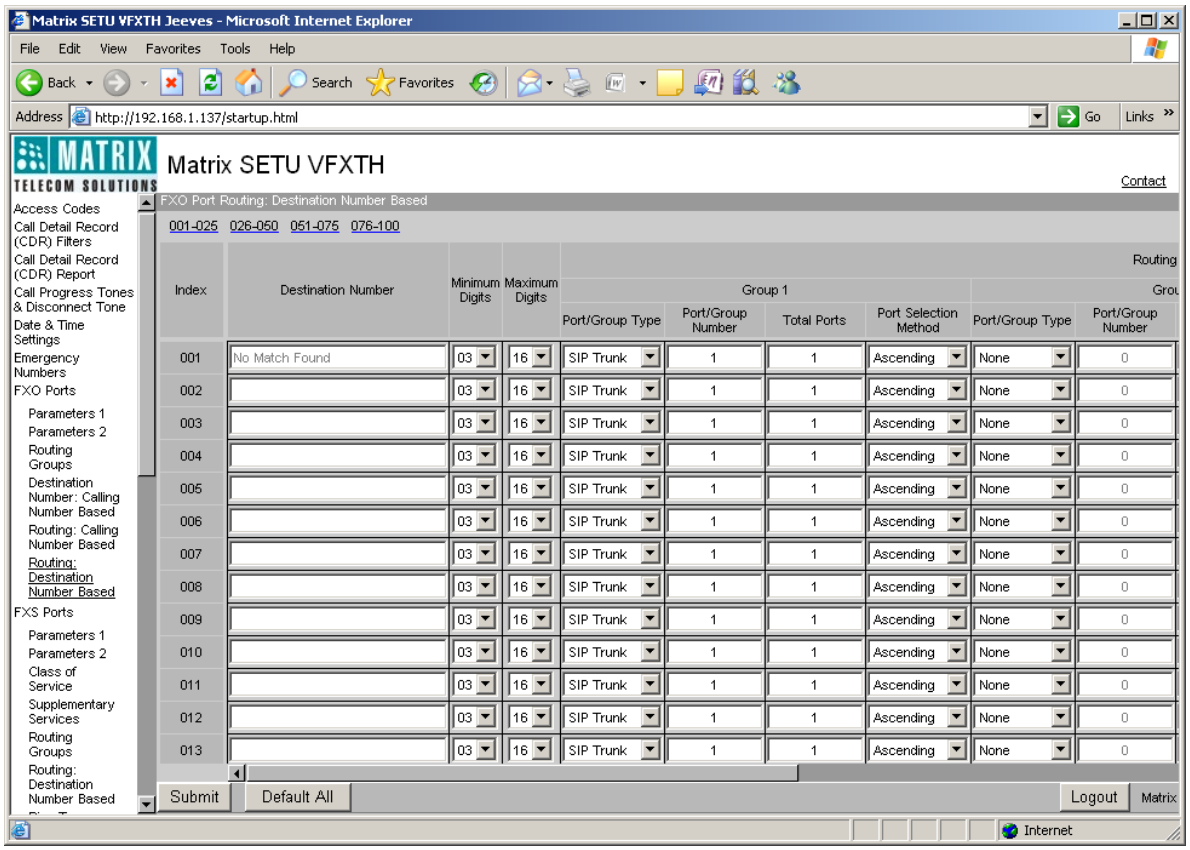
- **CLI number on FXS Port:** This field is applicable only when group type 'FXS' is used for routing the call. (Please refer "Number Presentation on FXS Port" for more details)

Improper selection of Destination Port Determination Method and Destination Number Determination Method may result in improper and undesirable routing of the calls.

Logic of FXO Port Routing: Destination Number Based

In destination number based routing; call is routed to the destination port on the basis of the destination number. FXO Port Routing: Destination Number based table supports 100 entries. Each entry has following parameters:

- Destination Number
- Minimum Length
- Maximum Length
- Routing Groups: Group1, Group2 and Group3.
- CLI Number on FXS Port



Let us understand FXO Port Routing: Destination Number Based logic with the help of an example:

Suppose following entries are made in the FXO Port Routing: Destination Number Based table:

Index	Destination Number	Min. Length	Max. Length	Routing Groups						CLI Number on FXS Port
				Group 1		Group 2		Group 3		
				Group Type	Group No.	Group Type	Group No.	Group Type	Group No.	
001	No Match Found	2	16	SIP	1	None	0	None	0	Received Calling Party
002	2	7	7	FXO	1	None	0	None	0	Received Calling Party
003	28	7	8	FXO	2	None	0	None	0	Received Calling Party
004	98	3	15	FXO	1	None	0	None	0	Received Calling Party
005	9	5	10	SIP	2	None	0	None	0	Received Calling Party
006	09	10	12	SIP	3	None	0	None	0	Received Calling Party

SETU VFXTH checks 'FXO Port Routing: Destination Number Based' table when the source port is FXO, Destination Number Determination Method is Manual Dial and Destination Port Determination Method is

Destination Number Based. Best Match logic will be applied, while routing the call to the destination port using this table.

Consider the following cases:

Case 1: Number dialed on FXO port is 2830555. In this case, dialed number best matches index 002 and 003 but it best matches the minimum and maximum length criteria of index 002 therefore SETU VFXTH will route the call as per the programming done in index 002.

Case 2: Number dialed on FXO port is 6530300. In this case, dialed number does not match any of the entry and thus SETU VFXTH checks minimum and maximum length criteria of 'No Match Found' entry i.e. index 001. As the dialed number matches the length criteria of 'No Match Found', gateway routes the call as per the programming done in index 001.

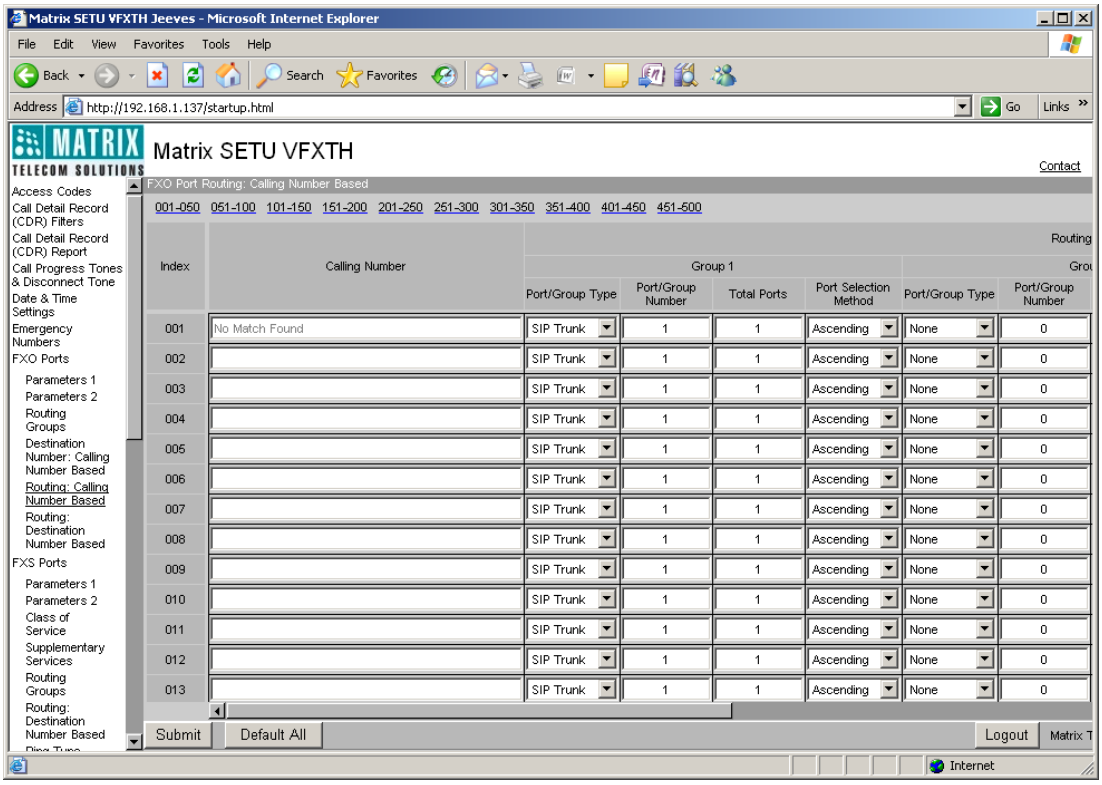
Case 3: Number dialed on FXO port is 09898098. In this case, dialed number best matches destination number programmed in index 006 but it does not satisfies minimum and maximum length criteria of that entry and therefore the gateway rejects the call and give error tone to the user.

Case 4: Number dialed on FXO port is 98980148240. Dialed number matches two entries: entry number 004 and 005. But it nearly matches the maximum and minimum length criteria of entry number 004. SETU VFXTH will strip off the last digit of the dialed number and will dial out the number as per the programming done in index 005.

Logic of FXO Port Routing: Calling Number Based

In calling number based routing the call is routed to the destination port on the basis of the calling number. FXO Port Routing: Calling Number based table supports 500 entries. Each entry has following parameters:

- Calling Number
- Routing Groups: Group1, Group2 and Group3.
- CLI Number on FXS Port



Let us understand FXO Port Routing: Calling Number Based logic with the help of an example:

Suppose following entries are made in the FXO Port Routing: Calling Number Based table:

Index	Calling Number	Routing Groups						CLI Number on FXS Port
		Group 1		Group 2		Group 3		
		Group Type	Group No.	Group Type	Group No.	Group Type	Group No.	
001	No Match Found	FXO	1	None	0	None	0	Received Calling Party
002	02	FXO	1	None	0	None	0	Received Calling Party
003	02652630555	FXO	2	None	0	None	0	Received Calling Party
004	02668263173	FXO	3	None	0	None	0	Received Calling Party
005	919898123456	SIP	1	None	0	None	0	Received Calling Party
006	919426594265	SIP	1	None	0	None	0	Received Calling Party

SE has programmed the Calling Number in 'FXO Port Routing: Calling Number Based' table in such format as supported by the PSTN exchange. Now if the call land on the FXO Port of SETU VFXTH and Destination Port Determination Method programmed by the SE is 'Calling Number Based' then SETU VFXTH will check 'FXO Port Routing: Calling Number Based' table and routes the call as per the programming done in this table.

Case 1: Call is received from 02668263173. Steps followed by SETU VFXTH are as follow:

a. Matches the calling number received with the 'FXO Port Routing: Calling Number Based' table.

b. As the calling number received perfectly matches with the entry made at index number 004, it routes the call to the destination port as per the programming done in index 004.

Case 2: Call is received from 6530300. Steps followed by SETU VFXTH are as follow:

a. Matches the calling number received with the 'FXO Port Routing Calling Number Based' table.

b. As the calling number received does not matches any of the entry of the table programmed, it routes the call as per the programming done for 'No Match Found' case.

Case 3: Call is received from 02669260001. Steps followed by SETU VFXTH are as follow:

a. Matches the calling number received with the 'FXO Port Routing: Calling Number Based' table.

b. The calling number best matches the entry number 002 and thus SETU VFXTH routes the call as per programming done for index 002.

SETU VFXTH applies best match logic while comparing the calling party's number with the 'FXO Port Routing: Calling Number Based' table to route the call, as shown in the above example.

SE should program the calling number in the 'FXO Port Calling Number Based Routing' table in such a way as supported by the PSTN exchange.

Relevant Topics:

1. “Call Processing” 47
2. “Number Presentation on FXS Port” 87
3. “Port Parameters-FXO” 104
4. “Routing Group” 135

Routing Mechanism on FXS

From SETU VFXTH's perspective, a call originates on a port and terminates on a port. A call from FXS Port of the SETU VFXTH can be routed through any of the other ports of SETU VFXTH, to any number.

When a call is made from FXS Port (originates on the FXS Port), two decisions are taken related to routing the call. Where to route this call and through which port to route this call? Two parameters are defined in SETU VFXTH for this purpose viz. 'Destination Number Determination Method' and 'Destination Port Determination Method'.

Destination Number Determination Method:

SETU VFXTH supports only 'Manual Dial' method for determining the destination number for calls originated on FXS Port.

- **Manual Dial** - In this method, the caller is given freedom to dial his desired number. (PIN authentication can be enabled/ disabled on FXS Ports.)

If destination number to be dialed out is an IP Address then SETU VFXTH will not check Destination Port Determination Method. Instead it will route the call using the SIP Trunk Group programmed for IP Dialing. (Please refer "IP Dialing" feature for more details)

Destination Port Determination Method:

SETU VFXTH supports two different methods of determining the destination port for the calls originated on FXS Port viz.

1. **Fixed** - In this method, calls received on a FXS Port (originating on a FXS Port) are routed through fixed port.

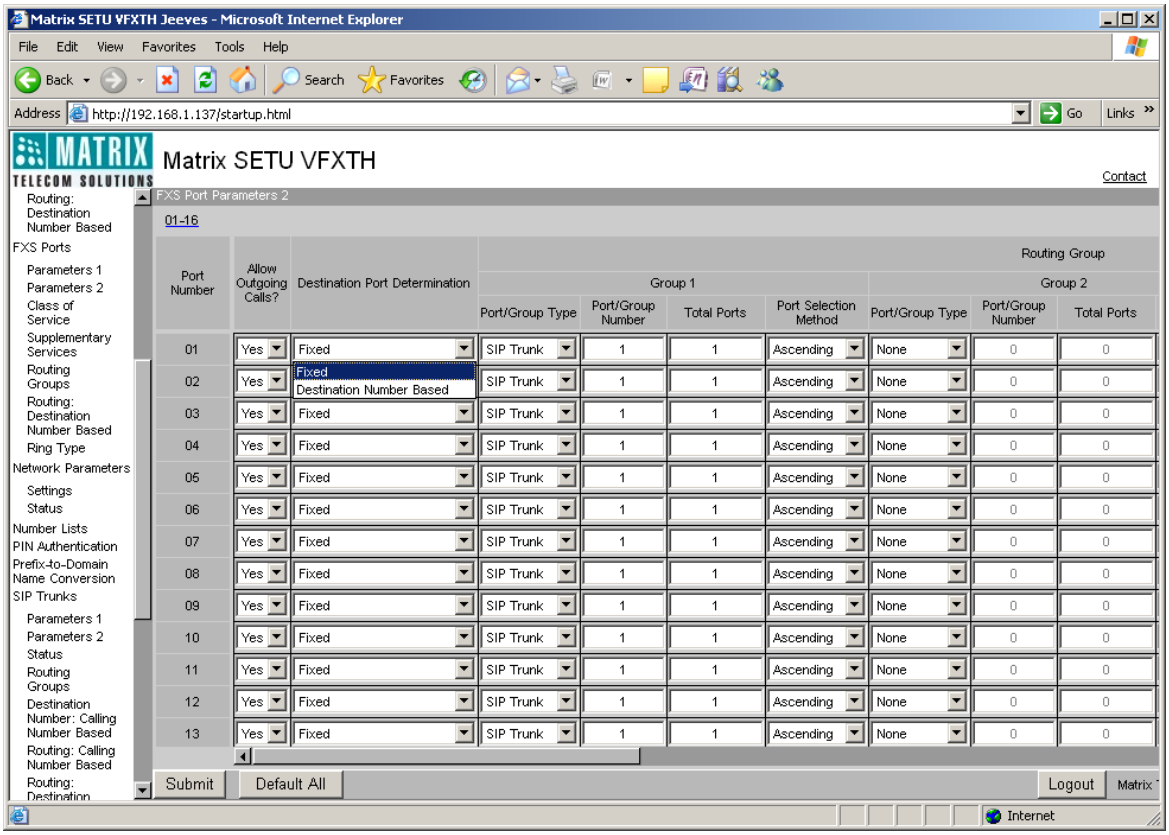
2. **Destination Number Based** - In this method, calls received on FXS Port are routed on the basis of the number dialed by the caller. To use this method, 'FXS Port Routing: Destination Number Based' table should be programmed. (Please refer "Logic of FXS Port Routing: Destination Number Based" at the end of this topic.)

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")

- Click on '**FXS Port Parameters 2**' and select Destination Port Determination Method for each FXS Ports. Two methods are supported viz. 'Fixed' and 'Destination Number Based'. By default, 'Fixed' is selected.

- If Destination Port Determination Method selected is "Fixed" then program following parameters of Routing Groups (1, 2 and 3):



- **Port/Group Type:** Select desired port or group for routing the call originated on FXS Port in this field for each FXS Ports. You can choose one of the following options:

  - None
  - FXS Port
  - FXO Port
  - SIP Trunk
  - FXS Group
  - FXO Group
  - SIP Group

  By default, 'SIP Trunk' is selected for all FXS Ports.

- **Port/Group Number:** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

  - For FXS and FXO Port: 01-16
  - For SIP Trunk: 1-32
  - For FXS and FXO Group: 01-16
  - For SIP Group: 1-9

  By default, Port number '1' is selected for all FXS Ports.

- **Total Ports:** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

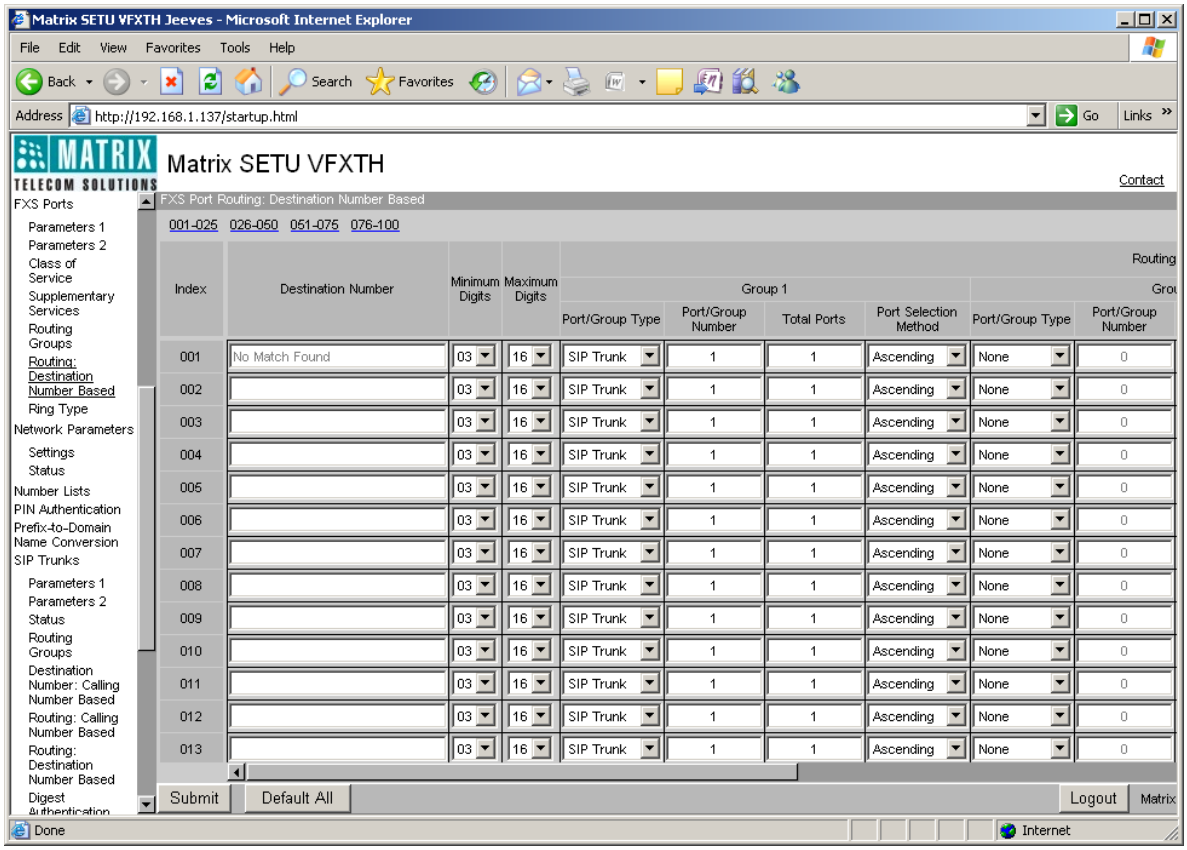
  - For FXO and FXS Port: 01-16
  - For SIP Trunk: 1-9

  By default, 'Total Ports/Channels' is set to '1' for all FXS Ports.

- **Port Selection Method:** Select desired Port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

- **CLI number on FXS Port:** This field is applicable only when group type 'FXS' is used for routing the call. (Please refer "Number Presentation on FXS Port" for more details)

- If Destination Port Determination Method selected is 'Destination Number Based' then click on '**FXS Port Routing: Destination Number Based**' link and program the following parameters:



- **Destination Number:** Program frequently called numbers in this field. Destination number can be of maximum 24 digits. Digits 0 to 9, #, *, (.) dot are allowed. By default, it is blank.

- **Minimum Length:** Program minimum length of destination number, required to be dialed by the user in this field. If the destination number dialed by the user is less than the minimum length programmed for that FXS Port then SETU VFXTH will reject the call. Range of minimum length of destination number is 01 to 24. By default, it is 03.

- **Maximum Length:** Program maximum length of destination number that can be dialed by the user in this field. If user dials a number greater than the number programmed for that FXS port then SETU VFXTH shall strip off the extra digits and route the call. Range of maximum length of destination number is 01 to 24. By default, it is 16.

- **Routing Group:** Program Routing Groups (1, 2 and 3) parameters for Destination Port Determination Method: Destination Number Based.

  - ***Port/Group Type:*** Select desired port or group for routing the call originated on FXS Port in this field for each FXS Ports. You can choose one of the following options:

    - None
    - FXS Port
    - SIP Trunk
    - FXO Port
    - FXS Group
    - FXO Group
    - SIP Group

    By default, 'Port/Group Type' is set to 'SIP Trunk'.

  - ***Port/Group Number:*** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

    - For FXS and FXO Port: 01-16
    - For SIP Trunk: 1-32
    - For FXS and FXO Group: 01-16
    - For SIP Group: 1-9

    By default, Port number '1' is selected.

  - ***Total Ports:*** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

    - For FXO and FXS Port: 01-16
    - For SIP Trunk: 1-9

    By default, it is set to '1'.

  - ***Port Selection Method:*** Select desired Port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

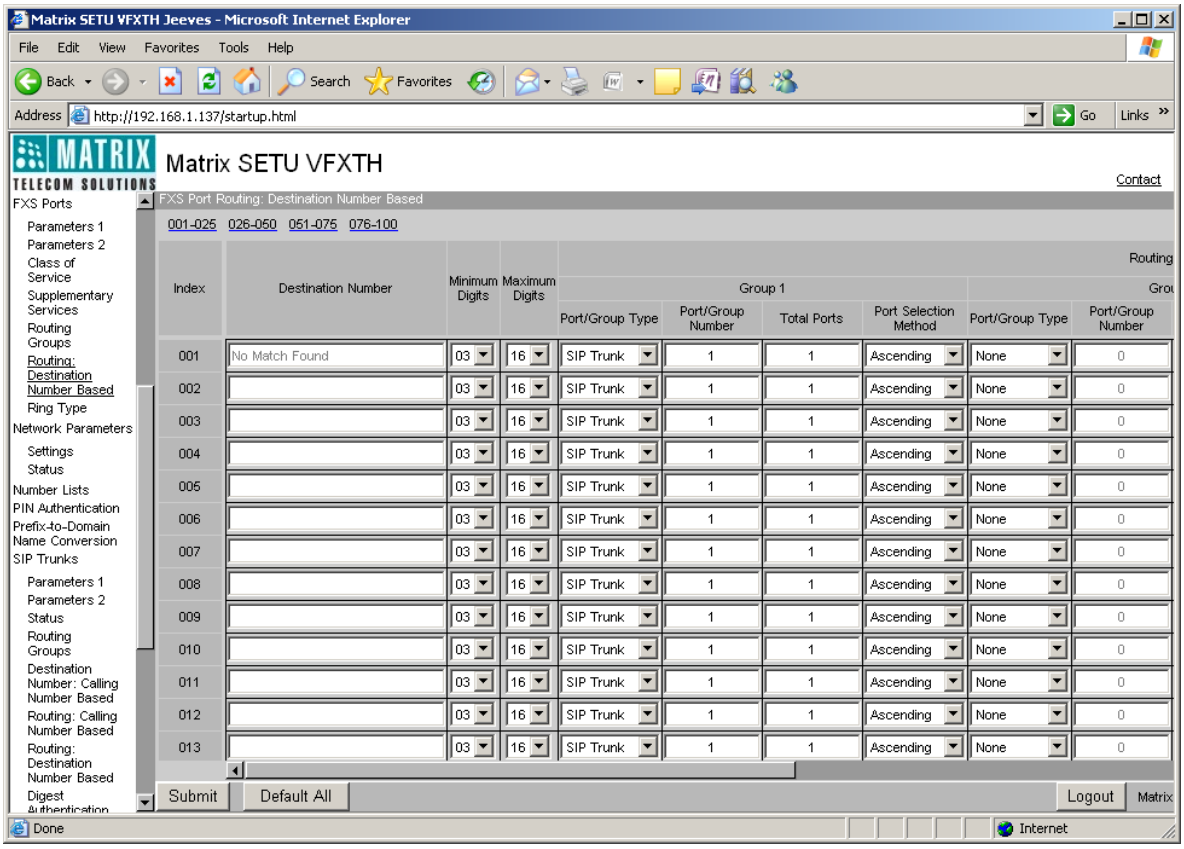
- ***CLI number on FXS Port:*** This field is applicable only when group type 'FXS' is used for routing the call. (Please refer “Number Presentation on FXS Port” for more details)

Improper selection of Destination Port Determination Method and Destination Number Determination Method may result in improper and undesirable routing of the calls.

Logic of FXS Port Routing: Destination Number Based

In destination number based routing; call is routed to the destination port on the basis of the destination number. FXS Port Destination Number based table supports 100 entries. Each entry has following parameters:

- Destination Number
- Minimum Length
- Maximum Length
- Routing Groups: Group1, Group2 and Group3.
- CLI Number on FXS Port



Let us understand FXS Port Routing: Destination Number Based logic with the help of an example:

Suppose following entries are made in the FXS Port Routing: Destination Number Based table:

Index	Destination Number	Min. Length	Max. Length	Routing Groups						CLI Number on FXS Port
				Group 1		Group 2		Group 3		
				Group Type	Group No.	Group Type	Group No.	Group Type	Group No.	
001	No Match Found	3	16	SIP	1	None	0	None	0	Received Calling Party
002	2	7	7	FXO	1	None	0	None	0	Received Calling Party
003	28	7	8	FXO	2	None	0	None	0	Received Calling Party

Index	Destination Number	Min. Length	Max. Length	Routing Groups						CLI Number on FXS Port
				Group 1		Group 2		Group 3		
				Group Type	Group No.	Group Type	Group No.	Group Type	Group No.	
004	98	3	15	FXO	1	None	0	None	0	Received Calling Party
005	9	5	10	SIP	2	None	0	None	0	Received Calling Party
006	09	10	12	SIP	3	None	0	None	0	Received Calling Party

SETU VFXTH checks 'FXS Port Routing: Destination Number Based' table when the source port is FXS, Destination Number Determination Method is Manual Dial and Destination Port Determination Method is Destination Number Based. Best Match logic will be applied, while routing the call to the destination port using this table.

Consider the following cases:

Case 1: Number dialed on FXS port is 2830555. In this case, dialed number best matches index 002 and 003 but it best matches the minimum and maximum length criteria of index 003 therefore SETU VFXTH routes the call as per the programming done in index 003.

Case 2: Number dialed on FXS port is 6530300. In this case, dialed number does not match any of the entry and thus SETU VFXTH checks minimum and maximum length criteria of 'No Match Found' entry i.e. index 001. As the dialed number matches the length criteria of 'No Match Found', system routes the call as per the programming done in index 001.

Case 3: Number dialed on FXS port is 09898098. In this case, dialed number best matches destination number programmed in index 006 but it does not satisfies minimum and maximum length criteria of that entry and therefore SETU VFXTH rejects the call and give error tone to the user.

Case 4: Number dialed on FXS port is 98980148240. Dialed number matches two entries: entry number 004 and 005. But it nearly matches the maximum and minimum length criteria of entry number 004. SETU VFXTH will strip off the last digit of the dialed number and will dial out the number as per the programming done in index 004.

Relevant Topics:

1. “Call Processing” 47
2. “Number Presentation on FXS Port” 87
3. “Port Parameters-FXS” 112
4. “Routing Group” 135

Routing Mechanism on SIP

Routing mechanism on SIP is the process of routing the call originated on the SIP Trunk to the desired destination port. In SETU VFXTH, routing mechanism involves two steps:

1. Determination of Destination Number
2. Determination of Destination Port

Destination Number Determination Method:

Some number is to be dialed on the source port to route the call to the destination port. The method of collecting destination number on the source port is called Destination Number Determination Method.

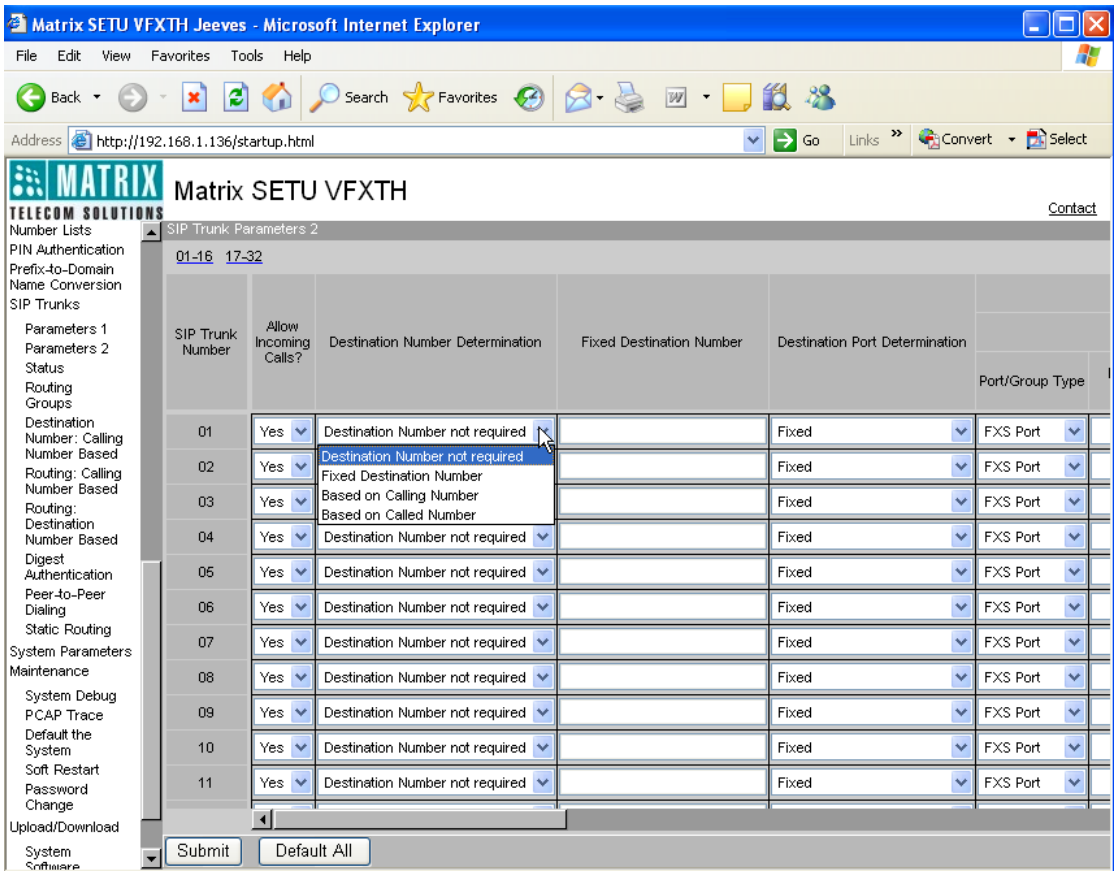
SETU VFXTH supports five different methods of determining the destination number for the calls originated on SIP Trunk viz.

1. **Number Not Required:** In this method, destination number is not required. The call originated on a SIP Trunk is simply routed to a fixed port programmed for that SIP Trunk. No specific number is to be dialed in this method.
2. **Fixed Destination Number:** In this method, a call received on the SIP Trunk is routed to a fixed destination number programmed for that SIP Trunk.
3. **Based on Calling Number:** In this method, a call is routed to a specific number depending upon the calling party's number i.e. SETU VFXTH will route the call to the destination number programmed for the calling party in SIP Trunk Destination Number Determination: Calling Number Based table.
4. **Based on Called Number:** In this method, a call is routed to the desired number depending on the called party number received in the SIP ID of Request URI of the INVITE message.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

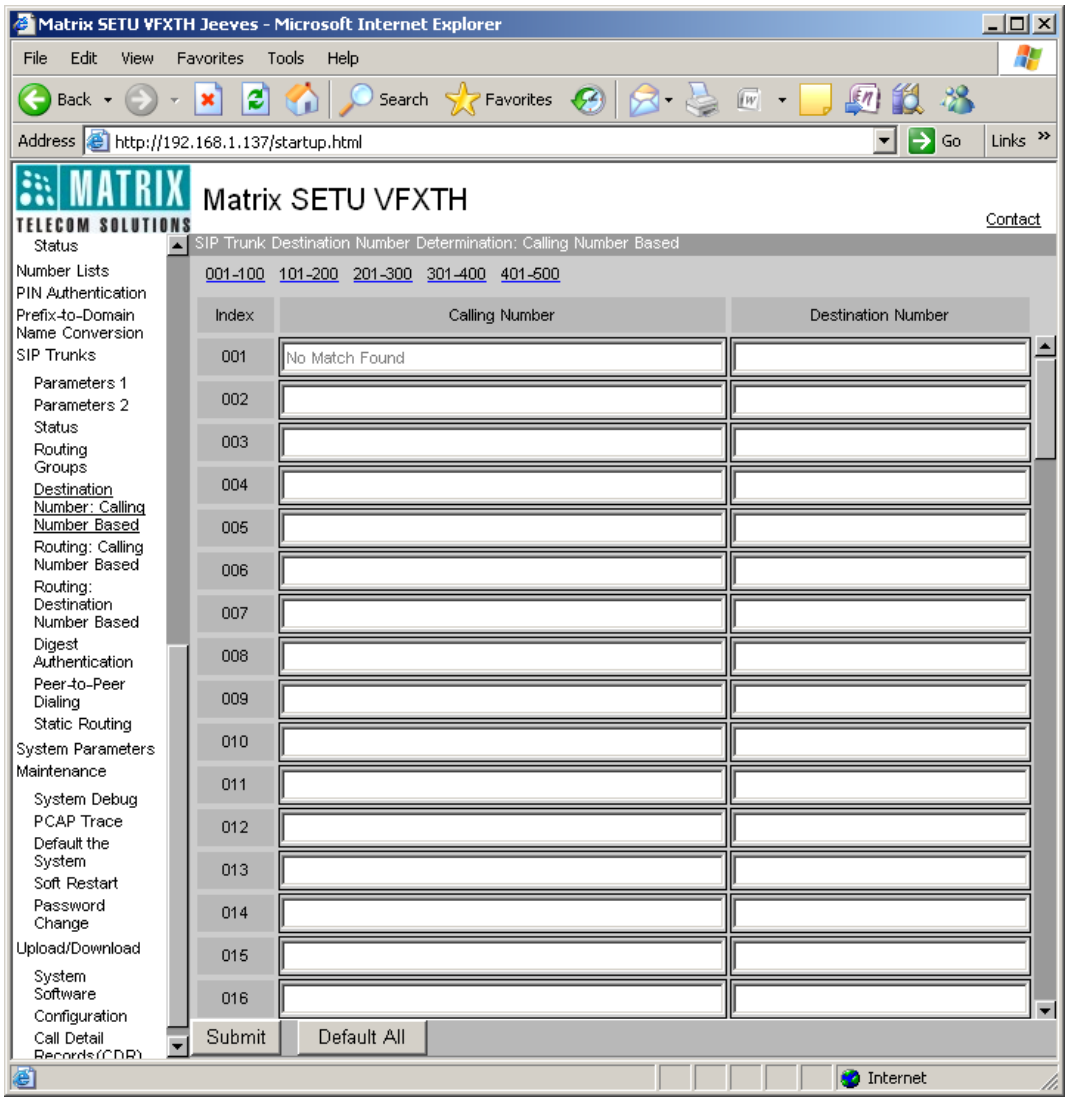
- Click on '**SIP Trunk Parameters 2**' and select destination number determination method for each SIP Trunk. By default, 'Number Not Required' is selected.



- If destination number determination method selected is 'Fixed Destination Number' then program destination number for the SIP Trunk in 'Fixed Destination Number' field in 'SIP Trunk Parameters-2'. Destination number can be of maximum 24 digits. Valid digits are 0 to 9, *, # and (.) dot. By default, it is blank.

Fixed destination number should be less than or equal to the maximum number of digits allowed to be dialed on the SIP Trunk.

- If Destination Number Determination Method selected is 'Based on Calling number' then click on 'SIP Trunk Destination Number Determination: Calling Number Based' link and program the following parameters:



- **Index:** In this field index number is given. Total 500 entries can be made in this table starting from 001 to 500.

- **Calling Number:** Program calling party number in this field. Destination number can be of maximum 40 characters. All ASCII characters are allowed. By default, first entry is used for 'No Match Found' case and other entries are blank.

- **Destination Number:** Program destination number in this field. Destination number can be of maximum 24 characters. Digits 0 to 9, *, # and (.) dot are allowed. By default, it is blank.

Destination Port Determination Method

When the call originates on the source port, it should be routed to the destination port. The method of finding destination port for routing the call is called Destination Port Determination Method. Destination Port Determination Method is applicable only on source port.

SETU VFXTH supports following three methods for Destination Port Determination.

1. **Fixed:** In this method, calls received on a SIP Trunk (originating on SIP Trunk) are routed through a fixed port irrespective of the number dialed on the source port.

2. **Destination Number Based:** In this method, the call is routed to the destination port on the basis of the destination number dialed by the caller i.e. called party number. To use this method, program 'SIP Trunk Routing: Destination Number Based' table. SETU VFXTH will check this table and place the call on the port programmed for the destination number.

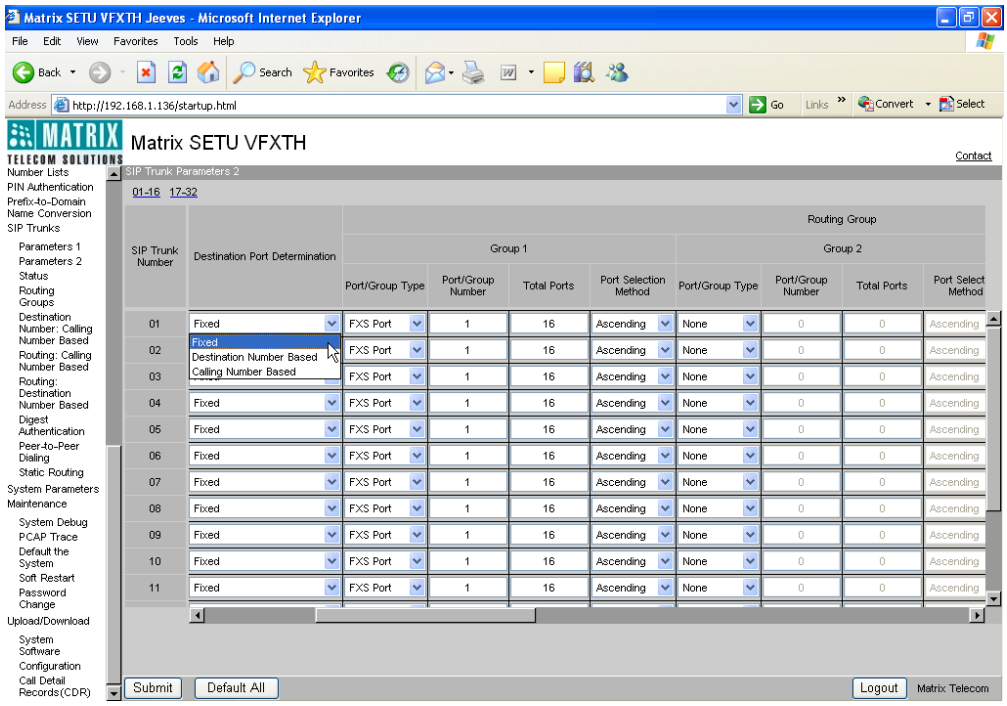
3. **Calling Number Based:** In this method, call is routed to the destination port on the basis of the calling party's number. To use this method, program 'SIP Trunk Routing: Calling Number Based' table. SETU VFXTH shall check this table while routing the call to the destination port and place the call on the port programmed for the calling party's number.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**SIP Trunk Parameters 2**' and select Destination Port Determination Method for each SIP Trunks. By default, 'Fixed' is selected.

- If Destination Port Determination Method selected is "Fixed" then program following parameters of Routing Groups (1, 2 and 3):



  - **Port/Group Type:** Select desired port or group for routing the call originated on SIP Trunk in this field. You can choose one of the following options:

    - None
    - FXS Port
    - FXO Port
    - FXS Group

- FXO Group

By default, 'FXS Port' is selected for all SIP Trunks.

- **Port/Group Number:** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

  - For FXS and FXO Port: 01-16
  - For FXS and FXO Group: 01-16

  By default, Port number '1' is selected for all SIP Trunks.

- **Total Ports:** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

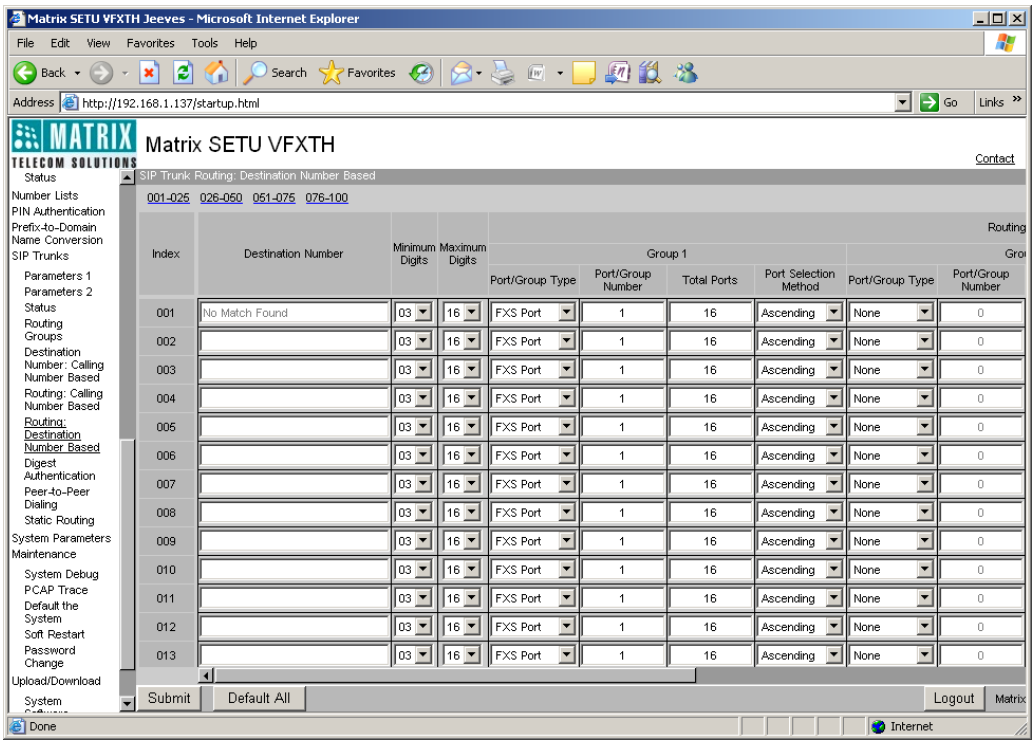
  - For FXO and FXS Port: 01-16

  By default, it is '2' for all SIP Trunks.

- **Port Selection Method:** Select desired port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

- **CLI number on FXS Port:** This field is applicable only when port/group type 'FXS' is used for routing the call. (Please refer "Number Presentation on FXS Port" for more details)

- If destination Port determination method selected is 'Destination Number Based' then click on 'SIP Trunk Routing: Destination Number Based' link and program the following parameters:



- **Destination Number:** Program frequently called numbers in this field. Destination number can be of maximum 24 characters. Characters 0 to 9, *, # and (.) Dot are allowed. By default, it is blank.

- **Minimum Length:** Program minimum length of destination number required to be dialed by the user in this field. If number string dialed by the user is less than the minimum length programmed for that trunk then SETU VFXTH will reject the call. Range of minimum length of destination number is 01 to 24. By default, it is 03.

- **Maximum Length:** Program maximum length of destination number string that can be dialed by user in this field. If user dials a number string greater than the number length programmed in this field then SETU VFXTH shall strip off the extra digits and route the call. Range of maximum length of destination number is 01 to 24. By default, it is 16.

- **Routing Group:** Program Routing Groups (1, 2 and 3) parameters for Destination Port Determination Method: Destination Number Based.

  - ***Port/Group Type:*** Select desired port or group for routing the call originated on SIP Trunk in this field. You can choose one of the following options:
    - None
    - FXS Port
    - FXO Port
    - FXS Group
    - FXO Group

    By default, 'Port/Group Type' is set to 'FXS Port'.

  - ***Port/Group Number:*** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

    - For FXS and FXO Port: 01-16
    - For FXS and FXO Group: 01-16

    By default, Port number '01' is selected.

  - ***Total Ports:*** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

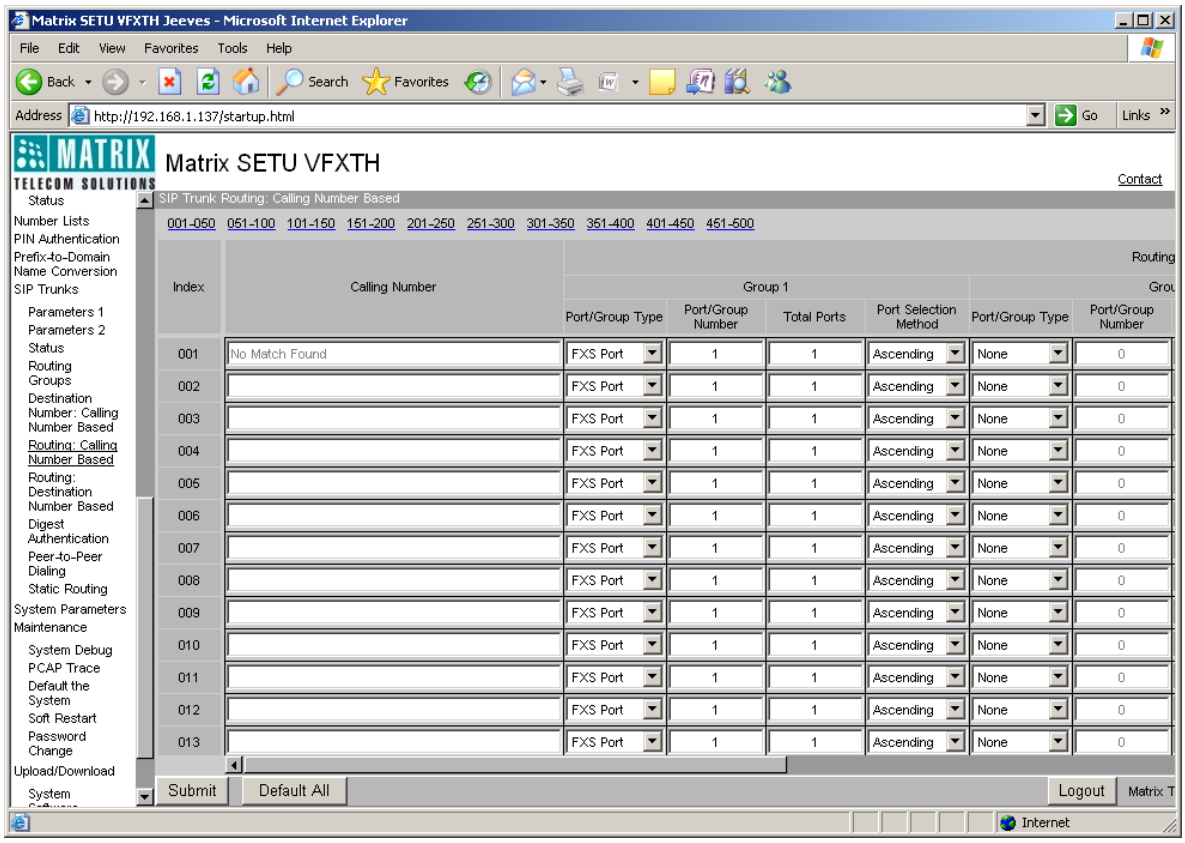
    - For FXO and FXS Port: 01-16

    By default, 'Total Ports' is set to '2'.

  - ***Port Selection Method:*** Select desired port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

- **CLI number on FXS Port:** This field is applicable only when port/group type 'FXS' is used for routing the call. (Please refer “Number Presentation on FXS Port” for more details)

- If destination Port determination method selected is 'Calling Number Based' then click on 'SIP Trunk Routing: Calling Number Based' link and program the following parameters:



- **Calling Number:** Program number of the calling party, who often calls SETU VFXTH, in this field. Calling party's number can be of maximum 24 digits. All ASCII characters are allowed. By default, it is blank.

- **Routing Group:** Program Routing Groups (1, 2 and 3) parameters for Destination Port Determination Method: Calling Number Based.

  - ***Port/Group Type:*** Select desired port or group for routing the call originated on SIP Trunk in this field. You can choose one of the following options:

    - None
    - FXS Port
    - FXO Port
    - FXS Group
    - FXO Group

    By default, 'Port/Group Type' is set to 'FXS Port'.

  - ***Port/Group Number:*** Assign Port Number, if Port Type is selected in Port/Group Type field. Assign Group Number, if Group Type is selected in Port Group Type field. You may select from the following options:

    - For FXS and FXO Port: 01-16
    - For FXS and FXO Group: 01-16

    By default, Port number '01' is selected.

- ***Total Ports:*** Program total ports to be checked while routing the call, if first port selected is busy. You have following options:

  - For FXO and FXS Port: 01-16

  By default, it is set to '2'.

- ***Port Selection Method:*** Select desired port selection method in this field. SETU VFXTH will select the port to place the call as per this method. SETU VFXTH supports two port selection methods viz. Ascending and Descending. By default, it is 'Ascending'.

- **CLI number on FXS Port:** This field is applicable only when port/group type 'FXS' is used for routing the call. (Please refer "Number Presentation on FXS Port" for more details)

SETU VFXTH also supports Remote Call Forward, Remote Held and Remote Transfer on SIP Trunks.

- ***Remote Call Forward:*** *When a remote party forwards his calls to another external party and a user connected to SETU VFXTH calls such a remote party then it is a case of 'Remote Call Forward'.*

- ***Remote Held:*** *When a remote party puts the user connected to SETU VFXTH on hold, it is called 'Remote Held'. During remote held condition, SETU VFXTH user can hold the remote held call and access any supplementary feature except blind transfer and attended transfer.*

- ***Remote Transfer:*** *When the remote party transfers the call of the user connected to SETU VFXTH, to a third party, it is termed as remote transfer. During remote transfer, if transfer target is busy or is not responding then SETU VFXTH will play the tone as per the message received to the transferee and then disconnects the call.*

Relevant Topics:

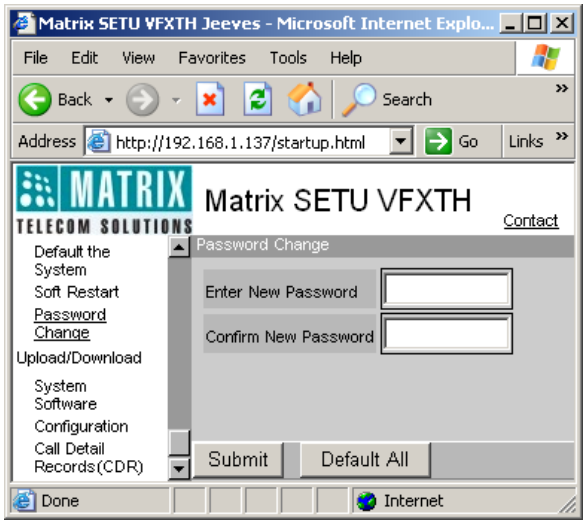
1. "Call Processing" 47
2. "Number Presentation on FXS Port" 87
3. "Port Parameters-SIP" 117
4. "Routing Group" 135

SE Password

Programming of SETU VFXTH can be done using Web JEEVES only. It does not support programming using telephone instrument except few Network Port Parameters.

For programming the system, SE must enter programming mode using SE password. SE password is of four digits and by default, it is 1234. You can change SE password by following the steps as shown below:

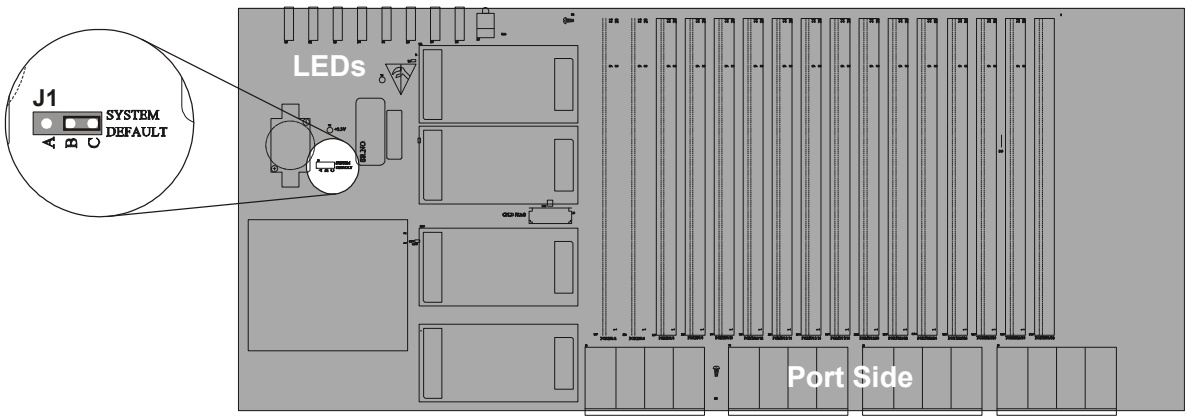
- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)



- Click on '**Password Change**' link. To change the current password, enter new password in 'Enter New Password' field. Re-enter the new password in 'Confirm New Password' field and submit the page.

You can also default SE password in case you forget it by changing the jumper position. To default the SE password, follow the steps given below:

- Switch Off SETU VFXTH.
- Locate a mini jumper J1 on the card. It is in BC position by default.



- Change the jumper position from BC to AB.
- Switch On the system and wait till it gets initialize.
- SE password gets default to 1234.

- Switch Off SETU VFXTH and restore the jumper in its original position i.e. BC.
- Switch On the system again.

Please note that when you default SE Password by changing Jumper settings, following parameters will also get default.

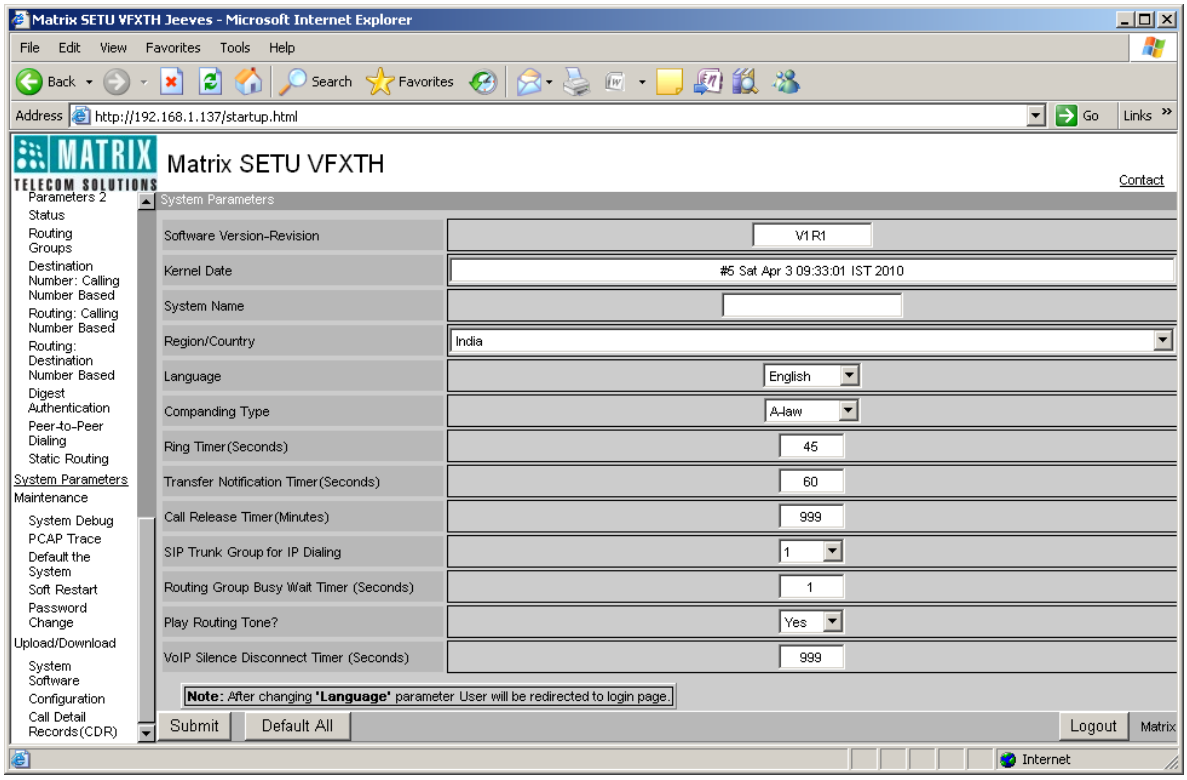
- *Network Connection Type (Static)*
- *IP Address (192.168.001.136)*
- *Subnet Mask (255.255.255.000)*
- *VLAN*

Software Upgrade

Due to continuous feedbacks from the customer and their requirements, software change becomes inevitable. SETU VFXTH provides facility to upgrade the software to latest version just by clicking a button. You can check the current software version revision of the system and upgrade the same to the latest version through Web JEEVES.

To check Software Version-Revision:

- Click on '**System Parameters**' link. Current Software version revision of SETU VFXTH will be displayed as shown below.



File Transfer Protocol (FTP) is a standard Internet Protocol that is used to exchange files between computers on the IP network. SETU VFXTH supports an embedded FTP server which can be used for Uploading and Downloading System files. FTP server of the system give access to all currently used configuration files, system software and call detail records.

Follow the steps given below to upgrade SETU VFXTH:

- Click on '**System Software**' link. Following executable files/folders will be displayed:

  - Setu VFXTH_V1R1
  - VoPP
  - wdtserver
  - Webs

  Delete the files which are to be upgraded and copy the new files.

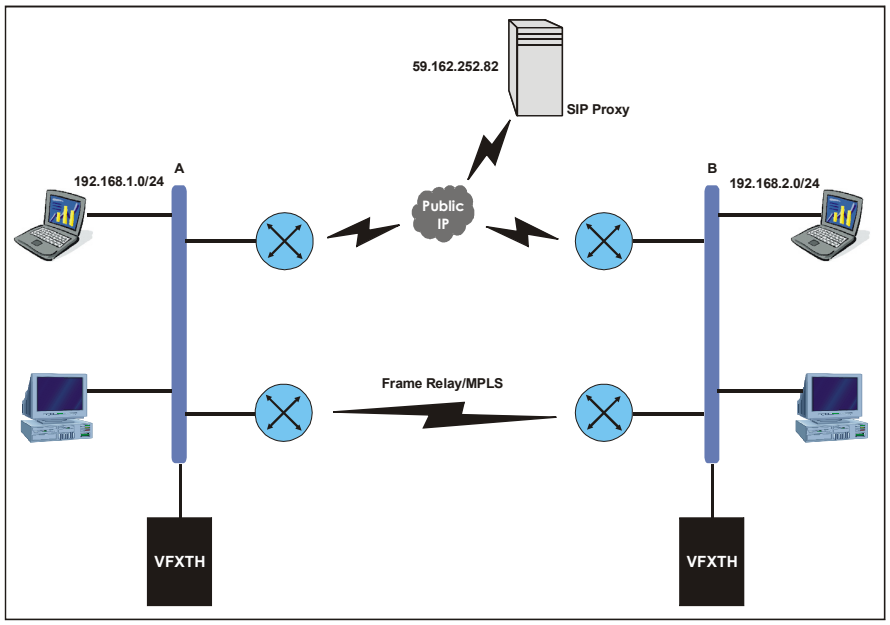
- Click on '**Configurations**' link. Configurations files will be displayed. Delete the current files. On next start up new configuration files will be created. You may also copy new files (only for supported system version) to change system's configuration.

- Click on '**Call Detail Records**' link. Copy and save the CDR files to keep records of the calls.

Restart the system after upgrading the files. The new configurations will take place only after the system restarts.

Static Routing

Static routing table helps you routing the call between point to point sites (connected through MPLS, Frame Relay etc) and to public internet at the same time.

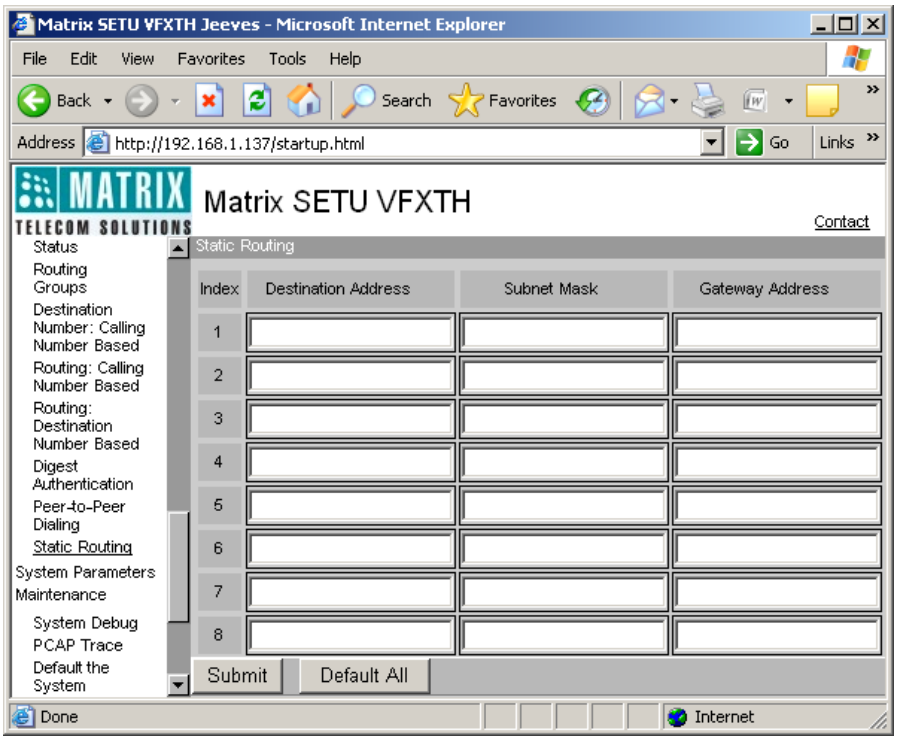


Generally, Static routing is not required to be configured when SETU VFXTH is connected behind the NAT router. LAN interface of NAT router acts as default gateway for SETU VFXTH and all calls initiated from SETU VFXTH get routed from LAN port of NAT router. But, if you have connected multiple offices through MPLS, Frame Relay etc and want to make/ receive Peer to Peer calls between various offices and Proxy calls to public internet at the same time, you need to configure the Static Routing Table in the SETU VFXTH.

As in the above figure, two offices are connected through MPLS or Frame Relay. User wants to make Peer to Peer calls between two offices through Frame Relay/ MPLS network. At the same, time user also wants to make proxy calls through public internet. Refer 'How to program?' topic below to know about the programming of such configuration.

How to program?

- Open Web Jeeves of SETU VFXTH. (Refer “Accessing Web JEEVES”) Click on the 'Static Routing' link. The following screen will appear:



- Total 8 entries can be programmed in Static Routing Table.
- Each entry has three parameters viz. Destination Address, Subnet Mask and Gateway Address.
- Destination Address is the address of final destination where the call is to be made. This can be IP address of called SIP device or network address where called SIP device resides.
- Subnet Mask is the mask to be applied on destination address.
- Gateway Address is the IP address of LAN interface of your router from where you want to route the call for specific destination. This address must be in the same subnet where calling (remote) SETU VFXTH resides.

  By programming this table, SETU VFXTH can route the calls on the basis of destination address. Calls for specified network can be routed to specific gateway.

- With reference to the above figure, following programming should be done in the Static Routing Table of SETU VFXTH at location A.

Index	Destination Address	Subnet Mask	Gateway Address
1	192.168.2.0	255.255.255.0	192.168.1.1
2			
3			
4			
5			

- Only one entry in Static Routing Table needs to be configured.
- Destination Address viz. 192.168.2.0 specifies the network address of location B.
- Subnet Mask specifies the mask to be applied on destination address.
- Gateway Address viz. 192.168.1.1 specifies the LAN address of router which connects location A and location B.
- IP address of LAN interface of router which connect location A to public internet should be configured as Default Gateway as in normal scenario. Refer “Network Port Parameters”.
- Doing above programming, all calls made by SETU VFXTH to 192.168.2.0/ 24 will be routed to router which connects location A to location B. Whereas, all calls made by SETU VFXTH to address other then 192.168.2.0/ 24 will be routed to default gateway.
- Similar programming is required in SETU VFXTH at location B to enable calling from location B to location A.

Relevant Topics:

1. “Peer to Peer Calling” 97
2. “Network Port Parameters” 72

Supplementary Services of Service Provider

Exchange (CO) supports many advance features like call waiting, call forward, call hold etc. User should dial specific access codes to use these features. Exchange provides access codes to use these features. But dialing of these access codes creates problem if SETU VFXTH or PBX is connected between user and exchange.

SETU VFXTH provides access codes for using supplementary services of service provider. Supplementary Services of Service Provider can be used only if call hold feature is allowed to the FXS Port user. These features are allowed to be used for a fixed time i.e. for Service Provider Access Code Wait Timer after dialing access code for Using Supplementary Services of Service Provider. By default, access code assigned for Using Supplementary Services of Service Provider is #4. Dialing this access code enables the user to directly access the supplementary services provided by exchange.

How to use?

- Suppose A (User) is in speech with B (Called Party) and A wants to use supplementary services of service provider.
- A dials Flash (Call Hold access code) to keep B on hold and then dials 'Using Supplementary Services of Service Provider' access code.
- A is in speech with B again.
- Dial desired feature code (given by Service Provider) within Service Provider access code wait timer to use service provider's features.

Programming Access Code for Using Supplementary Services of Service Provider:

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on '**Access Code**' link and program access code for Using Supplementary Services of Service Provider. By default, Access Code for using Supplementary Services of Service Provider is ***#4***.

Relevant Topics:

1. “Access Codes” 25
2. “Port Parameters-FXO” 104
3. “Port Parameters-FXS” 112
4. “Port Parameters-SIP” 117

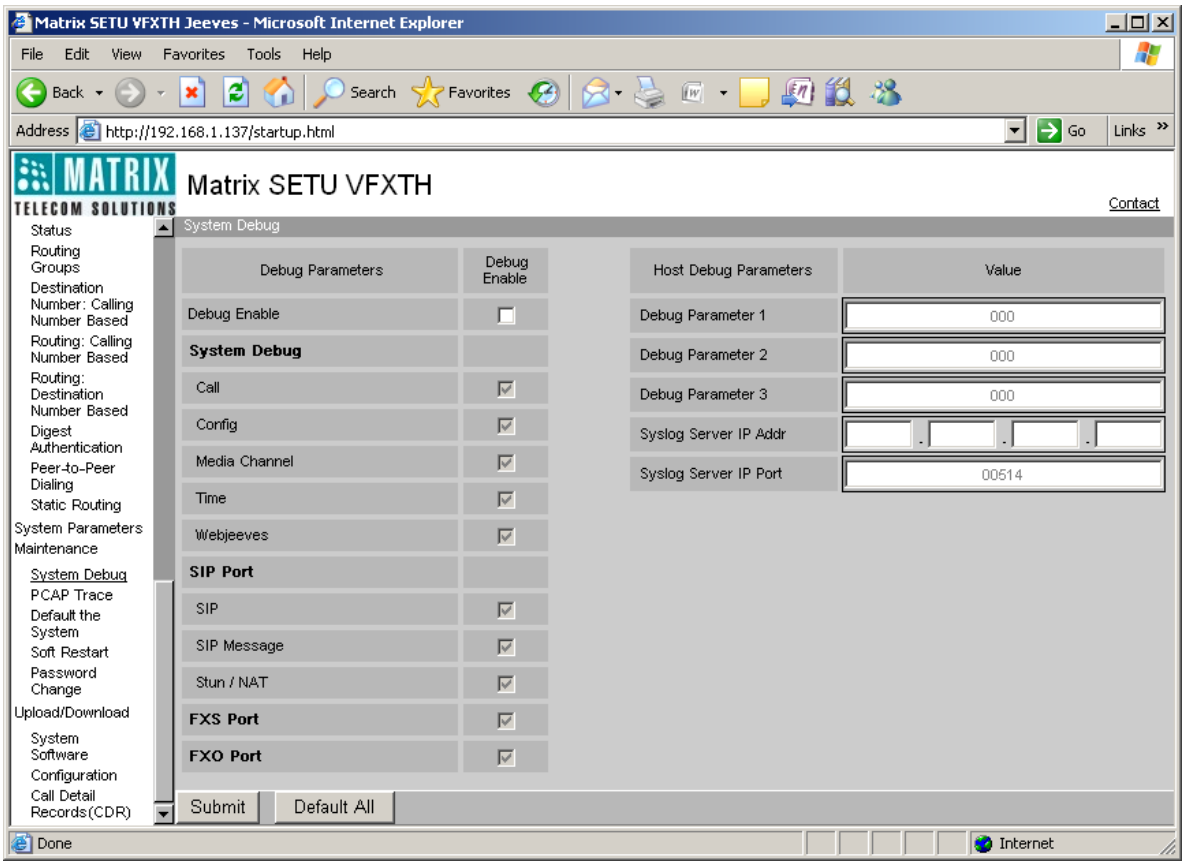
System Debug

Logging is a method used to record actions and events of the system. Debugs are the primary record keepers of the system and network activity. Logging has several benefits which include troubleshooting, security and system administration.

Debug messages are sent to remote server on IP network for finding and reducing the number of bugs or defects from a system. Syslog is one of the protocols used extensively for sending debug messages on IP network. It is a client/server protocol that uses UDP as transport protocol for debugging process. SETU VFXTH has in-built syslog client that enables it to send the debug messages.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")
- Click on '**System Debug**' link and program the following parameters.



Debug Parameters:

- **Debug Enable:** Tick this option to enable the debug. By default, Debug flag is disabled.
- **Debug Levels:** If 'Debug Enable' flag is selected then tick desired options for debugging.

- Following options are supported for System Debug:

  i. Call
  ii. Config
  iii. Media Channel
  iv. Time
  v. Webjeeves

- Following options are supported for SIP Port:

  i. SIP
  ii. SIP message
  iii. STUN/ NAT

- FXS Port

- FXO Port

By default, all the above mentioned debug levels are enabled.

If debug is enabled, at least one debug level should be selected. If no debug level is selected, SETU VFXTH will gives following error message "Please select the Debug level".

Host Debug Parameters:

- **Syslog Server Address:** Enter Syslog server address in this field if debug is enabled. Only IP address programming is allowed and it can be of maximum 15 characters. Valid range for syslog server address is 000 to 255 and dot (.). Blank is also allowed. By default, it is blank.

- **Server Port:** Enter Syslog server Port in this field. Valid range of server port is 514, 1024 to 65535. By default, it is 514.

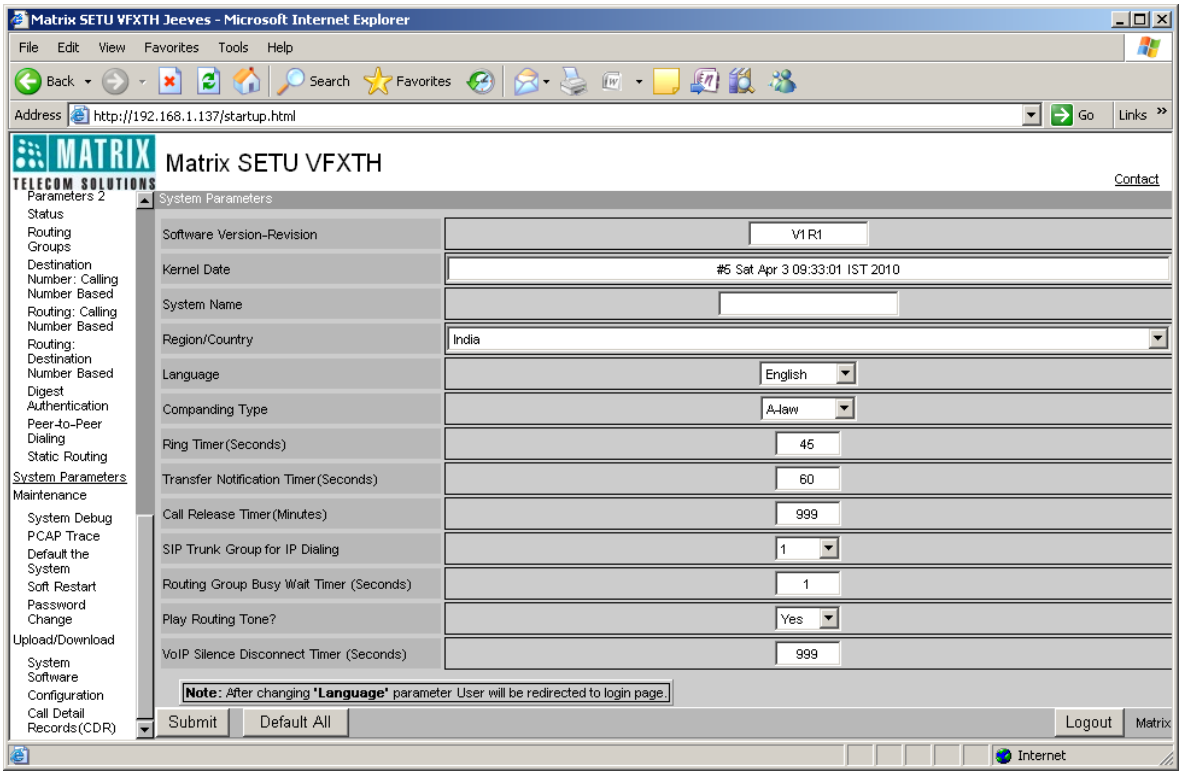
Debug Parameter 1, Debug Parameter 2 and Debug Parameter 3 are reserved for future use and at present are to be ignored.

System Parameters

In SETU VFXTH, certain global parameters are to be configured which are related to the entire system and not a specific port. These parameters are called System Parameters.

To program System Parameters, open Web Jeeves of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**System Parameters**' link and program all the parameters on the page as shown below:



- **Software Version-Revision:** Current Software Version-Revision of SETU VFXTH is displayed in this field.

- **Kernel Date:** The Date of current Kernel of the system is displayed in this field.

- **System Name:** You can assign a name to your SETU VFXTH. This name is known as 'System Name'. This name has significance only when two or more SETU VFXTHs are connected in a LAN network.

  System name can be of maximum 40 characters. All ASCII characters are allowed. By default, this field is blank.

- **Region/Country:** Select the Country or Region in which SETU VFXTH is installed. By default, India is selected in this field. (Refer “Region Selection” feature for more details).

- **Language:** Web JEEVES of SETU VFXTH supports eight languages viz. English, French, Italian, German, Portuguese and Spanish. You can choose the language of your choice in this field. The contents of the JEEVES will be displayed in the language selected in this field. By default, language selected is 'English'.

- **Companding Type:** Companding is a method of improving the signal-to-noise (S/N) ratio resulting from the pulse code modulation (PCM) process on voice calls. The analog signal's amplitude is compressed before it is quantized and transmitted.

  SETU VFXTH supports two algorithms for Companding the voice band signals: A-law and -law. A-law is generally used in countries that use E1 at 2.048 mbps; while -law is used in countries that use T1 at 1.544 mbps. By default, A-law is selected.

- **Ring Timer (Seconds):** It is the duration for which SETU VFXTH will play ring on the FXS Port to indicate an incoming call. Ring timer is loaded when the call is placed on FXS Port i.e. either there is a ring event on FXS Port or call waiting beeps in case FXS Port is busy.

  Valid range of Ring Timer is 01 to 99 seconds. By default, it is programmed as 45 seconds.

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

  This timer is loaded as soon as user performs any transfer activity and transferor is notified within this timer of the status of transfer activity.

  Valid range of Transfer Notification Timer is 001 to 999 seconds. By default, it is programmed as 60 seconds.

- **Call Release Timer (Minutes):** It is the duration for which SETU VFXTH will wait before releasing the ports involved in an active call. 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.

  Valid range of Call Release Timer is 001 to 999 minutes. By default, Call Release Timer is 999 minutes.

- **SIP Trunk Group for IP Dialing:** Select 'SIP Trunk Group for IP Dialing' in this field. When an IP Address is dialed from any port of SETU VFXTH, it is routed to the desired destination through the SIP Trunk group programmed for IP dialing.

  Range of this parameter is 'None' and SIP Trunk Group 1 to 9. By default, SIP Trunk Group selected for IP Dialing is 1.

  (Refer “IP Dialing” for more details)

- **Routing Group Busy Wait Timer (Seconds):** It is the duration for which SETU VFXTH would search for free destination port to place the call from all the routing groups. Routing Group Busy Wait Timer is loaded when from all the routing groups no free destination port is found for routing the call on the destination port.

  Valid range of Routing Group Busy Wait Timer is 01 to 99 seconds. By default, it is 01 seconds.

- **Play Routing Tone:** Routing Tone is played when Destination Port group for routing the call is found. During outgoing call, routing tone indicates that the call is in progress. Select 'No' in this field to disable routing tone at the time of routing the call to the destination port. By default, it is set to 'Yes'.

- **VoIP Silence Disconnect Timer:** The duration (in seconds) after which SIP call is disconnected, if continuous silence (No RTP Packets) is detected for this time period. VoIP Silence Disconnect Timer is

loaded as soon as silence is detected during a SIP call. SIP call is disconnected after the expiry of this timer, if continuous silence (No RTP Packets) is detected for this time period. This timer is applicable for all type of calls received or made through SIP Trunk.

Valid range of VoIP Silence Disconnect Timer is 001 to 999 seconds. By default, it is 999 seconds.

Relevant Topics:

1. “Call Processing” 47
2. “Call Progress Tones” 48

Telephony Features

Attended Call Transfer

Transferring the call after consulting the party to whom the call is to be transferred is called Attended Call Transfer. User can use this feature by dialing an access code programmed in the access code table.

SETU VFXTH enables transferor to know the result of the transfer activity i.e. whether the call has been transferred successfully or not, in case of SIP to SIP attended transfer. SETU VFXTH supports Transfer Notification Timer for this.

Transfer Notification Timer is loaded as soon as attended call transfer access code is dialed while performing attended call transfer activity. SETU VFXTH intimates the transferor of the result of the transfer activity within this timer.

How to use?

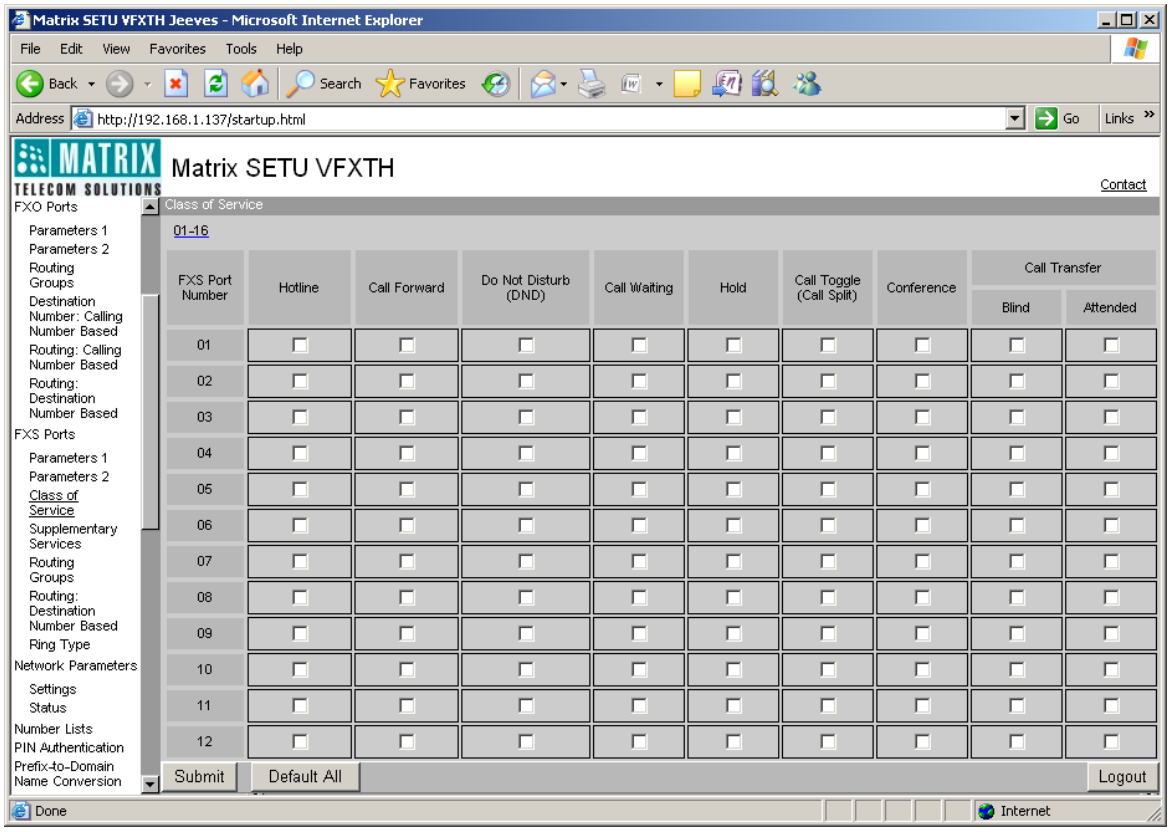
- Suppose A (transferor) is in speech with B and A wants to transfer B's (transferee) call to C (transfer target).
- A dials Flash (Call Hold access code) to put B on hold and then dials C's number.
- A goes On-Hook (Default Access Code for attended call transfer being (^) "On-Hook") after C's number start ringing or after speech with C.
- B is in speech with C.

You can change the default attended call transfer access code. If access code programmed for using Attended Call Transfer feature is other than (^) On-hook then after dialing C's number, A should dial attended call transfer access code to transfer B's call to C.

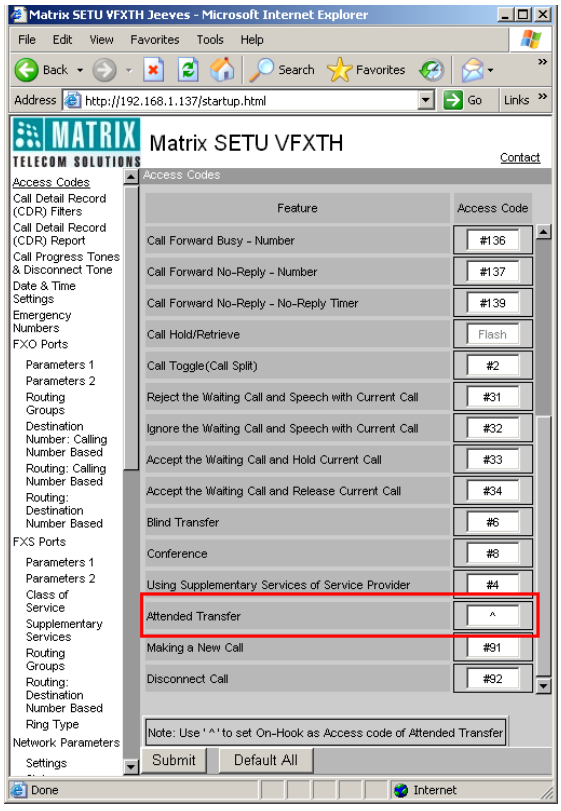
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

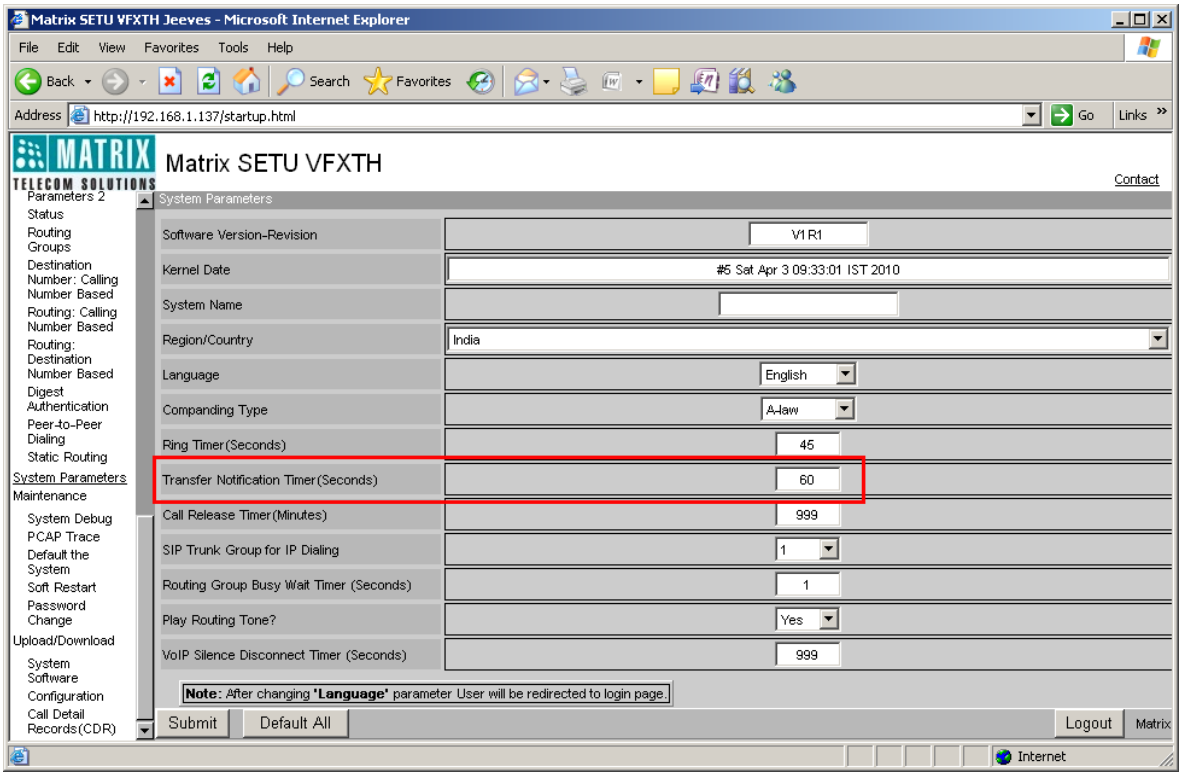
- Click on '**Class of Service**' link. Select 'Yes' to enable Attended Transfer feature. By default, it is set to 'No' i.e. disable.



- Click on '**Access Codes**' link and program access code for Attended Call Transfer. By default, '^' (On hook) is programmed as access code for Attended Call Transfer.



- Click on '**System Parameters**' link and program Transfer Notification Timer. Range of Transfer Notification Timer is 001 to 999 seconds. By default, it is 60 seconds.



- *Only FXS Port user can perform Attended Call Transfer. This feature works only if 'Attended Call Transfer' and 'Call Hold' is enabled in the Class of Service.*
- *Attended Call Transfer is not allowed if user dials transfer access code after holding the remote held call.*
- *SETU VFXTH notifies the transferor only in case of SIP to SIP call and when the attended call transfer access code is other than (^) i.e. On-Hook.*

Relevant Topics:

1. “Access Codes” 25
2. “Call Hold” 190
3. “Call Processing” 47
4. “Class of Service” 55

Blind Call Transfer

Transferring the call without consulting the party to whom the call is to be transferred is called Blind Call Transfer. User can use this feature by dialing an access code programmed in the access code table.

SETU VFXTH enables transferor to know the result of the transfer activity i.e. whether the call has been transferred successfully or not. SETU VFXTH supports Transfer Notification Timer for this.

Transfer Notification Timer is loaded as soon as blind call transfer access code is dialed while performing call transfer activity and it is stopped if transferor goes On-Hook. SETU VFXTH intimates the transferor of the result of the transfer activity within this timer.

How to use?

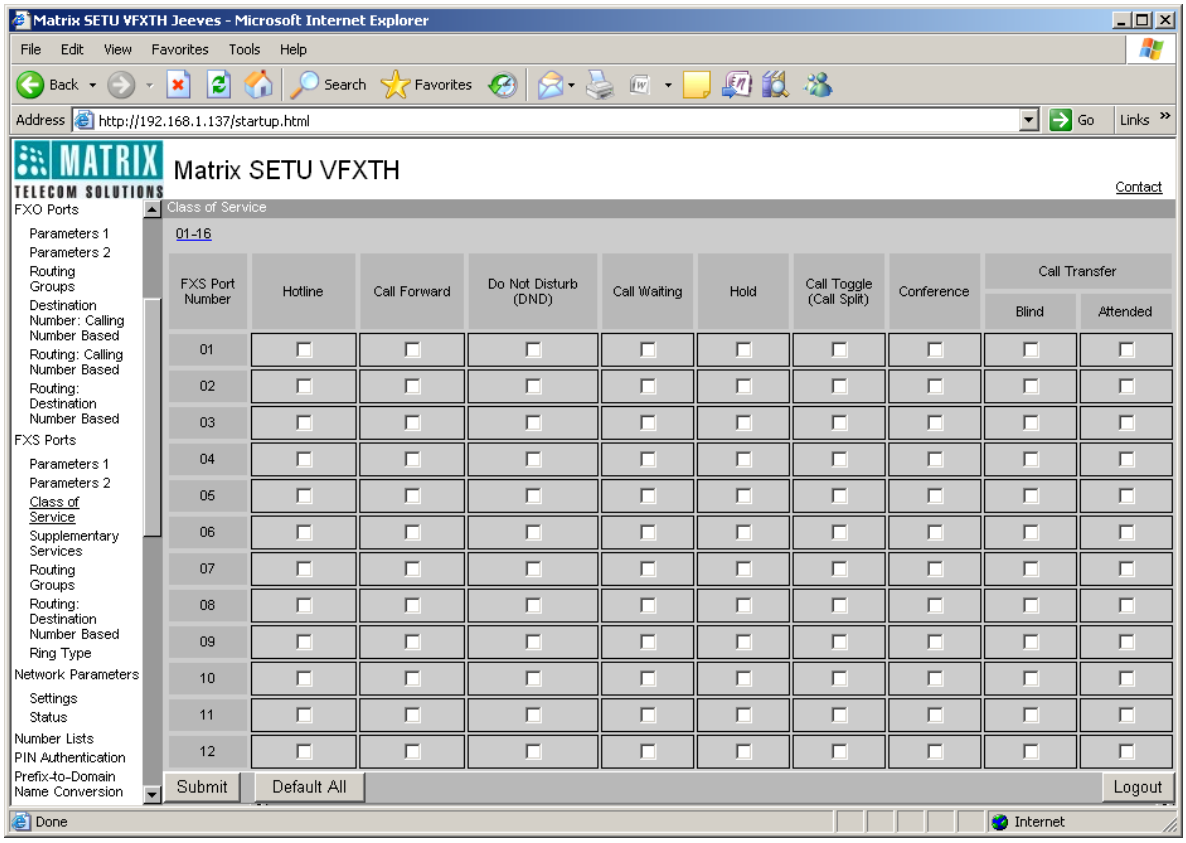
- Suppose A (transferor) is in speech with B (transferee) and A wants to blind transfer B's (transferee) call to C (transfer target).
- A dials Flash (Call Hold access code) to keep B on hold and then dials Blind Call Transfer access code.
- A dials number of C and B's call is transferred to C.

To know the call transfer results, A should remain off-hook after dialing C's number. One of the following results may occur:

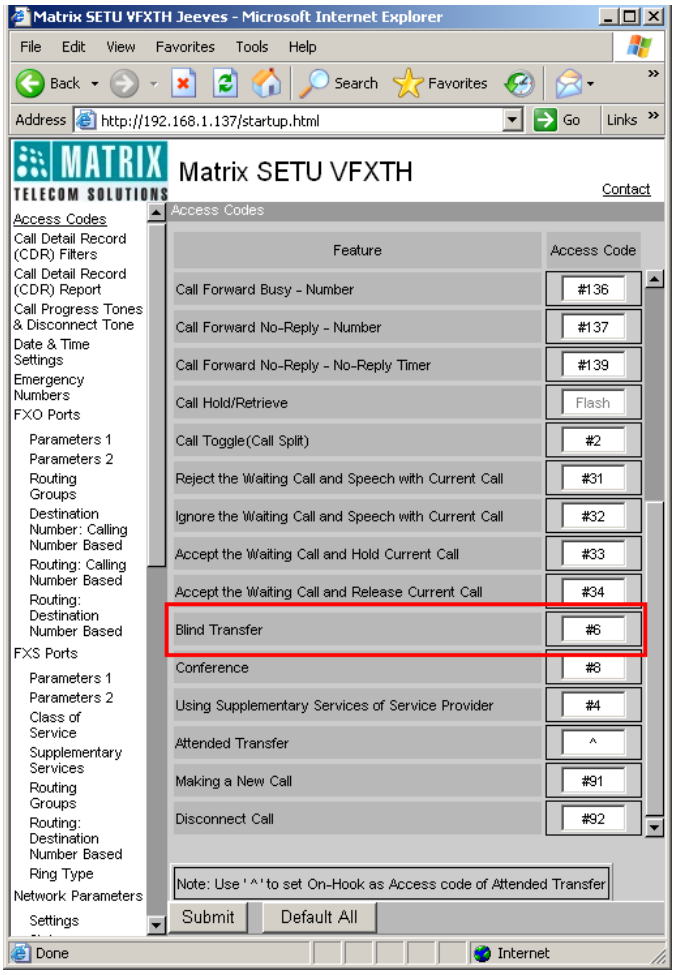
- If transfer is successful, A (transferor) gets confirmation tone for confirmation tone timer followed by error tone for error tone timer followed by system standby.
- If transfer target is busy, A (transferor) gets busy tone followed by speech with transferee.
- If no message is received during transfer notification timer, A (transferor) gets error tone for error tone timer followed by system standby.

How to Program?

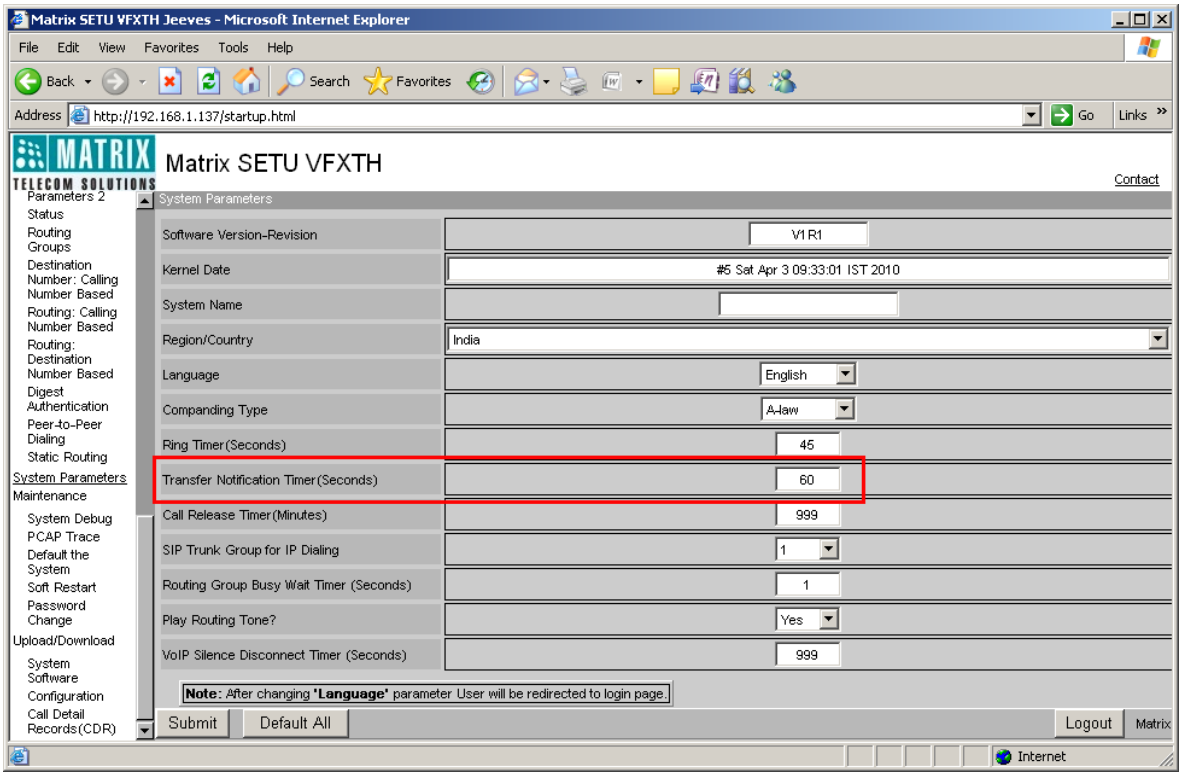
- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on '**Class of Service**' link. Select 'Yes' to enable Blind Transfer feature. By default, it is set to 'No' i.e. disable.



- Click on '**Access Codes**' link and program access code for Blind Call Transfer. By default, #6 is programmed as access code for Blind Call Transfer.



- Click on '**System Parameters**' link and program Transfer Notification Timer. Range of transfer notification timer is 001 to 999 seconds. By default, it is 60 seconds.



- *Only FXS Port user can perform Blind Call Transfer. This feature works only if 'Blind Call Transfer' and 'Call Hold' is enabled in the Class of Service.*
- *Blind Call Transfer is not allowed if user dials blind transfer access code after holding the remote held call.*

Relevant Topics:

1. "Access Codes" 25
2. "Call Processing" 47
3. "Class of Service" 55
4. "Call Hold" 190

Call Forward

Call Forward feature enables the user to forward his calls to another destination number when he is busy or when he is not available. SETU VFXTH supports three types of call forward:

- Call Forward: Unconditionally
- Call Forward: Busy
- Call Forward: No Reply

To use Call Forward feature, it has to be enabled in Class of Service. Only incoming calls on FXS Ports can be forwarded.

Call Forward- Unconditional means all incoming calls on FXS Ports are forwarded unconditionally to the destination number programmed.

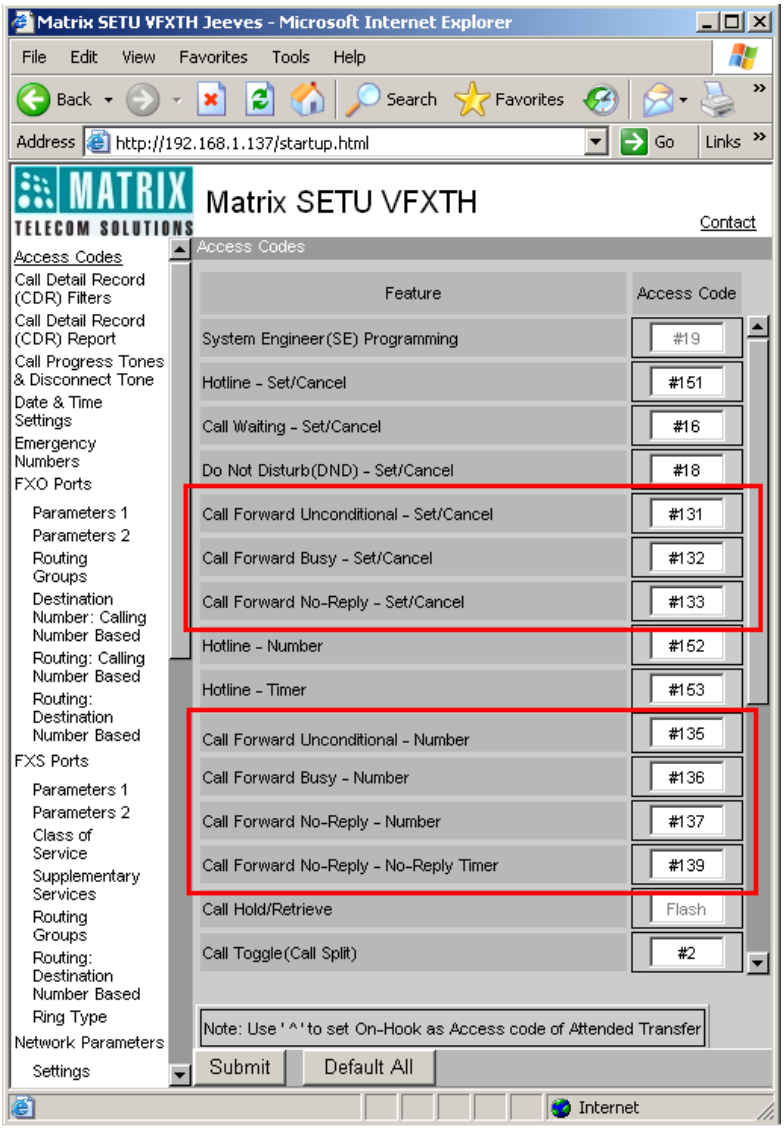
Call Forward-When busy means all incoming calls on FXS Port are forwarded to the destination number programmed if the FXS Port user is busy on another call.

Call Forward-When No Reply means all incoming calls on FXS Port are forwarded to the destination number programmed if there is no reply on the dialed FXS Port. In this type of call forward, SETU VFXTH should wait for the Call forward- No reply timer before forwarding the call.

The programming of access code can be done using Web JEEVES only. After programming the access codes, the feature can be set/cancel using telephone instrument. Also programming of the destination number to which the call is to be forwarded and programming of Call Forward- No Reply Timer can be done using telephone instrument.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer "Accessing Web JEEVES")
- Click on '**Access Codes**' link and program access codes for all three types of Call Forward.

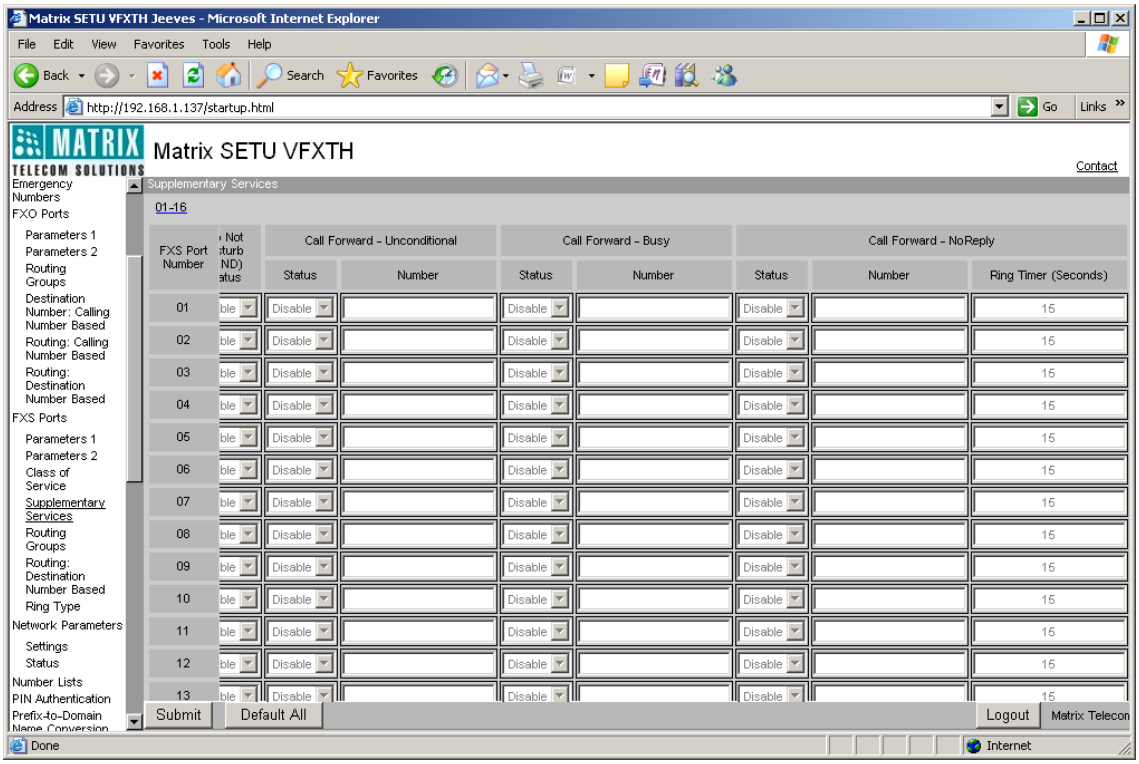


- Default Access Codes to set/cancel Call Forward-Unconditional/Busy/No Reply and to program Call Forward-Destination Number and No-Reply Ring Timer are mentioned in the table given below:

Call Forward-Unconditional	
To Set	***#131-1***
To Cancel	***#131-0***
To program call forward- unconditional number	***#135-Destination Number-End of Dialing***
Call Forward-Busy	
To Set	***#132-1***
To Cancel	***#132-0***
To program call forward- busy number	***#136-Destination Number-End of Dialing***
Call Forward-No Reply	
To Set	***#133-1***
To Cancel	***#133-0***
To program call forward- no reply number	***#137-Destination Number-End of Dialing***
To program No-Reply Ring Timer	***#139-Seconds; Default =15 secs.***

- *Access code to set/cancel the Call Forward feature and access code for programming destination number, to which the calls are to be forwarded, can be programmed using Web JEEVES only.*
- *Using telephone instrument, you can only set/cancel the feature and program destination number and Call Forward No-Reply Timer with the help of access code programmed in the access code table.*

- Click on '**Supplementary Services**' link and program the following parameters:



- **Call Forward-Unconditional:** Program the following parameters to use the Call Forward-Unconditional feature.
  - ***Status:*** Select 'Enable' in the status field to use this feature. By default, it is disabled.
  - ***Number:*** Program the number to which the call is to be forwarded unconditionally, in this field. Number can be of maximum 24 characters. Allowed characters are 0 to 9, *, # and (.) dot. By default, it is Blank.
- **Call Forward- Busy:** Program the following parameters to use the Call Forward- Busy feature.
  - ***Status:*** Select 'Enable' in the status field to use this feature. By default, it is disabled.
  - ***Number:*** Program the number to which the call is to be forwarded when FXS Port is busy, in this field. Number can be of maximum 24 characters. Allowed characters are 0 to 9, *, # and (.) dot. By default, it is Blank.
- **Call Forward-No Reply:** Program the following parameters to use the call forward- no reply feature.
  - ***Status:*** Select 'Enable' in the status field to use this feature. By default, it is disabled.
  - ***Number:*** Program the number to which the call is to be forwarded when there is no reply on the FXS Port, in this field. Number can be of maximum 24 characters. Allowed characters are 0 to 9, *, # and (.) dot. By default, it is Blank.
- **Ring Timer (Seconds):** Program the time after which the call is to be forwarded, to the number programmed. Range of Ring Timer is 01 to 99 seconds. By default, it is set to 15 seconds.

- *To use Call Forward feature from any FXS Port, it has to be enabled in the class of service for that FXS Port.*
- *Call Forward destination number should be less than or equal to the maximum number of digits allowed to be dialed on the FXS Port.*
- *Call Forward-When Busy has priority over Call Waiting feature in all conditions except in call mature state i.e. when FXS Port is in speech with other port.*

Relevant Topics:

1. “Access Codes” 25
2. “Call Hold” 190
3. “Call Waiting” 194
4. “Call Processing” 47
5. “Class of Service” 55
6. “Port Parameters-FXS” 112
7. “Prefix to Domain Name Conversion” 124
8. “Routing Mechanism on FXS” 155

Call Hold

Call Hold feature allows user to hold the remote party and retrieve the held call. It also allows making of second call or answering a waiting call by keeping current call on hold. It is possible to hold maximum two calls. Call Hold feature is applicable only for FXS Ports. This feature will work only if it is enabled in Class of Service and Subscriber Type is programmed as Gateway.

After holding a call, user can access following features:

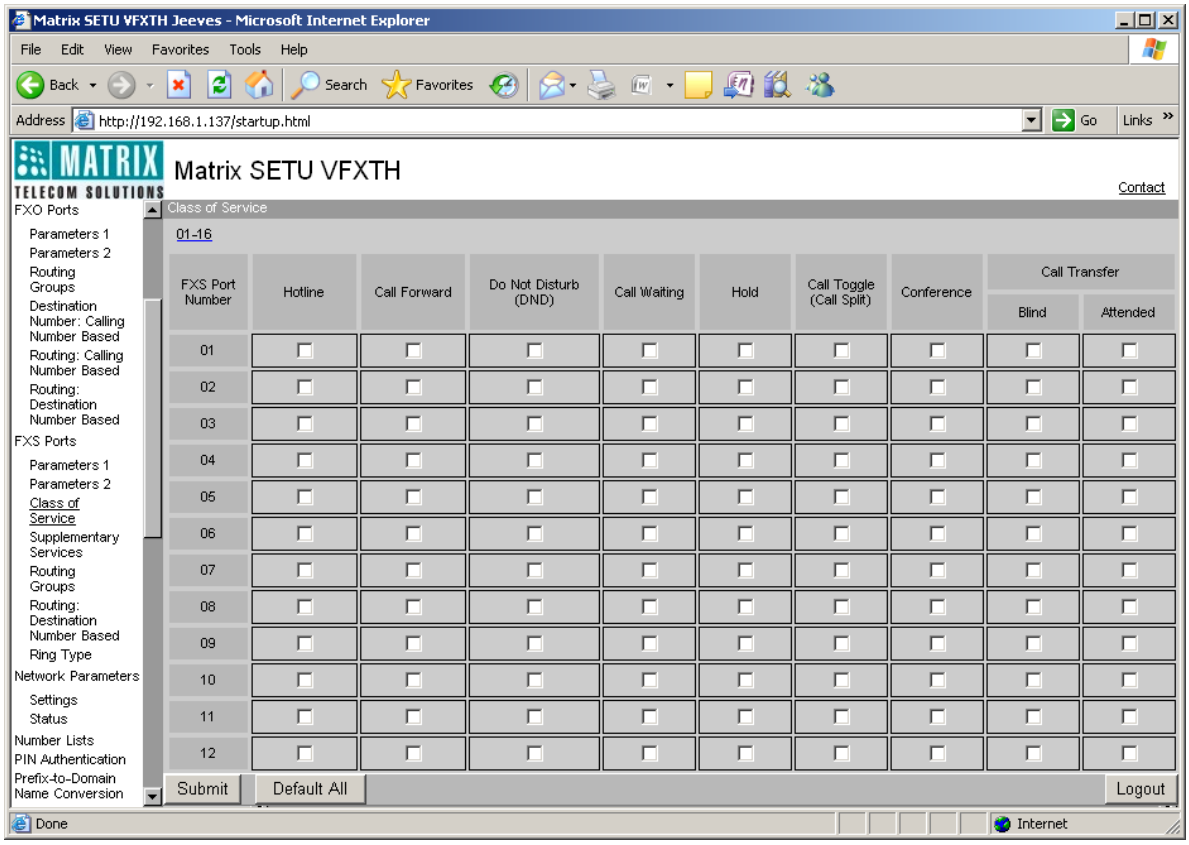
- Retrieve Held Call
- Make a Second Call
- Call Toggle
- Call Conference
- Blind Transfer
- Attended Transfer
- Accept Waiting call and Reject Current call
- Accept Waiting call and Hold Current call
- Reject Waiting call and speech with Current call
- Ignore Waiting call and speech with Current call

If Call Hold feature is disabled in Class of Service, user will not be able to access the above features.

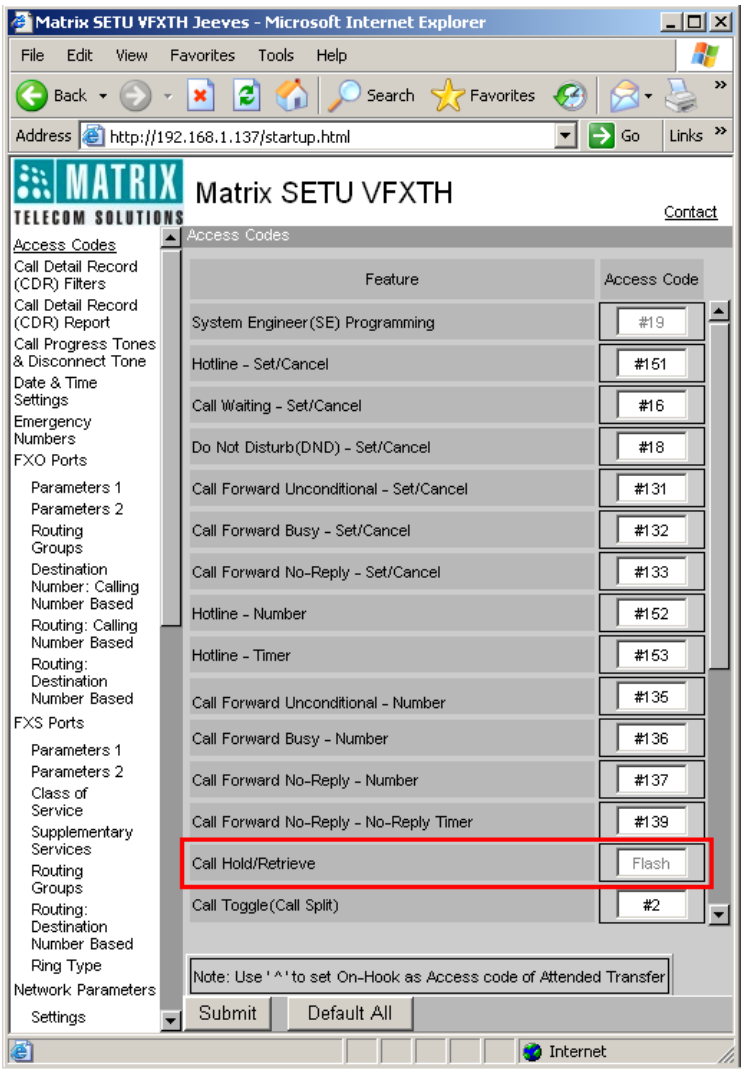
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**Class of Service**' link. Select 'Yes' to enable call hold for the ports on which it is to be allowed. By default, 'Call Hold' feature is set to 'No' in Class of Service i.e. it is disabled.



- Click on '**Access Codes**' link. By default access code for call hold/retrieve is Flash and it is un-editable.



Relevant Topics:

1. “Access Codes” 25
2. “Attended Call Transfer” 182
3. “Blind Call Transfer” 184
4. “Call Processing” 47
5. “Conference” 198
6. “Call Toggle (Call Split)” 192
7. “Call Waiting” 194
8. “Class of Service” 55
9. “Port Parameters-FXS” 112
10. “Routing Mechanism on FXS” 155

Call Toggle (Call Split)

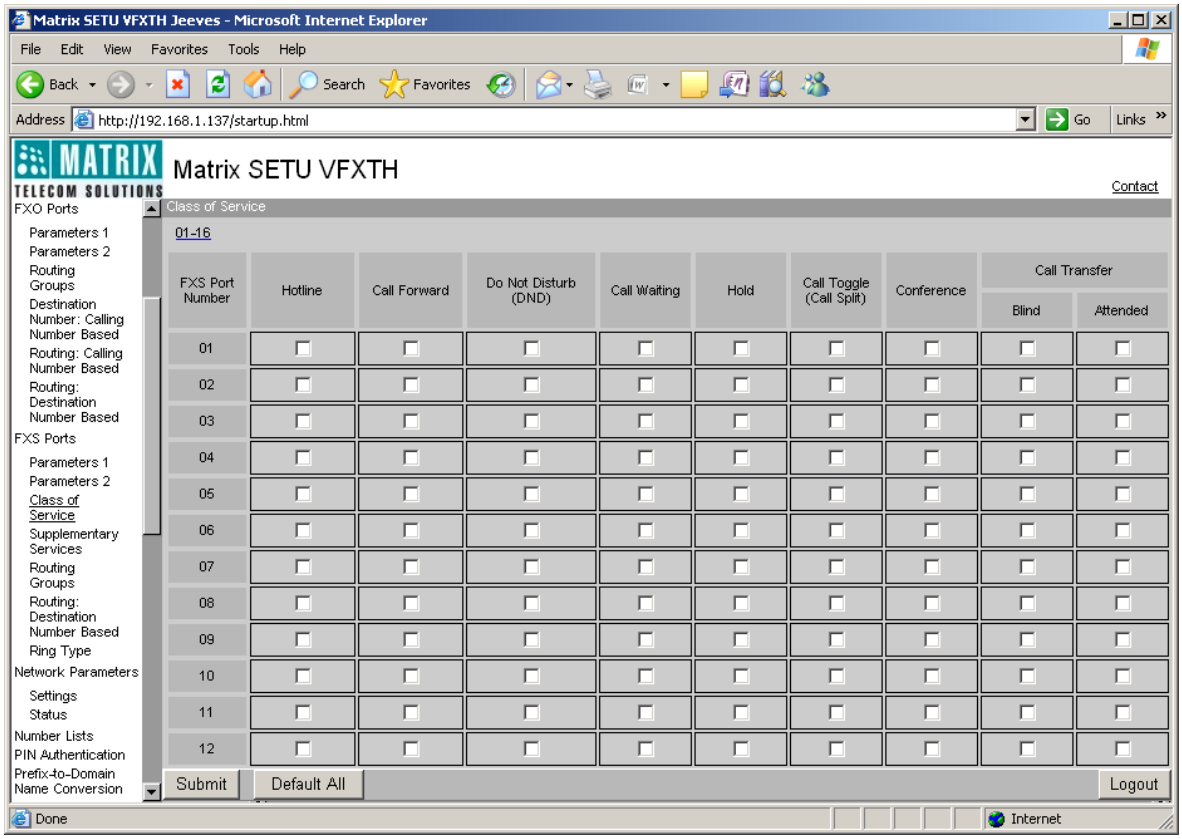
SETU VFXTH enables user to switch between an active call and a held call. User should dial call toggle access code to attend the held call and hold the active call.

Call Toggle feature will not work if it is disabled for the FXS Port in the Class of Service. Call Toggle (Call Split) feature can be used only if two held calls are present on the FXS Port.

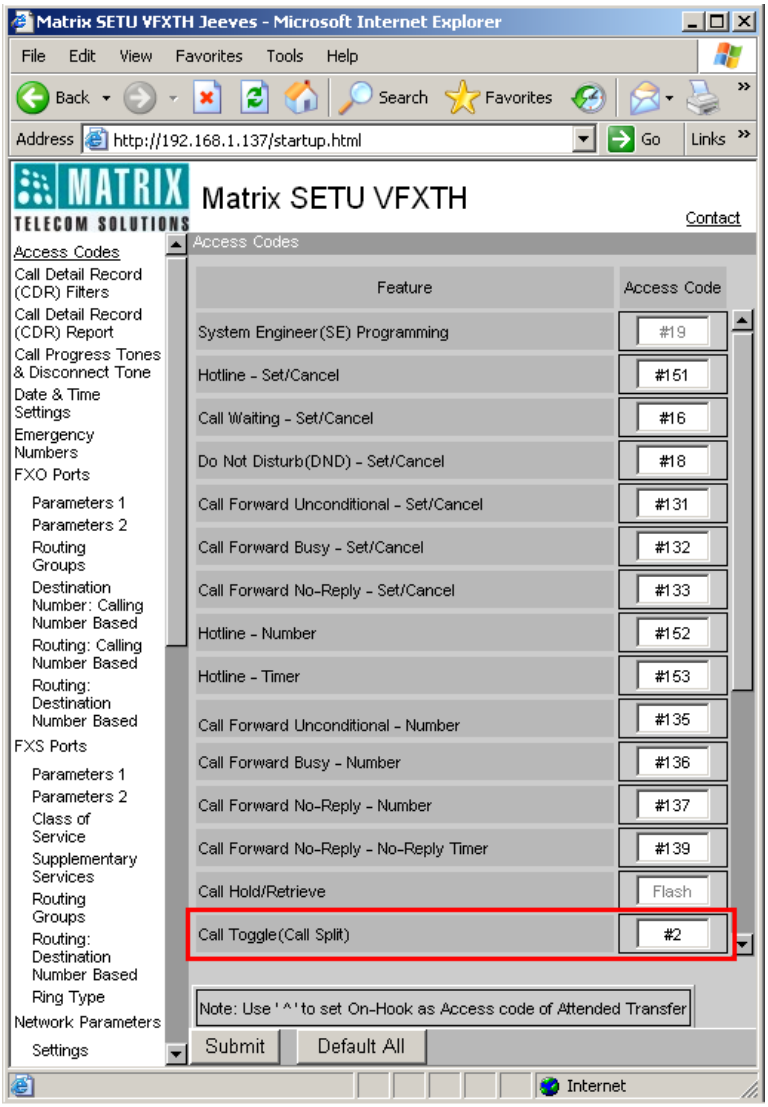
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**Class of Service**' link and enable Call Toggle (Call Split) for the ports on which it is to be allowed. By default, Call Toggle feature is set to 'No' i.e. it is disabled.



- Click on '**Access Codes**' link and program Call Toggle access code to use this feature. Default access code for Call Toggle (Call Split) is #2.



Call Toggle (Call Split) feature can be used during call conference also. To use Call Toggle (Call Split) feature user should dial call toggle access code to toggle between the calls in conference.

Relevant Topics:

1. “Access Codes” 25
2. “Call Hold” 190
3. “Call Waiting” 194
4. “Class of Service” 55
5. “Conference” 198

Call Waiting

Call Waiting feature notifies the user of another incoming call on the FXS Port, when he is busy on a call. User will get beeps during speech when there is a waiting call for him.

He has four options when there is a waiting call for him.

1. Reject the waiting call.
2. Ignore the waiting call.
3. Accept waiting call and hold the current call.
4. Accept waiting call and disconnect the current call.

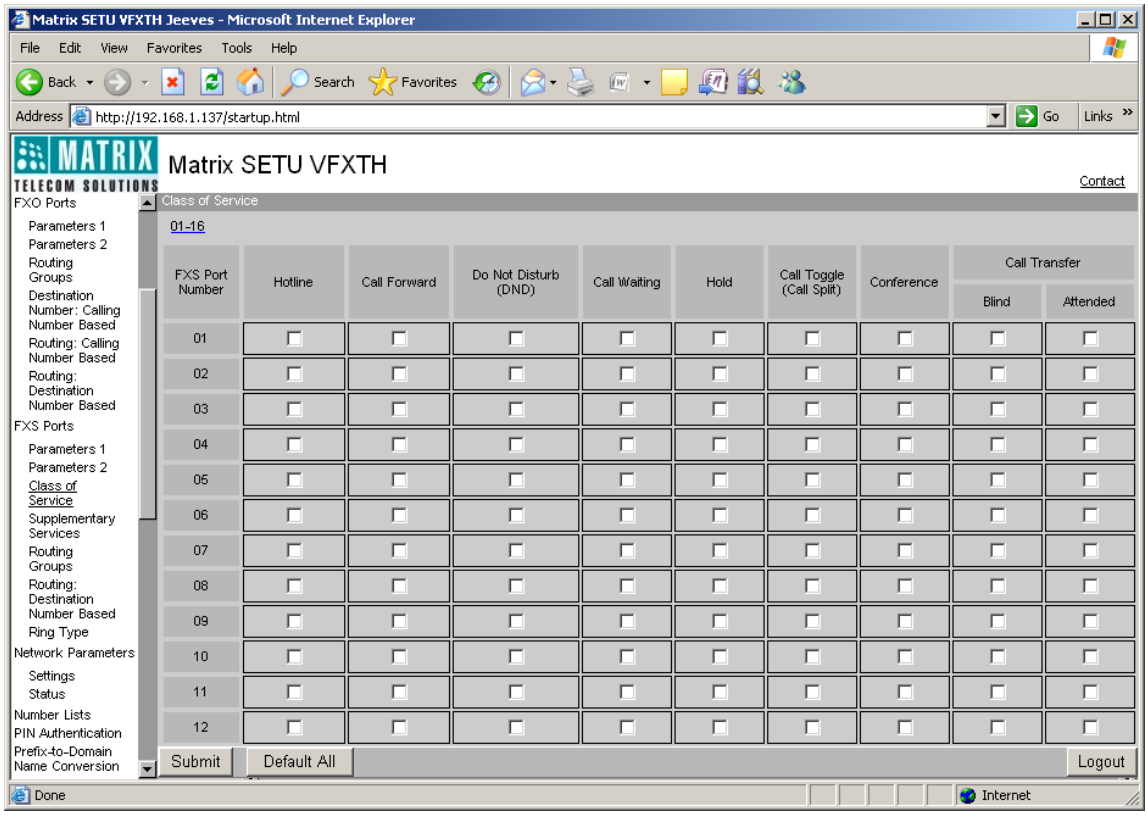
How it works?

A is in speech with B	
C calls A	A gets beeps while talking to B
Reject the waiting call	
A dials ***Flash-#31***	Beeps stop. C's call is rejected. A is in speech with B
Ignore the waiting call	
A dials ***Flash-#32***	Beeps stop. C will search for another free member in the group. A is in speech with B
Answer the waiting call and hold the current call	
A dials ***Flash-#33***	B goes on Hold. Speech is established between A and C.
Disconnect the current call and accept the waiting call	
A dials ***Flash-#34***	B gets disconnected. Speech is established between A and C.

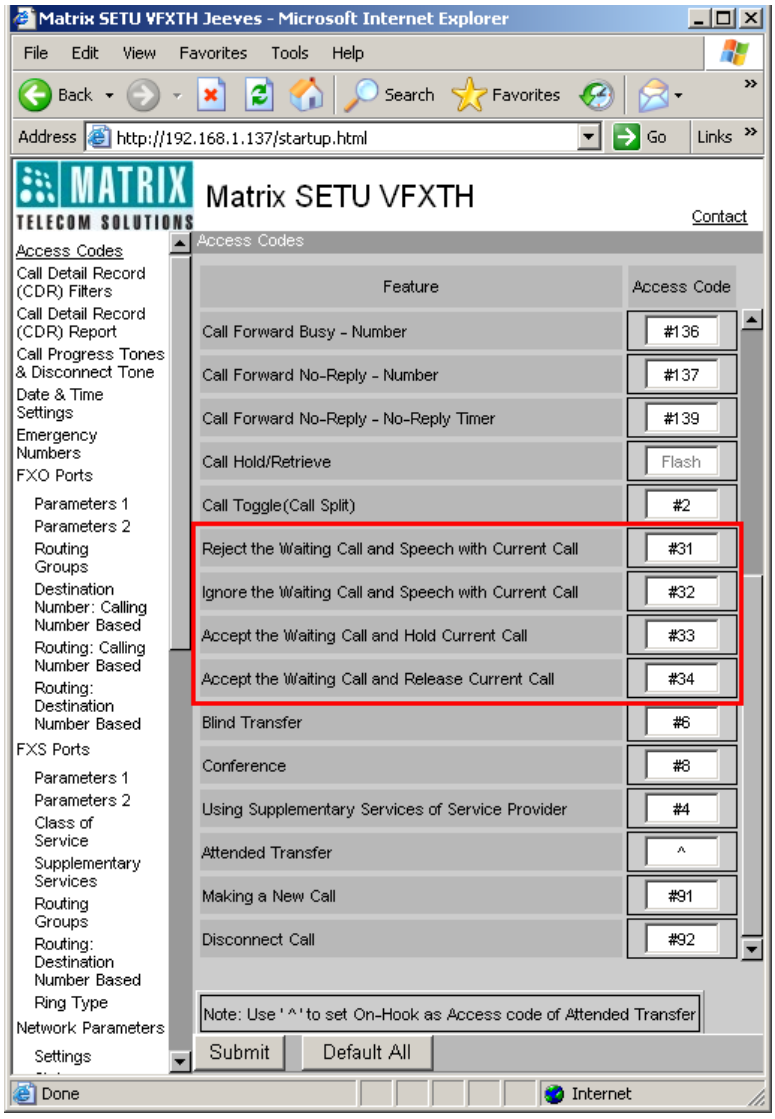
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**Class of Service**' link. Select 'Yes' to enable Call Waiting for the FXS Ports on which it is to be allowed. By default, Call Waiting is set to 'No' i.e. disable.



- Click on '**Access Codes**' link. Program access codes to enable/ disable Call Waiting for FXS Ports.

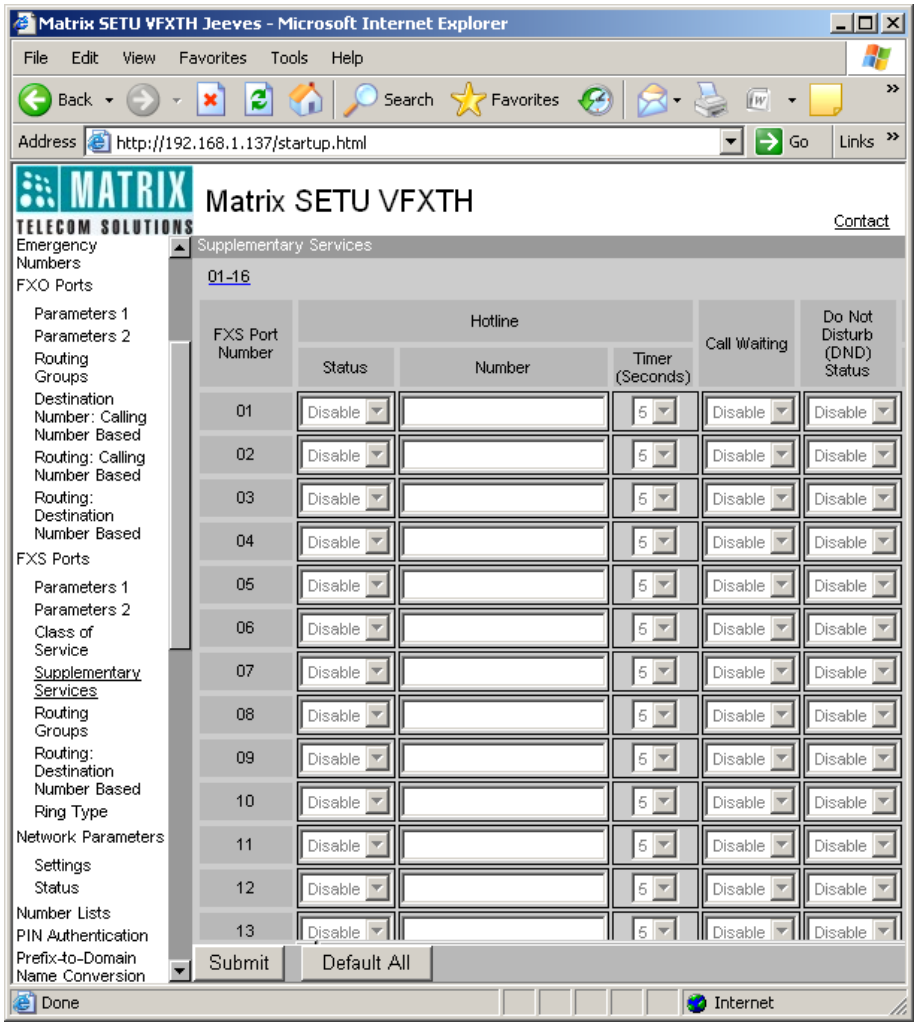


- Also program access codes for: Reject the Waiting Call and Speech with Current Call, Ignore the Waiting Call and Speech with Current Call, Accept the Waiting Call and Hold Current Call and Accept the Waiting Call and Release Current Call.

Default Access Codes to use call waiting feature is shown in the table given below:

Feature	Access Code
Set Call Waiting	***#16-1***
Cancel Call Waiting	***#16-0***
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***

- Click on '**Supplementary Services**' link and enable the Status of Call Waiting feature to use it. By default, it is disabled.



a. *Call waiting can be enabled/ disabled using telephone instrument as well as Web JEEVES but access codes can be programmed only using Web JEEVES.*

b. *This feature can be enabled/ disabled independently on all FXS Ports in Class of Service. When disabled, the user does not get waiting beeps for the waiting call.*

c. *Call Waiting feature has priority over Call Forward-When Busy in call mature state i.e. when the FXS Port is in speech with other port.*

d. *Call Waiting feature does not apply:*

i. *If Call Waiting feature is disabled.*
ii. *If Waiting Call is ignored.*
iii. *If already one Waiting Call is present.*
iv. *In Programming Mode.*
v. *In Conference.*
vi. *In Remote Held condition.*
vii. *In Dial state, Routing state and in Disconnect state.*

Relevant Topics:

1. “Access Codes” 25
2. “Call Processing” 47
3. “Call Progress Tones” 48
4. “Class of Service” 55

Conference

Conference feature enables user to talk to two parties simultaneously i.e. three people can talk to each other at a same time. User can also toggle between two remote parties during or after the Conference. This feature is applicable on FXS Ports only.

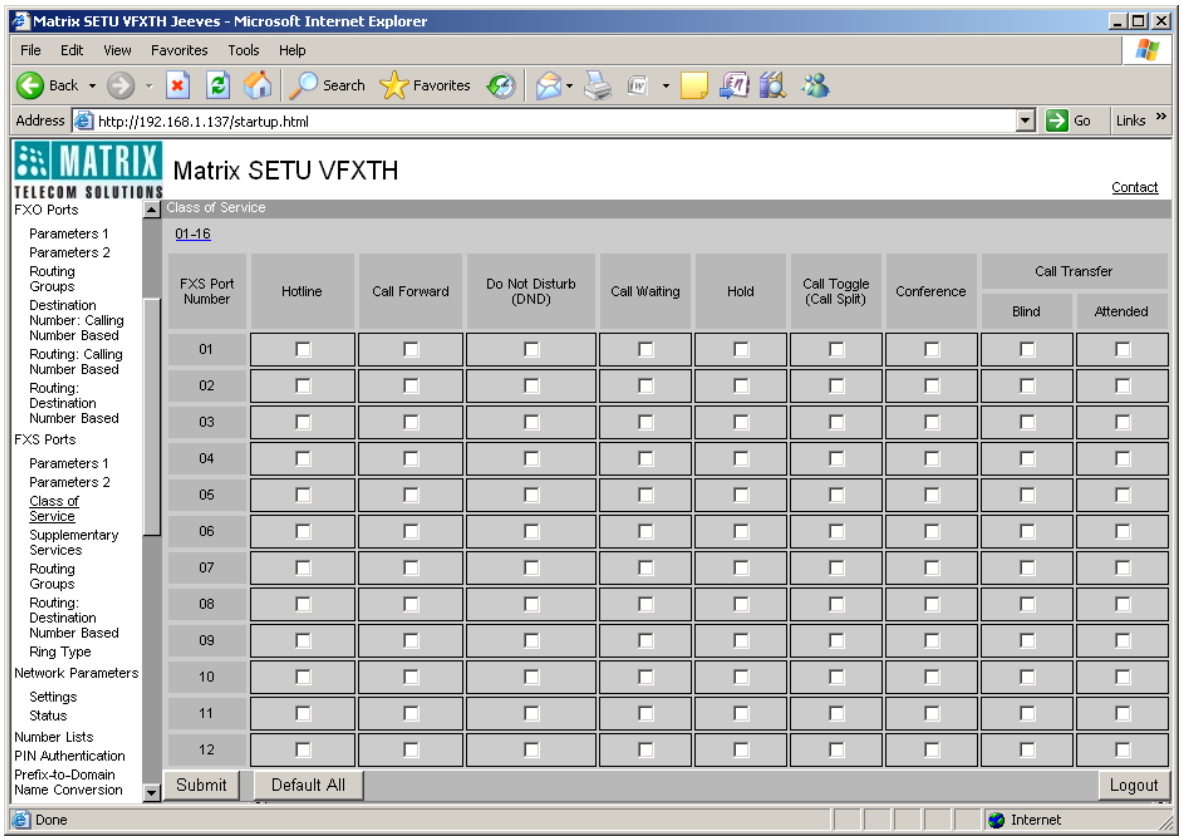
How it works?

A in speech with B	
To conduct Call Conference	
A dials ***Flash***	B goes on hold. A gets dial tone
A dials C	A in speech with C
A dials ***Flash-#8***	A in speech with B and C (3-Party Conference begin)
To terminate Call Conference	
A goes On-hook	Calls with both B and C is released
To convert Call Conference to Call Toggle	
A dials ***Flash-#2***	A in speech with B. C goes on hold (3-Party Conference converted to Call Toggle)

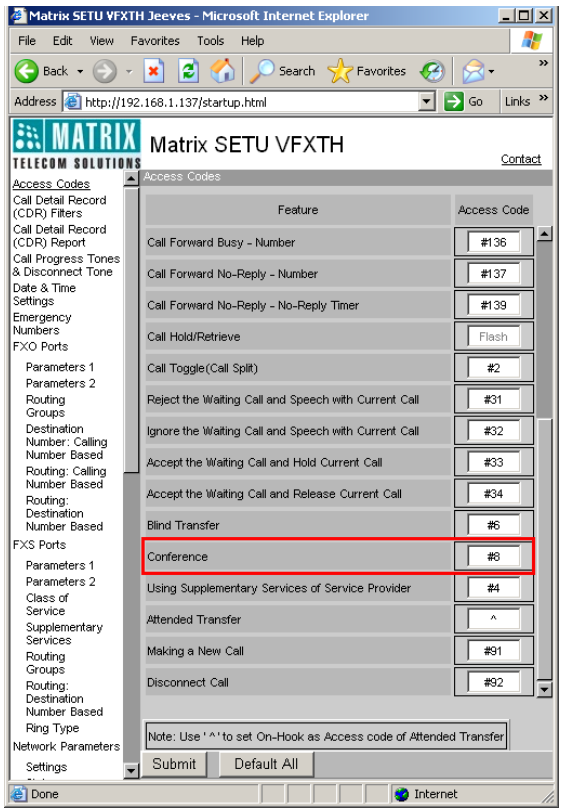
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**Class of Service**' link and enable conference for the ports on which it is to be allowed.



- Click on '**Access Codes**' link and program Conference access code to use this feature. Default access code for Call Conference is #8.



- *3-Party Conference can be converted to Call Toggle by dialing Call Toggle (Call Split) access code. User will be connected to one of the parties and the other party goes on hold.*
- *Conference is not allowed to the FXS Port, which is already in conference.*

Relevant Topics:

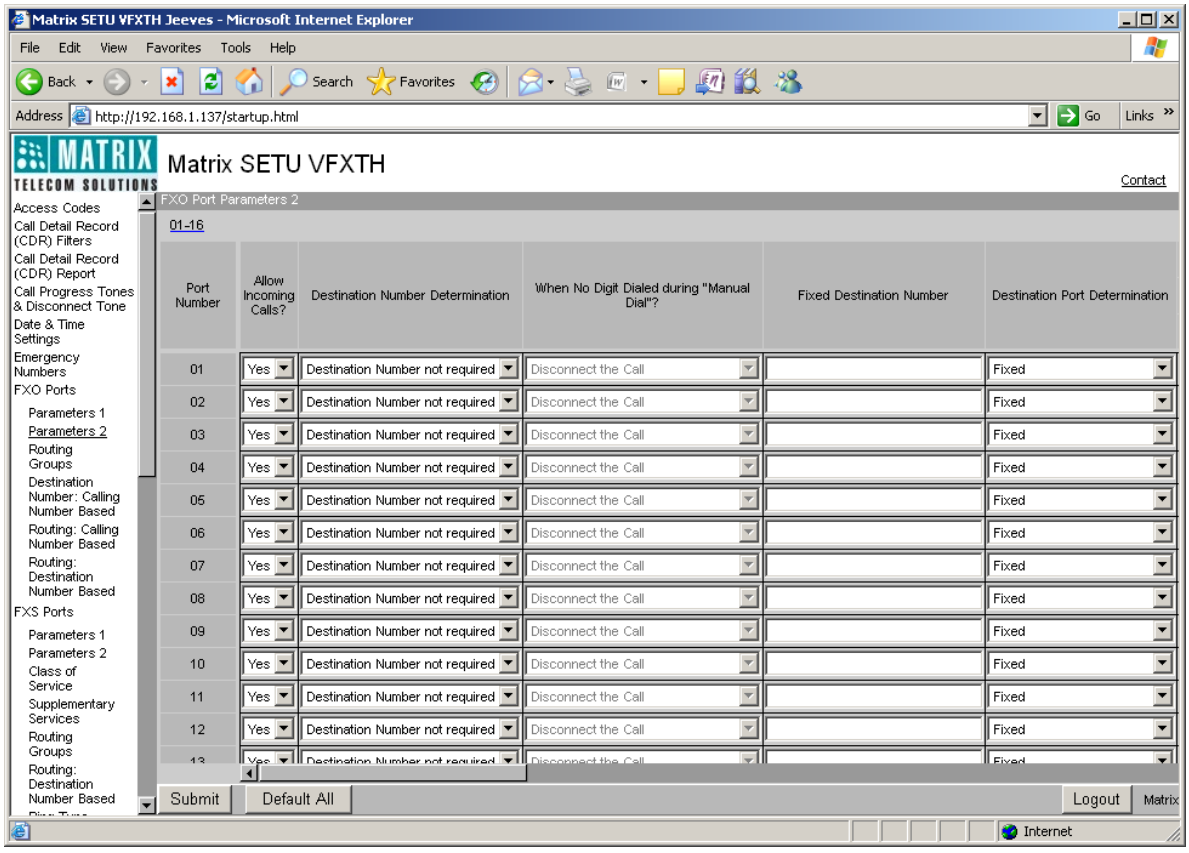
1. “Access Codes” 25
2. “Call Hold” 190
3. “Call Toggle (Call Split)” 192
4. “Class of Service” 55

Disconnect call using Access Code

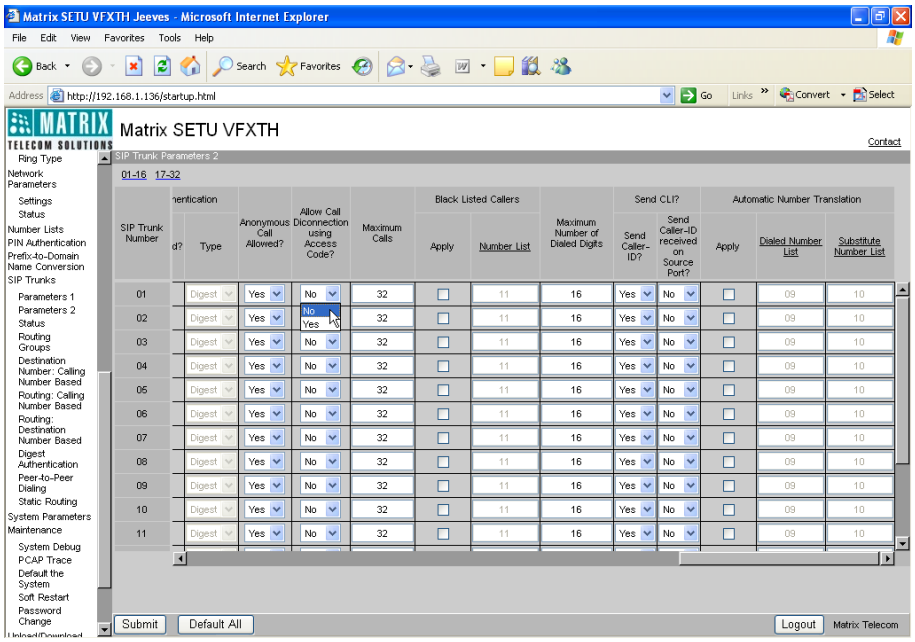
SETU VFXTH enables user to disconnect the call using access code. When the call disconnect access code is dialed, SETU VFXTH releases the port engaged in the call. Disconnecting call using access code is applicable only on FXO Port and SIP Trunk.

How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on '**Access Codes**' link and program Disconnect Call access code. By default, it is #92.
- Click on '**FXO Port Parameters 2**' and select 'Yes' in 'Allow Call Disconnection using Access Code?' field to enable call disconnection using access code. By default, it is 'No'.



- Click on '**SIP Trunk Parameters 2**' and select 'Yes' in 'Allow Call Disconnection using Access Code?' field to enable call disconnection using access code. By default, it is 'No'.



Relevant Topics:

1. “Access Codes” 25
2. “Port Parameters-FXO” 104
3. “Port Parameters-SIP” 117

Do Not Disturb (DND)

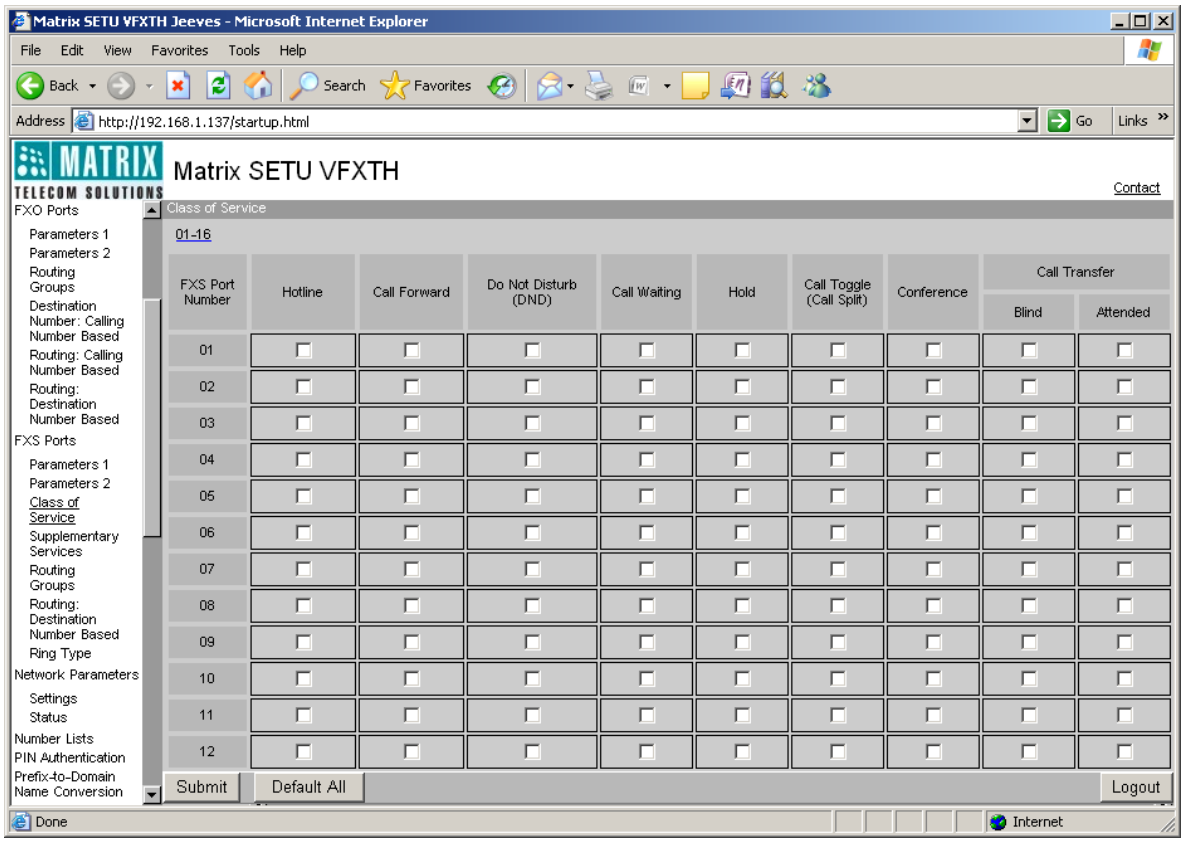
Do Not Disturb (DND) feature enables the user to have privacy for some duration i.e. it is useful to the user when he does not want to receive any calls for a particular time period but enables him to make outgoing calls.

Do Not Disturb (DND) is applicable on FXS Ports only and FXS Port user can enable and use this feature only if it is enabled in Class of Service for that port.

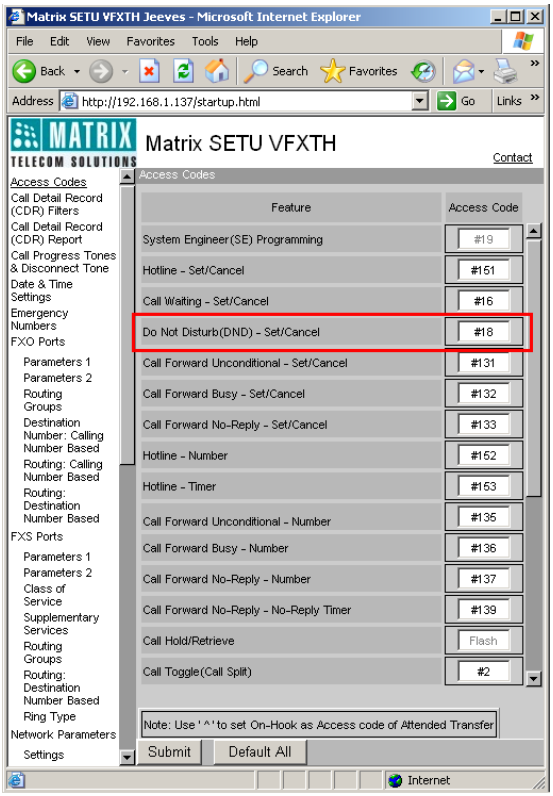
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

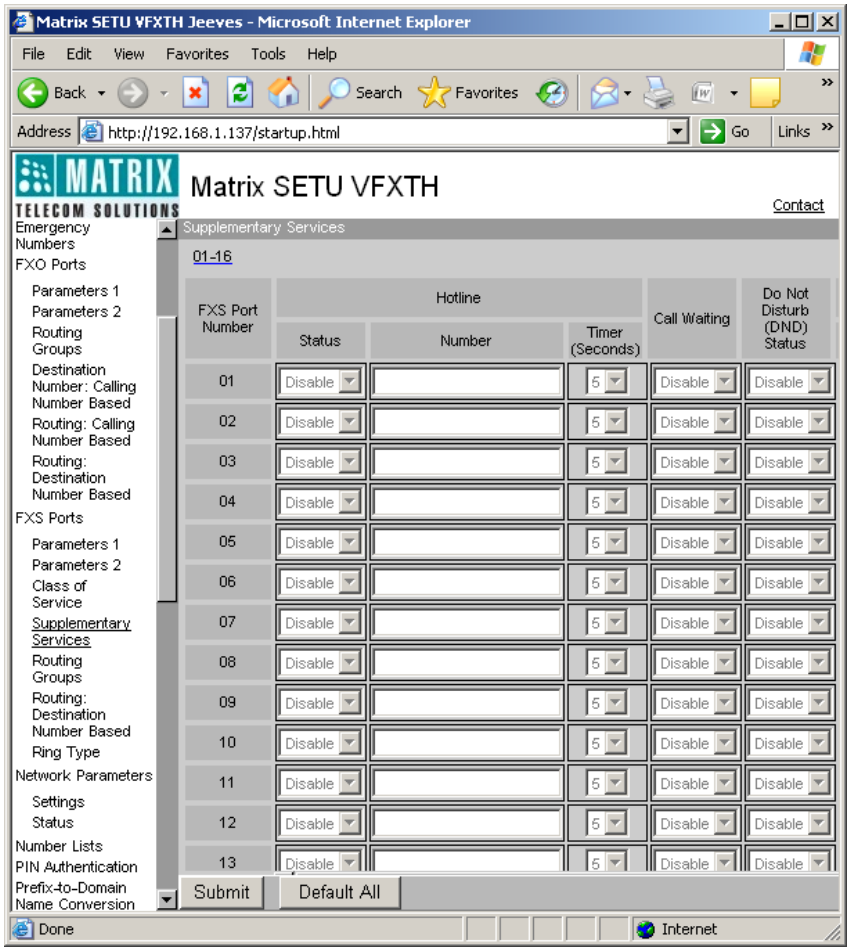
- Click on '**Class of Service**' link and enable Do Not Disturb (DND) on the port on which it is to be applied. By default, it is disabled.



- Click on '**Access Codes**' link and program access code to enable and disable this feature. Default access code to set Do Not Disturb (DND) is ***#18-1*** and to cancel Do Not Disturb (DND) is ***#18-0***.



- Click on '**Supplementary Services**' link and enable the Status of Do Not Disturb (DND) feature to use it. By default, it is disabled.



- *If Do Not Disturb (DND) and Call Forward-Unconditional, both are set then Call Forward-Unconditional shall have priority over Do Not Disturb (DND).*
- *Do Not Disturb (DND) have priority over Call Forward- No Reply and Call Forward- Busy.*

Relevant Topics:

1. “Access Codes” 25
2. “Call Processing” 47
3. “Class of Service” 55

Hotline

Every organization has fix customers, suppliers, dealers etc. with whom the people of organization frequently interact. Dialing same long number again and again is time consuming and frustrating. SETU VFXTH provides facility of Hotline feature as a solution to this problem.

Hotline feature relieves the caller from dialing long frequently dialed numbers again and again. This feature is applicable for FXS Port only. To use this feature, SE shall program frequently dialed number as Hotline numbers so that whenever the port goes Off-Hook, the number is dialed out automatically after a short delay. The number is dialed out after the expiry of hotline timer programmed by SE. Hotline timer is the time for which the FXS Port should wait after going Off-Hook to dial the hotline number.

To use this feature, SE should enable 'Hotline' in Class of Service for all FXS Ports. By default, Hotline feature is disabled for all FXS Ports in Class of Service.

How to Program?

Hotline feature can be programmed either using telephone instrument or using Web JEEVES.

Programming Hotline using telephone instrument:

To Set/Cancel Hotline:
Dial ***#151-1*** to set Hotline.
Dial ***#151-0*** to cancel Hotline.
By default, Hotline is cancelled for all FXS Ports.

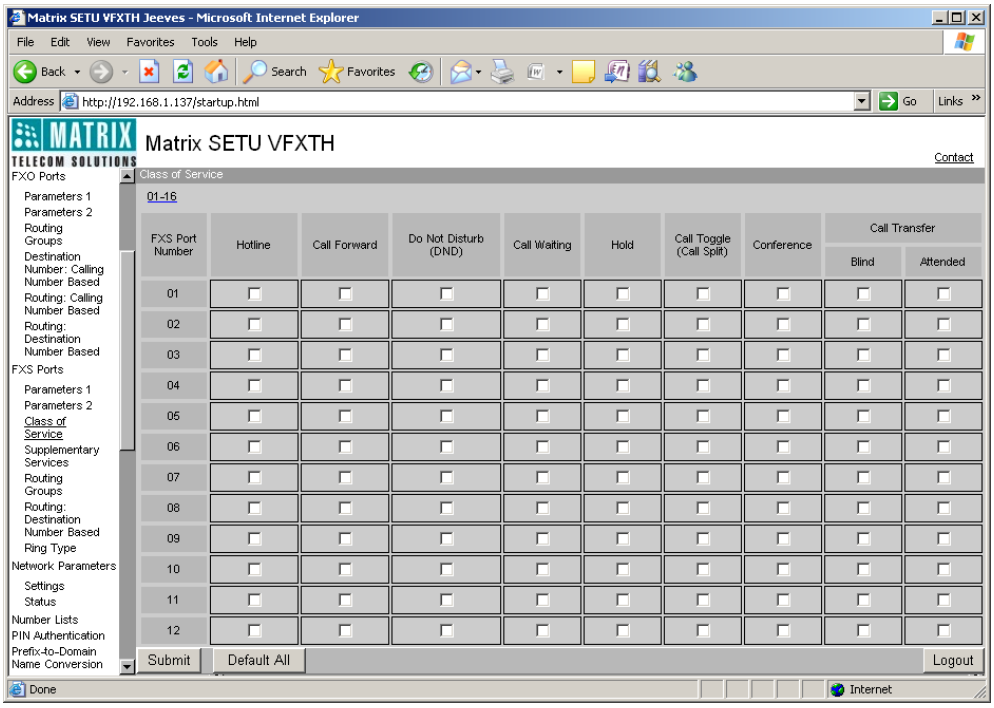
To program Hotline number:
Dial ***#152-Destination Number-End of Dialing***
Where,
Destination number can be of maximum 24 digits. Digits 0 to 9, *, #, dot (.) are allowed.
By default, it is blank for all FXS Ports.

Hotline number should be less than or equal to the maximum number of digits allowed to be dialed on the FXS Port.

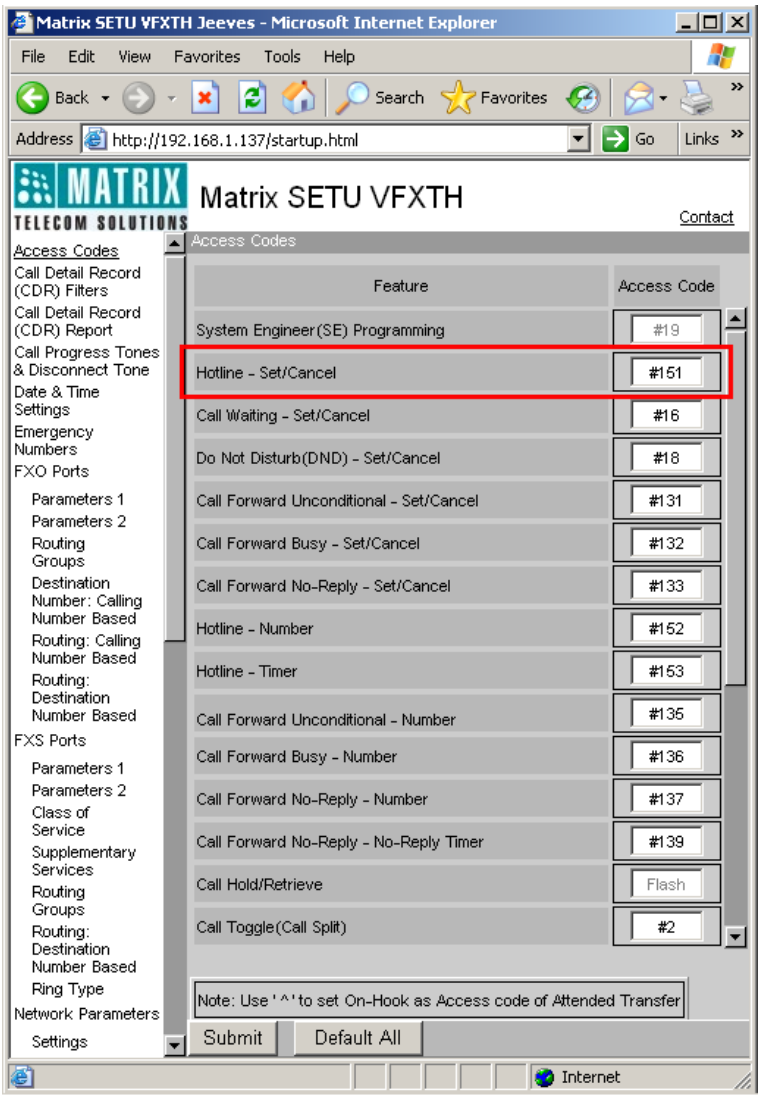
To program Hotline Timer:
Dial ***#153-X***
Where, X is the timer value ranging from 1 to 9 seconds.
By default, it is 5 seconds.

Programming Hotline using Web JEEVES:

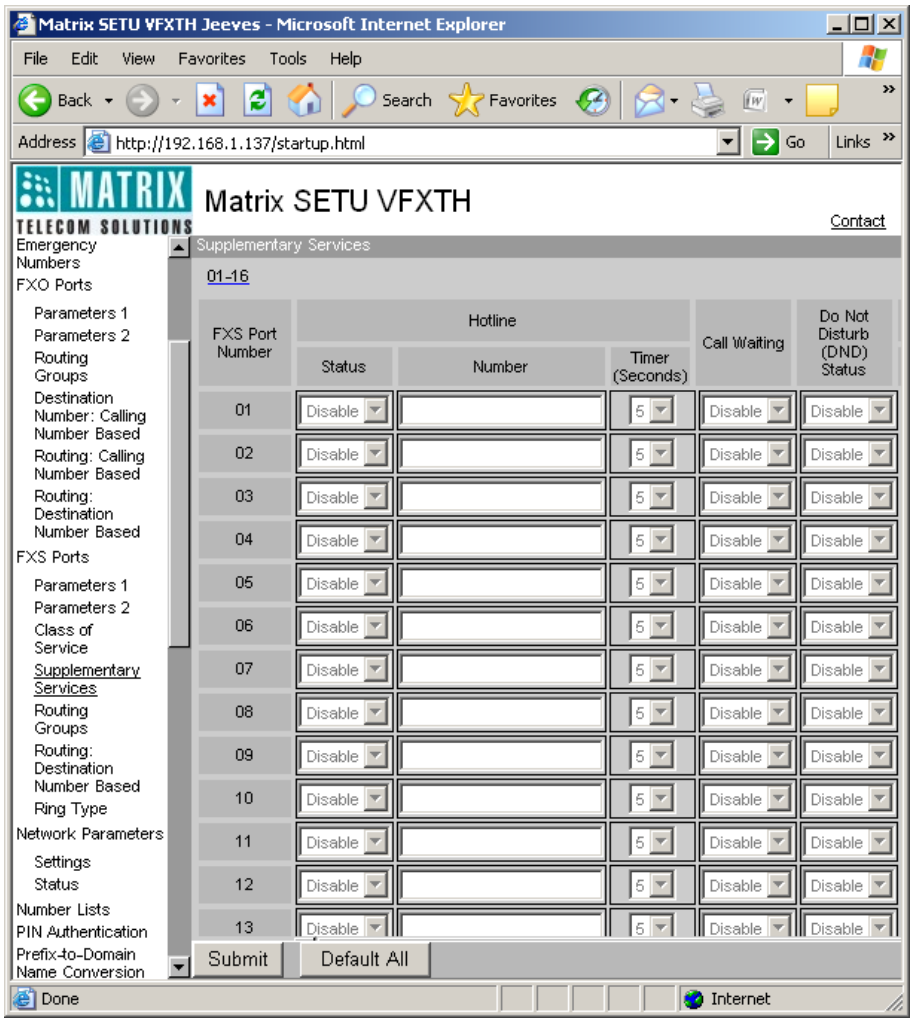
- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)
- Click on '**Class of Service**' link and enable hotline feature for the FXS Ports on which hotline is to be applied. By default, it is disabled for all ports.



- Click on '**Access Codes**' link and program access code to set/cancel hotline feature. Also program access code for hotline number and hotline timer in access code table.



- Click on '**Supplementary Services**' link and program the following parameters.



- **Status:** Enable the hotline status to use this feature. By default, it is disabled.
- **Number:** Program the hotline number for each FXS Port which is to be out-dialed from that port. By default, it is blank.
- **Timer (Seconds):** Program hotline timer for each FXS Port after which hotline number is to be out-dialed after going Off-Hook. By default, it is 5 seconds.

- *Allowed-denied number logic is applied on the hotline number.*
- *Hotline feature is allowed on FXS Port only if it is enabled in 'Class of Service' for that port.*

Relevant Topics:

1. “Allowed-Denied Numbers” 28
2. “Class of Service” 55
3. “Call Processing” 47
4. “Emergency Number Dialing” 66
5. “Routing Mechanism on FXS” 155

Making New Call

Making new call feature enables the caller to disconnect the current call and make a new call using SETU VFXTH, from a remote place, without calling SETU VFXTH and dialing PIN number and Password again and again.

This feature is applicable only on FXO Port. Making New Call access code is allowed to be dialed only from source port. To use this feature, program Destination Number Determination Method as 'Manual Dial' for the source port.

Let us understand this feature with an example:

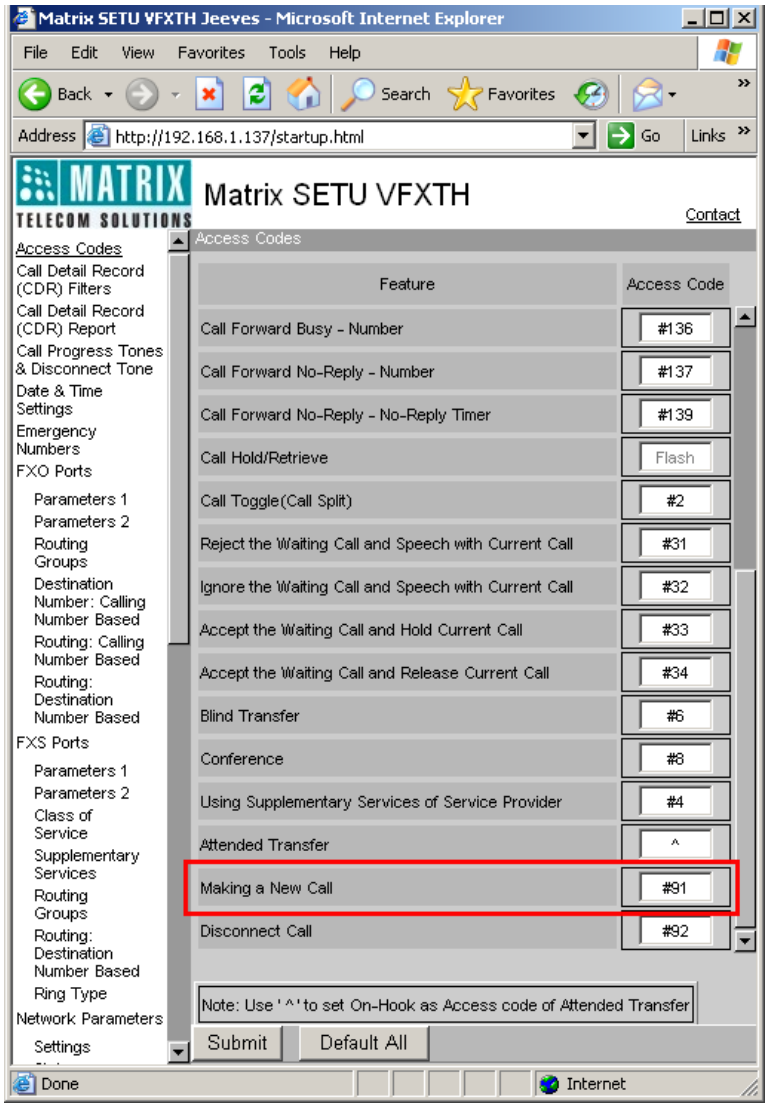
- A Cyber-Cafe owner has installed SETU VFXTH at his cafe and programmed it in such a way that it can be used for providing international call services to the end users.
- A residential user has subscribed for such services.
- Cyber-Cafe owner gives to the residential user, a number to call SETU VFXTH using which he can make calls to the desired international numbers from his home.
- In this scenario, the user has to call SETU VFXTH and dial PIN number and password every time he wants to make international call. This is very irksome and cumbersome process.
- To relieve user from this burden, SE at cyber cafe shall enable 'Making New Call' feature and program access code for the same.
- This access code is given to the user to enable him to make a new call after disconnecting the current call i.e. without calling SETU VFXTH and dialing PIN number and Password again.
- However, if remote end disconnects the call during speech then SETU VFXTH will give error tone for 4 seconds followed by dial tone to enable user to make new call directly i.e. without dialing making new call access code.

  This is how making new call feature works.

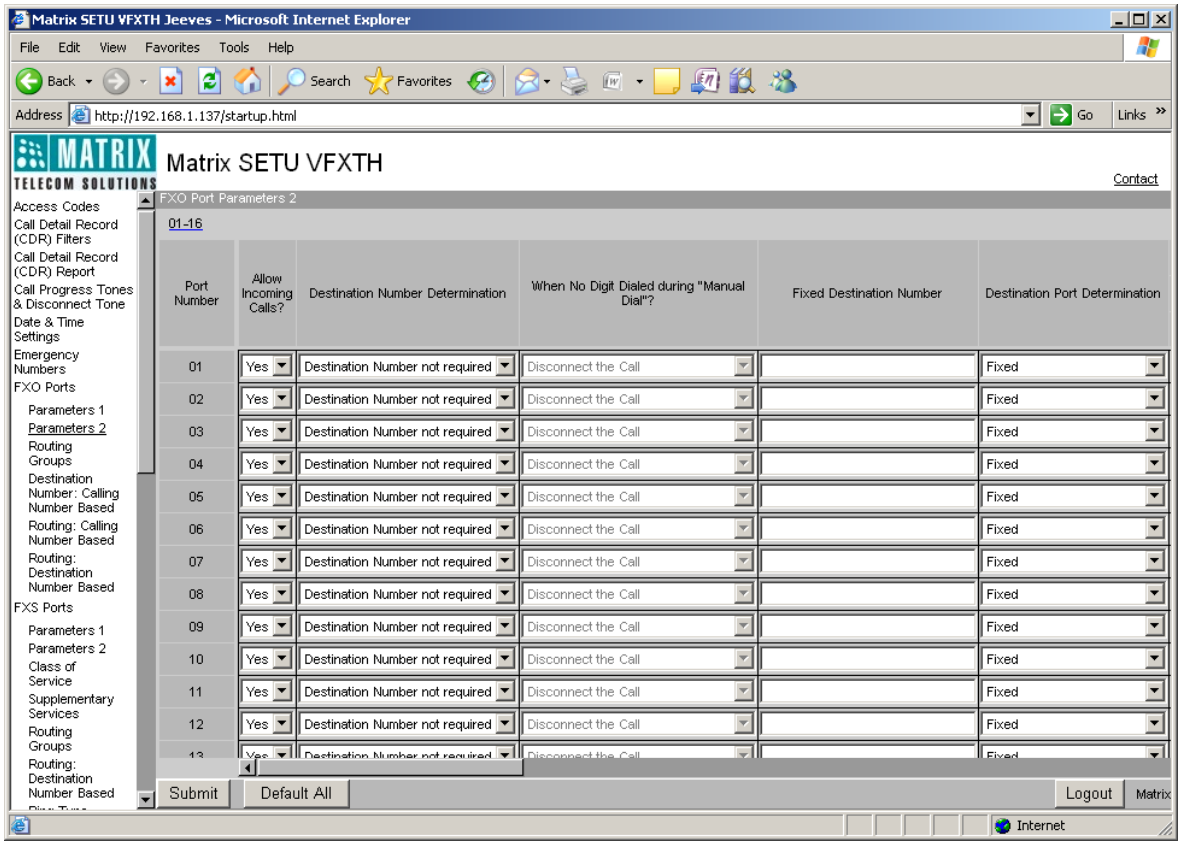
How to Program?

- Open Web JEEVES of SETU VFXTH. (Refer “Accessing Web JEEVES”)

- Click on '**Access Codes**' link and program Access Code for 'Making New Call'. By default, it is #91.



- Click on '**FXO Port Parameters 2**' link. Select 'Yes' in 'Allow New Call using Access Code?' field to activate this feature for the desired FXO Ports. By default, it is set to 'No'.



- *Making New Call access code can be used only after collecting the destination number string from the source port and in the matured state from the source port.*
- *Making New Call access code will be ignored if Destination Number Determination Method is programmed other than the Manual Dial.*
- *While using Making New Call feature, if user does not dial any digit during Manual Dial then 'Route to Fixed Destination Number' option is not applicable i.e. the call will be disconnected.*

Relevant Topics:

1. “Access Codes” 25
2. “Port Parameters-FXO” 104

Making Second Call

Making Second Call feature enables the user to make a second call by keeping the first call on hold. This feature is applicable on FXS Port only.

How it works?

A and B are in speech, in between A wants to talk to C. (make second call)

A is in speech with B	
A dials Flash	B goes on hold. A gets feature tone.

A dials C	A in speech with C
C goes on-hook	A gets connected to B
A in speech with B	

- *To use making second call feature, call hold feature should be enabled in Class of Service.*
- *After making second call, user can toggle between first and second call using call toggle access code and can also conduct conference by dialing conference access code.*

Relevant Topics:

1. “Call Processing” 47
2. “Call Hold” 190

Appendix

Acronyms

ARP	Address Resolution Protocol
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

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
SME	Small and Medium Enterprise
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

Features at Glance

Feature Description	Feature Code
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 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 Attended Transfer	**^**
For Making a New Call	***#91***
To Disconnect Call	***#92***

a. Dial # as end of dialing if end of dialing digit is programmed or wait till expiry of inter digit wait timer.

System Commands

Description	System Commands
To Enter Programming Mode	***#19-User Password***
To Exit Programming Mode	***00#****
To Program Network IP Address	***11-IP Address-#****
To Program Network Subnet Mask	***12-Subnet Mask-#****
To Program the Connection Type	***10-Code-#****
To Enable/Disable VLAN tag	***31-Code-#****
To Display the Connection Type	***20-#* - Go On-Hook***
To Display the Network IP Address	***21-#* - Go On-Hook***
To Display the Network Subnet Mask	***22-#* - Go On-Hook***
To Display the Network Gateway Address	***23-#* - Go On-Hook***
To Display the DNS Address	***24-#* - Go On-Hook***
To Display the Status of SIP Trunks	***27-SIP Trunk-#****

Product Specifications

Port Description

Port Name	Application	Connector Type
FXS Port	To connect analog phone or fax machine	RJ11
FXO Port	To connect with PSTN or to FXS ports of PBX	RJ11
SIP Trunks	To connect to Internet Network for VoIP	RJ45 (Ethernet)

Different Configurations supported in the SETU VFXTH

Sr. No.	Configuration	VoIP Channels	FXO Ports	FXS Ports
1	SETU VFXTH-0016	16	0	16
2	SETU VFXTH-0024	24	0	24
3	SETU VFXTH-0032	32	0	32
4	SETU VFXTH-0800	8	8	0
5	SETU VFXTH-1600	16	16	0
6	SETU VFXTH-2400	24	24	0
7	SETU VFXTH-3200	32	32	0
8	SETU VFXTH-0808	16	8	8
9	SETU VFXTH-1212	24	12	12
10	SETU VFXTH-1616	32	16	16

FXS Port Parameters

Signaling	Loop Start
Connector	RJ45
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 (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: -3dB to +6dB; Rx Gain: -3dB to +6dB
Answer Signaling on FXS	Battery Reversal
Disconnect Signaling on FXS	Battery Reversal and Open Loop Disconnect

FXO Port Parameters

Signaling	Loop Start
Connector	RJ11
Off-Hook Line Impedance	600Ω/900Ω/Complex
Loop Limit	1200Ω
Pulse Dialing	10PPS and 20PPS
DTMF Dialing and Reception	ITU-T Q.23 and Q.24
CLI Reception	DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	Over Voltage and Over Current Secondary Protection
Return Loss	>18dB
Longitudinal Balance	>50dB
Transmission Level Adjust	Tx Gain: -15dB to +10dB; Rx Gain: -15dB to +10dB
Call Maturity	Delay and Polarity Reversal
Answer Supervision on FXO	Battery Reversal
Disconnect Supervision on FXO	Battery Reversal and Open Loop Disconnect

VoIP Parameters

Connector	RJ45
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833)
Network Protocols	IPv4, TCP, UDP, DHCP, SNTP, STUN, HTTP
SIP	32 SIP Trunks with Out Bound Proxy Support
NAT	STUN and NAT Keep Alive
Voice Codec's	G.711 (a-Law and mu-Law), G.729ab, G.723-L, G.723-H, 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

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 VFXTH
- Black Listed Callers
- Call Disconnect Tone
- Call Forwarding
- Call Hold
- 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

Time Settings

Synchronizing with specific Time Server

Provisioning, Administration and Maintenance

- Configurable using Access Codes
- Programmable using Web Jeeves
- Software Upgrade (Using FTP Server)

LED Indication (Total 34 LEDs)

- Power = 1
- Status = 1
- Port = 32

Packing

- **Dimension (W x H x D):** 40.7 x 5.1 x 17.2 cm (16.0" x 2.0" x 6.8")
- **Unit Weight:** XX kg.
- **Shipping Weight:** XX kg.
- **Mounting:** Wall Mounting or Table-Top

Power Supply

- **External Adaptor:** 24V DC @2.5A
- **Power Consumption:** 60 Watts
- **Connector:** DC Power Jack

Environmental

- **Operating Temperature:** -10°C to 50°C (14°F to 122°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)

Warranty Statement

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4. Products damaged due to operation of product outside the products' specifications or use without designated protections.

Warranty valid only if:

- Primary protection on all the ports provided.
- Mains supply is within limit and protected.
- Environment conditions are maintained as per the product specifications.

Warranty Card:

When the product is installed, please return the warranty card with:

- Date, signature and stamp of the customer.
- Date, signature and stamp of the channel partner.

Matrix assumes that the customer agrees with the warranty terms even when the warranty card is not signed and returned as suggested.
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