SETU VFX88L
SETU VFX44L
System Manual

TELECOM SOLUTIONS
▶ www.matrixtelecom.hu ▶▶

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Documentation Information

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Contents

Section 1: Introduction

Welcome

Welcome to the world of telecom solutions from Matrix and thanks for purchasing a Matrix product.

We want you to get the maximum performance from our product. If you run into technical difficulties, we are here to help. But please consult this system manual first.

If you still can't find the answer, gather all the information or questions that apply to your problem and with the product close to you, call your dealer. Matrix dealers are trained and ready to give you the support you need to get the most from your Matrix product. In fact, most problems reported are minor and can be easily solved over the phone.

In addition, technical consultation is available from Matrix engineers every business day. We are always ready to give advice on application requirements or specific information on installation and operation of our products.

The system manual is divided in following sections:

Section 1: ***Introduction***
Section 2: ***Features and Facilities***
Section 3: ***Appendices***

We suggest the first time users to read this system manual in following sequence and then remaining chapters:

- Section 1
- Section 2 (In hierarchy given below):

The words 'SETU VFX88L/VFX44L' and 'System' and 'Gateway' are used interchangeably in the manual.

Packing List

The ideal sales package for SETU VFX88L/VFX44L is as mentioned below:

Sr.	Accessories	Qty.
01	SETU VFX88L/VFX44L	1
02	System Manual	1
03	Adaptor 12V, 2.0Amp.	1
04	Mounting Screw 30/7	2
05	Screw Grip	2
06	Warranty Card set	1
07	Support Card	1
08	RJ11 Cable	9
09	RJ45 Cable	1
10	Mounting Template	1
11	Technical Document CD	1

- Please make sure that these components are present.
- In case of short supply or damage detection, contact the source from where you have purchased the system.

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Warranty Statement

Matrix warrants to its consumer purchaser any of its products to be free of defects in material, workmanship and performance for a period of 15 months from date of manufacturing or 12 months from the date of installation which ever is earlier. During this warranty period, Matrix will at its option, repair or replace the product at no additional charge if the product is found to have manufacturing defect. Any replacement product or part/s may be furnished on an exchange basis, which shall be new or like-new, provided that it has functionality at least equal to that of the product, being replaced. All replacement parts and products will be the property of Matrix. Parts repaired or replaced will be under warranty throughout the remainder of the original warranty period only.

This limited warranty does not apply to:

1. Products that have been subjected to abuse, accident, natural disaster, misuse, modification, tampering, faulty installation, lack of reasonable care, repair or service in any way that is not contemplated in the documentation for the product, or if the model or serial number has been altered, tampered with, defaced or removed.
2. Products which have been damaged by lightning storms, water or power surges or which have been neglected, altered, used for a purpose other than the one for which they were manufactured, repaired by customer or any party without Matrix's written authorization or used in any manner inconsistent with Matrix's instructions.
3. Products received improperly packed or physically damaged.
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.

The Purchaser shall have to bear shipping charges for sending product to Matrix for testing/rectification. The product shall be shipped to the Purchaser at no-charge if the material is found to be under warranty. The Purchaser shall have to either insure the product or assume liability for loss or damage during transit.
Matrix reserves the right to waive off or make any changes in its warranty policy without giving any notice.

If Matrix is unable to repair or replace, as applicable, a defective product which is covered by Matrix warranty, Matrix shall, within a reasonable time after being notified of the defect, refund the purchase price of the product provided the consumer/purchaser returns the product to Matrix.

In no event will Matrix be liable for any damages including lost profits, lost business, lost savings, downtime or delay, labor, repair or material cost, injury to person, property or other incidental or consequential damages arising out of use of or inability to use such product, even if Matrix has been advised of the possibility of such damages or losses or for any claim by any other party.

Except for the obligations specifically set forth in this Warranty Policy Statement, in no event shall Matrix be liable for any direct, indirect, special, incidental or consequential damages whether based on contract or any other legal theory and where advised of the possibility of such damages.

Neither Matrix nor any of its distributors, dealers or sub-dealers makes any other warranty of any kind, whether expressed or implied, with respect to Matrix products. Matrix and its distributors, dealers or sub-dealers specifically disclaim the implied warranties of merchantability and fitness for a particular purpose.

This warranty is not transferable and applies only to the original consumer purchaser of the Product. Warranty shall be void if the warranty card is not completed and registered with Matrix within 30 days of installation.

Protecting the System

Installation Precautions:

- Do not install in direct sunlight and excessive cold or humid places.
- Do not install at places where sulfuric gases are produced and in areas where there are thermal springs, etc. because it may damage the equipment or contacts.
- Do not install at places where shocks or vibrations are frequent or strong.
- Do not install at dusty places or places where water or oil may come into contact with the system.
- Do not obstruct area around the system (for reasons of maintenance and inspection be especially careful to allow space for cooling above and at the sides of the system).

Important Safety Instructions:

When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock and injury to persons, including the following:

- Read and understand all instructions.
- Unplug this product from the wall outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a dry and soft cloth for cleaning.
- Do not use this product near water. For example, near a bathtub, wash bowl, kitchen sink, or laundry tub, in a wet basement, or near a swimming pool.
- Do not open the system in power ON condition.
- Do not place this product on an unstable cart, stand or table. The product may fall, causing serious damage to the product.
- Interfacing cables should not touch the exposed power line cable.
- This product should be operated with proper supply voltage. If you are not sure about supply voltage, contact authorized dealer. It is advisable to give proper, stabilized power.
- Do not allow anything to rest on the power cord of product or AC-DC Adapter. Do not place this product where the cord will be misused by people walking on it.
- Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.

- To reduce the risk of electric shock, do not disassemble this product. Take it to a qualified serviceman when some service or repair work is required. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock when the appliance is subsequently used.
- Unplug this product from the wall outlet and contact qualified service personnel under the following conditions:
  a) When the power supply cord or plug is damaged or frayed.
  b) If liquid has been spilled into the product.
  c) If the product has been exposed to rain or water.
  d) If the product does not operate normally by following the operating instructions. Adjust only those controls, which are covered by the operating instructions because improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to normal operation.
  e) If the product has been dropped or the cabinet has been damaged.
  f) If the product exhibits a distinct change in performance.
- Do not use the telephone of the product to report a gas leak in the vicinity of the leak.

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Introducing the System

Introduction:

- The Matrix gateway ‘SETU VFX88L/VFX44L’ is a SIP based VoIP product. Sending voice signals over Internet is called Voice Over IP (VoIP), Session Initiated Protocol (SIP) is an internationally recognized standard for implementing VoIP.
- This is an innovative enterprise gateway that offers a rich set of functionality and sound quality. They are fully compatible with SIP industry standard and can interpret with many other SIP compliant devices and software on the market.
- The ‘VFX88L/VFX44L’ supports 1-Ethernet Port, 8-FXS Ports (4 FXS ports for VFX44L) and a Lifeline port for connecting a PSTN line for making OG call.
- It supports multiple SIP Accounts so that a powerful routing option from FXS to IP at very low cost is possible. The SIP account can be considered as a SIP trunk. VFX88L/VFX44L can be configured using Web pages.
- User can select any SIP trunk out of nine-trunks for IC or OG call. Maximum 8-calls (4 calls for VFX44L) can be set up simultaneously. If there is network problem, number can be dialed from Lifeline port also. But call can not be received from this port.
- The gateway also supports peer-to-peer calling and Emergency number dialing.
- User can use Access Codes for some features like Hotline and Do Not Disturb, by using phone programming.

Terminology used:

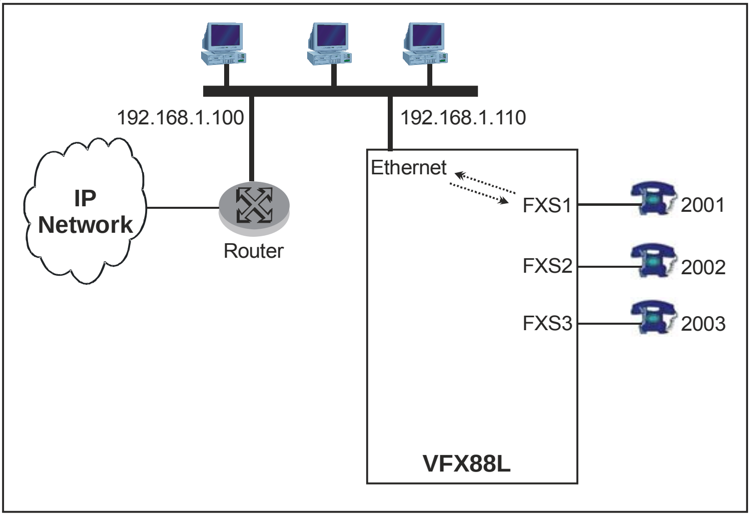
- The port on which the call originates is termed as “Originating Port” or “Source Port”.
- The port on which the call terminates is termed as “Terminating Port” or “Destination Port”.
- SE is System Engineer who programs the gateway.

Configuration-Examples:

Following illustrations show how you can connect the VFX88L to PBX or different phones. Similarly VFX44L, can be used.

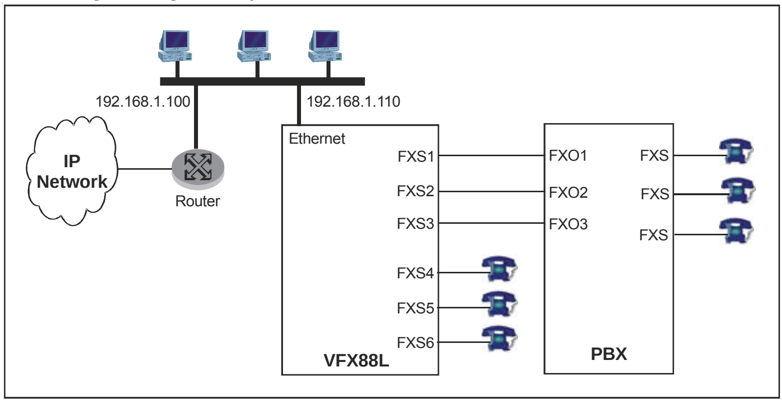
Application 1: Standalone

- You can make OG Calls and at the same time configure the gateway using PC connected at Ethernet port.



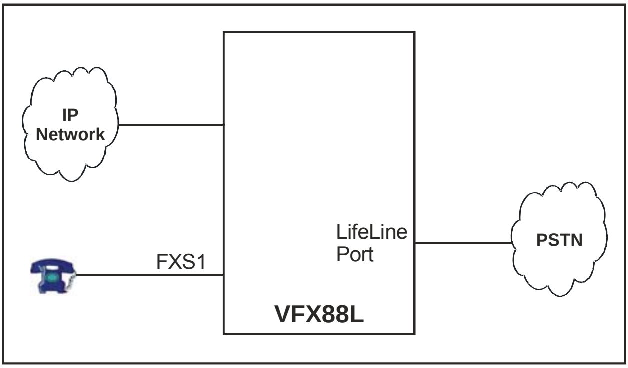
Application 2: Behind the PBX

- More extensions of the PBX can also access the IP Network through the gateway and make calls.



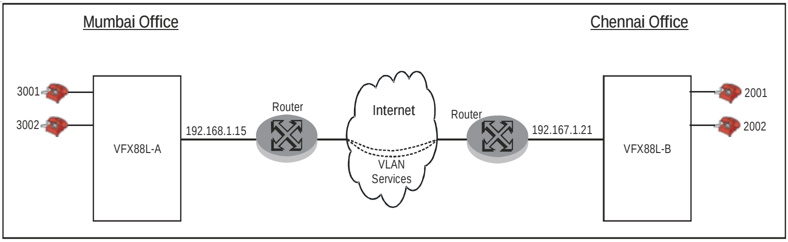
Application 3: Life Line

- In case of power down VFX88L can be used for OG call from the lifeline port (virtual FXO).



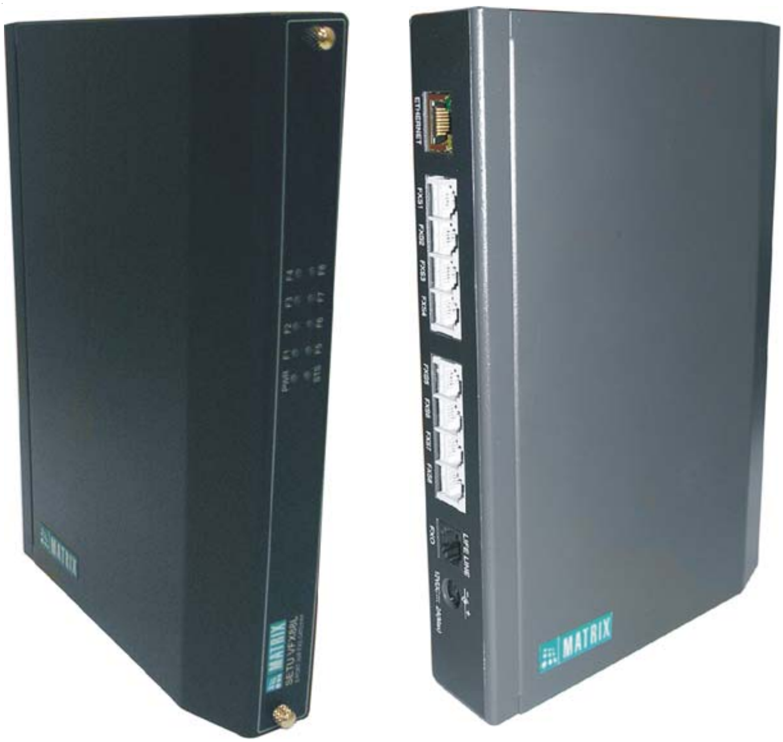
Application 4: Peer-to-Peer Calling

- You can dial IP Address to make call using non-proxy server to call another Gateway, using peer-to-peer calling feature.



Before proceeding further please ensure that you have an Internet access and at least one P2P SIP Account already set up. The default IP Address for the gateway is 192.168.001.156.

SETU VFX88L/VFX44L Photograph:



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Getting Started

Overview

- This chapter explains how VFX88L can be configured and used. Similarly VFX44L can also be used.
- The Matrix Gateway VFX88L allows you to use the analog telephone to make phone calls over the Internet.

This chapter gets the Gateway up and running quickly. The details which we have skipped to make this brief can be found elsewhere in the manual. It is divided into four sections:

1. Getting to know the Gateway.
2. Instruction for connecting the Gateway.
3. Basic steps for configuration.
4. Making phone calls.

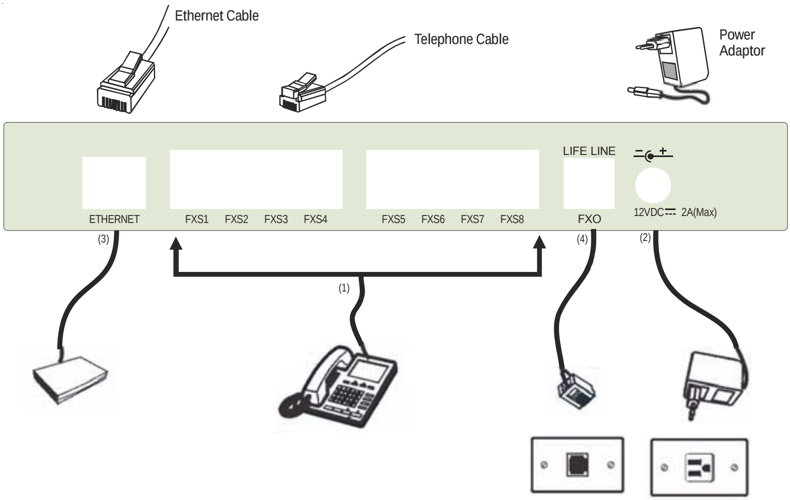
Getting to Know the Gateway

The Matrix Gateway‘s ports are located on side panel and LEDs are on front panel. Both are explained below:

- Left Side panel ports.
- Right Side panel LEDs.

Left Side Panel Ports:

Following picture shows how the ports are connected to various interfaces:



No.	Port Name	Connector	Description
1	FXS 1 to FXS 8	RJ 11	To connect analog telephone/fax.
2	Power	DC Jack	Power Adaptor, 12 Volt 2.0 Amp. DC.
3	Ethernet (WAN)	RJ 45 with two LED’s	This is a WAN Port. Connection to cable modem/ADSL modem or router or LAN switch. Ethernet activity LED (Green) is glowing. PC can be connected from Ethernet switch to access the Gateway.
4	Lifeline	RJ 11	For PSTN bypass/Lifeline feature.

Right Side Panel LEDs:
The VFX88L has ‘10’ LEDs; These LEDs are used to signify various events.

LED Label	Number of LED
PWR	1
F1 to F8	8 (One for each FXS port)
SYS	1

The Reset Sequence:

- At power ON, 'PWR' LED is glowing continuously, whereas all other LEDs glow for few seconds time.
- On successful completion of initialization cycle, each LED is glowing as per the normal conditions.

LEDs indication of the Gateway:
Reset Sequence:

Time	Response
At Power ON	All LEDs glow Green for 500ms, Red for 500ms, and Orange for 500ms once
Reset (Initialization) cycle	Takes 2-3 minutes (approx)

FXS Port LEDs During Normal Functioning of the Gateway:

Event/State/Status	Color	Cadence (in msec)			
		ON	OFF	ON	OFF
Port Idle + SIP Unregistered	-	OFF			
Port Idle + SIP Registered	Green	200	4800		
Port Idle + SIP Authentication failed	Green	200	200	200	4400
Incoming Ring Event	Red	1000	4000		
Off-Hook Event	Red	Continuous			
Speech	Red	Continuous			

Note:

- If FXS port is disabled, user will not get LED indication. It will be same as Port Idle condition.

SIP Account Status LED:

- The SIP Account Status is displayed in Idle condition of FXS port.
- Thus, 8 SIP Account Status (one for each FXS port) will be displayed on LED. 9th SIP Account status will not be displayed.
- If any event is detected on FXS Port, system will stop indicating SIP Account status for that SIP Account and start indicating FXS Port LED as per the event detected.
- If FXS port Hardware fail condition is detected, SIP account status will not be indicated.
- Status of 'Registered' will be displayed if SIP Account is registered to Proxy or if SIP Account is Non-proxy.
- Status of 'Authentication failed' will be displayed if SIP account is not registered due to invalid authentication i.e authentication user name or password is not correct.
- Status of 'Unregistered' will be displayed in following conditions:
  - Disable
  - Trying
  - Un-registering
  - Message Send Failed
  - Register Timer failed

System LED:

LED Status	Meaning
GREEN ON Continuously	At power ON. Waiting for config parameter.
GREEN Blink 500ms ON, 500ms OFF, 500ms ON 2 sec OFF	Try to get network Parameters
GREEN Blink, 1 sec ON -1 sec OFF	Gateway started Successfully.
Red ON, Continuously	Fail to download program in DSP device. {Contact Matrix}
RED Blink, 1sec ON- 1sec OFF	PPPoE User Password or other data invalid.
RED Blink 500ms ON-500ms OFF-500ms ON-2sec. OFF	Communication between VoPP and Processor failed

Instruction for Connecting the Gateway:
This section describes the instructions on, how to connect the Matrix Gateway to internet network and telephones.

Unpack the box. Get satisfied with the contents and the condition of the parts. Refer to ‘Packing List’. If parts are OK, proceed with connections as shown below:

- Mount your Gateway in a safe and convenient location where cables for your network and phone system are accessible. For this, mechanical drawing can be used as a reference, which is provided at end of the chapter.
- Connect a standard analog telephone instrument to required FXS-ports out of 8-ports, using telephone cables provided.
- Insert one end of the Ethernet cable into the WAN port of Gateway and connect the other end of the Ethernet cable to Ethernet Switch which is connected to router or DSL Modem.
- Connect the PC using cross cable.
- Check the voltage from the power point from where the supply is to be accessed. It should be between 90-265VAC, 47-63Hz.
- Insert the DC output terminal of power adapter into the ‘Power’ jack of the Gateway and connect the 230VAC pins of the power adapter to a wall outlet for 230VAC.
- When you power ON the gateway, observe that all LED’s are ON for few seconds and then gateway system LED is flashing.

Basic Steps for Configuration
WAN Port:

- Default WAN IP Address for Gateway is 192.168.001.156.

ABSOLUTELY NEEDED !
Use WAN IP Information from your LAN Admin. or ISP • IP Address • Subnet Mask • Gateway IP Address

- Note that subnet of WAN and Internet (Router) are same.
- Read the IP of PC, by clicking on Network Neighborhood → Properties → TCP/IP Connection → Properties for Windows98.
- Choose the unique IP address of the PC such that its subnet is

same as subnet of IP address of Gateway, provided by ISP.

PLEASE CHECK !
The subnet of your PC IP Address must be same as of Gateway.

- For example, if IP address of Gateway is 192.168.5.10, then PC IP Address port should be programmed as 192.168.5.X. where ‘X’ can be any number from 001 to 254, except 10. By doing this, subnet remains same.
- Pick up the handset of the analog phone connected to Gateway.
- Enter the programming access code #19 followed by password 1234. You get programming tone.
- Enter the WAN IP Address, using command 11-<WAN IP Address>-#*. For e.g. to enter IP Address 192.168.1.120 enter command 11-<192168001120>-#*.
- You get confirmation tone followed by system restarting.
- Start Internet Explorer (IE6 with SP2, Service Pack) of PC connected to the Ethernet Switch. Type in the IP Address programmed above in the Address field. On the login page, enter default password ‘1234’.
- Click on SIP Account from the menu provided on left side of the web page.

Get Information from your ITSP for SIP Account
• SIP ID • Registrar Server Address • Registrar Server Port • Authentication User ID • Authentication Password • Outbound Proxy Server Address • Outbound Proxy Server Port

- Enter above information in SIP Account1 field. If these information is not available, keep the values as default. For e.g. Registrar Server Port = 5060.
- Enable SIP Account for SIP1 and also enable Outbound Proxy, if information is programmed.
- If you have a second internet phone Service Account (SIP2), then

use above fields for SIP2 port and enable. It allows you to use second phone (or Fax) connected at FXS2 port. Similarly program other SIP Account numbers Submit page.

- Ping your gateway IP address or default the system and start again, if you cannot establish connectivity from your Gateway.

Making Phone Calls:

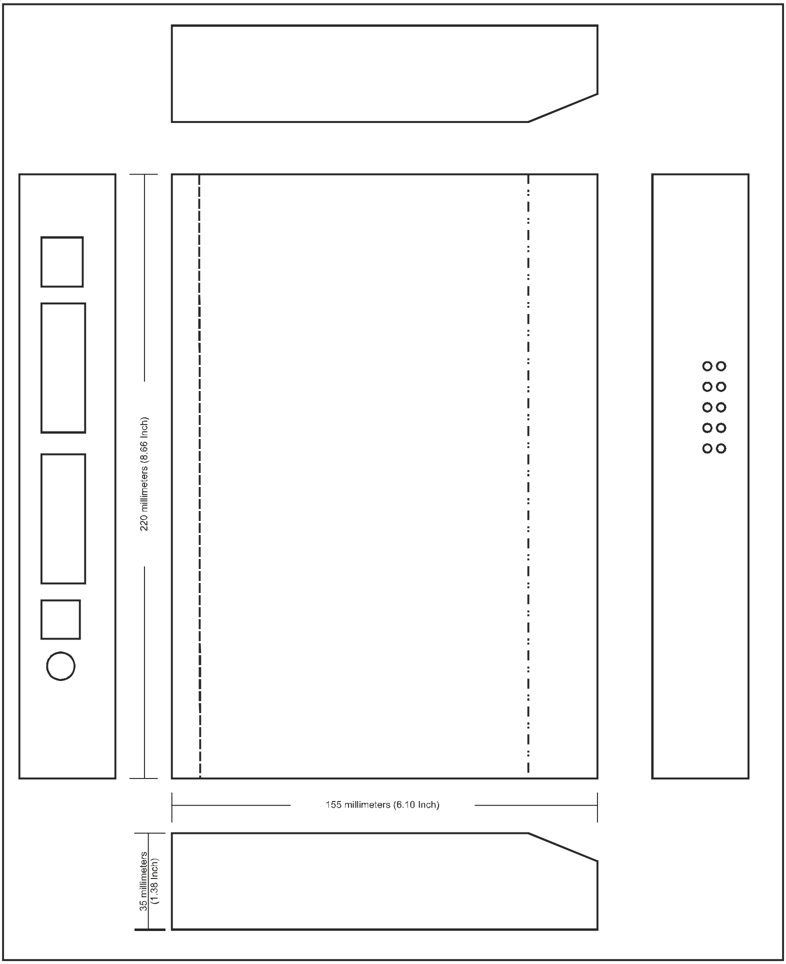
OG Call:

- After gateway LED is ON, just dial the SIP ID numerical number e.g. 4567 which is registered to the same ITSP provider.
- If SIP account is 4567@provider.com then the SIP ID number is 4567 and SIP service domain is ‘Provider.com’.
- If you want to use a specific SIP Trunk, enter that number in the Routing Group at first index and set ‘Rotation’ OFF to dial a number through this trunk.
- Assign supplementary feature to any phone to use the feature like Call Forward, Call Wait etc.
- You can also call another Gateway or IP-Phone by dialing its IP Address. For e.g. Just dial 192*168*1*21, to call the IP number 192.168.1.21.

IC Call:

- When an IC call for the gateway comes it will land on the extension Phone. You can also make it land on a specific Phone connected to the Gateway.

Mechanical Dimension of the SETU VFX88L/VFX44L:



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Section 2: Features and Facilities

Access Codes

What's this?

- To access some important feature the Matrix gateway supports a string of digits which can be dialed by user. This is called 'Access Code'.
- The system supports two different types of Access codes viz. Dial State and Matured State Access codes.
- **Dial State** Access Code table consists of following features by default:

• SE Programming	***#19***
• Emergency Number (4 entries)	***Blank***
• Lifeline Port Access	##
• Set Hotline	****41***
• Cancel Hotline	****42***
• Set Call Forward-Unconditional	****51***
• Cancel Call Forward-Unconditional	****52***
• Set Call Forward-Busy	****53***
• Cancel Call Forward-Busy	****54***
• Set Call Forward-No Reply	****55***
• Cancel Call Forward-No Reply	****56***
• Set Do Not Disturb (DND)	****61***
• Cancel Do Not Disturb (DND)	****62***
• Set Call Waiting	****71***
• Cancel Call Waiting	****72***
• Retrieve Hold Call	****81***

- **Matured State** Access code table consists of following features by default:

• Hold Call	***Flash 1***
• Blind Transfer	***Flash 2***
• Release Call Waiting Call	***Flash-3***
• Accept Call Waiting Call	***Flash-4***
• Reject Remote Held Call	***Flash-5***

How it works?

- Access codes for different features are programmed by the user by the Web Jeeves and this number can be used irrespective of allowed-denied numbers or Automatic Number Translation.

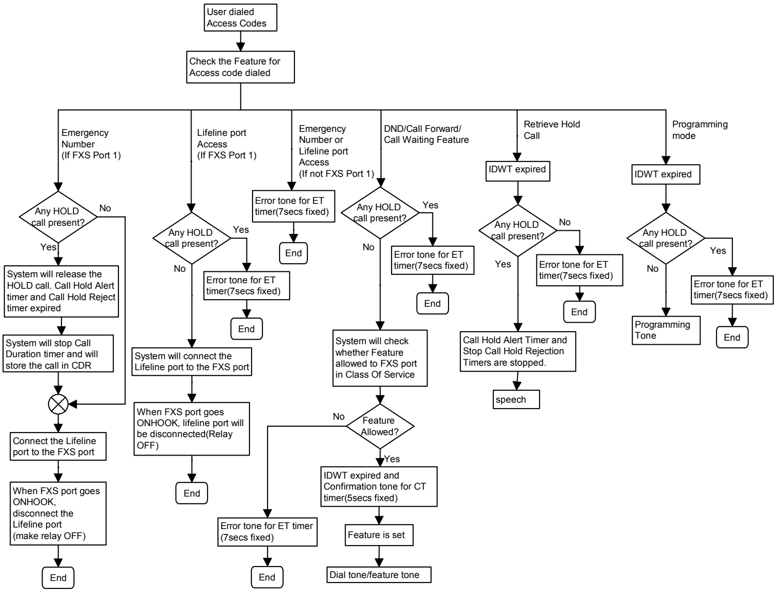
- After the number is dialed by user, the gateway first checks the access code table and then checks features like Allowed-Denied Number logic and Routing option. But End of Dialing Digit is not checked for ‘Dial State Access Code’.
- After dialing blind transfer access code system will check access code table for blind transfer number dialed by the user.
- System will not check ‘End of Dialing Digit’ while checking ‘dial state’ access codes.
- Access code can be dialed through FXS port only.
- The Access codes is of Maximum of 3 digits.
- Dial state Access codes for SE Programming is fixed. By default it is ***#19***.
- For all other features of Dial state, Access code is programmable.
- Matured State Access code is fixed i.e. not programmable.

How to program?

- Refer VFX88L/VFX44L Web Jeeves Page.
- Click on “**Access Codes**”.
  - **Feature:** This is a name of Dial State feature for which Access code is to be programmed.
  - **Access Code:** Enter here the digit string of maximum 3-digits for each feature except for “SE Programming” For Matured state Access code is fixed.
    Range : Blank, 0-9, #, *,
    Default = as per topic “What’s this?”.

Important Point:

- Conflict numbers programming is not allowed. For example, while programming the access codes, if Access code is programmed as 456 than you cannot dial any number starting with 456.



IDWT = Inter Digit Wait Timer
ET = Error Tone

Relevant Topics:

1. Supplementary Services 161
2. Emergency Number Dialing 96
3. Lifeline Port 110
4. Class of Service 86

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Allowed-Denied Numbers

What's this?

- The gateway supports a feature by which some users out of eight FXS ports, can be prevented to dial the specific number strings. This is called Allowed Denied numbers feature.
- For example, for some phone it may be required to allow only few numbers which can be dialed. This is possible by suitable programming of two Number Lists; Allowed Number List, Denied Number List.

How it Works?

- The feature will be 'enabled' first and then the programmed Number List-number is entered for Allowed or Denied list for the specific Source port, as per the application.
- If user has programmed same number in both allowed and denied lists, it will be dialed out.
- If the dialed number is not there in any of he lists, it can be dialed.
- If the dialed number string matches with any Access codes or Emergency Numbers, Allowed-Denied number logic will not be applied.
- This feature is not applicable for 'Hotline' number programmed or Blind Transfer Number dialed.

How to program?

Refer VFX88L/VFX44L Web Jeeves Pages.

- Click on 'Number List' to program both types of number-Lists and remember those List Numbers.
- Click on "**FXS Port Parameters**".
- Apply: Click here to disable the feature. Default = enabled.
- Allowed Number List; Enter the list number programmed previously. Range = 01-24, ***Default = 01***.
- Denied Number List; Enter the list number programmed previously. Range = 01-24, ***Default = 02***.

For Example:

- To prevent dialing of any string, enter the number string '4678' in the number list-08 and enter this list number-08 in the field for

Denied number list.

- If this entry is done for FXS2 port, then FXS2 user can not dial 4678 and he will get ‘Error Tone’, if he has dialed this string.

Relevant Topics:

1. Number Lists 119
2. FXS Port Parameters 103
3. Access Codes 29
4. Emergency Number Dialing 96
5. SIP Account Parameters 149

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Answer Signaling on FXS Port

What's this?

- As general application, telecom equipment like PCO machine is connected to the FXS port of the system. Now whenever the called party (remote party) answers i.e. goes off-hook it is required to inform the FXS port so that the PCO machine can consider the call as matured and start billing. In absence of this signal, the call is never considered as matured and hence no billing will be generated.
- To avoid such problems the system supports 'Answer Signaling'. It is a signal which will be generated on FXS port, which indicates that the called party has answered and the call is matured. This helps in accurate billing, avoids billing of unanswered and unsuccessful call attempts.
- It is generated in the form of:
  - None
  - Polarity (Battery) Reversal

How it works?

- Answer Signaling is applicable for OG call made from FXS.
- When call is made from FXS port to any other port, system will wait for the call to get matured.
- When the call gets matured, the system will check the Answer Signal programmed for the FXS. The options are as explained below:
  - **None:** This option is used when no answer signaling is to be generated on the FXS port.
  - **Polarity Reversal:** This option is used when answer signaling is to be generated in the form of Polarity Reversal on the FXS port. The Battery polarity of the FXS port will get reversed. For example, if the battery polarity of the FXS port is +ve for TIP and -ve for RING in speech condition then after call maturity, TIP will become -ve and Ring will become +ve.

How to program?

Refer VFX88L/VFX44L Web Jeeves Pages.

Click on '**FXS Port Parameters'**

Answer Signaling:

- Select signaling option from ‘None’ or ‘Battery Reversal’ for the FXS port. Program for remaining FXS ports.
  By default, Battery Reversal.

Relevant Topics:

1. Disconnect Signaling on FXS Port 93
2. FXS Port Parameters 103
3. Call Processing 66

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Automatic Number Translation

Introduction

- This feature is used to translate the dialed number string to the number string which will be actually dialed out by the gateway. It is applicable on Destination Port only. This feature is also called Automatic Number Translation (ANT).
- This feature is useful for:
  - Entering short number string for the frequently used number of long number string.
- It is also useful for the application as mentioned below:
  - If user wants to remember and dial Access code for only one domain (say abc.com) instead of other Access code for some other domain (say Pulver.com) while making an IP-Call.
- ANT is applicable for SIP accounts only.

For Example:

- Suppose the Gateway is so programmed that all the calls made to abc.com are routed through the SIP trunk of Pulver.com. Now when a customer makes a call to a number, say *1235678 (*123 is the access code given by SE, for the domain viz. abc.com and 5678 is the subscriber number), the system determines that the called party is a subscriber of abc.com from the access code dialed.
- However, pulver.com does not consider *123 as the access code for abc.com. Rather pulver.com specifies *777 as an access code for abc.com which is not known to the caller. Hence, *1235678 needs to be replaced by *7775678, and ANT will be used for this.

How it works?

- Two number Lists are programmed: Dialed Number list and Substitute Number list, programmed for the destination port.
- The number string in Substitute Number List is programmed at the corresponding Index number of the Dialed Number List. For example, if the number string programmed at the index “3” in the dialed number list matches with the dialed number then the part of the matched number string will be replaced with the number string programmed at the index “3” of the substitute number list.
- After selecting the destination port for routing the call, the number

is dialed out if ANT is **‘Enabled’** on the destination port.

- If ANT is **disabled** on the destination port, then the number in the Dialed Number List is dialed without any modification
- If the dialed number string doesn’t matches with any number of the number strings in the Number list (Dialed Number String) assigned to the destination port, same number string without any modification is dialed.
- ANT is not applicable on the Emergency numbers.
- Blank number cannot be replaced by number through ANT logic.

How to program?

Refer VFX88L/VFX44L Web Jeeves Pages.

- Click on ‘Number List’ to program both types of number-Lists and remember those List Numbers.
- Click on SIP Account Parameters-2.
- Apply: Click here to disable the feature. ***Default = enabled***.
- Dialed Number List; Enter the list number programmed previously. Range = 01-24, ***Default = 04***.
- Substitute Number List; Enter the list number programmed previously. Range = 01-24, ***Default = 05***.

Example1:

- For the above example given in the Introduction:
- Program *123 number string in the number list say 01, Index 01 and assign this number list as Dialed Number List.
- Program *777 number string in the number list say 02, Index 01 and assign this number list as Substitute Number List.
- Thus, when the user dials the number *1235678, this number will be replaced by *7775678 through ANT logic and will be dialed out.

Example2:

- In Peer-to-Peer Calling this can be used for stripping off the prefix digit. For e.g. User dials the UK number as 0044-XXX where XXX is number. But ITSP requires only 44 for UK number dialing. Thus, 00 can be stripped off from the number string to be dialed from the gateway.
- The string ‘44’ is programmed in ‘Substitute Number List whereas ‘0044’ is programmed in ‘Dialed Number List’.

Relevant Topics:

1. Number Lists 119
2. SIP Account Parameters 149

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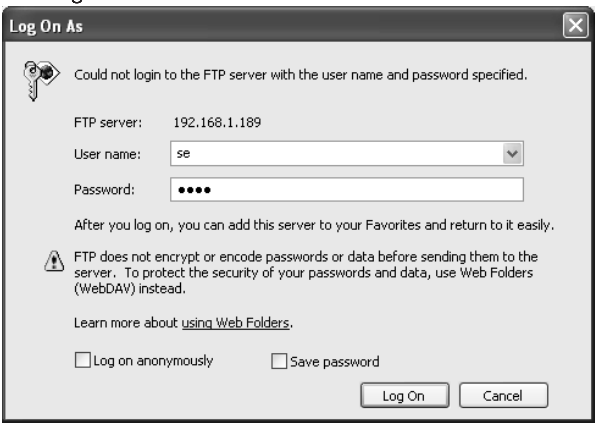
Backup-System CDR

What's this?

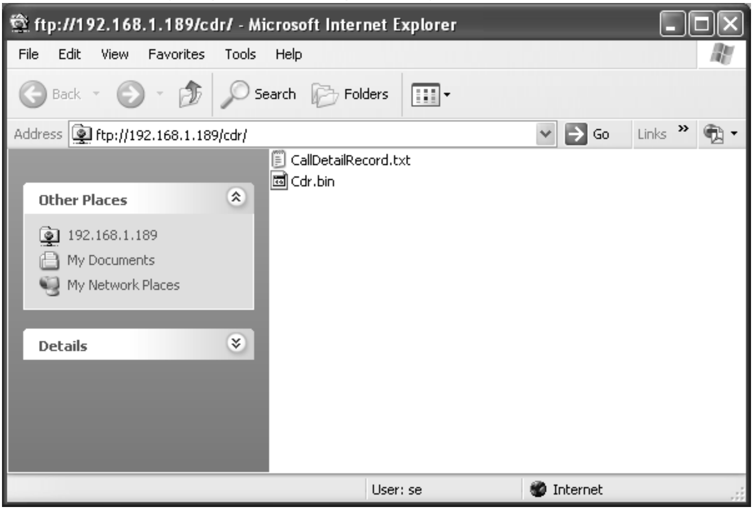
- In SETU VFX88L/VFX44L there is an embedded FTP server which can be used for Backup of SMDR call records.
- File Transfer Protocol (FTP), a standard Internet protocol, is the simplest way to exchange files between computers on the Internet. Like the Hypertext Transfer Protocol (HTTP), which transfers displayable Web pages and related files, and the Simple Mail Transfer Protocol (SMTP), which transfers e-mail, FTP is an application protocol that uses the Internet's TCP/IP protocols. FTP is commonly used to transfer Web server for everyone on the Internet. It's also commonly used to download program and other files to your computer from other servers.

How to use it?

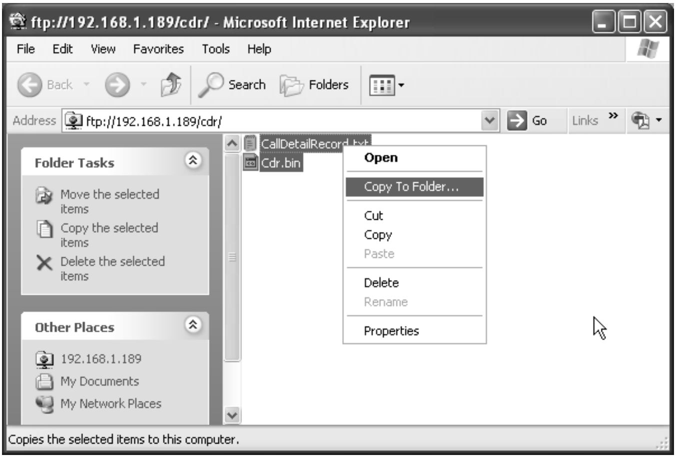
- Click on **Start** > **Programs** > **Internet Explorer**. Enter the IP address in Address bar. Let us say IP address 192.168.1.189 is entered at Address bar on opening page of **Microsoft Internet Explorer**. Click on 'Upload/Download Call Detail Records'. No need to enter User Name and Password for this process.
- User gets 'Log On As' Window on screen. Enter the user name (se) and Password (1234) in 'User Name' and 'Password' fields. Click on 'Log On' button.



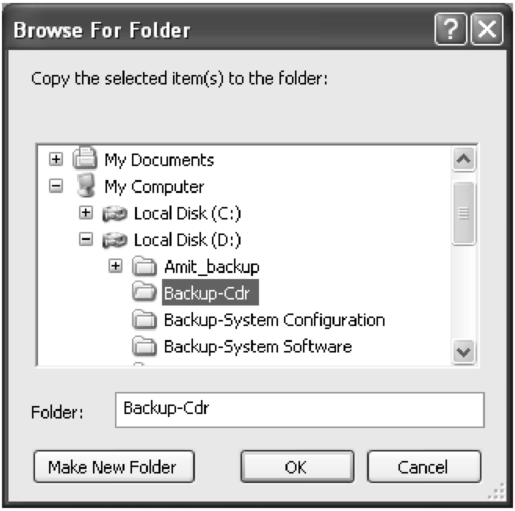
- On successfully log In following display appears on the screen.



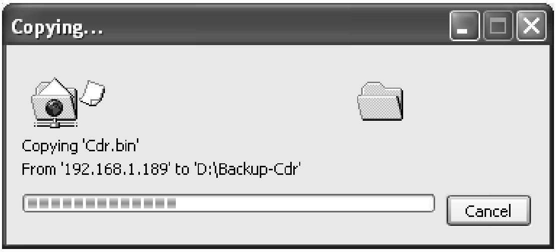
- Select all files and right click the mouse button. Select 'Copy to Folder' option.



- Select a path where you want to save backup files on your system. Click OK button.



- CDR Backup process will be standard.



Important Points:
- For uploading a CDR files in the system, first remove current file present in the system and then copy the new file from computer (backup source) to system.
- User can also get the stored report by opening the FTP through DOS.

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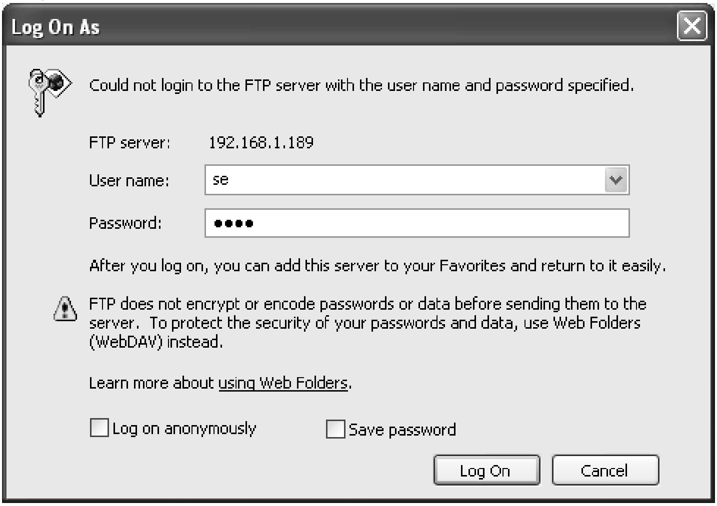
Backup-System Configuration

What's this?

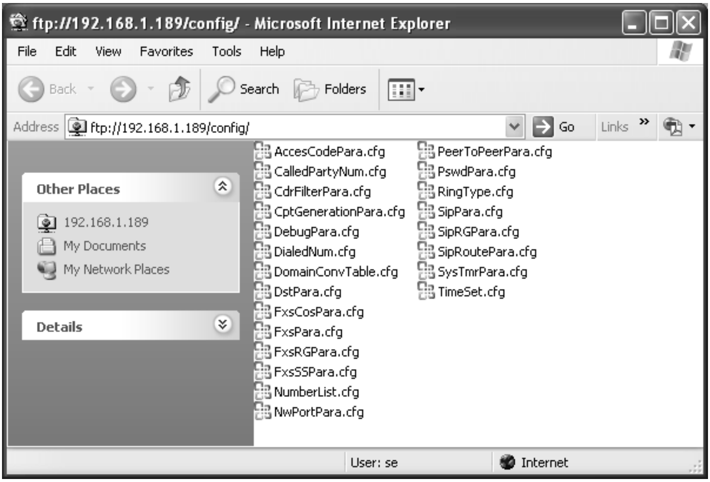
- In SETU VFX88L/VFX44L there is an embedded FTP server which can be used for Backup of System Configuration files.
- For details on FTP, refer chapter "Backup-System CDR".

How to use it?

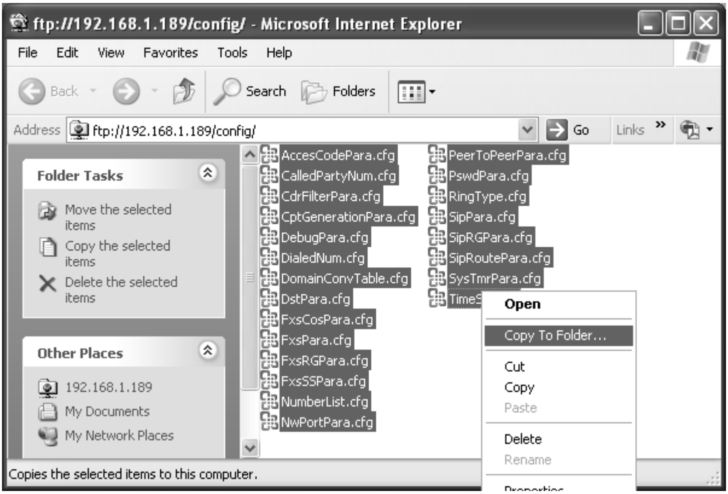
- Click on **Start** > **Programs** > **Internet Explorer**. Enter the IP address in Address bar. Let us say IP address 192.168.1.189 is entered at Address bar on opening page of **Microsoft Internet Explorer**. Click on 'Upload/Download Configurations'. No need to enter User Name and Password for this process.
- User gets 'Log On As' Window on screen. Enter the user name (se) and password (1234) in 'User Name' and 'Password' fields. Click on 'Log On' button.



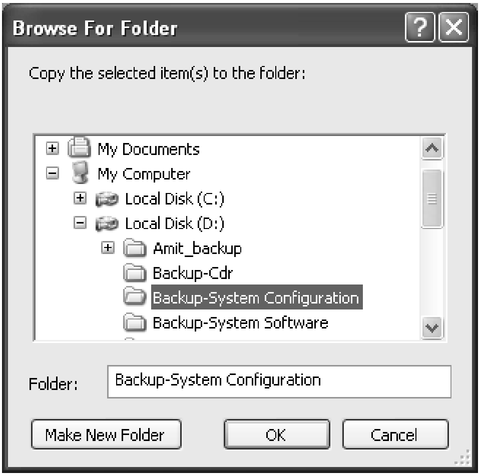
- On successfully log In following display appears on the screen.



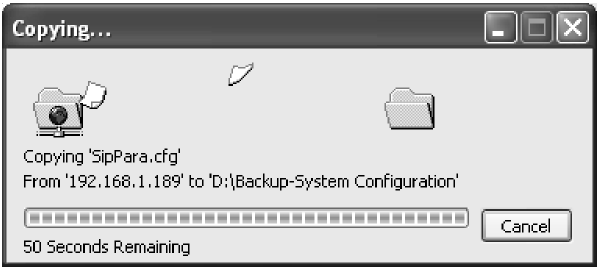
- Select all files and right click the mouse button. Select ‘Copy to Folder’ option.



- Select a path where you want to save backup files on your system. Click OK button.



- System Configuration Backup process will be on progress as shown below:



Important Points:

- For uploading a configuration files in the system, first remove current file present in the system, and then copy the new file from computer (backup source) to system.
- User can also get the stored report by opening the FTP through DOS.

Relevant Topic:

1. Backup-System CDR 39

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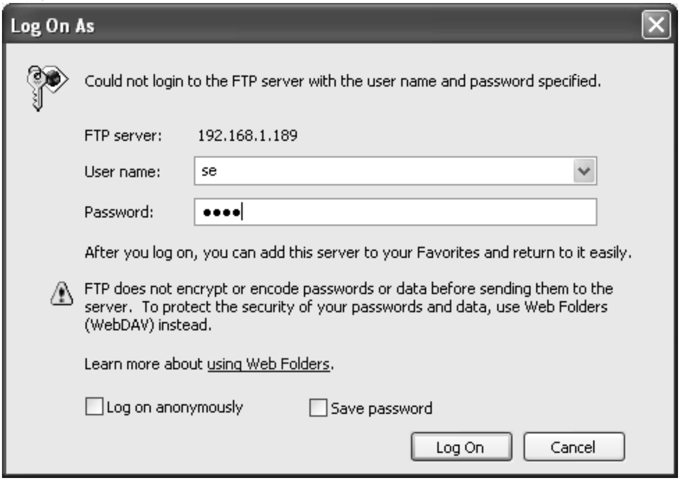
Backup-System Software

What's this?

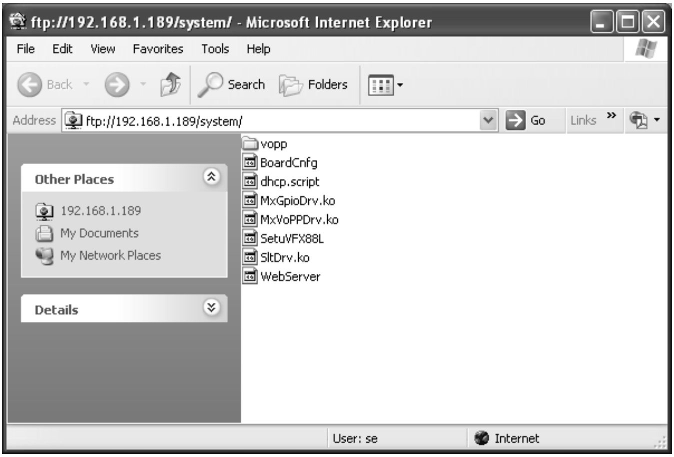
- In SETU VFX88L/VFX44L there is an embedded FTP server which can be used for Backup of system software files.
- For details on FTP, refer chapter "Backup-System CDR".

How to use it?

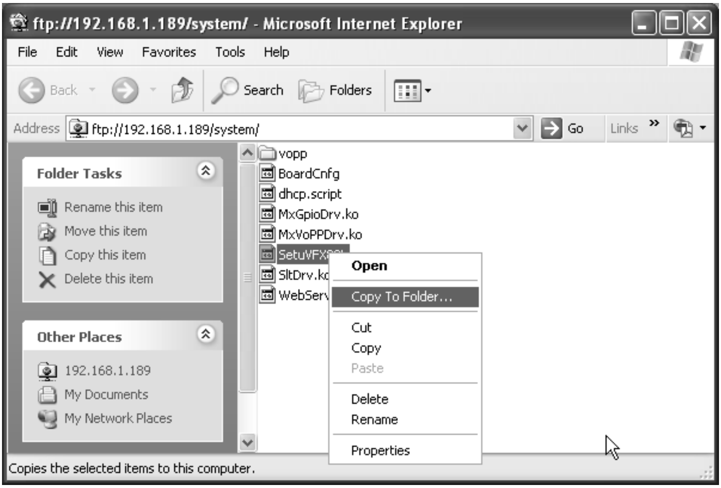
- Click on **Start** > **Programs** > **Internet Explorer**. Enter the IP address in Address bar. Let us say IP address 192.168.1.189 is entered at Address bar on opening page of **Microsoft Internet Explorer**. Click on 'Upload/Download System Software'. No need to enter User Name and Password for this process.
- User gets 'Log On As' Window on screen. Enter the user name (se) and password (1234) in 'User Name' and 'Password' fields. Click on 'Log On' button.



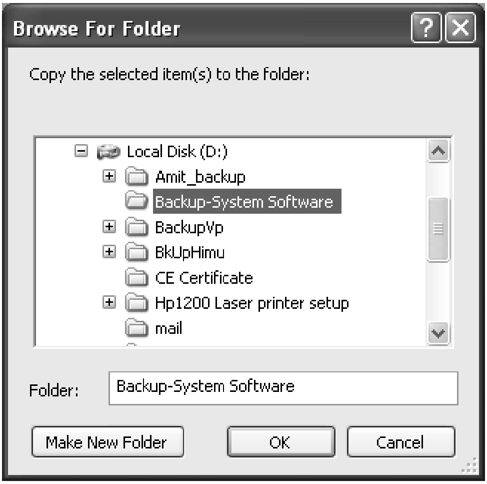
- On successfully log In following display appears on the screen.



- Select all files and right click the mouse button. Select 'Copy to Folder' option.



- Select a path where you want to save backup files on your system. Click OK button.



- System Software Backup process will be on progress as shown below:

Important Points:

- For uploading a software file in the system, first remove current file present in the system and then copy the new file from computer (backup source) to system.
- User can also get stored report by opening the FTP through DOS.

Relevant Topic:

1. Backup-System CDR 39

=X=X=

Blind Call Transfer

What's this?

- In call transfers, three parties are involved. Hence, following terms are used to explain the feature.
    - **Transfer Target:** The remote user to whom the call is to be transferred (Remote User 2).
    - **Transferer:** The user who transfers the call.
    - **Transferee:** The user who is to be transferred (Remote User 1).
- The gateway supports a feature of transferring a call by which the user will be able to transfer the call without talking to the transfer target. This is called a Blind call-transfer.
- The system supports to notify the user about 'transfer' if remote user1 also supports Notification.
- Notification Timer is programmable.

How to use it?

- **Enable** the feature for required FXS port number from Class of Service.
- When you are in speech, dial the Access code **(Flash-2)** for Blind Call Transfer. You will get Dial tone (Feature Tone) and remote user1 will get 'Hold'. If there is any waiting call, it will be released by the system.
- **Dial a Number** of the Transfer target to transfer the caller (Transferee) to the Transfer target.
- If transferee supports notification, you will get **NOTIFICATION** by transferee about the transfer action i.e. failure/successful.

How it works?

If you go ON-Hook the call will be disconnected. If you are OFF-Hook, during time of notification following options are possible:

- If you are OFF-Hook and do not get notification from remote user till Notification-Timer expires, you will get, a quick confirmation tone for confirmation tone timer followed by Dial tone and the remote user1 (transferee) will be released.
- If remote user1 is 'busy' you will get Busy-tone for 5-sec. After expiry of this timer you will get error tone.
- If dialed number is invalid-string, you will get 'Error-tone' for period

of error tone timer and remote user1 will be released.

- If you are ON-Hook during any error condition, you will get ‘Ring’ indication for period of ‘ring timer’. If you don’t pick up the call, during this time the remote user1 will get released.
- If the valid number is dialed, and if there is no error condition, further call processing will be as per flowchart of SIP to FXS Call in chapter ‘Call Processing’.

How the Notification Timer works?

- Transfer Notification timer is programmable (Only from ‘Web’).
- When call transfer is initiated the notification timer starts.
- After notification is received, this timer is stopped.
- If the transferee disconnects the user, the notification timer is stopped and now the you can make a new call.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on **‘Class of Service’** to Enable the feature.
- Click on ‘**Access Code’** to know the Access code.

  Click on “System Parameters” to program ‘Transfer Notification timer’.

  Range = 001-999.

  Default = 030 sec.

Important Points:

- It is not required to check Access Code table because the user is going to dial the number which is to be transferred to the remote end.
- System will also not check the Allowed-Denied logic because the number dialed by the user for Blind Transfer is not to be dialed by the system. Whereas Allowed-Denied logic is to be applied when it is to be dialed by the system.
- After dialing **Flash** (i.e dialing the matured state access code), if other request is received from the remote end, that request will be rejected.
- After dialing **Flash** (i.e dialing the matured state access code), if there is another call for the same user, that call is not considered as Call waiting and the call will be routed to the next port as per the routing group for routing the call.

- After dialing **Flash** if the user goes ON-Hook within Inter digit wait timer, he will not get the ‘ring’ and the call will get disconnected.

Remote Transfer:

- Matrix VFX88L supports Remote Transfer feature as explained below:
    - Suppose ‘A’ has dialed a number of ‘B’. ‘A’ and ‘B’ are in speech with each other.
    - ‘B’ is connected to Matrix Gateway ‘VFX88L’.
    - Now this call can be transferred by ‘A’ so that ‘B’ will be in speech with 3rd party ‘C’. This is called ‘Remote Transfer’.
- This can be done by following steps:
    - ‘A’ will send ‘Hold’ Message to ‘B’. ‘B’ will get-Remote Hold Tone.
    - Then ‘A’ will send number of ‘C’ in his message to ‘B’ by dialing the number of ‘C’. But if there is any waiting call, ‘B’ will not accept any request of remote transfer from ‘A’.
    - The gateway of ‘B’ will dial number of ‘C’.
    - If ‘C’ attends the call, ‘B’ will be in speech with ‘C’ and Remote Transfer will be completed.

Relevant Topics:

1. Access Codes 29
2. Class of Service 86
3. Call Processing 66
4. System Parameters 165

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Black Listed Callers

What's this?

Some times it is required to prevent the IC calls from the specific callers. For this the gateway supports a feature for blocking of incoming calls from specific addresses (Contacts).

- This feature is applicable for IC calls on SIP Accounts only.
- It does not affect the OG calls.

How it works?

- It can be Enabled/Disabled for each SIP Account.
- Program the Numbers to be blocked in the specific Number Lists and assign it as Black Listed Callers Number list, to the SIP account.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on 'Number List' to program a list for numbers to be blocked.
- Click on '**SIP Account Parameters-2**'.
- **Apply:** Select to disable this feature. ***Default = Enable.***
- **Number List:** Enter the list number containing numbers to be blocked, for IC call.
  Range = 01-24. ***Default = 03.***
- Program for all required SIP Accounts.

Important Point:

- If IC call is NOT, anonymous Call, the system will check this feature. If number is matched with number in the number list, the call will be rejected, provided feature is enabled.

Relevant Topics:

1. Number Lists 119
2. SIP Account Parameters 149

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Call Detail Record (CDR)

What's this?

- It is a kind of report for the calls, containing information about the gateway's usage when call was made. With Call Detail Record (CDR) feature it is possible to know following details of a call:
  - Source Port (Maximum 5 characters)
  - Terminating Port (Maximum 5 characters)
  - Calling Party Number (Maximum 24 digits)
  - Called Party Number (Maximum 24 digits)
  - Date of call origination (Maximum 11 digits)
  - Time of the call origination (Maximum 5 digits)
  - Duration of call (Maximum 4 digits)-Seconds
- It is also possible to filter out the details like, calls made to specific numbers, calls received from specific numbers and based on the duration of call, etc. Refer a CDR report at the end of chapter.

How it works?

- The feature works only if "Apply Filter" is enabled on the Web page.
- Maximum of 2000 call records can be stored.
- Call record entries are stored in FIFO logic.
- A call is stored when it gets matured.
- The number list containing Called party number and another Number List containing Calling party numbers are assigned for the Called or Calling Number List Filter. Now when the filter is applied, the calls with the number programmed in the number list only will be displayed in CDR.

For Example:

- **If source port is FXS port:**
  - Called Party Number stored in the Call records will be the number dialed by the caller or the Hotline Number.
  - The Calling Party Number stored in the Call records are the number programmed in the FXS Port Parameters for that port.

- **If source port of SIP Account:**
  - Called Party Number stored in the Call records will be the Called Party Number received in the SIP Message.

- CDR report is displayed on Web Jeeves as per the filters set.

CDR Report:

The report contains some terminology as shown below:

- SPORT= Source Port (A port on which call initiates).
- DPORT= Destination Port (A port from which call terminates).
- Calling Number= Number of Calling Party.
- Called Number= Number of Called Party.
- Date is displayed in DD-MMM-YYYY format, for example 01-JAN-2006 and time in HH:MM format.
- Time=Time of call.
- Duration (Sec.)= Duration of call.
- Remarks of the call will be displayed in 'REM' column, for normal call : Blank, for Blind Transfer Call : **BT**, for Attended Transfer = **AT**, for Attended Transfer and Received : **ATR**, for Anonymous Incoming Call : **A.**

Example:

- For anonymous call, Calling Party Number = Blank, remarks = '**A**'.

How to Program?

Refer VFX88L/VFX44L Web Jeeves.

For Call Detail Record Filters settings.

- Click on "**Call Detail Records Filters**".
- Click on 'Apply' to deselect the report. ***Default = report-enable.***
- Default Filters for CDR are All FXS Ports (1-8), All SIP Accounts (1-9),
- Date from 01-Jan-2007 to Current date, Time 00:00 to 23:59, Calling Party Number list 01, Called Party Number list 01, Duration more than 001 sec.

For Call Detail Records Report.

- Click on **"Call Detail Record Report"**.
- You will find 4 options (buttons) on the top left hand side- First, Previous, Next and Last.
- By default First and Previous will be disabled (non-editable) i.e Call records (First) from 1-20 will be displayed when you open this page.
- If you click on 'Next', Call records from 21-40 will be displayed and

First and Previous button will be enabled.
- If you click 'Last' button, last 20 records will be displayed and Next button will be disabled.
- If there are no call records, you will get display of Alert message "**No Calls**".

Parameters for CDR Filters:
- To clear or delete records, click "Clear Call Record" The system will show alert message "**This will clear all the call records from the system. Do you want to continue?**". Select 'Yes' to delete the record.

Refer the Notes below, for programming all parameters on the Web-page:
1. If 'Apply' Filter is enabled, than only the From-To fields will be editable.
2. Enter the Current date of the system in the 'To' Field of the filter 'Calls made between date'.
3. 'Apply' filter can not be edited for following filters:
   a. Calls made between date
   b. Calls made between time
   c. Called Party Numbers matching with Number List
   d. Calling Party Numbers matching with Number List
   e. Call Duration greater than (Seconds)

Call Originated From FXS Ports
- Enter the FXS port number range (from and to), which are the source ports for all the calls for which report is desired. The range is from 1-8. For example, we need a report for the calls which were originated from FXS port number from 1 to 2. Then, enter values "**1-2**" in the field of this first parameter.

Call Originated From SIP Accounts
- Enter the SIP Account numbers range (from and to) which are the source ports, for all the calls for which report is desired. Range is from 1-9.

Call Terminated on FXS Ports
- Specify the FXS port number range (from and to), which are the

destination port for all the calls, for which report is required. Range is 1-8.

Call Terminated on SIP Accounts

- Enter the SIP Account number range (from and to), which are the destination port for all the calls for which report is required. Range is 1-9.

Calls Made between Dates:

- Enter from which date to which date report is required in format: DD-MM-YYYY. For example, you may need report from 2-10-06 to 2-10-06 dates. Then enter: 02-Sep-2006 to 02-Sep-2006. ‘To’ field will be programmed for current date of the system.

Calls Made between Times:

- Enter from which time to which time report is required, in the format HH:MM. This time is the time recorded in report for each call.

Called Party Number Matching with Number List

- Enter the Called party Number List-Number for which report is required. List number range is 01-16. Enter Called party list number in the box.

Calling Party Number Matching with Number List

- Enter the Calling party Number List-Number for which report is required. List number range is 01-16. Enter Calling party list number in the box.

Call Duration Greater than Seconds:

- Get report of all calls of specific duration. Range is from 0001 to 9999 seconds. For example, if you need report for all calls which were of 300 seconds duration. Then, enter here 0300.

Important Points:

- Numbers stored are not the Numbers after applying the ANT logic.
- CDR filters are applicable only for CDR report and not for storing a call record.
- Call Billing information is not supported.
- Scheduled Report Generation is not supported.

- If you click on ‘CDR Report’ link in upload/download, the text file for CDR report for FTP will be generated.
- The filters are not applicable if CDR is through FTP server.
- If any or all index of number list is blank which is set as filter for Called or Calling Number, than Called or Calling number with blank also will be displayed.

Relevant Topic:

1. RTC Parameters 146

```
                         CALL DETAIL RECORD REPORT as on 01-Mar-2007
-----------------------------------------------------------------------------------------------
SRN  S-Port D-Port Calling Number            Called Number             DATE        TIME  DUR  REM
-----------------------------------------------------------------------------------------------
   1 FXS-8  SIP-6  2008                      192.168.1.156             01-Mar-2007 01:50   10  AT
   2 SIP-6  FXS-1  2008@192.168.1.156        2001                      01-Mar-2007 01:50   10
   3 FXS-7  SIP-6  2007                      192.168.1.156             01-Mar-2007 01:51   10
   4 SIP-6  FXS-2  2007@192.168.1.156        2002                      01-Mar-2007 01:51    9
   5 FXS-6  SIP-6  2006                      192.168.1.156             01-Mar-2007 01:51   11
   6 SIP-6  FXS-3  2006@192.168.1.156        2003                      01-Mar-2007 01:51   10  BT
   7 FXS-5  SIP-6  2005                      192.168.1.156             01-Mar-2007 01:52    7
   8 SIP-6  FXS-4  2005@192.168.1.156        2004                      01-Mar-2007 01:52    7
   9 FXS-4  SIP-6  2004                      192.168.1.156             01-Mar-2007 01:52    8
  10 SIP-6  FXS-5  2004@192.168.1.156        2005                      01-Mar-2007 01:52    8
  11 FXS-3  SIP-6  2003                      192.168.1.156             01-Mar-2007 01:53    6 ATR
  12 SIP-6  FXS-6  2003@192.168.1.156        2006                      01-Mar-2007 01:53    6
  13 FXS-2  SIP-6  2002                      192.168.1.156             01-Mar-2007 01:53    8
  14 SIP-6  FXS-7                            2007                      01-Mar-2007 01:53    8   A
  15 FXS-1  SIP-6  2001                      192.168.1.156             01-Mar-2007 01:53    8
  16 SIP-6  FXS-8  2001@192.168.1.156        2008                      01-Mar-2007 01:53    8
-----------------------------------------------------------------------------------------------
Matrix Setu VFX88L V1R1
```

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Call Forward

What's this?

- The gateway supports few supplementary features like 'Call Forward (CF), Call Hold etc.
- You can use following types of call-forwarding:
  - **Forward-Unconditionally:** Used when desired destination number is dialed without knowing the status of destination-busy or not replying
  - **Forward-Busy:** Used when desired destination number is dialed which is Busy with some other call.
  - **Forward-No Reply:** Used when desired destination number is dialed which is not attending the incoming call and user is getting Ring Back Tone for specified time. For this user will be required to program "No-Reply Timer".
- You can forward calls to different destination numbers in different cases. For e.g. you will be able to forward calls to destination number 1 when busy and to destination number 2 when No Reply.
- Using programmed Access Code, you can Set/cancel Call Forward -Unconditional, Call Forward-Busy and Call Forward-No Reply.

How to use it?

- First enable the feature for a specific FXS port in Class of Service.
- Go OFF-Hook.
- You will get 'Feature Tone' instead of 'dial tone'.
- Then dial following code to Set/Cancel Call Forward feature:
  - ***51**-To Set Call Forward-Unconditional
  - ***52**-To Cancel Call Forward-Unconditional
  - ***53**-To Set Call Forward-Busy
  - ***54**-To Cancel Call Forward-Busy
  - ***55**-To Set Call Forward-No Reply
  - ***56**-To Cancel Call Forward-No Reply

How it works?

- Access code for Set/Cancel Call Forward of any type is programmable.
- When call has to be routed on any FXS port, Call Forward Status

has to be checked first.
- Allowed/Denied number logic is not applied for the CF number programmed.
- Call Forward Status can be changed through telephone instrument or Web Jeeves (Supplementary service page).
- Call Forward Number can be programmed through Web Jeeves in Supplementary Service page.

How to program?
- Refer VFX88L/VFX44L Web Jeeves.
- Click on “Access Codes” to Set/Cancel Call Forward Access code
- Click on “Supplementary Services” to program Call Forward Number and Status.
- Program following parameters for all types of call forward for the FXS-Port number as required.
- **Status:** Select ‘Enable’ if you want to work with the feature. Default = disable.
- **Number:** Enter here the Destination number on which call forward is desired.
  Maximum 24 digits using 0-9, #,*, @.
  Default = blank
- **Time:** Enter this only for No-reply type. Enter the time for which No-reply period is required before forwarding the call.

For Example:
- Enable both Busy and No reply types.
- Enter 2345 for FXS1 in Number field for type ‘Busy’ and enter ‘6789’ for FXS2 in Number field of type ‘No reply’.
- Thus you can make IC call on FXS1, forwarded to the number ‘2345’ if your number is ‘Busy’.
- And also IC call on FXS2 will be forwarded to ‘6789’ if user of FXS2 does not reply till 15-seconds.

Important Points:
- Call Forward-Unconditional is given priority over the DND but if DND and Call Forward-Busy or Call Forward-No Reply is set, than DND will be given the priority, by the system.
- Call Forward-Busy is given priority over the Call Waiting feature.

Relevant Topics:

1. Access Codes 29
2. Class of Service 86
3. Prefix to Domain Name Conversion 123
4. Supplementary Services 161

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Call Hold

What's this?

- Using this feature you can keep a call on hold and retrieve that call when required.
- The held call is disconnected, if it is not retrieved before specified time.
- For this, the gateway supports two types of Timers: Call Hold Alert Timer and Call Hold Reject Timer.
- You can get indication of 'held' call if you go ON-Hook, by getting 'Ring' for period of Ring Timer (programmable) to indicate that there is held party present.

How to use it?

User will follow the steps as mentioned below:

- Enable the feature from **'Class of Service'**.
- Program the Alert timer and Reject timers.
- During speech if you want to hold a remote user, dial the access code ***Flash-1*** in active call condition. If you go ON-Hook, you will get 'Ring' indication after expiry of Alert timer.
- Dial ****81*** to retrieve the Held Remote user.

How it works?

- The feature works only if enabled from Class of Service for the specific FXS port.
- Call can be held, by dialing the access code only during 'connected' state speech condition.
- Access code for Call Hold is ***Flash-1*** and it is not programmable.
- If you dial access code and feature is allowed, you will get confirmation tone till confirmation tone timer and you are in dial state.
- In this state, you cannot make another call, but you can do following:
  - Can dial retrieve Hold Call Access Code.
  - Can dial Emergency Number.
- If you dial access code or number other than above, you will get 'Error Tone'.
- If you dial Emergency Number, after holding the call, the call will be released.

- Any waiting call will be released.

Call Hold Alert timer:
- It is applicable when you have put someone on hold Call Hold Alert Timer expires and you go ON-Hook, you will get a **‘Ring’** to indicate that there is a held party.
- If Call Hold Alert Timer expires and you are still OFF-Hook:
  - System will wait for your port (FXS port) to go ON-Hook and wait for expiry of Call Hold Reject timer.
  - If you are still not ON-Hook and Call Hold Reject timer gets expired, the HOLD call will be released.

Call Hold Reject Timer:
- It is applicable when you have put someone on hold and you don’t retrieve the held call. After expiry of this timer, held party will be disconnected automatically.
- If call hold reject timer expires and you are OFF-Hook, you will get ‘Error Tone’ on dial state.
- Now you can make a call.
- If held caller goes ON-Hook during ‘Hold’ state, and you (FXS Port) are OFF-Hook, then you will get ‘Error Tone’ in dial state. (i.e. dialing the emergency number or retrieve hold call access code or any other number)
- Programming of the timer, access code and feature enable/disable is through Web Jeeves.
- Both timers are programmable.
- Refer flow chart for more information.

How to program?
- Refer VFX88L/VFX44L Web Jeeves.
- Click on “**Class of Service**” to allow/deny Hold Feature for the FXS.
- Click on “**Access Codes Table**” to get the code for Retrieve/Hold.
- Click on “**System Parameters**” to program timers.
  Range for timers is 001-999 seconds.
  Default timer value for Call Hold Alert timer is 60 secs, for Call Hold Reject Timer is 180 secs. and Remote Held Reject Timer is 180 secs.

Remote Held:

- When a remote user puts you on ‘Hold’ it is called ‘Remote Held Call’. During this you can not ‘Hold’ that remote user.
- You can get indication of ‘Remote Held’ call by following ways:
  - Getting beeps (cadence: 500ms ON 3sec OFF and freq. 400Hz).
  - By any voice message if received from the remote server/user. System supports Remote Held Reject Timer which is programmable.
- This timer will be started when Remote user sends HOLD message and the call will be released on expiry of this timer. Default is 180 seconds with a range of 001-999.
- System will check Call Waiting only if user is OFF-Hook.
- If you receive a new call (Call Waiting call) than it is indicated by the “Remote Held + Call Waiting Tone”.
- If you go ON-Hook in Remote Held state and there is any Call Waiting, the Remote Held call will be disconnected and you will get ‘ring’ on FXS port. Refer the flowchart.

How to disconnect Remote Held call?

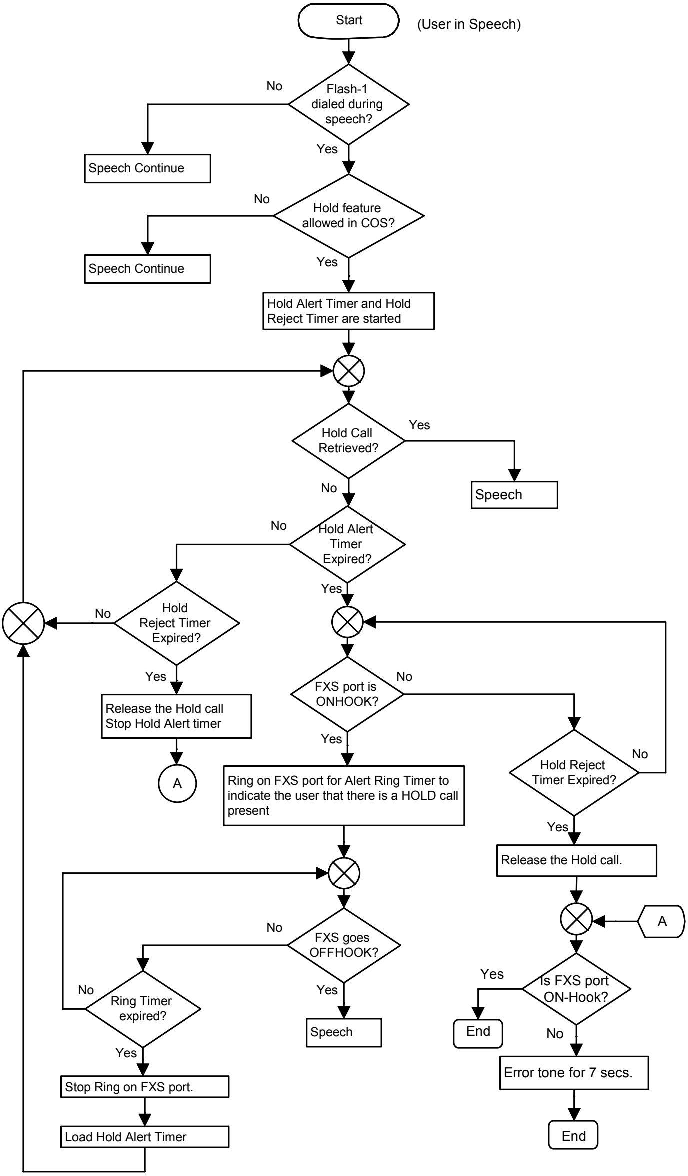
- Dial the access code **‘*Flash-5*’** for rejecting the Remote Held call. This access code is not programmable.
- The call will not be disconnected, if you go ON-Hook.

Important Points:

- It is advisable to program the value of “Call Hold Alert Timer” less than “Call Hold Reject Timer”.
- If “Call Hold Reject Timer” is programmed less than “Call Hold Alert Timer”, you will not get alert tone and call will be released after “Call Hold Reject Timer”.
- If programmed “Call Hold Reject Timer” is equal to “Call Hold Alert Timer”, you will get alert tone. If that port is busy (OFF-Hook) during alert tone, than call will get released. If that port is free and you do not pickup the call during alert tone timer, call will get released.
- After dialing Flash (i.e dialing the matured state access code), if other request is received from the remote end, the request will be rejected..
- After dialing Flash (i.e dialing the matured state access code), if there is another call for the same user, that user is not considered for Call waiting and the call will be routed to the next port from the

routing group for routing the call.

- After dialing Flash if the user goes ON-Hook within Inter digit wait timer, the call will get disconnected instead of giving ring.


Start
(User in Speech)
Flash-1 dialed during speech?
No
Yes
Speech Continue
Hold feature allowed in COS?
No
Yes
Speech Continue
Hold Alert Timer and Hold Reject Timer are started
Hold Call Retrieved?
Yes
Speech
No
Hold Alert Timer Expired?
No
Yes
Hold Reject Timer Expired?
No
Yes
Release the Hold call Stop Hold Alert timer
A
FXS port is ONHOOK?
No
Yes
Ring on FXS port for Alert Ring Timer to indicate the user that there is a HOLD call present
Hold Reject Timer Expired?
No
Yes
Release the Hold call.
A
Is FXS port ON-Hook?
Yes
End
No
Error tone for 7 secs.
End
FXS goes OFFHOOK?
No
Yes
Speech
Ring Timer expired?
No
Yes
Stop Ring on FXS port.
Load Hold Alert Timer

Relevant Topics:

1. Access Codes 29
2. Class of Service 86
3. FXS Port Parameters 103
4. Call Processing 66
5. Call Progress Tones 74
6. Call Waiting 80
7. System Parameters 165

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Call Processing

What's this?

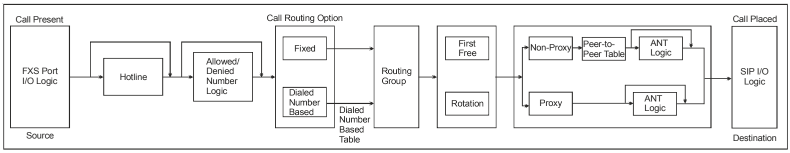
When the call originates on source port, call will be routed on the destination port through some process. This process for routing the call on the destination port is called Call Processing.

- When the call originates on a FXS port, it is required to place the call on the SIP Account.
- When the call originates on the SIP Account, it is required to route the call on the FXS port.
- Call cannot be placed directly on the destination port because there are multiple ports, supported by the gateway.
- Thus, some process is required in the Gateway through which it can decide on which port the call will be placed when the call originates on the port.

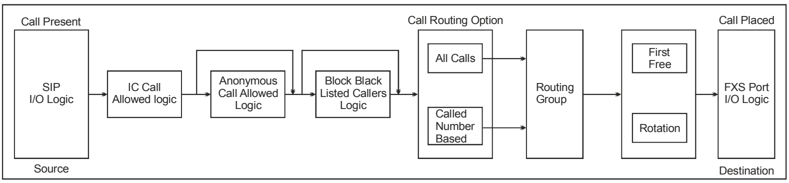
How it works?

The following block diagram shows the Call processing flow:

FXS to SIP Call



SIP to FXS Call



FXS to SIP Call:

- Before call processing, the gateway will check 'Class of Service' and 'Supplementary Features' to allow the feature.
- After you go OFF-Hook, from the Phone connected at FXS port, the system will first check whether that FXS port is enabled or disabled.

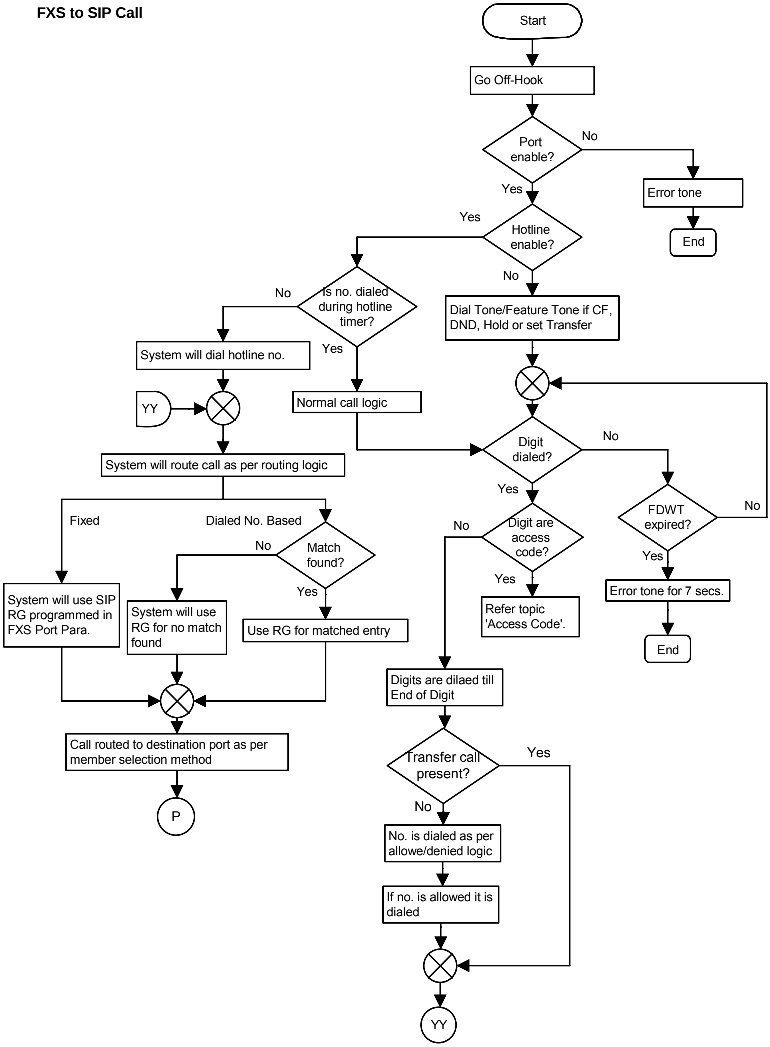
It gives ‘error tone if it is disabled and checks for any Hotline feature. If it is ‘set’, it will dial out the ‘Hotline number’ programmed. But if feature is not set, you will get ‘Dial tone’.

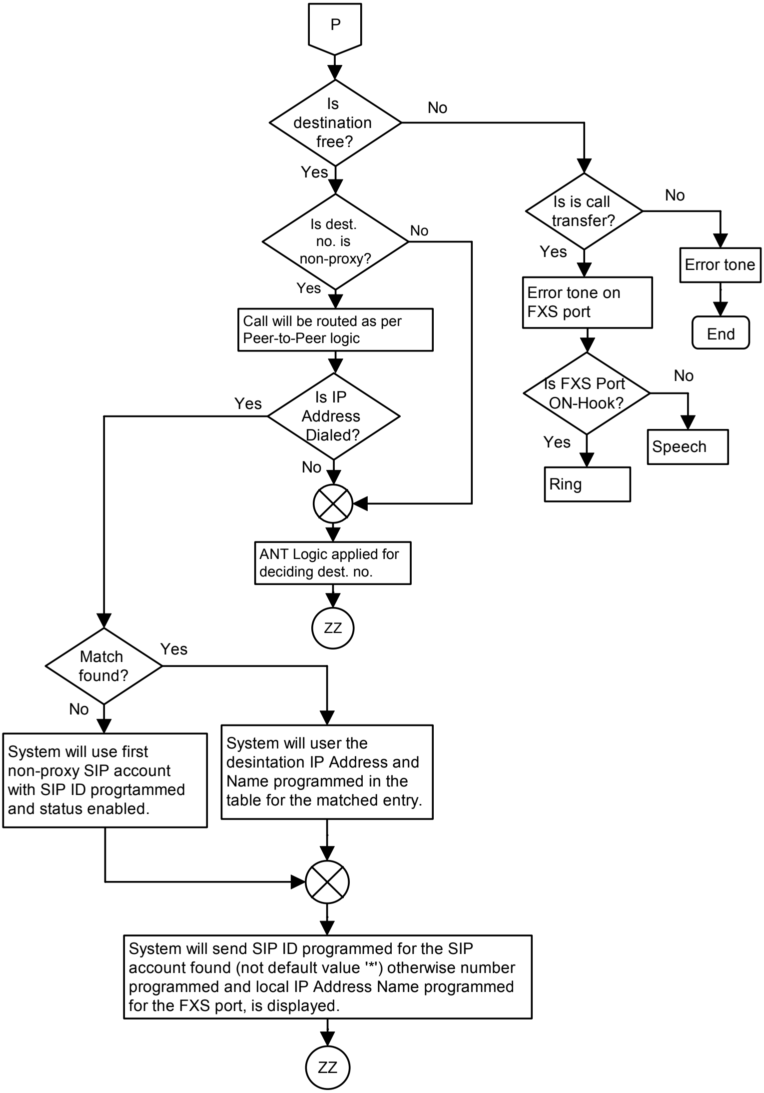
- The dialed number is checked for ‘Access code’. If it matches with the code, the feature is set as per the ‘code’ for example; Call forward, Call Waiting, DND etc. If any feature is set the gateway will process the call as per the feature otherwise the dialed number is checked for Allowed-Denied logic. The call will be routed as per call routing option programmed for the FXS port: Fixed or Dialed Number Based.
- The call is routed as per Routing Group assigned and the destination port is selected as per member selection logic either First free or Rotation.
- Now after selecting the destination port (SIP Account Number) from the Routing Group, if the dialed number is IP Number which should use SIP Proxy to route the call, then number will be dialed as per ANT logic and call will be established. But if IP Number is for non-proxy type, then Peer-to-Peer table is followed and then ANT logic is applied to dial out the number.
- Note that if he dialed number is Emergency Number or Access Code is ##, then the number will be dialed out from the Life Line port.

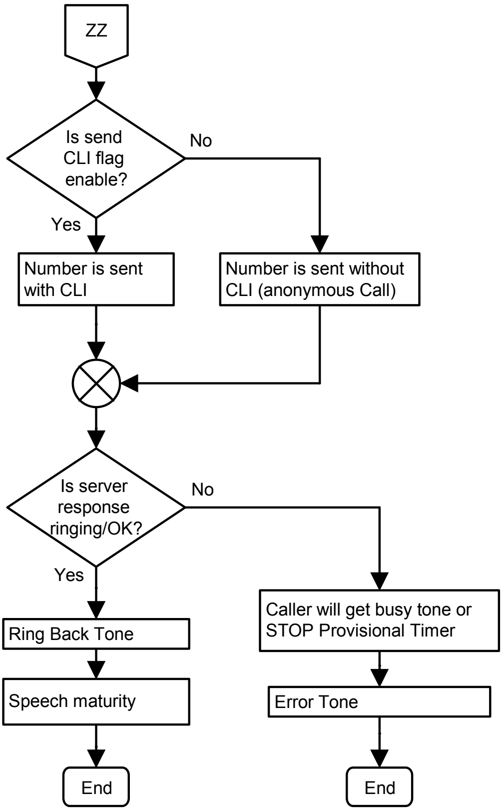
SIP to FXS Call:

- If you receive any IC call, the gateway will first check if the IC call is allowed or not and then follow the logic for Anonymous call and match with Black Listed Number to decide on blocking the IC call.
- If the number is allowed, the call will be routed as per the Routing Option set: All Calls or Called Number Based Routing.
- The call is routed as per Routing Group assigned and the destination port is selected as per Member selection logic, either First Free or Rotation.
- Now after selecting the destination port (FXS Port), from the Routing Group, the call is landed on the destination FXS port. The Phone connected on that FXS Port will ring with cadence for the Country-type programmed This will give indication for an IC call. Speech is established if FXS port goes OFF-Hook.
- Now if another IC call is there for the gateway and the feature Call Waiting is set then you will get Call Waiting Beeps. IC Call is not supported from the Life Line port.

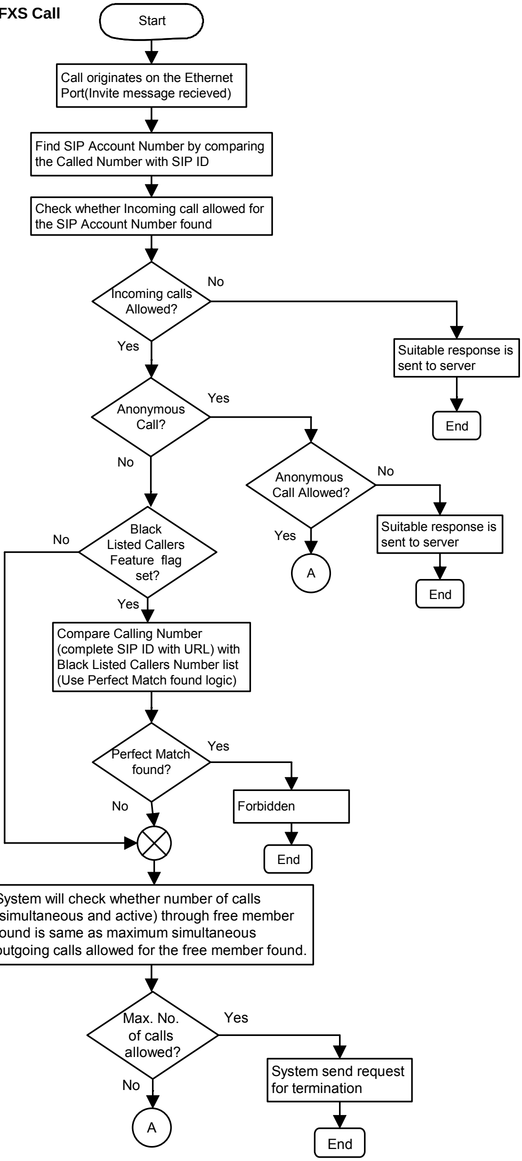
FXS to SIP Call

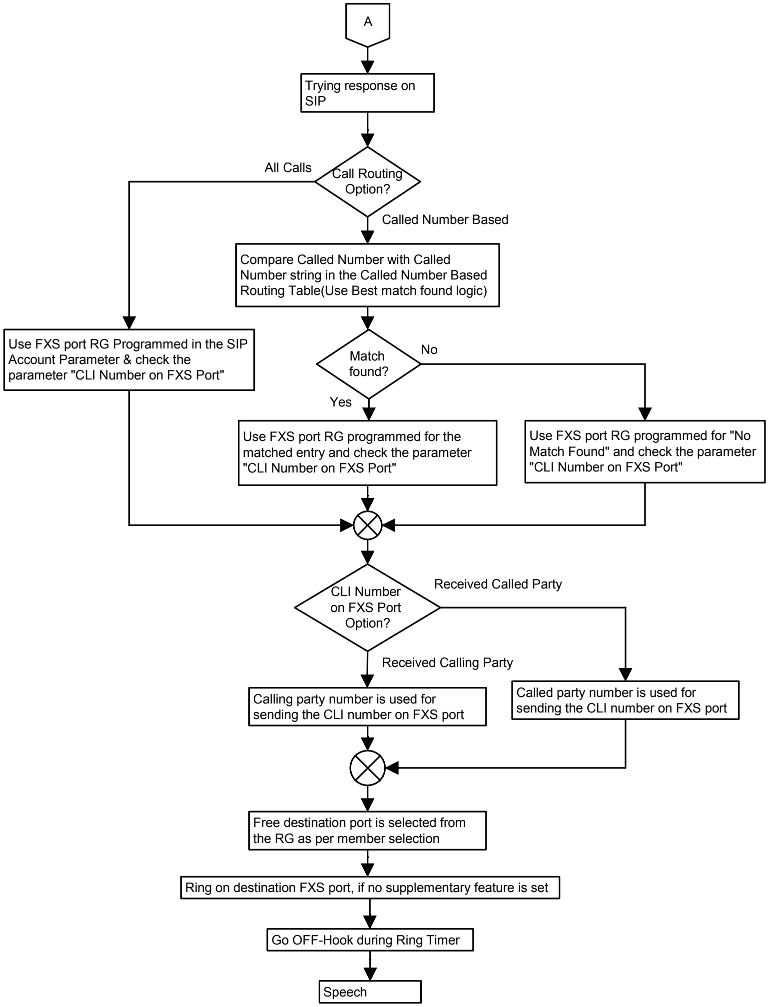



P
Is destination free?
No
Yes
Is is call transfer?
No
Yes
Error tone
End
Error tone on FXS port
Is FXS Port ON-Hook?
No
Yes
Speech
Ring
Is dest. no. is non-proxy?
No
Yes
Call will be routed as per Peer-to-Peer logic
Is IP Address Dialed?
Yes
No
ANT Logic applied for deciding dest. no.
ZZ
Match found?
Yes
No
System will use first non-proxy SIP account with SIP ID progrtammed and status enabled.
System will user the desintation IP Address and Name programmed in the table for the matched entry.
System will send SIP ID programmed for the SIP account found (not default value '*') otherwise number programmed and local IP Address Name programmed for the FXS port, is displayed.
ZZ


ZZ
Is send CLI flag enable?
No
Yes
Number is sent with CLI
Number is sent without CLI (anonymous Call)
Is server response ringing/OK?
No
Yes
Ring Back Tone
Speech maturity
End
Caller will get busy tone or STOP Provisional Timer
Error Tone
End

SIP to FXS Call





Relevant Topics:

01. SIP Account Parameters 149
02. FXS Port Parameters 103
03. Access Codes 29

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Call Progress Tones

What's this?

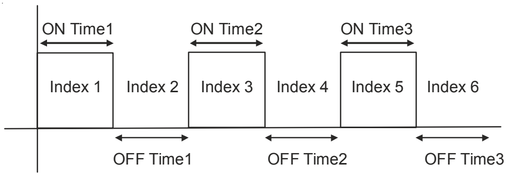
- To indicate response of various events while establishing the call, the gateway provides different types of tones.
- It is played only when FXS port is a Source port.
- These tones are not same for all the countries. Hence the gateway provides country-specific standard values of Frequency and Cadence for the tones as mentioned below:
    - Dial Tone
    - Ring Back Tone
    - Error Tone
    - Busy Tone
    - Confirmation Tone
    - Feature Tone
    - Routing Tone
    - Intrusion Tone
    - Remote Hold Tone
    - Remote Hold and Call Waiting Tone

How it works?

- Remote Hold Tone' and 'Remote Hold+Call Waiting Tone' are **fixed** for all countries and are not programmable.
- You can select from two options of Call Progress tones for remaining tones, Countrywise or Customized.
- If **Countrywise** option is selected, all Call Progress tone will be played as per the country selected.

Customized Option:

- The gateway supports programmable Frequency and Cadence for each tone. This is required if you want to set your own 'type' or if the country is not supported by default.
- Each call progress tone will have following parameters associated with it:
    - Frequency Set (Frequency 1 and Frequency 2) to be used
    - Six Cadence Periods
- Typical Cadence Index will look like as:



Where ON1 and OFF1 time is for Cadence1 and ON2 and OFF2 is for Cadence2.

- If Customized option is selected, all Call Progress tone are played as per the cadence and frequency programmed.
- Frequency is programmable from range of fixed values.

Condition for playing different tone:

- **Dial Tone:**
  - You will get Dial tone when you go OFF-Hook for period of First Digit Wait Timer programmed for the FXS port. First digit wait timer is programmable and refer FXS Port Parameters for programming. If any digit is dialed during this First Digit Wait Timer, dial tone will be stopped.
  - If Hotline feature with Hotline Timer=0, is set than you will not get dialtone and destination number will be dialed by the system.
  - If any feature like DND or Call Forward is set than you will not get dial tone.

- **Ring Back Tone:**
  - You will get Ring Back tone when there is Ringing response on the SIP Account.
  - Ring Back tone is played till call is rejected.

- **Error Tone:**
  - You will get Error tone when you perform some invalid operation or try to access some denied parameter.
  - For example:
    - Dialing the denied number, or
    - Routing is failed.
    - Error tone is played for the fixed time duration i.e Error tone timer=07 seconds. This timer is not programmable. But when you enter invalid command or invalid value in programming

mode, this tone is played for fixed duration i.e Programming Error tone timer = 7 seconds fixed.

- **Busy Tone:**
  - When a call is made by system and it gets response for 'busy', caller gets the 'Busy Tone for the fixed time duration i.e Busy tone timer=07 seconds. This timer is not programmable.

- **Confirmation Tone:**
  - You will get Confirmation tone, when you program the IP Address, correctly.
  - Confirmation tone is played for the fixed time duration i.e Confirmation tone timer = 07 seconds. Confirmation tone timer is not programmable.

- **Feature Tone/Programming Tone:**
  - You will get Feature tone instead of Dial tone when you set some features for e.g. Call Forward or DND. It will be played for the First digit Wait timer or Hotline timer.
    - **For Example:**
      - This tone is played when you enter the Programming code i.e ***#19.*** This tone will be played for fixed duration i.e Programming Digit Wait timer=7 seconds.

- **Routing Tone:**
  - Refer 'Call Routing Tone'.

- **Intrusion Tone:**
  - You will get Intrusion tone when you are in speech and there is another call for you.
  - Intrusion tone will be played till you go ON-Hook or incoming call is disconnected.

- **Remote Hold Tone:**
  - This tone will be played when remote party has sent a HOLD message and speech packets are not sent by the Remote end.

- **Remote Hold + Call Waiting Tone:**
  - This tone will be played when user is in Remote Held state and

there is another call present for the user (Call Waiting).

- This tone is played by stopping the Hold tone (send by the remote end or played locally).

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on "**Call Progress Tones**".
- Select option from: Countrywise or Customized. ***Default = India.***

Default table for Customized option.

Tone Type	Freq. 1 (Hz)	Freq. 2 (Hz)	Cadence					
			ON Time1 (msec)	OFF Time1 (msec)	ON Time2 (msec)	OFF Time2 (msec)	ON Time3 (msec)	OFF Time3 (msec)
Dial Tone	400	0	9999	0	0	0	0	0
Ring Back Tone	400	0	400	200	400	2000	0	0
Error Tone	400	0	250	250	0	0	0	0
Busy Tone	400	0	750	750	0	0	0	0
Confirmation Tone	400	0	100	100	0	0	0	0
Feature Tone/ Programming tone	400	0	100	900	0	0	0	0
Routing Tone	400	0	100	1900	0	0	0	0
Intrusion Tone	400	0	200	100	200	7500	0	0
Remote Hold Tone	400	0	500	3000	0	0	0	0
Remote Hold+Call Waiting Tone	400	0	70	70	70	70	70	1200

- Select either Countrywise or Customized option.
- If country wise option is selected you will get a list of countries and the CPTG table will be as per the values for the country presently selected. It is not programmable.
- If customized option is selected, you will not get list of country and the CPTG table will be programmable. The default values will be as per the last country selected.
- Enter the Frequency1 and Frequency2 from Range: 0,350,400,425, 440,450,480,620,950,1400.
- If Off time is not programmed, than the tone will remain continuous ON after that cadence i.e if Off time 3 is not programmed, than the tone will be played as per cadence programmed till On time 3 but than the tone will remain continuous ON.
- If Off time is programmed as last cadence, than the cycle will

repeat after that Off time programmed.

- Frequency F1 and Frequency F2 will be used as a juxtaposition of two frequencies F1 and F2 without modulation.
- For all countries Confirmation Tone is 400 Hz, 0.1 ON , 0.1 OFF, Programming Tone is 400 Hz, 0.1 ON, 0.9 OFF and Routing Tone is 400 Hz, 0.1 ON, 1.9 OFF.

For list of countries refer the table as under:

Code	Country	Dial Tone		Ring Back Tone		Error Tone		Busy Tone		Confirmation Tone		Intrusion Tone	
		Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence(sec)
		Hz	second	Hz	second	Hz	second	Hz	second	Hz	second	Hz	second
01	Argentina	425	cont.	425	1.0on 4.0 off	425	0.3on 0.4off	425	0.3on 0.2off	400	0.1on 0.1off	425	0.3on 10.0off
02	Australia	425*25	cont.	400*25	0.4on 0.2off 0.4on 2.0off	425	0.375on 0.375off	425	0.375on 0.375off	400	0.1on 0.1off	425	0.2on 0.2off 0.2on 4.4off
03	Belgium	425	cont.	425	1.0on 3.0off	425	0.167on 0.167off	425	0.5on 0.5off	400	0.1on 0.1off	1400	0.175on 0.175off 0.175on 3.5off
04	Brazil	425	cont.	425	1.0on 4.0 off	425	0.25on 0.25 off	425	0.25on 0.25off	400	0.1on 0.1off	425	0.05on 1.0off
05	Canada	350+440	cont.	440+480	2.0on 4.0off	480+620	0.25on 0.25off	480+620	0.5on 0.5off	400	0.1on 0.1off	440	0.3on 10.0off
06	China	450	cont	450	1.0on 4.0off	450	0.7on 0.7off	450	0.35 on 0.36off	400	0.1on 0.1off	450	0.4 on 4.0off
07	Egypt	425*50	cont	425*50	2.0on 1.0off	450	0.5on 0.5off	425*50	1.0on 4.0off	400	0.1on 0.1off	425*50	0.1on 0.1off 0.1on 2.7off
08	France	440	cont	440	1.5on 3.5off	440	0.25on 0.25off	440	0.5on 0.5off	400	0.1on 0.1off	440	0.3on 10.0off
09	Germany	425	cont	425	1.0on 4.0off	425	0.24on 0.24off	425	0.48on 0.48off	400	0.1on 0.1off	425	0.2on .2off .2on 5.0off
10	Greece	425	0.2on 0.3off 0.7on 0.8off	425	1.0on 4.0off	425	0.15on 0.15off	425	0.3on 0.3off	400	0.1on 0.1off	425	0.3on 10.0off 0.3on 10.0off
11	India	400*25	cont	400*25	0.4on 0.2off 0.4on 2.0off	400	0.25on 0.25off	400	0.75on 0.75off	400	0.1on 0.1off	400	0.2on 0.1off 0.2on 7.5off
12	Indonesia	425	cont.	425	1.0on 4.0off	425	0.25on 0.25off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.15on 0.15off 0.15on 10.0off
13	Iran	425	cont	425	1.0on 4.0off	425	0.25on 0.25off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.2on 0.2off 0.2on 10.0off
14	Israel	400	cont	400	1.0on 3.0off	400	0.25on 0.25off	400	0.5on 0.5off	400	0.1on 0.1off	400	0.5on 10.0off
15	Italy	425	cont.	425	1.0on 4.0off	425	0.2on 0.2off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.4on 0.1off 0.25on 0.1off 0.15on 5.0off
16	Japan	400	cont	400*20	1.0on 2.0off	400	0.25on 0.25off	400	0.5on 0.5off	400	0.1on 0.1off	400*25	0.5on 2.0off 0.05on 0.45off 0.05on 3.45off
17	Kenya	425	cont	425	0.67on 3.0off 1.5on 5.0off	425	0.2on 0.6off	425	0.2on 0.6off 0.2on 0.6off	400	0.1on 0.1off	425	0.1on 0.1off 0.1on 2.7off
18	Korea	350+440	cont	440+480	1.0on 2.0off	480+620	0.3on 0.2off	480+620	0.5on 0.5off	400	0.1on 0.1off	350+440	0.25on 0.25off 0.25on 3.25off
19	Malaysia	425	cont	425	0.4on 0.2off 0.4on 2.0off	425	0.5on 0.25off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.1on 0.1off 0.1on 2.7off
20	Mexico	425	cont.	425	1.0on 4.0off	425	0.25on 0.25off	425	0.25on 0.25off	400	0.1on 0.1off	425	0.1on 0.1off 0.1on 2.7off
21	New Zealand	400	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.25on 0.25off	400	0.5on 0.5off	400	0.1on 0.1off	400	0.2on 3.0off 0.2on 5.0off
22	Phillippines	425	cont.	425+480	1.0on 4.0off	480+620	0.25on 0.25off	480+620	0.5on 0.5off	400	0.1on 0.1off	440	0.3on 10.0off
23	Poland	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.15on 0.15off 0.15on 4.0off
24	Portugal	425	cont.	425	1.0on 5.0off	450	0.33on 1.0off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.2on 0.2off 0.2on 5.0off
25	Russia	425	cont.	425	0.8on 3.2off	425	0.25on 0.25off	425	0.4on 0.4off	400	0.1on 0.1off	950	0.333on 1.0off
26	Saudi Arabia	425	cont.	425	1.2on 4.6off	425	0.25on 0.25off	425	0.5on 0.5off	400	0.1on 0.1off	425	0.15on 0.2off 0.15on 10.0off
27	Singapore	425	cont.	425*24	0.4on 0.2off 0.4on 2.0off	425	0.25on 0.25off	425	0.75on 0.75off	400	0.1on 0.1off	425	0.3on 0.2off 0.3on 3.2off
28	South Africa	400*33	cont.	400*33	0.4on 0.2off 0.4on 2.0off	400	0.25on 0.25off	400	0.5on 0.5off	400	0.1on 0.1off	400*33	0.4on 4.0off
29	Spain	425	cont.	425	1.5on 3.0off	425	0.25on 0.25off	425	0.2on 0.2off	400	0.1on 0.1off	425	0.175on 0.175off 0.175on 3.5off
30	Thailand	400*50	cont.	400	1.0on 4.0off	400	0.3on 0.3off	400	0.5on 0.5off	400	0.1on 0.1off	400	0.1on 0.1off 0.1on 2.7off
31	Turkey	450	cont.	450	2.0on 4.0off	450	0.2on 0.2off 0.6on 0.2off	450	0.5on 0.5off	400	0.1on 0.1off	450	.2on .6off .2on 8.0off
32	UAE	350+440	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.4on 0.35off 0.225on 0.525off	400	0.375on 0.375off	400	0.1on 0.1off	350+440	0.1on 0.1off 0.1on 2.7off
33	UK	350+440	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.4on 0.35off 0.225on 0.525off	400	0.375on 0.375off	400	0.1on 0.1off	350+440	0.1on 0.1off 0.1on 2.7off
34	USA	350+440	cont.	440+480	2.0on 4.0off	480+620	0.25on 0.25off	480+620	0.5on 0.5off	400	0.1on 0.1off	440	0.3on 10.0off

Relevant Topic:

1. Call Routing Tone 79

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Call Routing Tone

What's this?

- When the call is routed on the SIP Account, the caller hears silence for sometime till the called party sends the Ringing response and user hears 'Ring'.
- During this silence period, the gateway provides specific tone indication to the user. This is called 'Call Routing Tone".
- It is applicable only if Source port is FXS and destination port is SIP Account.

How it works?

- Routing Tone is played on the FXS port only when call is initiated on any SIP Account.
- It is programmable with frequency and cadence just as other tones.
- This tone is stopped when gateway receives suitable response from the SIP Account and hears 'ringing'.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on **'Call Progress Tones'**.
- Select Country. ***Default = India.***

Relevant Topics:

1. Call Progress Tones 74
2. Routing Options 133

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Call Waiting

What's this?

- When the user is busy with one active call and there is another call for the same user, another call is considered as Call Waiting.
- User can Set/Cancel Call Waiting feature only if Call Waiting feature is enabled in Class of Service and Supplementary services.
- Feature can be set/cancel by the Access code as given below (by default):
  - Set Call Waiting ***71***
  - Cancel Call Waiting ***72***
- Access code for Accept Call Waiting call and release call waiting call are as given below:
  - Release Call Waiting ***Flash-3***
  - Accept Call Waiting ***Flash-4***

How to use it?

- You will get indication of a new Incoming call when user will get the Call Waiting Beeps (Intrusion tone) during speech condition. To attend the waiting call proceed as described below:

Option 1:

- Go ON-Hook.
- Active call will be disconnected.
- You will get Ring on the telephone.
- Go OFF-Hook, during ring.
- You will be connected to the waiting call.

Option 2:

- You can dial **Flash-4.**
- Now you will be directly connected to the Waiting call and current call will be released.
- When user hears the Call Waiting Beeps(Intrusion tone) during speech condition and if user wants to stop Call Waiting beeps i.e do not want to accept the Waiting call,
  - User has to dial **Flash-3**.
  - Now user will remain connected to the current call and Call Waiting beeps will be stopped.

How it works?

- Access code for Set/Cancel Call waiting is programmable.
- System will not check Call waiting feature in following cases:
  - In dial state (user is dialing the number)
  - In programming mode
  - In matured state, if user is dialing Matured state access code
  - In Remote Held, if user is ON-Hook
  - In Hold condition i.e user has put someone on 'Hold'.
- System gives Call Waiting beeps in following cases:
  - If user is in speech with another call.
  - If user is Held by Remote party and user is in OFF-Hook condition.
- Only one Waiting call will be allowed.
- If Call waiting call is disconnected or if remote party has gone ON-Hook, before the user has answered the call, Call Waiting beeps (Intrusion tone) are stopped.
- But if the user goes ON-Hook, the current call is disconnected and Call waiting beeps are stopped. User will get Ring.

How to program?

Refer VFX88L/VFX44L Web Jeeves.

- Click on **'Class of Service'** to Allow the feature, for the FXS port.
- Click on **'Supplementary Services'** to Enable the feature.
- To set the feature, using phone: Dial the programmed Access Code to set/cancel the feature.
- Click on Access Codes Table to know or program a new Set/Cancel Call Waiting code.

Important Points:

- System will check Call Waiting after Call Forward-When Busy feature.
- After dialing ***Flash*** (i.e dialing the matured state access code), if there is another call for the same user, that user is not considered for Call waiting and the call is routed to the next port from the routing group for routing the call.

Relevant Topics:

1. Access Codes 29
2. Call Progress Tones 74
3. Class of Service 86
4. Supplementary Services 161

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Calling Line Identification Presentation (CLIP)

What's this?

- Displaying the calling party number on the LCD of the phone is called Calling Line Identification Presentation (CLIP).
- When the call originates on the source port i.e. SIP Account, calling party number as sent by the service provider is detected by the system and is presented on the destination port i.e. FXS port.

How it works?

- Each FXS Port can be assigned any of the following CLIP type of signaling for presenting on the LCD display of the phone connected to that port:
  - None
  - DTMF
  - V.23 FSK
  - Bellcore FSK

None:

- If this type is selected, the gateway will not present the CLI on the FXS port. This is useful when the telephone instrument on FXS port does not have the LCD to display the number or you do not want to know the calling party number.

DTMF:

- If this type is set on the FXS port, only 'Number' is displayed on the LCD display of the phone. Phone connected to FXS port should also support detection of DTMF type CLI.

CLIP:

- If V.23 FSK or Bellcore FSK is set on the FXS port number and Name are displayed on the LCD display of the phone. If only number is received, then the number is displayed in the 'Name' field of the LCD display. This is because, the 'Number' field of the phone do not support any character like '.' or '@'. It supports only numerical value.

How to program?

- Click on "**FXS Port parameter**".
  Refer chapter "FXS Port Parameters" for more details.

CLIP Type:

- Select the signaling type to be supported for FXS Port from options: None, DTMF, V.23 FSK or Bellcore FSK.
  Default = DTMF, for all FXS ports.

Relevant Topics:

1. FXS Port Parameters 103
2. Routing Option-Called Number Based 135

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Change Password

What's this?

- For system security it is advisable to change the default 'System password' after configuring the required parameters.
- For this user has to just enter New password at two fields.
- ***Default Password is '1234'.***

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on 'Password Change'.
- Enter new password: Enter here the password which will be used every time to Log-in the system. Use maximum 4-digits, 0-9.
- Confirm new password: Enter same password as entered above.
- If user has lost the changed Password, the only way to default the Password is by using Hardware default method as explained below:
  - Switch OFF the Gateway. Change the Jumper **'J10'** in position 'AB' and Power 'ON' the Gateway. Wait for few seconds. The Password will get default value.
  - Switch OFF the Gateway and put back the Jumper **'J10'** in position 'BC'.

Important Point:

- For the password, digits entered are displayed as "*".

Relevant Topic:

1. Default the Configuration 91

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Class of Service

What's this?

- The gateway supports multiple FXS Ports. It also supports many features like call forward, Hot line etc. But some users may not be allowed to use the feature which is enabled in the gateway.
- For this, the feature 'Class of Service (COS)' is supported by the gateway. It is used to decide which feature will be allowed and which feature will not be allowed to a particular user.

Note:

- COS is not applicable for SIP Accounts.

- Following features are provided for Class of Service:
  - Hotline
  - Call Waiting
  - Call Hold
  - Blind Transfer
  - Call Forward
  - Do Not Disturb (DND)
- Refer corresponding chapter for details of each feature.

How it works?

- Class of Service table is checked when any user tries to use access code of following features:
  - Call Waiting
  - Call Hold
  - Blind Transfer
- For each FXS port, the feature can be Enable/Disable from COS.
- If the feature 'Access Code' is disabled in Class of Service than you will get error tone while using that access code.
- If it is enabled in class of service then user will get 'Confirmation Tone' if the access code is dialed.

How to program?

Refer VFX88L/VFX44L Web Jeeves.
Click on '**Class of Service**'.

FXS Port: This is non programmable. Enter status for each port to enable/disable the feature.

Features: Select Enable/Disable for each feature as required. For example if you need to enable ‘Hot Line’ for FXS4, then select ‘Enable’ for FXS4, for feature ‘Hot Line’.

Important Points:

- When a feature is disabled from Class of Service its ‘status’ is also changed in Supplementary Services. For Example, if Hotline, Call Forward and DND is changed from Enable to Disable for any FXS port, then is also ‘disabled’ in the status page for supplementary service.
- Similarly, Call Waiting feature was enabled in Class of Service. If now you change to disable, in Class of Service. In this case if you try to Cancel the Call Waiting, you will get error tone because the feature is disabled in the Class of Service.
- Thus, the status also will be changed for this feature when Class of Service is changed to disable.

Relevant Topics:

1. Supplementary Services 161
2. Call Forward 57
3. Do Not Disturb (DND) 95

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Daylight Saving Time

What's this?

- The Gateway supports various features like CDR etc., which are dependant on the current date and time settings of the gateway. Such facilities work properly only if the Gateway is set with correct date and time value. For local time, refer chapter <u>RTC Parameters</u>.
- DST is Daylight Saving Time, which is the new local time, a region is assigned for a period of the year, usually an hour forward from its standard official time. Gateway supports the facility.
- While using this feature, many countries' clocks will be advanced at the beginning of the daylight saving time and delayed at the end of day light saving time.

How to program?

- Click on '**Daylight Saving Time Adjustment**'. Program the parameters as per the country DST rules, where gateway is installed.

Forward Time Adjustments:

Type: Select the type from:

- None
- Day-Month wise: Program this option only for a country where DST is observed as per Day-Month wise every year
- Date-Month wise: Program this option only for a country where DST is observed as per Date-Month wise every year.
  Default = None.

Day-Month Wise:

- **On:** Program 'day' and 'month' on which DST will be applied every year. For this, program Ordinal, Day and Month. To program this 'Type' should be selected as 'Day-Month wise'. **Default = 1st. Sunday January.**
- **Ordinal:** Program 1st, 2nd, 3rd, 4th, or 5th. For example if DST starts on first Sunday of January every year, then program '1st' here.

Note:

- For last Sunday (or other day) of the month, always set Ordinal

Number = 5th. For e.g. If the month has 4 Sundays in a particular Calendar year then the last Sunday will be automatically the fourth one and if the month has 5 Sundays in a particular month of the Calendar year then the last Sunday will be automatically the fifth one, and since while programming it will not be known that in a given year, the last Sunday will be fourth or fifth one, it should be programmed as 5th.

- **Day:** Program the day on which DST starts. Select from Sunday to Monday of the week. In above example, select 'Sunday'
- **Month:** Program the month on which DST starts every year. Select the month from January to December of the year. In above example, select 'January'.
- **Change Time From:** It is the current time settings at which DST will start to change. The time is in HH:MM (Hours-Minutes) format, where HH=00 to 23 and MM=00 to 59. **Default = 00:00.**
- **To:** Enter forwarded time here at which DST will be forwarded. Enter the time in HH:MM (Hours-Minutes). **Default = 00:00**

Date-Month Wise:

- **On:** Program 'date' and 'month' on which DST will be applied every year. For this, program Date and Month. For this option for 'Type' should be selected as 'Date-Month wise'. **Default = 01 January.**
- **Date:** Program the date on which DST starts. Select from '01' to '31' of the month. For example if first January is the DST date, then select '01'.
- **Month:** Program the month on which DST starts every year. Select the month from January to December. In above example, select 'January'.
- **Change Time From:** It is the current time settings at which DST will start to change. The time is in HH:MM (Hours-Minutes) format. **Default = 00:00**
- **To:** Enter forward time here at which DST will be forwarded. Enter the time in HH:MM (Hours-Minutes). **Default = 00:00.**

Backward Time Adjustments:

- Program all the parameters as explained for 'Forward Time Adjustment' except the 'time for parameter 'To'. Default values are same.
- **To:** Enter the backward (delayed) time here, for the country.

Note: If for some country, the Backward Time Adjustments is at 00:00 hours, use previous date with “from” time = 23:59 and “to” time as required.

Above programming for forward and backward time adjustment are explained with following examples for the country.

- Hour-Minute, in the parameter ‘Change Time From’ is for current time settings whereas the Hour-Minute in ‘To’ is for time to which the clock should be forwarded to.

Example 1:

- In New Zealand, the DST starts on Last Sunday of October. The clock changes from 02:00 to 03:00. Hence, parameter ‘On’ will be as 5th Sunday, October as per Day-Month wise option, of ‘Forward Time Adjustment’.

Example 2:

- In New Zealand, the DST ends on Third Sunday of March. The clock changes from 03:00 to 02:00. Hence, parameter ‘On’ to affect DST in New Zealand, will be programmed as 3rd Sunday, March as per Day-Month wise option of ‘Backward Time Adjustment’.

Example 3:

- In Cuba, the DST starts on 1st April of every year. The clock changes from 01:00 to 02:00. Hence, parameter ‘On’ will be programmed as ‘01 April’ as per Date-Month wise of ‘Forward Time Adjustment’.

Example 4:

- In Cuba, the DST ends on last Sunday of October every year. The clock changes from 23:59 (in fact 00:00 midnight) to 23:00. Hence, parameter ‘On’ will be programmed as 5th Sunday, October as per option ‘Day Month wise’ of ‘Backward Time Adjustment’.

Relevant Topic:

1. Default the Configuration 91

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Default the Configuration

What's this?

Some times, user needs to get back all factory set values for the parameters when he is unable to analyze the problem while configuring. For this the Gateway supports 'default' feature.

How to use it?

- When the user clicks on the link "**Default System**", it shows Alert message:
  '***This option will assign default values to all the programmable parameters and will restart. Do you want to continue***?"
- If the user clicks 'Yes', the system will get default.

When the system gets default, following parameters will get default values:

- Access Codes
- Call Detail Records Filters
- Called Number Based Routing Table
- Class of Service
- Day light Saving Time
- Dialed Number Based Routing Table
- FXS Port Parameters
- FXS port routing group
- Network Parameters
- Number Lists
- Password
- Peer-to-Peer Table
- Prefix to Domain Name Conversion Table
- SIP Account Parameters-1
- SIP Account Parameters-2
- Supplementary Services
- Time setting parameters except RTC Time and Date

Note:

- Call Detail Records will not get cleared, when the system gets default.

Important Point:

- When the system is default, the system will Reset and the Reset sequence of LED will be as per chapter 'Getting Started'.

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Disconnect Signaling on FXS Port

What's this?

- As a general application, telecom equipment like PCO machine is connected to the FXS port of the system. Now whenever the called party (remote party) disconnects or goes on-hook it is required to inform the FXS port so that the PCO machine can consider the call as complete and stop billing. In absence of this signal, the call is considered as complete when the caller goes on hook. But this will result in inaccurate billing.
- To solve such problems, the system supports 'Disconnect Signaling'. When a call is made from FXS to SIP or SIP to FXS, if SIP disconnects, Disconnect Signal is generated on the FXS port.
- It can be in the form of:
    - None
    - Polarity Reversal
    - Open Loop Disconnect

How it works?

- 'Disconnect Signaling' is applicable for both incoming and outgoing calls.
- When a call is made from FXS port to SIP or SIP to FXS and call gets matured, the system will wait for 'disconnect message' on SIP port.
- If the message is received on SIP, the system will check the Disconnect Signaling programmed for the FXS port from where call has been originated or terminated.
- This parameter can be set for either one of the following options:
    - ***None***: It is used when no signaling is to be generated on FXS for call disconnection. When call is disconnected, user will get Error tone.
    - ***Polarity Reversal***: It is used when the call disconnection is to be signaled in the form of Polarity Reversal. The Battery polarity of the FXS port will be reversed. For example, if the battery polarity of the FXS port is +ve for TIP and -ve for RING in speech condition then on disconnection on other port, TIP will become -ve and Ring will become +ve. When call is disconnected, user will get Error tone. When FXS port goes ON-Hook, the Battery

polarity of the FXS port will get again reversed and user will not get error tone.

- ***Open Loop Disconnect***: It is used when call disconnection is to be signaled in the form of Open Loop Disconnect pulse. During this:
  - The Battery voltage on FXS port will be removed for time of Open loop disconnect timer programmed for that FXS port and will be restored again.
  - But the Polarity of the FXS port battery voltage will not be changed. When call is disconnected, user gets Error tone.
  - "Open Loop Disconnect Timer" can be programmed and is applicable only if Open Loop Disconnect option is selected for the Disconnect Signal.
  - User will get Error tone after disconnect signal is generated on the FXS port.

How to program?
Refer VFX88L/VFX44L Web Jeeves Pages
Click on '**FXS Port Parameters'**

Disconnect Signaling:

- Select the signaling option from 'None' or 'Battery Reversal' or Open Loop Disconnect for the FXS port. Program for remaining FXS ports.
  By default, Battery Reversal for all FXS Ports.

Open Loop Disconnect Timer (msec):

- It is applicable only if Open Loop Disconnect option is selected for Disconnect Signaling. Range = 001- 999 msec.
  By default, 500 msec. for all FXS Ports.

Relevant Topics:

1. Answer Signaling on FXS Port 34
2. FXS Port Parameters 103
3. Call Processing 66
4. Call Progress Tones 74
5. Ring Type 130

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Do Not Disturb

What's this?

- Do Not Disturb (DND) feature is for users who wishes privacy for some duration and do not want to receive any calls.
- But at the same time he can make outgoing calls.
- User can set/cancel DND.

How it works?

- This feature is for IC Calls only.
- It can be set/cancel for FXS port only if DND feature is enabled in Class of Service for that FXS port.
- If DND and Call Forward-Unconditional is set, than system will check first, Call Forward-Unconditional.
- After setting DND feature, you will get Feature tone when you go OFF-Hook as an indication of the feature 'set'.
- Telephone instrument or Web Jeeves can be used for set/cancel.

How to program?

- Click on **'Class of Service'**. Select set/cancel DND as required. ***Default = Cancel.***
- By Phone:
  - Go OFF-Hook.
  - Dial ***61** access code to set the feature and dial ***62** access code to Cancel DND.
- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**Supplementary Services**' to Enable/Disable the feature.

Relevant Topics:

1. Access Codes 29
2. Class of Service 86
3. Supplementary Services 161

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Emergency Number Dialing

What's this?

- The gateway provides a Lifeline port which is not a 'true' FXO Port, for dialing the emergency numbers. No IC calls can be received on his port. The number must be preprogrammed in the system.
- The gateway does not support Emergency call through SIP port.
- Emergency Number can be dialed through FXS1 port only.

How to use Lifeline Port?

- Go OFF-Hook from phone at FXS1 port. You will get dial tone.
- Dial the access code ## or as per the Access Code table.
- You will again get the Dial tone if any line is connected to the Lifeline port.
- Dial the number. Speech is established, if call is matured.
- Go ON-Hook. Lifeline port is disconnected.

How to make Emergency Call?

- Go OFF-Hook from phone at FXS1 port.
- Dial programmed Emergency number. Lifeline port is accessed and you will get PSTN dial tone.
- Dial again Emergency number.
- The number will be dialed out.
- Go ON-Hook. Call is disconnected.

How it works?

- If there is perfect match for dialed number with programmed number in 'Account Code Table' that number is considered for emergency number.
- Allowed-Denied Number logic is not applicable for this feature.
- Maximum 4 Emergency Numbers can be programmed.
- Maximum 3 digits can be entered for Emergency Number.

How to program?

Refer VFX88L/VFX44L Web Jeeves.

- Click on Access Codes. Get the access code for dialing emergency number from 'Lifeline Access' field. ***Default = ##.***

- Program maximum 4 Emergency numbers at Emergency Number 1 to 4, using maximum 3 digits (0-9, #, *). ***Default = Blank.***

Relevant Topics:

1. Lifeline Port 110
2. Access Codes 29

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End of Dialing

What's this?

- End of dialing is a mechanism by which end of number string is interpreted and the gateway can start out dialing faster. It will reduce delay for the inter digit wait timer.
- The gateway support following mechanisms for using as "End of Dialing" for the FXS-Port:
  - End of Dialing Digit
  - Number length
  - Inter Digit Wait Timer
- Length of digit programmed is useful for some features. For Example, for Peer-to-Peer Calling it is required that fixed 4 digit extension number has to be dialed. Hence Number of digits programmed will be '4' and after entering 4-digits, the gateway will dial out the number.
- Similarly if '#' is programmed then dialing "#" after entering the number, will be required for faster dialing. Only * or # can be considered as 'End of Dialing Digit'
- ***By default, '#' digit is considered as End of dialing digit for each FXS port.***
- After entering few digits if you do not dial any digit and time Inter Digit Wait Timer is expired, then system will consider it as end of dialing and system will dial out few digits only.

How it works?

- The option of enable/disable can be programmed for any of the above mechanisms.
- If all are enabled, end of dialing is interpreted on the basis of first trigger received. For example, '#' is set for End of dialing digit, number length is set to 5 and inter digit wait timer is set to 6 seconds, then end of dialing is considered using '#' if received after 3 digits. Likewise, end of dialing is interpreted if the inter digit timer expires after few digits or end of dialing is interpreted if 5 digits are received.
- If all options are disabled, "Inter Digit Wait Timer" is considered as End of Dialing method.
- End of Dialing is programmable for each FXS port.

- Number length is programmable.
- Inter digit wait timer is also programmable.

How to program?
Refer VFX88L/VFX44L Web Jeeves.
Click on '**FXS Port Parameters**'.

Number of Digits:
Apply: Click here to disable the feature. ***Default is enabled***.
Number of Length: Enter how many digits which are dialed will be considered as End of dialing process by the gateway. Range = 01=24, ***default = 16.***

End of Dialing Digit:
Apply: Click here to disable the feature.
Digit: Enter the digit to work as EOD. Range = # or *, ***default = #.***

Inter Digit Wait Timer:
Apply: Click here to disable the feature.
Timer: Enter the period of time after the time of last digit entered, which the gateway will consider as End of entering digit and the number will be dialed out. Range = 01-99, ***default = 5 Sec.***

Important Points:
- If you dial a number string then 'First Digit' of the string is not considered as End of Dialing Digit.
- End of Dialing digit will be * or # only because, other digit on the phone are 0-9 which is generally required while dialing the number.

Relevant Topics:
1. FXS Port Parameters 103
2. Routing Option-Fixed 142
3. Routing Option-Dialed Number Based 139
4. Inter Digit Wait Timer 107

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First Digit Wait Timer

What's this?

- It is the Time period for which the system waits for receiving a first digit after going OFF-Hook or answering the call when originated on any port.
- It is programmed for the FXS port of the gateway. After this time the call is processed further.

How it works?

- First Digit Wait Timer is assigned to each FXS port.
- Range for the First Digit Wait Timer will be 01-99 sec.
- This timer is not used if Hotline feature is enabled for the FXS port.
- If any digit is dialed before expiry of timer, the system will start the Inter digit wait timer and stop the first digit wait timer.
- If no digit is dialed before expiry of timer, you will get Error tone.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on **'FXS Port Parameters'**.
- Enter the time period up to which system should wait for entry of first digit after going OFF-Hook.
  Range = 01-99.
  Default = 15 sec.

Relevant Topic:

1. FXS Port Parameters 103

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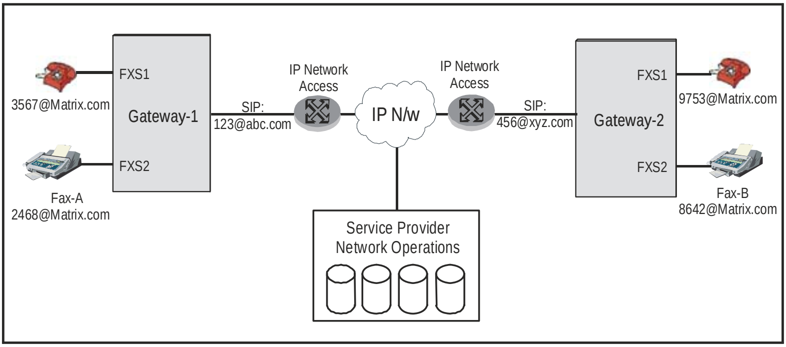
Fax over IP (FoIP)

What's this?

- This chapter describes the application of gateway for using the Fax feature. The gateway supports Fax Over IP (FoIP).
- The gateway can be used for sending Fax message as explained below. Following example explains how fax calls will be handled by the Gateway.

Example:



How to use it?

- User will select type of FAX that will be supported by the gateway, from: T.38 (UDPL), T.38 (RTP) and Pass through. It is preferable that this selection is same as type of Fax settings at 'remote end'.
- User of Gateway-2 operates 'Fax B' to make fax-call by dialing 123@abc.com.
- If Gateway-1 supports 'T.38', it acknowledges the request. Otherwise, it sends 'CANCEL' message to Gateway-2.
- The Gateway -2 re-negotiates with Gateway-1 for another vocoder (Normally G.711). This time, it is responded by Gateway -1 with same vocoder G.711, and the Fax-Call is established. This is called 'Fax Pass Through'.
- Similarly the fax-call is set when user of Gateway -1 initiates the call for Fax B by dialing 456@xyz.com.

How to program?

- Refer chapter "SIP Account Parameters" for more details

Important Point:

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

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FXS Port Parameters

What's this?

For interfacing telephones or Fax etc. at the FXS Port of the gateway, it is required to set some port parameters for each port. For example, facility can be provided to enable/disable a Port. Disabling port is required in case of Hardware failure.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**FXS Port Parameters**'.
- Program following parameters for all 1 to 8 FXS-ports (1-4 for FXS Ports for VFX44L) as required.

Status:

- Select disable if you want any FXS Port not to be used.
- Gateway will route the call from the port only if the port is enabled.
- When port is disabled and you go OFF-Hook, you will get error tone. The Gateway will not allocate the port for routing the call if port is disabled.

  By default, all port will be Enable.

Name:

Enter here the Name of maximum 12 characters, may be with ASCII.

- It is used as Display Name for making an outgoing call on any SIP Account.
- The name entered here is sent on the SIP Account as the Display name in the SIP message.

  Default: Blank.

Number:

- Enter here Maximum 4 digits number of the FXS port using digits Blank, 0-9, *, #.
- This number is used only when there is a Non-Proxy SIP Account call.
- It is send on the Non-proxy SIP Account call in the SIP message

  Default: 2001 to 2008 for FXS1 to FXS8 respectively (VFX88L).

  Default: 2001 to 2004 for FXS1 to FXS4 respectively (VFX44L).

Flash Timer

- Enter the time for ‘Flash’ period required.
  Range = 083msec to 999msec.
  Default: 600msec.

Refer related chapter for programming other FXS port parameters mentioned below:

- CLIP Type
- First Digit Wait Timer
- End of Dialing
- Answer Signaling-refer Answer Signaling on FXS Port
- Disconnect Signal-refer Disconnect Signaling on FXS Port
- Open Loop Disconnect Timer-refer Disconnect Signaling on FXS Port
- Allowed-Denied Numbers
- Routing Type-refer Routing Group
- SIP Account Routing Group (refer Routing Group)

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Hotline

What's this?

- Using this feature long numbers which are frequently dialed, can be dialed by the system when you go OFF-Hook for specified time period.
- To use this feature first enable it from 'Class of Service' for a specific FXS Port number.
- Also you need to program Hot line number and Hotline timer.
- This feature can be set/cancel by Access code or Web-page.

How to use it?

- Go OFF-Hook.
- Dial required number during Hotline timer period (if programmed) or Programmed number will be immediately dialed out by the gateway.

How it works?

- This feature is applicable only for FXS port and Hotline Number can be programmed for each FXS port.
- Hotline Timer is used if you want to dial any other number instead of Hotline number. The number should be dialed during this time.
- The timer is programmable for each FXS port.
- The Hot line number is not checked for Allowed-Denied Number list while dialed by the system.

How to program?

- Refer VFX88L/VFX44L Web Jeeves. Program following parameters for all required FXS ports.
- Click on **'Class of Service'**.
- Select 'Enable' to allow the feature by the system.
- Click on "Access code": To know the Access code for set/cancel of the Hot line feature.
- Click on '**Supplementary Services**'.
  - **Status:** Select to enable the feature for FXS Port, ***default = disable.***
  - **Number:** Enter maximum 24-digit number using 0-9, *, #, '.'.
  - **Timer:** Enter the time period after which the system will dial out

the Hot line number. Range = 0-9 sec. If ‘0’ is programmed, the number will be dialed out immediately after you go OFF-Hook. ***Default = 0.***

Relevant Topics:

1. Class of Service 86
2. Supplementary Services 161

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Inter Digit Wait Timer

What's this?

- It is the time period between dialing of two digits for which the system waits before out dialing the dialed digits.
- It is useful to decide that if during this time you do not dial anything, the system will dial out the number string dialed so far.

How it works?

- Inter Digit Wait Timer (IDWT) will be assigned to each FXS port.
- When the caller dials the first digit (if routing option is Fixed or Dialed Number Based), stop the first digit wait timer and start the Inter digit wait timer.
- If dialed digit is End of Dialing digit, the Inter digit wait timer is stopped.
- If Inter digit wait timer expires, the system will stop considering the digits dialed further.
- System will check the routing option for routing the call.

For Example:

- IDWT is programmed as 20 seconds. Now if you dial 567 and wait for 21 seconds before dialing digit '8'. Then number string 567 will be dialed out by the gateway, as per routing group and routing type, programmed.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on FXS Port Parameters.
- Refer chapter '<u>End of Dialing</u>'.

Relevant Topics:

1. FXS Port Parameters 103
2. Routing Option-Fixed 142
3. Routing Option-Dialed Number Based 139

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IP Dialing

What's this?

- This chapter describes some important conventions for dialing IP number in different condition and general information regarding parameters sent and received during IP call.
- IP number can be dialed with dot '.' as entered by '*' while dialing it. (If IP dialing is to be used, don't program '*' as the End of Dialing digit).
- For e.g. to dial IP address 192.167.100.1 dial as: 192*167*100*1 from the Phone at FXS.
- The system considers '*' as **'.'** , only when * is not dialed as first digit i.e. if user dials first digit as *, it will be considered as '*' dialed.
- While programming the IP number in the Number list for Allowed-Denied Number logic, enter directly **'.'**. For this, IP Number is considered as Normal Number, dialed.
- If IP address dialed is allowed, the system will use the first Non-proxy account whose SIP ID value is as default value ('*') and 'Status' parameter is enabled. No Routing option and Routing group selection logic will be applied.
- If SIP ID is programmed for all the Non-proxy SIP trunks, the system will use the first non-proxy SIP account whose 'Status' is enabled.
- The ANT logic is not applied after selecting the Non-proxy account.

Outgoing call from any SIP Account:

- When an OG call is made through any SIP Account (destination), the gateway will send following values to the destination (SIP Account) for further call processing:
  - SIP Account Number.
  - Complete Called Party Number with Called Party Name if present. e.g.: Sumer, 2001@192.168.1.10 (if Name and Destination IP address is found from the Peer-to-Peer Table for the dialed number) or 2001 only i.e. dialed number by user (if Name and Destination IP address is not found from the Peer-to-Peer Table for the dialed number) or 192.168.001.010 if only IP address is dialed by user.
  - Calling Station Number and Name(Number and name of the FXS

port who has dialed the number).

Incoming call on any SIP Account:

- When there is an IC call on any SIP Account, the SIP Account will send following parameters for call processing:
  - SIP Account Number (after checking and comparing the Called Number with the SIP ID).
  - Called Number and Name.
  - Calling Number and Name (if received).
- The gateway routes the call depending on the parameters and routing types applicable for that SIP Account number.

Relevant Topics:

1. Getting Started 19
2. End of Dialing 98
3. Programming Using Conventional Phone 126
4. SIP Account Parameters 149
5. Peer-to-Peer Calling 120

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Lifeline Port

What's this?

- Some times even if there is IP Network problem dialing of important number is required. For this the Gateway supports to dial such numbers from PSTN line connected to a port known as 'Life Line port'. It is not a true FXO port and IC calls are not supported.
- This feature is useful in following conditions:
  - If the Ethernet link is down.
  - If the SIP account fails to get registered.
  - If an emergency number is to be dialed.
  - If you want to dial a number using PSTN.
  - If you are unable to achieve toll quality speech through IP due to some reasons.
- The Gateway will switch back to Normal mode on completion of the call through Life line port.
- Refer chapter 'Emergency Number Dialing' for using Lifeline port.

How it works?

- The system compares the dialed number string with the numbers in the Access Code Table and if a perfect match is found it connects the user to the PSTN (Lifeline port).
- The user will listen to the PSTN dial tone and will get prompted to dial the number again.
- When user dials the number, same will be dialed out by the system.

Important Point:

- This port can be accessed only, from FXS1 port.

Relevant Topics:

1. Access Codes 29
2. Emergency Number Dialing 96

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Member Selection Method

What's this?

This is a method used for selecting a member from multiple member ports of the Routing Group.

- The gateway supports two methods for selection:
  - First Free
  - Rotation

For example:

- There are 8 numbers of members in the Routing Group. After selecting the Routing group for placing a call, it is required to select a member from 8 numbers of members for placing a call. Now if for last call the Nth. Member is selected then now it will select (N+1)th. Member of the group. This is called "Rotation".
- But the gateway can also select a port which is free starting from 1st port for call routing. This is called 'First Free' Method.

How it works?

- Member selection method is programmable for each Routing Group.
- Select the destination port for placing the call on the basis of the Member selection method assign to the Routing group.
- Programming will be done through Web Jeeves.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**SIP Account Routing Group**'.
- Click on '**FXS Port Routing Group**'.

Member Selection Method (FXS Port RG):

- Select the method from Rotation or first free for 16 routing groups. ***Default method is First Free for Routing Group 1 to 8 and Rotation for Routing Group 9 to 16.***

Member Selection Method (SIP Account RG):

- Default method is First Free for 1-8 Routing Group and Rotation for Routing Group 9.

Important Points:

- For SIP Account Routing group i.e. for FXS to SIP Account call, it is advisable for the SE to program the member selection method as ‘Rotation’ if more than 1 SIP Account is registered to the proxy and assigned to same Routing Group.
- If only 1 SIP Account is registered or non-proxy call has to be made i.e. only 1 member is programmed in the Routing Group, then member selection method ‘First Free’ should be assigned to that Routing Group.

Relevant Topic:

1. Routing Group 143

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Network Port Parameters

What's this?

- This chapter describes how to configure basic SIP settings and some of the advanced SIP and RTP parameters. For example, Static IP Address, Quality of Service (QoS) etc.
- You can set only 'IP address and subnet mask of Network port' using Web Jeeves as well as conventional phone instrument. Other parameters can not be programmed using conventional Phone.
- ***Default static IP address of Gateway is 192.168.001.156.***
- This Network port will be connected to the LAN Switch/Hub/Router/ Broadband modem.
- Change the IP address of the Network port when connected to LAN of the organization as per LAN addressing scheme. The LAN might be using static addresses or dynamic addresses. If static addresses are used, get the static address of the Network port from the LAN administrator and program it as the IP address for the Network port of Gateway. Refer chapter Getting Started.
- If DHCP server is used in the LAN, then enable DHCP server for the Network port. Doing so, the DHCP server in the LAN will assign an IP address dynamically to Network port of the Gateway whenever Gateway is restarted. This IP address assigned dynamically to the Network port of Gateway by the DHCP server can be known from the Web page of Network parameters.
- Refer topic 'Programming using Conventional Phone' for programming Network IP Address through Conventional phone.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**Network Port Parameters**'.

Notes:

- If Connection Type = DHCP, followings fields will be non-programmable:
  - PPPoE User ID
  - PPPoE User Password
  - IP Address
  - Subnet Mask

- Gateway IP Address
- The current values are displayed in the above fields except PPPoE User Password which will be displayed as ******.

Note:
- Gateway will restart after submitting this page.

Name:
- Enter 'Name' of WAN port. It will be used as a Tag only using maximum 12 characters and ASCII.
  Default: Blank.

Connection Type:
- Enter the type of getting IP Address which your gateway will implement when switched ON. The option of Connection Types are:
  - DHCP
  - PPPoE
  - Static IP

Note:
- If DHCP is enabled then IP address, Subnet Mask, Gateway Address, DNS IP Address will be replaced by the dynamic values sent by the DHCP server. Also, the PPPoE functionality shall be redundant.
  Default = Static IP.

PPPoE User ID:
- Enter this if Connection Type = PPPoE is selected using maximum 16 characters and extended ASCII Character set.
  Default: Blank.

PPPoE Password:
- Enter this if Connection Type = PPPoE is selected using a string of maximum 16 characters and extended ASCII Character set.
  Default: Blank.

IP Address:
- Enter this as provided by the ISP when Connection Type = Static IP is selected using string of maximum 15 characters.

Default: 192.168.001.156.

Subnet Mask:

- Enter this if Connection Type = Static IP is selected using maximum 15 characters.
  Default: 255.255.255.000.

Gateway IP Address:

- Enter this if Connection Type = Static IP is selected maximum 15 characters.
  Default: Blank.

DNS IP Address:

- The string of maximum 15 characters.
  Default: Blank.

DNS Domain Name:

- The string of maximum 40 characters with ASCII characters.
  Default: Blank.

MAC Address:

- This field is **NOT** programmable. It is an unique number programmed in the gateway.
- This will be displayed in Status page.

STUN Server's Address:

- This is used when there is a NAT router between gateway and SIP server.
- Enter the address of the STUN server. It can be IP address also.
- Maximum of 40 characters with extended ASCII characters.
  Default: Blank.

STUN Server's Port:

- Enter the STUN Server's listening port for SIP.
- Change this field if the ITSP provides STUN Server port number other than the default.
- Valid range for this field is 1024-65535.
  Default: STUN Server port is 3478.

NAT Keep Alive Status:
- It is Network Address translation. Enable it if required.
- On enabling this field, the gateway will send the NAT keep Alive message to the active registrar.
  Default: NAT is disabled.

NAT Keep Alive Interval:
- This specifies the time after which the Gateway will send SIP notify messages to the SIP server.
- Range = From 001 to 999 seconds.
  Default: NAT Keep Alive Interval is 120 seconds.

SIP QoS Type:
- Types of SIP QoS are as follows:
  - Precedence (ToS)
  - DiffServ
- This field specifies the QoS type viz. ToS (Precedence, also called Priority) or DiffServ for voice traffic.
- The Gateway will send all the voice packets with this QoS setting.
- If one is selected, the other one is not programmable.

SIP QoS Level:
- Types of SIP QoS Levels are as follows:
  - Precedence
  - DiffServe
- ToS uses 3 bits and hence has valid range from 0 to 7.
  Default = 5.
- DiffServ uses 6-bits and hence has valid range from 00 to 63.
  Default = 26.

RTP QoS Type:
- Types of RTP QoS are as follows:
  RTP QoS
  - Precedence (ToS)
  - DiffServe
  - This field specifies the QoS type viz. ToS (Precedence, also called Priority) or DiffServ for voice traffic. The Gateway will send all the voice packets with this QoS setting.
  - If one is selected, the other one is not programmable.

RTP QoS Level:

Types of RTP QoS Levels are as follows:

- Precedence (ToS)
- DiffServe
- ToS uses 3 bits and hence has valid range from 0 to 7.
  Default=5.
- DiffServ uses 6-bits and hence has valid range from 00 to 63.
  Default = 46.

SIP INVITE Timer:

- This timer is used when call is initiated and there is no response from the server.
- This timer will stop when OK response or Reject response is received.
- On expiry of this timer, user will get error tone.
- Valid range for this field is 001-999 seconds.
  Default: 180 seconds.

SIP Provisional Timer:

- This timer is used when call is initiated.
- If there is any provisional response like ‘trying’ or ‘remote-ringing’ etc. response comes from the server, this timer will be stopped.
- On expiry of this timer, user will get error tone.
- Valid range for this field is 001-999 seconds.
  Default: 60 seconds.

SIP Listen Port:

- Valid range for this field is 1024-65535.
  Default: SIP Listen port is 5060.

RTP Listen Port:

- Program only even number for this parameter.
- Valid range for this field is 1024-65502 (for VFX88L).
- Valid range for this field is 1024-65518 (for VFX44L).
  Default: RTP Listen port is 8000.

Important Point:

- The gateway will restart after submitting this page.

Relevant Topics:

1. VoIP Basics 166
2. Glossary 182
3. Getting Started 19
4. Web Jeeves 171

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Number Lists

What's this?

- Number List is a data structure which has maximum 24 maximum number of Number List and each Number list has maximum 64 maximum number of different numbers programmed in it.
- It is a common table which has different Number list and each list is programmed with different number string.
- The gateway makes use of these lists for allowed/denied list, automatic number translation, etc. features.
- Each Number list can be assigned to each port for above applications.

How it works?.

- You can enter Maximum 24 digits number in each entry using digits: 0-9, *, #, +, ASCII character.
- It can be used to Allow/Deny a number dialed on a source port.
- It is also used for Automatic Number Translation on a destination port and for Blocking Black Listed Callers.
- Programming the Number list will be done with Web Jeeves.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on **'Number Lists'**.
- Program maximum 64 number strings in each Number list and remember this list number to assign to the port as required.
  Default = Number list 01 is programmed with digits 0-9, *, # for first 12 index. All other number list are blank.

Relevant Topics:

1. Allowed-Denied Numbers 32
2. Automatic Number Translation 36
3. Black Listed Callers 51
4. SIP Account Parameters 149

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Peer-to-Peer Calling

What's this?

- Making a call on the VoIP port without going through any proxy is called Peer-to-Peer (P2P) Calling.

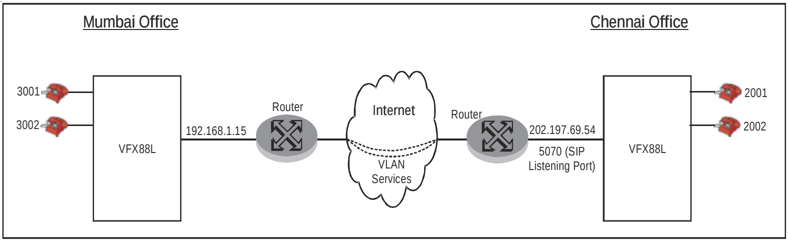
For example:

- Two gateways are installed at one Mumbai office and the other at Chennai office as shown below These two gateways are connected to each other by the virtual LAN (VLAN).
- Now if you want to call from Mumbai office to Chennai office without proxy server to route the call following information for the gateway at Chennai office, will be required:
  - Extension number
  - IP address
  - SIP Listen Port Number
- Program the Peer-to-Peer call table for the gateway at Mumbai office, using above information in Number, Destination Address and Destination Port column respectively.
- You have to dial just '2002' to call the gateway of Chennai office.
- Similarly networking in with other branch office in different cities can be done.

The table will look as shown below:

Index	Number	Destination Address	Destination Port	Name
001	No Match found (**Note1**)	192.168.001.156	5060	
002	2002	202.197.69.54	5070	Chennai

Note 1: As shown in table, if no match is found for the dialed number, then gateway will use first-entry for call routing. For this, the Destination Address is 192.168.001.156, Destination port is 5060, Number field is uneditable and Name field is blank.



How it works?

- Peer-to-Peer table is used in the gateway while making a call from FXS port to SIP Account i.e. when destination is SIP Account and not for SIP Account to FXS port call routing.
- The table consists of Number, Destination Address, Destination port and Name field for each entry.
- If the destination port is found to be a Non-Proxy SIP Account, then dialed number is matched with the entries in the Number field in the Peer-to-Peer table. The system will use the Number of best match found for making Peer-to-Peer call.
- The Destination address and Destination port corresponds to the address and SIP listening port, of the gateway to which the call has to be made.
- Thus, while making Peer to Peer call, gateway uses the Destination Port as programmed in the P2P-Table. But, while receiving the call, the gateway uses the SIP listening port as programmed in the Web page 'Network Port Parameters'.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**Peer-to-Peer Dialing**'. Program maximum 500 entry as required. First entry is for No match found case.
- **Number:** Enter here the number of maximum 24-digits with blank, 0-9, *, #.
- **Destination Address:** Program the Public IP Address of the peer gateway. The address of maximum 40 ASCII characters. ***By default, Blank***.
- **Destination Port:** Program the SIP listening port of the peer gateway. Range is 1024 to 65335 (Blank is not allowed). ***By***

default, 5060.

- **Name:** Enter here maximum 12 characters name of peer Gateway.

Important Points:

- The ‘Name’ field in the Peer-to-Peer Table is used as ‘tag’ only for the entry. There is no use of ‘Name’ parameter in the system for any application.
- User should get the information of Number, Destination Address and Destination Port from networking personal before configuration the P2P table.

Relevant Topics:

1. Network Port Parameters 113
2. SIP Account Parameters 149
3. Call Processing 66
4. Routing Option-Dialed Number Based 139

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Prefix to Domain Name Conversion

What's this?

- The Gateway routes the call to the IP network using the SIP Account number from different Domain (SP) determined by the routing mechanism.
- This Domain Name is identified by some prefix code number.
- The prefix code information is provided by the SE.

For Example:

- Suppose the Gateway is so programmed that all the calls made to abc.com from FXS port are routed through the SIP Account Pulver.com. Now when a user makes a call to a number, say *2349874. Here *234 is the prefix code for the domain, abc.com which is provided by SE and 9874 is the subscriber number.
- Thus by the prefix code dialed, the gateway determines that the called party is a subscriber of abc.com.
- SE provides a table for prefix code to domain name to the user.

The table will look as shown below:

Index	Prefix	Domain Name
01	*234	abc.com
02	Blank	Blank
:	Blank	Blank
64	Blank	Blank

How it works?

- The Table consists of entries for Prefix code and Domain Name. This Table is not checked for making an outgoing call.
- This table will be checked when some FXS port has set Call Forward and only Number is programmed i.e. not with Domain name or IP address or user is doing Blind Transfer.
- The number dialed is compared with the 'number' in table using 'Best Match' found logic.
- If the match found, the digit is stripped off and domain name is added to the number.

For Example:

- User has programmed Call Forwarded number as *1239874.
- In the table, Prefix code is programmed as *123 and Domain Name is programmed as abc.com.
- Now when the number is matched with the entry programmed, number will be replaced as 9874@abc.com. Where the prefix *123 is removed and domain abc.com is added to the number 9874.
- Thus the table is used for converting prefix code to the domain name. The table is not used for making an OG call.
- If best match is not found, refer chapter 'Call Processing'

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on **'Prefix to Domain Name Conversion'.** You can enter following parameters for maximum 64 entries as required.
- **Prefix:** Enter prefix code corresponding to Domain name of maximum 4-digits using blank, 0-9,*, #.
- **Domain Name:** Enter name of maximum 40 characters using Blank and ASCII.

Relevant Topics:

1. Blind Call Transfer 48
2. Call Forward 57
3. Call Processing 66

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Programming the System

- The gateway can be programmed by two methods:
  - Using the Telephone
  - Using Web Jeeves
- Some parameters like Network port IP Address can be programmed by the Phone and by Web Jeeves. Other parameters can be programmed only by Web Jeeves. For this refer chapter 'Web Jeeves' and log-in using the password.
- For programming through phone, refer chapter "Programming using Conventional Phone".
- ***Default Static IP address of Network port is 192.168.001.156.***
- You can not program the MAC Address of the gateway. It is programmed at factory for each gateway.

Relevant Topics:

1. Web Jeeves 171
2. Programming Using Conventional Phone 126

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Programming using Conventional Phone

What's this?

- Following parameters for the Gateway can be programmed using Phone:
  - Static IP address for the Network port
  - Subnet Mask
- For this user it is required to enter the Programming Mode using phone.

How to enter Programming Mode using Phone?

- Get the IP address for the WAN port from the administrator.
- Go OFF-Hook.
- Enter the Programming Mode by: Dial Access Code **'#19'** followed by Password 1234 (default).
- During programming mode, if you go ON-Hook, you will get Continuous ring. This is used to indicate that you are in Programming mode. If you go off hook, you will get programming tone (feature tone).
- If any feature is successfully set/cancel, you will get confirmation tone for 5 seconds followed by programming tone.
- You will also get Error tone if invalid command or invalid value is entered by the SE. The 'error tone' will be for 5 seconds followed by programming tone.
- You can dial programming command during confirmation tone or error tone.
- Enter **'*00#**'*** to get out of programming mode.

How to Program IP Address of Network port?

Use following command to program the Network port IP address:

11-XXX XXX XXX XXX-#*

Where,

XXX XXX XXX XXX is the Network IP address. Network IP Address is of 12 digits maximum

Each octet is of three digits. If only single digit is used then convert to three digits and then enter. For e.g. To enter IP address 192.168.9.10, enter command ***11-192168009010-#*.***

Valid range for each octet is 000-255.

- If Network IP address is valid, you will get confirmation tone for Programming Confirmation Tone Timer (7 secs. fixed).
- After expiry of this timer, the gateway will Restart.
- After restart of the gateway, you will be in normal dial state.
- After entering valid IP address, Connection type will change to Static IP mode.

Use following command to program the Subnet Mask:
12-XXX XXX XXX XXX-#*
Where,
XXX XXX XXX XXX is the Subnet mask of Max.12 digits. Each octet is of three digits.
Range = 000-255
By default = 255.255.255.000.

Example:
To program Subnet Mask address 255.255.1.0, enter the command:
12-255255001000-#*

If the programmed Subnet Mask is a valid, user will get Confirmation tone for the period of Programming Confirmation tone timer (5 sec. fixed). After expiry of this timer, the gateway will restart. At the end of ‘restart’, the user will be in normal dial state.

Display of IP Address on Phone:
Use following command to display the Network IP address on Phone:
21-#*
- You will get Confirmation tone for Programming Confirmation tone timer.
- If you go ON-Hook during this period, than the IP Address is displayed on the Phone instrument and you will get continuous ring to indicate that it is in programming mode.
- But if you go ON-Hook after expiry of Programming Confirmation tone timer, you will not get display of IP address but will get continuous ring to indicate that it is in programming mode.

Use following command to display subnet mask:
22-#*

Use following command to display the Gateway Address:
23-#*

Use following command to display the DNS Address:
24-#*

Important Points:

- If DTMF CLIP type is set for the FXS port, than you will get display of IP address as: 192168001100.
- If FSK CLIP type is set for the FXS port, than you will get display of IP address as: 192.168.001.100, and IP, in the Name field.

Relevant Topic:

1. Programming the System 125
2. Calling Identification and Presentation 83

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Restart Gateway

What's this?

- This feature is just to Switch OFF and ON, the gateway without going to the system and doing physically power OFF and ON.
- By operating this feature, the gateway is not default.

How to restart?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on "**Soft Restart**".
- The gateway will show an Alert message when this link is clicked: "This will Restart VFX88L/VFX44L. Do you want to continue?"
- Click on 'YES' to restart the gateway. Wait to complete the restart process.

Relevant Topic:

1. Default the Configuration 91

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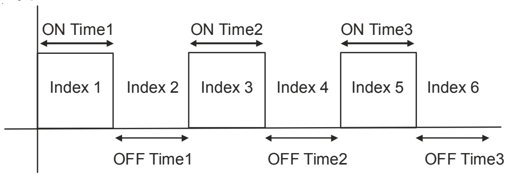
Ring Type

What's this?

- For the indication of Incoming calls, the user will get 'ring' cadences which are not programmable. The system has the cadence-data for 24 countries, pre-programmed.
- When country-name is entered, the specific 'ring' for IC call for that country is played by the system.
- It also supports 'customized' option for programming of required cadence. Refer Table1 for Country name and Ring type at the end of chapter.

How it works?

- The gateway supports two options for selecting the Ring Type:
  - Countrywise
  - Customized
- If 'Country wise' is selected default values of Ring cadence as per the country will be played.
- If 'Customized' option is selected, the Cadence for ring will be programmable. This is required if the user wants to set on his own or if some country is not supported by default.
- Six cadence periods can be programmed for Ring.
- Typical Cadence Index will look like as shown below:



Default table for customized option:

ON Time1 msec.	OFF Time1 msec.	ON Time2 msec.	OFF Time2 mesc.	ON Time3 mesc.	OFFTime3 msec.
400	200	400	2000	0	0

How to program?

- Refer VFX88L/VFX44L Web Jeeves.

- Click on “**Ring Type**”.
- Select option, either Country wise or Customized.
- If Country wise option is selected than you can only select the country and can not program the cadence.
- If Country wise option is selected. Ring Cadence will display the values for the country selected after submitting the Country.
- If Customized option is selected, Country can not be selected and Ring Cadence can be programmed. Default values will be as per the last country selected.
- When you select the Customized option and page is Submitted, Country selected Ring Cadence values are programmable. Range = 0000-9999 msec.
- Click on ‘**System Parameters**’ to program ‘Ring Timer’ to set time period for the ring played by the gateway. Range 01-99. ***Default = 45 seconds***.

Important Points:

- If Country wise option is selected, Ring is played as per the country selected.
- If Customized option is selected, Ring is played as per the cadence programmed.

Code	Country	Frequency (Hz)	CADENCE (In Seconds)			
			TON1	TOFF1	TON2	TOFF2
01	Australia	25	0.4	0.2	0.4	2.0
02	Belgium	25	1.0	3.0		
03	Brazil	25	1.0	4.0		
04	Canada	25	2.0	4.0		
05	China	25	1.0	3.0		
06	Egypt	25	2.0	4.0		
07	France	25	1.5	3.5		
08	Germany	25	3.5	5.5	0.79	1.1
09	Greece	25	1.0	4.0		
10	India	25	0.4	0.2	0.4	2.0
11	Israel	25	2.0	3.0		
12	Italy	25	1.0	4.0		

13	Japan	25	1.0	2.0		
14	Korea	25	1.0	3.0		
15	Malaysia	25	2.0	3.0		
16	New Zealand	25	2.0	3.0		
17	Poland	25	2.0	3.0		
18	Portugal	25	1.0	5.0		
19	Russia	25	1.0	3.0		
20	Singapore	25	0.4	0.2	0.4	2.0
21	South Africa	25	0.4	0.2	0.4	2.0
22	Spain	25	1.5	3.0		
23	Thailand	25	2.0	3.0		
24	UAE	25	2.0	3.0		
25	UK	25	0.4	0.2	0.4	2.0
26	USA	25	2.0	4.0		

Relevant Topic:

1. System Parameters 165

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Routing Options

What's this?

- Routing Option or Routing Type is a method to route the call to the destination port when the call is originated on the source port.
- There can be different options for routing the call to the destination port.
- Each port can be assigned different option for routing the call.

How it works?

- The Routing option is programmable for each port.
- The gateway supports following Routing options when a call is originated on the **FXS Port**:
  - Fixed Routing
  - Dialed Number Based Routing
- Following Routing option are supported when a call is originated on the **SIP Account**:
  - All Calls
  - Called Number Based Routing
- Refer corresponding topic for each routing option for more details.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on "**SIP Account Parameters-2**" for selecting routing option for the SIP Accounts, 1 to 9.
- Click on '**FXS Port Parameters**" for selecting routing option for the FXS port, 1 to 8 (FXS Port 1 to 4, for VFX44L).

Relevant Topics:

1. SIP Account Parameters 149
2. FXS Port Parameters 103
3. Call Processing 66
4. Routing Option-All Calls 134
5. Routing Option-Fixed 142
6. Routing Option-Dialed Number Based 139
7. Routing Option-Called Number Based 135

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Routing Option-All Calls

What's this?

- If this Routing Option is selected, the gateway routes all calls to the Fixed Routing group irrespective of the called number or Calling Number.
- When the call originates on the SIP Account, it routes all calls to the fixed FXS port without answering the call, because in this case, FXS port Routing Group is assigned.
- Allowed/denied logic is not checked for this option.

How it works?

- 'All Calls' Routing Option is applicable for SIP Account, when it is a Source port.
- When the call originates on the SIP Account, the gateway will not give 'dial tone' by answering the call.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on **'SIP Account Parameters-2'**.
- **Routing Type:** By default it is 'All Calls'. You can keep same for all required SIP accounts if call is routed to FXS port as per fixed Routing group.
- **FXS Port Routing Group:** Enter Routing Group number from 01 to 16 in which your required destination port is programmed.

Important Point:

- First program the Routing group and member selection method. Then enter the group number. ***Default = 09***.

Relevant Topics:

1. Routing Options 133
2. Routing Group 143
3. Call Processing 66

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Routing Option-Called Number Based

What's this?

- If this Routing Option is selected, the gateway will route the call on the destination port as per the called party number without answering the call.
- When call is originated on the SIP account, called party number is received by the system and is used for deciding the FXS Port routing group number.

How it works?

- This Routing Option is applicable only for SIP Account, when it is a Source port.
- For this, the gateway is programmed with Called Number Based Routing Table.
- Each entry is programmed with three parameters: Called Number String, FXS Port Routing group, and CLI Number on FXS port.
- While comparing the Called party number in the Called number based Routing table:
  - If the Called party number is not found from the table, the gateway will use the Routing group programmed in the "No Match Found" of the 1st entry of the table.
  - This entry is useful for routing the call when the called Party number doesn't match with any entry in the table. Thus Called Number String for the 1st entry is not programmable.
  - If the Called party number is found in the table, the gateway will use the Routing group programmed for that number.
- Now the parameter "CLI number on the FXS port" is checked before placing the call on the free destination port from the Routing group found.
- "CLI Number on FXS Port" is programmed by 2 options: Received Calling Party or Received Called Party.
- If Received Calling Party option is selected, the call is placed on the free destination port from the Routing group found and CLI number received in SIP Message is sent on FXS port and displayed.
- If Received Called Party option is selected, the call is placed on the free destination port from the Routing group found and called party

number received in SIP message is sent on FXS port and displayed.
- FXS port Routing group is assigned for each index.

The table will look as below:

Index	Called Number String	FXS Port RG	CLI Number on FXS Port
01	No Match Found Note1	01	Received Calling Party
:	:	:	:
99	:	01	Received Calling Party

Note1:
- This entry is useful for routing the call when the called Party number doesn’t match with any entry in the table. Thus Called Number String for the 1st entry is not programmable.

How to program?
- Refer VFX88L/VFX44L Web Jeeves.
- Click on ‘**Called Number Based Routing**’.
- **Called Number String:** Enter maximum 24 characters. With Blank, ASCII. Refer Note1.
- **FXS Routing Group:** First program the Routing Group (01-16) with required port numbers then enter RG number here. ***Default=01.***
- **CLI Number on FXS Port:** Depending on which number is required to be displayed on the extension’s display, enter the option:
  - Received Calling Party Number
  - Received Called Party Number

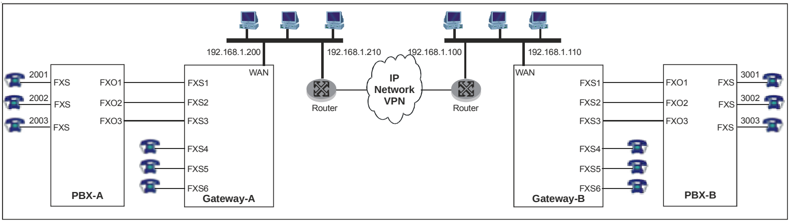
  For 1st entry of no match for ‘Called Number String is found’, this option is only ‘Received Calling Party’.

Important Points:

How to program ‘CLI Number on FXS Port?’:
- This parameter is programmed to get suitable CLI display on the phone.
- Two gateways are connected through VPN or public IP address and thus Peer-to-Peer call is made between this two PBX.

- When PBX-A user 2001 dials 3001 after grabbing the trunk (i.e. which is connected to the FXS port of the Gateway), the call will be routed directly to the 3001 user of PBX-B and 2001 will get RBT.

Calling to Extension Number:



- To get the Display on Extension number of the calling party, this parameter is programmed by the called party user, from two options i.e. Received Calling Party and Received Called Party.
- Following values will be displayed on the LCD of the Phone connected to the FXS port as per the CLIP type (DTMF or V.23 or Bellcore) set on the FXS port and whether Received Calling party or Received Called party is to be used. Date and Time will be displayed as per the system settings of the VFX88L

When Received Calling Party option is selected:

All Parameters in the SIP message (FROM)	**DTMF type**	**FSK (V.23 or Bellcore) type**
<Display Name> SIP ID @ IP address or Domain Name	'SIP ID'	'SIP ID' and 'Display Name'
SIP ID@IP address or Domain Name	'SIP ID'	'SIP ID' and 'SIP ID @IP address' or 'Domain Name'
IP address	'IP Address'	IP Address

When Received Called Party option is selected:

All Parameters in the SIP message (Request URL)	DTMF type	FSK (V.23 or Bellcore) type
SIP ID @ IP address or Domain Name	‘SIP ID’	‘SIP ID’ and ‘SIP ID @ IP address’ or ‘Domain Name’
IP address	‘Blank’ **(Note A)**	‘Blank’ and ‘IP Address’ **(Note A)**

Note A:

- Number field is displayed ‘Blank’ because while routing the call, system will check only SIP ID. But for this case there is no SIP ID.

Relevant Topics:

1. Routing Options 133
2. Routing Group 143
3. Call Processing 66
4. Calling Line Identification and Presentation 83

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Routing Option-Dialed Number Based

What's this?

- If this option is selected, call is routed to the SIP Account number on the basis of the number dialed by the user.
- The call is routed using the SIP account of the routing group, as per this 'dialed number'.
- You can program the SIP Account numbers in the Routing Group such that for some numbers it uses cheapest SIP trunk whereas for some other numbers it uses another routing group with little costlier SIP Trunk. Thus this feature can be used as Least Cost Routing (LCR), application.

How it works?

- Dialed Number Based Routing will be applicable for each FXS port.
- You will get 'dial tone' when you go OFF-Hook.
- The Dialed Number String Table is prepared with two parameters as shown in table1.
  - Dialed Number String
  - SIP Account Routing Group
- If Routing option is selected as Dialed Number Based, the dialed number is compared with the Dialed Number based Routing table and Best match is found.
- If the Dialed number is matched with the Number programmed in the Dialed Number Based Routing table, the SIP Account Routing group programmed for that number is used, by the gateway for call routing.
- If the Dialed number is not found in the Dialed Number Based Routing table, the SIP Account Routing group programmed for the No Match Found i.e. 1st entry in the table is used for call routing.
- The number is dialed using Store and Forward dialing method.
- First digit wait timer, Inter digit wait timer, End of Dialing digit is applicable, while dialing the number by the gateway.
- Emergency Number table and Allowed and Denied number feature are also checked.

Table 1 : Dialed Number String Table.

Index	Dialed Number String	SIP Account Routing Group
01	No Match Found (Note1)	1
:	:	:
99		1

Note1:

- This entry will be useful for routing the call when the Dialed number doesn't matches with any entry in the table. Thus Dialed Number String for the 1st entry is non-editable.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**Dialed Number Based Routing Table**'.
- **Dialed Number String:** Enter maximum 24 characters. With Blank, ASCII. Refer Note1.
- **SIP Account Routing Group:** First program the RG with required SIP account numbers then enter number here. Range = 1-9, ***Default =1.***

For example:

- This example describes the use of Dialed Number String table and Automatic Number Translation as Phone book or speed dialing:
- Suppose you want to dial 09825065266, *712123 and *712456 from the gateway using routing groups 1 and 2, then program the numbers for ANT, as mentioned below:

Automatic Number Translation list for SIP Account 1:

Index	No. String of Dialed No. List	No. String of Substitute No. List
1	1	9825065266

Automatic Number Translation list for SIP Account 2:

Index	No. String of Dialed No. List	No. String of Substitute No. List
1	801	*712123
2	802	*712456

Now, Program the Dialed number string table as given below:

Index	No. String of Dialed No. String	SIP Account Routing Group
1	1	1
2	8	2

- In SIP Account Routing Group 1 only Non-proxy trunk is programmed i.e. SIP Account 1 which is used for making Peer-to-Peer call.
- In SIP Account Routing Group 2 only Proxy trunk which is registered to Pulver.com is programmed i.e. SIP Account 2. Assign these groups to the FXS port from which numbers are required to be dialed.
- Thus, just dial 1, for dialing ‘9825065266’ and dial ‘801’ for dialing ‘*712123’.
- This is also called speed dialing or abbreviated dialing using SIP Account number.
- If Speed Dialing feature is to be used for making Peer-to-Peer call, program the number which is used for speed dialing in the ‘Destination Address’ of Peer-to-Peer Table. For example, program ‘1’ for dialing on ‘SIP Account 1’.

Relevant Topics:

1. Routing Options 133
2. Routing Group 143
3. Call Processing 66
4. Allowed-Denied Numbers 32
5. Peer-to-Peer Calling 120

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Routing Option-Fixed

What's this?

- If this routing option is used, then call will be routed irrespective of the number that is dialed.
- For example, if you dial from the FXS1 port of the gateway, it will be dialed out using fixed SIP Account 1 only programmed in the routing group. This Routing group number is used which is assigned to the FXS1 port.

How it works?

- Fixed Routing Option is applicable only for FXS port type used as source port.
- You will get 'dial tone' when you go OFF-hook.
- If Routing option is selected as Fixed Routing, the gateway will use the SIP Account Routing group programmed for that FXS Port in 'FXS Port Parameters' for routing the number.
- The number will be dialed using Store and Forward dialing method.
- First digit wait timer, Inter digit wait timer, End of Dialing digit and Allowed and Denied number feature are applicable, while dialing the number by the gateway.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on '**FXS Port Parameters**'.
- Routing Type: Select 'Fixed'.
- SIP Account Routing group: Enter here the routing group number which contains the required SIP account number for routing the dialed number. Default : 9.
- Program above parameters for required FXS ports from 1-8 (1-4 for VFX44L).

Relevant Topics:

1. Routing Options 133
2. Routing Group 143
3. Call Processing 66
4. Allowed-Denied Numbers 32

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Routing Group

What's this?

It is the group of different destination ports programmed.

- When the call originates on the source port, the call will be routed to the destination port programmed in the specific routing group for the source port.
- The gateway supports two types of routing groups:
  - FXS Port Routing Group
  - SIP Account Routing Group
- The gateway supports maximum 16-Routing Groups for FXS Port RG and 9 Routing Groups for SIP Account RG and 8-members can be programmed in each group.

How to use it?

- Use Web Jeeves for programming the members in the group.
- For FXS Port Routing Group program only FXS Port in the Member list.
- For SIP Account Routing Group program only SIP Account number in the Member list.
- If no port has to be assigned for that Member, Program 'Null' option.
- Enter FXS port routing group on web pages 'Called Number Based Routing Table' and 'SIP Account Parameters' for All Calls routing option.
- Enter SIP Account Routing group on web pages 'Dialed Number Based Routing Table' and 'FXS Port Parameters' for All Calls routing option.
- Assign the Member Selection method to select the Destination port (Member) from the Routing group refer chapter 'member selection method'.

How it works?

- When call originates on the FXS port, SIP Account Routing group is selected according to the Routing option programmed for that port and when call originates on the SIP Account, FXS Port Routing group is selected according to the Routing option programmed for that Account.

- While finding free destination port, following conditions can occur:
  - All members are disable.
  - All members are programmed as Null.
  - Some member are busy and some member are disable or programmed Null.
  - Free member found.
- If all members in the routing group are disable or if all members are ‘Null’, you will get error tone for 7-seconds and the call will be disconnected.
- For the ‘busy’ member you will get busy tone for fixed 7 seconds, and than error tone for fixed 7 seconds. After period of error tone timer expires, call is disconnected.
- If any free member is found, call is placed on the destination port.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on “**SIP Account Routing Group**” for programming 8-SIP account numbers as member1.......member8.
- **Member Selection Method:** Select from ‘first free’ or ‘rotation’ as required. Refer related chapter. Program other groups (1-9).
- Click on **‘FXS Port Routing Group’** for programming 8-FXS Port numbers as member1.......member8 (1-4 FXS Ports for VFX44L).
- **Member Selection Method:** Select from ‘first free’ or ‘rotation’ as required. Refer related chapter. Program other groups (01-16). Refer corresponding chapters.

Default (VFX88L) FXS Port Routing Group is as shown below:

Routing Group	Member Selection Method	Member 1	Member 2	Member 3	Member 4	Member 5	Member 6	Member 7	Member 8
01	Rotation	1	2	3	4	5	6	7	8
02	First Free	1	NULL	NULL	NULL	NULL	NULL	NULL	NULL
03	First Free	2	NULL	NULL	NULL	NULL	NULL	NULL	NULL
04	First Free	3	NULL	NULL	NULL	NULL	NULL	NULL	NULL
05	First Free	4	NULL	NULL	NULL	NULL	NULL	NULL	NULL
06	First Free	5	NULL	NULL	NULL	NULL	NULL	NULL	NULL
07	First Free	6	NULL	NULL	NULL	NULL	NULL	NULL	NULL
08	First Free	7	NULL	NULL	NULL	NULL	NULL	NULL	NULL
09	First Free	8	NULL	NULL	NULL	NULL	NULL	NULL	NULL
10	Rotation	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL
:	:	:	:	:	:	:	:	:	:
16	Rotation	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL

Default (VFX44L) FXS Port Routing Group is as shown below:

Routing Group	Member Selection Method	Member 1	Member 2	Member 3	Member 4	Member 5	Member 6	Member 7	Member 8
01	Rotation	1	2	3	4	NULL	NULL	NULL	NULL
02	First Free	1	NULL	NULL	NULL	NULL	NULL	NULL	NULL
03	First Free	2	NULL	NULL	NULL	NULL	NULL	NULL	NULL
04	First Free	3	NULL	NULL	NULL	NULL	NULL	NULL	NULL
05	First Free	4	NULL	NULL	NULL	NULL	NULL	NULL	NULL
06	First Free	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL
07	First Free	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL
08	First Free	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL
09	First Free	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL
10	Rotation	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL
:	:	:	:	:	:	:	:	:	:
16	Rotation	NULL	NULL	NULL	NULL	NULL	NULL	NULL	NULL

Relevant Topics:

1. SIP Account Parameters 149
2. FXS Port Parameters 103
3. Routing Option-Dialed Number Based 139
4. Routing Option-Called Number Based 135
5. Member Selection Method 111

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RTC Parameters

What's this?

- The gateway supports various features (like CDR etc.) which use current date and time parameters. Such facilities work properly only if the Gateway is set with correct date and time value, using Real Time Clock (RTC) feature.
- The gateway also uses SNTP (Simple network Time Protocol) for network time. It ensures accurate local time keeping with reference to radio and atomic clocks located on the internet.

How it works?

- The Gateway has its own RTC. Thus, RTC parameters can be programmed through Web Jeeves.
- It also uses the Network time to update the RTC. For this you can program the Time Server.
- The system makes use of SNTP (Simple network Time Protocol) to get time from the Time Server.
- The gateway supports following three Time Servers:
  - Ntp1.cs.wisc.edu
  - Time.windows.com
  - Time.nist.gov
- User can select any one option among three to obtain the date and time or may enter his own choice.

GMT Time (Greenwich Mean Time):

- These servers give GMT times. Hence time has to be adjusted as per the time zone selected by the user (depending the country you are working) and use the time for displaying and storing the date, time and duration of each call.
- The VFX88L/VFX44L gets synchronized with the specified NTP server (IP Address), which is a global server and gets the date and global time. Based on the time zone, settings, Matrix VFX88L/VFX44L shows correct time for that country.

For example:

- If time shown by ntp1.cs.wisc.edu server for India (Culcutta, Chennai, Mumbai, New Delhi) is 10:44:50AM, then current time shown for Kabul is 09:44:50AM. Because for India GMT is ‘GMT +5.30’ but for Kabul it is ‘GMT +4.30’ (1 hour behind India). The date is displayed as 17May2006.

- All the countries for “Time Zone” are supported same as supported by Windows software of PC.
- Clock adjust logic is used as per the time zone to display the time.
- Calcutta, Mumbai, Chennai and New Delhi is set as default.
- Gateway RTC can be set to sync with the Time Server time i.e. Time received from time server will be updated to the RTC.
- Thus, for Auto RTC Sync with NTP, following options are provided in the Web page:
  - No
  - Daily
  - Weekly
  - Monthly
- Also the option ‘Yes/No’ to sync Gateway RTC with Time Server at Power-ON is provided. This parameter is also applicable when the system restarts due to any condition. For example, when default command is given from Web Jeeves.
- RTC parameters, Time Server and Time Zone can be programmed from Web Jeeves only.

How to program?

Refer Web Jeeves

Click on **“Date and Time Settings”**

- **Current Date:**
  DD-Month-YYYY: Set date as per this format DD=01-31, Month=January-December and YYYY = 2007-2099.
- **Current Time:**
  HH-MM-SS: Set current time as per his format. HH=00-23, MM=00-59, SS=00-59.
- **Day:** It is displayed but cannot be edited (Sunday to Saturday).
- **NTP Address:** Select the required option, or select ‘Blank’ to program the time server address of your choice, with maximum 40 ASCII characters.
  By default, Ntp1.cs.wisc.edu.
- **Time Zone:** Select ‘country’ where the gateway is installed. The list of countries is same as supported by ‘Windows’.

Auto RTC Sync with NTP

- Select suitable option from following:
  - No (RTC will not change with Server).
  - Daily (RTC will change daily at 00:00 Hr.)
  - Weekly (RTC will change at every Saturday night i.e. 00:00 Hr. of Sunday)
  - Monthly (Change at every 1st day of the Month at 00:00 Hr. i.e. when month change)

Note:

- If the option is selected other than ‘No’, user can check the ‘Time’ by refreshing this page after submitting.

RTC Sync with NTP on Power ON?

- Yes
- No
- ***By default, No.***

Relevant Topics:

1. Daylight Saving Time 88
2. Default the Configuration 91

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SIP Account Parameters

What's this?

- The gateway supports maximum of 9-SIP account numbers subscribed from the same or different Service providers. You can program basic SIP registration parameters as well as advanced parameters like Outbound Proxy, STUN and OG Vocoder parameters.
- This chapter describes programming of these parameters in two web pages: SIP Account Parameters-1 and SIP Account Parameters-2
- If SIP account is already registered and then it is disabled, then the SIP account will get unregistered.
- Refer Glossary and chapter VoIP Basics for more details.
- Refer Important Notes at the end of chapter regarding Non proxy server, SIP ID, OG calls and IC calls etc. Also refer chapter 'Call Processing'.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on 'SIP Account Parameters-1' or 'SIP Account Parameters-2'.
- Program following parameters for (1-9) SIP Accounts, as required.
- For this click on 'SIP Account Parameteres1'.

Status:

- Select enable if you want to use SIP Account for call routing as programmed.
- Select 'disable' if you do not want to use the SIP Account for routing the calls. Disabling SIP Account is required in cases like not registered, problem on the service provider side, etc.
- ***Default = Disable.*** For example, if SIP1 is Not registered with he Server due to some reason, disable it.
- Refer **note 1** and **4** of the end of chapter.

Name:

- It is used just for information only. Enter the service provider name or person name that is using this SIP Account.
- Enter the string of maximum 12-alpha numeric characters.
- ***Default = Blank.***

SIP ID:

- It is the user part of the full SIP URI. This can be a number or text. Maximum 24 characters including all ASCII characters. Blank is not supported. ***By default, '*' for each SIP Account.***

For example:

- A full SIP URL provided by the ITSP is 12345@abc.com than program 12345 in this field.
- When a caller calls this number, the call lands on the Network port and is routed as per the Call Routing Option assigned to the SIP Account.
- When an OG call is made through a SIP Account with SIP ID; SIP:12345@abc.com, SIP:12345@abc.com and the FXS port name are also sent in the SIP message. This helps the callee to identify the caller. This is because many ports may use the same SIP Account to make an OG call.
- Refer **Note 3** and **4** at the end of chapter.

Registrar Servers' Address:

- Enter here the address of the SIP Registrar. It can be IP address also.
- Maximum 40 characters including extended ASCII characters.
- ***Default = Blank.***
- Refer **Note 2** at the end of chapter.

Registrar Servers' Port:

- Enter the Registrar Server's listening port number for SIP. SE may change this field if the ITSP provides a SIP port number other then the default.
- This may be same as SIP Server's Port address.
- Range = 1024-65535.
- ***Default = 5060.***

Re-Registration Timer:

- As a part of normal process, the Register server deletes an entry of its client from its database on expiry of a fixed timer.
- Thus, this timer sets period for how long an entry remains registered with the Register Server.
- This timer is used as Expire timer when message is sent for

registration to the Register server.

- Register server can send a time period which is less than or equal to the specified expire timer in the registration message.
- The gateway will use Expiration timer which is send by the Register server as Re-registration timer.
- If Register server does not send any expiration timer, than this timer will be used.
- Hence, in order to be registered always, the Gateway will send a Registration request before the timer expires.
- This timer also signifies the time after which the Gateway will send Registration request again to be registered
- To be on safe side as a rule, the Re-registration request is sent when half of the configured time period or expiry time sent by the register server has expired.
- Range = 001 to 999 minutes.
- ***Default* = Re-Registration timer is 60 minutes.**

Registration Retry Timer:

- Enter the period between retries for registration.
- On expiry of Re-registration timer or failure of registration or registering for the first time, the gateway sends the registration request.
- If the registration attempt fails, this timer will be started.
- The gateway will send the registration request again on expiry of this timer. The gateway will keep sending the registration request till it gets registered and once it gets registered the gateway will start Re-registration timer.
- Range = 00001 to 65535 seconds.
- **Default = 10 seconds.**

Authentication User ID:

- It is the user name for registering the SIP account with the SIP register server. This is used for encryption. It is provided by ITSP.
- Enter maximum 40 characters including all ASCII characters.
- ***Default, Authentication User ID will be blank.***

Authentication User Password:

- It is the password associated with the user name above.
- Enter maximum 16 characters including all ASCII characters.

- ***Default, Authentication User Password will be blank.***

Outbound Proxy Server Status:
- Enable this field if the ITSP service provider has a SIP outbound server to handle calls.
- ***Default, Outbound Proxy Server is disabled.***

Outbound Proxy Server's Address:
- It is the address of the Outbound Proxy Server. It can be IP address also. This may be same as the SIP Server address since the SIP server might also be acting as Outbound Server.
- Enter maximum 48-characters including extended ASCII characters.
- ***Default = Blank.***

Outbound Proxy Server's Port:
- It specifies the Outbound Proxy Server's listening port number for SIP. SE may change this field if the ITSP provides an Outbound Proxy Server port number other than the default.
- Range = 1024-65535. This may be same as SIP Server's Port.
- ***Default = 5060.***

Use STUN:
- Use STUN if ITSP doesnot provide Outbound Proxy.
- If STUN is used, configure the STUN Server Address and STUN Server Port in Network Port Parameters.
- Select: Yes/No.
- ***Default: No.***

OG Preferred Vocoders:
- To reduce the packet band width occupied in the RTP for the speech, the gateway supports following types of Vocoders and it is default sequence:
  - G.729
  - iLBC
  - G.723
  - GSM
  - G.711 (μ-Law)
  - G.711 (A-Law)

- The Vocoders for the OG calls will be selected as per the preferences programmed.

Note:

- In case of IC call, this parameter will not be checked. Rather, the Vocoder requested by the remote end will be entertained if supported by the Gateway, else the call will be rejected.

DTMF Option:

- Select the DTMF Option from:
  - RTP (RFC 2833)
  - SIP INFO
  - In-Band
- This option is used for both incoming and outgoing calls.
- In case of OG call,
  - The Gateway will request the DTMF option selected to the remote end.
  - If in response from remote end, same DTMF option requested is sent, DTMF option selected will be used for DTMF dialing.
  - If in response from remote end, same DTMF option requested is not sent, In-Band DTMF option will be used by default.
  - For Example: If DTMF option ‘SIP INFO’ is selected and when OG call is made, SIP INFO is requested to the remote end the SIP message for DTMF dialing. Now if remote end responds with ‘SIP INFO’, it will be used for sending and receiving DTMF digits. If remote end responds with ‘RFC2833’ DTMF option, the Gateway will use ‘In-Band’ for sending and receiving DTMF digits.
- In case of IC call,
  - The Gateway will check the DTMF option requested by the remote end, with the DTMF option selected.
  - If DTMF option requested by the remote end is same as selected DTMF option, it will detect the DTMF digits in DTMF option selected.
  - If DTMF option requested by the remote end is not same as selected DTMF option, it will detect the DTMF digits in ‘In-Band’ only.
  - For Example: If DTMF option SIP INFO is selected and when IC call is received, if ‘SIP INFO’ DTMF option is requested by the remote end, then the Gateway will reply with ‘SIP INFO’ and

detect the digits received in ‘SIP INFO’ and also send the DTMF digits in SIP INFO. If request is received with ‘RFC2833’ DTMF option, then the system will not reply with ‘RFC2833’ option and will use the DTMF option ‘In-Band’ for sending and receiving the DTMF digits.

- ***Default*, DTMF Options = RTP (RFC 2833)**

FAX Option:

- FAX options are:
  - T.38 (UDPTL)
  - T.38 (RTP)
  - Pass-Through
- The option selected here signifies how FAX messages will be handled by the Gateway.
- Select Pass-through option if you need to send fax message over G.711. The peer device will also use G.711.
- Use T.38 if you need to send fax messages over IP as UDP or TCP/IP packets. The peer device should also preferably support. If Pass through option is selected, the user should use G.711 as Preferred coder for better results.
- ***Default: T.38 (UDPTL).***

Click on link ‘**SIP Account Parameters2**’ to program following parameters:

- When Routing Type is selected as Called Number Based, ‘FXS Port Routing Group’ and ‘CLI Number on FXS Port’ can not be programmed for that Port.
- If Check Box ‘Apply’ is disabled, than Number List field for ‘ANT’ and ‘Black Listed Callers Number’ can not be edited.
- Refer related chapter for programming following parameters:
  - Black Listed Callers-refer chapter ‘Black Listed Callers’.
    Default = ‘Enable’
    Default Number list=03 and range = 00-24
  - Automatic Number Translation-refer chapter ‘Automatic Number Translation’.
    Defaults = Enable,
    Dialed number list = 04, Substitute number list = 05
  - Routing Type-refer chapter ‘Routing Option-All Calls’ and ‘Routing Option-Called Number Based’.

Default = All Calls.
- FXS Port Routing Group-refer chapter 'Routing Group'
  Default = 01 and Range = 01-16
- CLI Number on FXS Port-refer chapter 'Routing Option-Called Number Based'.
  Select from: Received Called Party and Received Called Party.
  Default = Received Calling Party

Incoming Calls Allowed:
- This parameter is required if the SE doesn't want to allow any incoming call through any SIP account.
- When Enabled, Incoming calls will be routed as per the Routing type programmed for that SIP account.
- When Disabled, the incoming call on this SIP account will be rejected.
- ***Default: Enable.***

Send CLI:
- Enable or Disable, to send or Block the CLI.
- When Send CLI is enabled, both the Calling Party number (SIP Account number through which the call is being made) and the Name (Station Name) will be sent.
- When Send CLI is disabled, neither will be sent.
- ***Default = Enabled.***

Answer Anonymous Calls:
- When Enabled, Anonymous calls (calls without Calling Party information) this type of calls will be accepted and routed by the Gateway.
- When Disabled, Anonymous calls (calls without Calling Party information) this type of calls will be rejected.
- ***Default = Enabled.***

Maximum calls:
- This parameter is useful to route the call to the next SIP account if number of calls to particular SIP account is maximum number of calls supported by the service provider.
- It is applicable to each SIP Account for both incoming and outgoing call.

- For I/C call on SIP account: If Number of calls (incoming or outgoing) is less than programmed value, the call will be routed to the FXS port as per the routing option programmed for this SIP account. But if it is equal to the programmed value, the call will be rejected. Refer chapter 'Call Processing'.
- For OG call on SIP account:
  - When any free member is found from SIP account routing group, the system will check whether number of calls made through this free member found (SIP account) is equal to Maximum Simultaneous Outgoing call programmed for this free member found (SIP Account).
  - If Number of calls (incoming or outgoing) is less than programmed value, the call will be routed through this free member (SIP account).
  - But if Number of calls (incoming or outgoing) is equal to programmed value, the system will find another free member from the SIP account routing group.
- This parameter is not applicable for transfer call or call forward call.

  Range = 8 (4 for VFX44L)

  Default = 8 (4 for VFX44L)

Note 1:

- When Network port link is failed, all SIP Accounts are considered as disable.

Note 2:

- How to consider Peer-to-Peer Call (**Non-Proxy**):
  - If Registrar Server Address is programmed for any SIP Account, that SIP Account is considered as Proxy.
  - If Registrar Server Address is Blank, that SIP account is considered as the Non-Proxy.

Note 3:

- SIP ID parameter programming:
  - ***By default this parameter field is programmed as '*' for all SIP Accounts.***
  - SE should not program the same SIP ID for different SIP account otherwise the first exact match will be considered for every call

coming with that SIP ID even though the service provider is different for every call. For example:

- In SIP Account parameter table 123@abc.com has been programmed for SIP Account 1 and 123@pulver.com has been programmed for SIP Account 2.
- Now call has come for 123@pulver.com.
- Gateway will check the SIP ID 123 in the SIP Account parameter and as it has found the 123 for SIP Account 1 as first matched, SIP Account 1 parameter will be used entry, for routing the call even though the call has been made for SIP Account 2.
- Thus, the SE must not use the same SIP ID for different Registrar.

Note 4:

- SE will enable the 'Status' of only one SIP Account with SIP ID '*' if it is required to be used as Non-Proxy i.e. for Peer-to-Peer calling because even if more SIP Accounts, are enabled the first SIP account with SIP ID programmed with '*' will always be considered for routing the incoming call.

Relevant Topics:

1. Peer-to-Peer Calling 120
2. Getting Started 19
3. IP Dialing 108
4. Call Processing 66
5. Routing Group 143

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Software Version/Revision Display

What's this?

- This feature helps user to know the current operating Version and Revision of the system without opening the system.
- It is very useful for System Engineer (SE), to know about updation of latest features in the system.
- The Software version revision can be known from Web Jeeves page.

How to restart?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on "CDR" or any other parameters.
- Software version revision is displayed on each Web page at right hand side bottom corner.
- The display format is: **VxRy** (x is the version number and y is the revision number). For example, V1R2.

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Status

What's this?

- The gateway user can quickly refer the Network Status and SIP Account Status for each SIP Account on a web page, called 'Status' page.
- If required user can change the parameter from the respective Web Page. User can also get the probable reason for any failure of registration.

Following basic parameters are displayed on the 'Status' page.

- Refer VFX88L/VFX44L Web Jeeves Pages.
- Click on **'Status'.**

SIP Account Status: (This is displayed for all 9-SIP Account numbers which are programmed in the system).

- ***Registration Status:***
  - After gateway is powered ON, the system sends registration request to the server and suitable messages are displayed. Maximum 24 characters. e.g. Disable, Registered, Registering Authentication Fail, Unregistering, Message Sent Fail.
- ***Registration Time:***
  - This time is provided by server during registration. Maximum 4 digits.
- ***Registration Retry Count:***
  - This shows how many times after registration failure, Gateway has sent. Maximum 4 digits.

Network Status:

- **IP Address:** This displays the current system IP Address. It can be Static IP Address or DHCP or PPPoE
- **Subnet Mask:** This displays the system subnet mask.
  Default = 255.255.255.255.
- **MAC Address:** This displays unique MAC address for the gateway.
- **Gateway IP Address:** This displays parameter provided by the server.
- **DNS Address:** This display system gateway IP Address.
- **NAT Type:** If STUN is not enabled, it is blank. If None Type is

detected by the gateway, the option is displayed as ‘Unknown’. Maximum 20 characters. If STUN is enabled, then system will check the NAT type and according to the NAT Type, it will display the options as mentioned.

- OPEN
- CONE NAT
- RESTRICTED NAT
- SYMMETRIC NAT
- SYMMETRIC FIREWALL
- BLOCKED

- **Public IP:** When STUN is enabled and during NAT Type checking, if system, is able to get public IP Address then it will be displayed.
- **Mapped SIP Port:** This parameter displays the SIP port number which is used by the system when connected behind NAT. Maximum 5 characters.
- **Stack Status:** It displays status of ‘Stack’ like: Idle, DHCP Response Wait, PPPoE Response Wait, NAT Checking Response Wait, Construct, Error, etc.

Relevant Topics:

1. Network Port Parameters 113
2. SIP Account Parameters 149

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Supplementary Services

What's this?

- The gateway supports 8-FXS ports (4 FXS Ports for VFX44L) and 9-SIP accounts. The FXS port can also be interfaced with PBX. Hence the supplementary services supported by the gateway will be useful for users.
- The system supports following Supplementary Services Features:
  - Call Waiting
  - Call Forward
  - Hotline
  - Do Not Disturb
- You can set/cancel this service as per your requirement. Refer corresponding chapter for configuring the web page.
- For Call Forward-(Unconditional, Busy and No Reply), user can set or cancel any Call Forward type, using Access Code also.
- You can program the Call forward number and Hotline number through Web Jeeves only.

How to program?

- Refer VFX88L/VFX44L Web Jeeves.
- Click on "**Supplementary Services**".
- If any feature is 'Disable', then related parameter of this feature cannot be edited. For example, if 'Call Forward-Busy', is disabled the 'Number' cannot be changed.

Following parameters can be programmed through this page:

- Hotline-Status, Hotline Number and Hotline Timer.
- Do Not Disturb(DND)-Status.
- Call Waiting-Status.
- Call Forward-Unconditional-Status and Number.
- Call Forward-Busy-Status and Number.
- Call Forward-No Reply-Status, Number and No Reply Timer.
- Status: Select Enable/Disable, as required.
- Number: Enter here the number string of maximum 24 digits (0-9, *, #, @).
- Hotline Timer: Enter the time period after which the system will dial out the Hot line-number after going OFF-Hook from (1-9) seconds.

- No Reply Timer: Enter this timer up to which the system will play the ‘ring’ for IC call. After this time call will be forwarded to the ‘call forward number’.
  Range = 01-99 seconds.

Important Point:

- If any feature is disabled in Class of Service and SE tries to enable the status of that feature through Web Jeeves, system will not allow to change the ‘Status’ and will pop-up a suitable “Error message” on Web Jeeves for rejecting the change.

Relevant Topics:

1. Class of Service 86
2. Call Forward 57
3. Do Not Disturb 95
4. FXS Port Parameters 103
5. Hotline 105
6. Access Codes 29

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System Debug

What's this?

- System Engineer can view a log of debug for different parameters on the Terminal of Remote Server.
- For, the specific Card supplied by Matrix using serial port it is required to be used. For remove server, IP address of Remote Server will be programmed.
- The debug is supported for following DEBUG levels:
  - APPL
  - CONFIG
  - CALL
  - VOPP
  - CHNL
  - STACK
  - NAT
  - STUN
  - SIP
  - SNTP
  - REG
  - SYS
- As per the level selected, debug log will be generated. For example: if debug log of Call is required, user will enable Call level and disable all other debug levels. ***By default, all debug levels are enabled.***

How it works?

- SE can Enable/Disable the feature. By Default, it is disabled.
- If it is disabled, system will not send any debug data to the Remote Server IP address if programmed, or on serial port.
- If it is enabled, system will send the debug as per the Debug level enabled to the Remote Server IP address and Server Port of Remote Server Terminal programmed. Debug can be viewed on the Remote server terminal.

How to program?

Refer VFX88L/VFX44L Web Jeeves.

Click on **‘Debug’** and program following parameters:

- **Debug enable:** Click on the box to enable or disable the feature.
- **Remote Server Address:** If IP Address is ‘Blank’ and debug is enabled , then system will send debug data on serial port. Program the IP Address of maximum 15 characters. ***Default = Blank***. Range is 000 to 255.
- **Remote Server Port:** Program the port number from 1024-65535, Blank is not supported. ***Default = 50000.***
- **Debug Levels:** Enable the parameter for which debug-log is required. The levels are as mentioned in this chapter.
  By default, all are enabled.

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System Parameters

What's this?
The Gateway supports some important parameters like System name, Transfer Notification Timer etc. These parameters can be programmed through specific Web page.

How to program?
Refer VFX88L/VFX44L Web Jeeves.
Click on **'System Parameters'.**

Program following Parameters:
- **System Name:** Maximum 24 characters including all ASCII characters can be programmed. ***Default = Blank***.
- **Ring Timer:** Refer chapter 'Ring Type'
- **Call Hold Alert Timer:** Refer chapter 'Call Hold'
- **Call Hold Reject Timer:** Refer chapter 'Call Hold'
- **Transfer Notification Timer:** Refer chapter 'Blind Call Transfer'
- **Remote Held Reject Timer:** Refer chapter 'Call Hold'

Relevant Topics:
1. Blind Call Transfer 48
2. Call Hold 60
3. Ring Type 130

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VoIP Basics

VoIP is the sending of voice signal over Internet Protocol. Calling on VoIP allows user to make phone calls and send faxes over the internet at a negligible cost compared to the normal circuit switched Telephone network. The main components of VoIP are:

- SIP
- Network Servers
- RTP
- NAT
- STUN
- Outbound Proxy
- Important Voice Parameters: Jitter Buffer, Voice Compression and QoS.

SIP:

- The signaling protocol which is used to establish an IP call is called SIP-Session Initiation Protocol. It is an application layer protocol that handles the setting up, altering and tearing down of voice and multimedia sessions over the internet. Refer RFC 3261 for more information.
- SIP handles telephone calls and can interfere with PSTN network.
- A SIP account uses complete identity called, URI (Uniform Resource Identifier) or SIP Address, like e-mail address and is given by: SIP-Number@SIP service Domain.
- SIP number can use letters or numbers like vrajeshp@ITSP Provider.com or 73121@ITSP Provider.com. A SIP service domain is the domain name in the SIP URL. For example, if SIP address is 73121@VoIP Provider.com, then ‘VoIP Provider.com’ is the SIP service domain.
- In general, SIP follows the Client-Server Architecture. To support this, there are two main entities-user agent and network servers. The peers in a session are called User Agents (UAS). A user agent can function in one of the following roles:
  - User agent client (UAC)-A client application that initiates the SIP request.
  - User agent server (UAS)-A server application that contacts the user when a SIP request is received and that returns a response subject to user’s input.

Network Servers:
There are three types of **network servers** in a SIP network: proxy server, redirect server and registrar server.

Proxy Server: A SIP proxy receives a request, makes a determination about the next server to send it to, and forwards the request, possibly after modifying some of the header fields. As such, SIP requests can traverse many servers on their way from UAC to UAS. Responses to a request always travel along the same set of servers the request followed, but in reverse order.

Redirect Server: The redirect server does not forward requests to the next server. Instead it sends a redirect response back to the client containing the address of the next server to contact.

SIP Registrar: SIP Registrar is an entity where SIP users can get themselves registered. Registrar imparts mobility to the SIP users. A SIP user can register himself with a registrar. If the user changes his location, he has to register again with the registrar stating his latest contact information. Whenever a call is to be delivered to that user, Registrar can provide the information about the location where the user was active recently.

RTP:
After a VoIP call is established using SIP, the RTP (Real Time Protocol) is used to handle voice data transfer in the packets form. Refer RFC 1889 for more details.

NAT
NAT (Network Address Translation-RFC 1631), is the translation of IP address of a host in a packet. For example, the source address of an outgoing packet, used within one network is changed to a different IP address known within another network.

- One network is designated the inside network and the other is the outside. Typically, a company maps its local inside network addresses to one or more global outside IP addresses and unmaps the global IP addresses on incoming packets back into local IP addresses. This helps ensure security since each outgoing or

incoming request must go through a translation process that also offers the opportunity to qualify or authenticate the request or match it to a previous request.
- NAT also conserves on the number of global IP addresses that a company needs and it lets the company use a single IP address in its communication with the world.

STUN: (RFC 3489)

Note:

STUN is used if there is a NAT router between VFX88L/VFX44L and SIP Server (from ITSP).

- STUN stands for Simple Traversal of UDP over NAT. It is a protocol, which enables the gateway to detect the presence and type of NAT behind which the gateway is placed. The gateway that supports STUN can intelligently modify the private IP address and port in its SIP/SDP message by using the NAT mapped public IP address and port through a series of STUN queries against a STUN server located on the public Internet.
- This will allow SIP signaling and RTP media to successfully traverse a NAT without requiring any configuration changes on the NAT. STUN presents a working solution for most NATs that are not symmetric NAT, e.g., most of the SOHO routers have non-symmetric NAT and in this case, it is OK to use STUN. However, STUN does NOT work with symmetric NAT and if your routers have built-in symmetric NAT, do not use STUN.
- A STUN server can help facilitate traversing through most NATs, except for symmetric NATs.

Outbound Proxy

VoIP service provider can host a SIP outbound proxy server to handle all of the gateway VoIP traffic. User has to enable this feature if VoIP service provider has a SIP outbound server to handle voice calls. This allows the gateway to work with any type of NAT router and eliminates the need of STUN.

Important Voice Parameters:

Jitter Buffer:

- The clarity of voice heard from the called party is also distorted due

to non-uniform delay introduced by the Telephony network known as ‘Jitter’.
- In order to reduce the effect of ‘jitter’, some sort of ‘filters’ are used in the voice-channel called ‘Jitter Buffer’. The gateway has built-in adaptive buffer that helps to smooth out variations in delay (jitter) for voice traffic. This will ensure good voice quality. It is not programmable.

A-law

It is an ITU-T companding standard, used in the conversion between analog and digital signals in PCM systems. A-law is used primarily in European telephone networks and is similar to the North American μ-law standard. See also companding and μ-law.

Voice Compression:
- Voice compression is a technique to reduce the band width of voice channel, using suitable codec or Coder-Decoder. The codec converts voice signals from analog form to digital signals and vice versa. It uses the compression alongwith, CELP.
- Compression-decompression is a voice compression-decompression algorithm that defines the rate of speech compression, quality of decompressed speech and processing power requirements. It is called companding.
- Codec of suitable bandwidth is selected during the speech reception or transmission, to minimize the digital voice channel bandwidth.

QoS:

Quality of Service, QoS is the capability of a network to provide better service to selected network traffic over various technologies, including Ethernet and 802.1 networks, and IP-routed networks. Gateway supports Type of Service tagging and Differentiated Services tagging as mentioned below: (This allows the gateway to tag voice frames so they can be prioritized over the network)
1. Type of Service (ToS)
2. DiffServe

ToS:
- Network traffic can be classified by setting the ToS values at the

data source (at the gateway) so that a server can decide best method of delivery, such as the least cost, fastest route etc.
- This parameter allows you to configure Type of Service (ToS) bits by specifying the **precedence** and **delay** of audio and signaling IP packets

DiffServe:
DiffServe is differentiated Services, which defines a Class of Service model that makes packets so that they receive per hope treatment at DiffServe-compliant network services along the route, based on the application types and traffic flow. Packets are marked with DiffServe Code Point (DSCP) indicating the level of service desired.

Relevant Topic:
1. Glossary 182

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Web Jeeves

Introduction

Matrix supplies a special windows based software package to program the SETU VFX88L/VFX44L through the Internet, known as Jeeves. Description of following features related to web will help in programming the system:

IP Address

- An IP address (Internet Protocol address) is a unique number, similar in concept to a telephone number, used by network devices (computers, time-servers, FAX machines, some telephones). It is attached to a network to refer to each other when sending information through a LAN or a WAN or the Internet for example.
- An example of IP address is 207.142.131.236. Converting a number address to a more human-readable form called a domain address is done via the Domain Name System.
- The Internet Protocol (IP) knows each logical host interface by a number, the so-called IP address. On any given network, this number must be unique among all the host interfaces that communicate through this network.
- Users of the Internet are sometimes given a host name in addition to their numerical IP address by their Internet service provider.
- For all programs that utilize the TCP/IP protocol, the sender IP address and destination IP address are required in order to establish communications and send data.

Sub Net Mask Address

- It is a mechanism that is used to split a network into a number of smaller sub networks. It can be used to reduce traffic on each sub network by confining traffic to only the sub networks for which it is intended, thereby eliminating issues of associated congestion on other sub networks and reducing congestion in the network as a whole.
- Makes entire network more manageable.
- Each sub network functions as though it were an independent network, keeping local traffic local, and forwarding traffic to another sub network only if the address of the data is external to the sub

network. Such decisions are made on the basis of routing-tables contained within the various routers, with each table comprising an IP address table.

- Subnet is a portion of the network, which may be a physically independent network, which shares a network address with other portions of the network and is distinguished by a subnet number.
- The first octet of the IP address shows the network address, the second one shows the subnet number and the last two shows the host part.

Default gateway

- A default gateway is a node on a computer network that serves as an access point to another network.
- In enterprises, the gateway is the computer that routes the traffic from a workstation to the outside network that is serving the Web pages.
- In homes, the gateway is the ISP that connects the user to the internet.
- A default gateway is used by a host when an IP packet’s destination address belongs to someplace outside the local subnet (thus requiring more than one host of Ethernet communication).
- The default gateway address is usually an interface belonging to the LAN’s border router.

Domain Name System (DNS)

- DNS is a system that stores information about **host names** and **domain names** in a type of distributed database on networks, such as the Internet.
- Of the many types of information that can be stored, most importantly it provides a physical location (IP address) for each domain name, and lists the mail exchange servers accepting e-mail for each domain.
- The DNS provides a vital service on the Internet as it allows the transmission of technical information in a user friendly way.
- While computers and network hardware work with IP addresses to perform tasks such as addressing and routing, humans generally find it easier to work with hostnames and domain names (such as www.example.com) in URLs and e-mail addresses.

- The DNS therefore mediates between the needs and preferences of humans and of software.

Dynamic Host Configuration Protocol (DHCP)

- The DHCP is a client-server networking protocol.
- A DHCP server provides configuration parameters specific to the DHCP client host requesting, generally, information required by the host to participate on the Internet network.
- DHCP also provides a mechanism for allocation of IP addresses to hosts.

The Welcome page contains following Links. Click on any link to open the programming page.

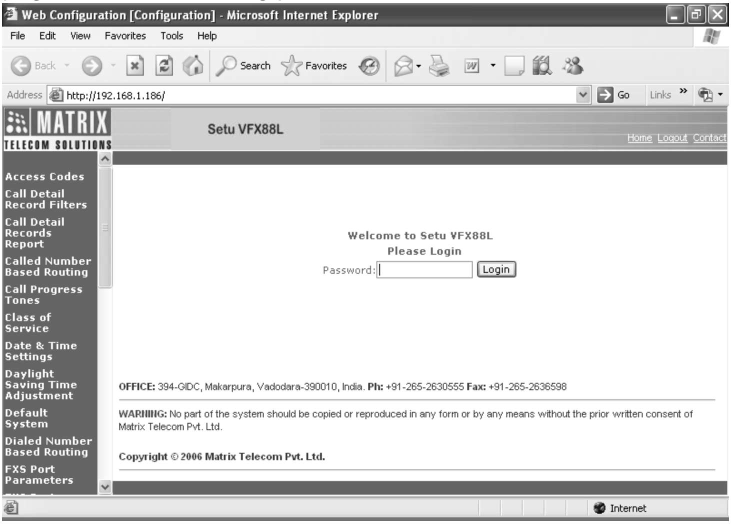
- Access Codes
- Call Detail Record Filters
- Call Detail Records Report
- Called Number Based Routing
- Call Progress Tones
- Class of Service
- Date and Time Settings
- Daylight Saving Time Adjustment
- Default System
- Dialed Number Based Routing
- FXS Port Parameters
- FXS Port Routing Groups
- Network Port Parameters
- Number Lists
- Password Change
- Peer-to-Peer Dialing
- Prefix to Domain Name Conversion
- Ring Type
- SIP Account Parameters-1
- SIP Account Parameters-2
- SIP Account Routing Groups
- Soft Restart
- Status
- Supplementary Services
- System Parameters

- Debug
- Upload/Download-System Software
- Upload/Download-Configuration
- Upload/Download-Call Detail Records

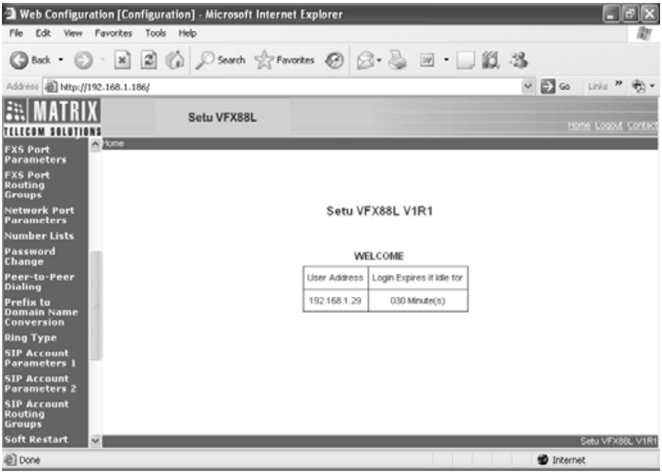
How to make changes in feature?
In order to make changes, click on any corresponding feature, from the list of the Web page and then refer related topic.

Web Pages:

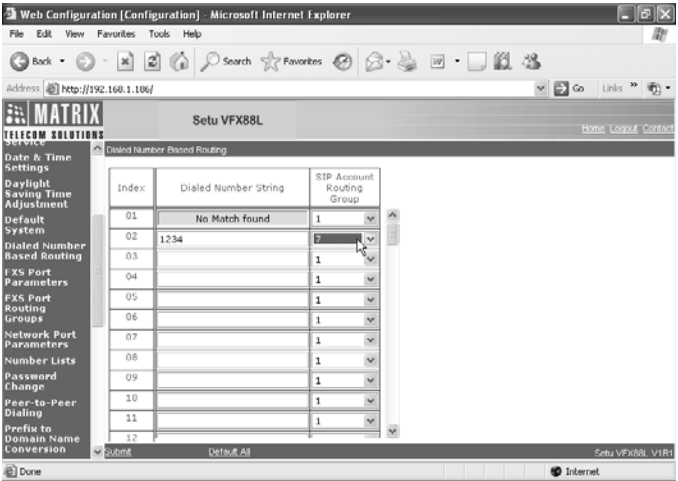
- After entering suitable IP Address in field of Internet Explorer 6 web page enter the following password as shown below:



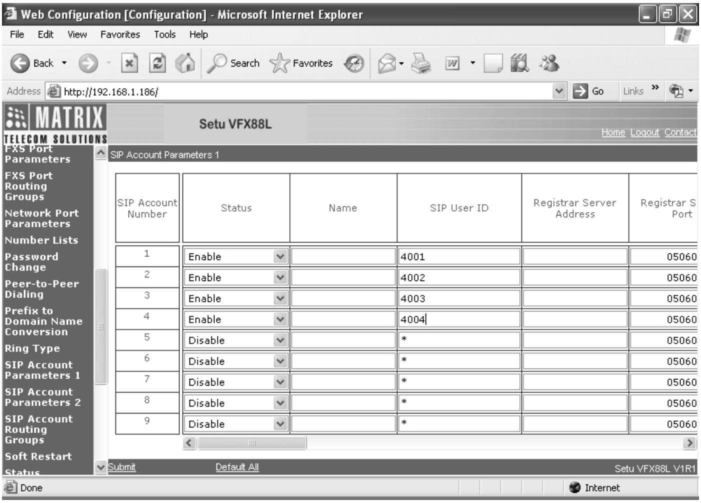
- The welcome page displays the time after which login expires as shown below:



- Click on the web link ‘Dialed Number Based Routing’. Program Dialed Number String and SIP Account Routing Group. Thus any dialed number starting with this string will be routed to the port as per routing group programmed.



- Click on web link ‘SIP Account Parameters-1’ and program ‘SIP User ID’ for SIP Account Number as shown below:


Web Configuration [Configuration] - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://192.168.1.186/
Go Links
MATRIX TELECOM SOLUTIONS
Setu VFX88L
Home Logout Contact
FXS Port Parameters
FXS Port Routing Groups
Network Port Parameters
Number Lists
Password Change
Peer-to-Peer Dialing
Prefix to Domain Name Conversion
Ring Type
SIP Account Parameters 1
SIP Account Parameters 2
SIP Account Routing Groups
Soft Restart
Status
SIP Account Parameters 1
SIP Account Number
Status
Name
SIP User ID
Registrar Server Address
Registrar S Port
1 Enable 4001 05060
2 Enable 4002 05060
3 Enable 4003 05060
4 Enable 4004 05060
5 Disable * 05060
6 Disable * 05060
7 Disable * 05060
8 Disable * 05060
9 Disable * 05060
Submit
Default All
Setu VFX88L V1R1
Done
Internet

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Section 3: Appendices

Appendix A: Technical Specifications

Configuration and Capacity:

	Application	No. of Ports	Connector Type
Maximum FXS Port	For connecting Analog Phone or Fax Machine	8(VFX88L) 4(VFX44L)	RJ 11
PSTN Life Line Port	For accessing PSTN line	1	RJ 11
Ethernet Port	To connect Ethernet cable through Router or Modem	1	RJ 45, 10/100 Base T
DC Jack	To connect the power adapter for DC supply voltage	1	DC Power Jack

VoIP and Networking Protocol:

- MAC Address (IEEE802.3)
- IPv4-Internet Protocol Version 4 (RFC791)
- ARP-Address Resolution Protocol
- DNS-A record (RFC1706), SRV Record (RFC 2782)
- DHCP Client-(RFC 2131)
- ICMP-Internet Control Message Protocol (RFC729)
- TCP-Transmission Control Protocol (RFC793)
- UDP-User Datagram Protocol (RFC768)
- RTP-Real Time Protocol (RFC1889) (RFC1890)
- RTCP-Real Time Control Protocol (RFC1889)
- DiffServe (RFC2475), Type Of Service TOS (RFC791/1349)
- SNTP-Simple Network Time Protocol (RFC 2030)

Voice Functionalities:

- SIP-Version 2
- Nine SIP Accounts can be programmed and 8-calls can be setup simultaneously (4 calls for VFX44L).
- Call Progress Tone Generation: Select as per the country
- QoS: RTP QoS and SIP QoS using DiffServ and ToS (Precedence)

- Full Duplex Audio
- Echo Cancellation: G.168
- Forward error correction (FEC)
- Voice Activity Detection (VAD)
- Fax using T.38

Voice Codecs: G.711 (a-Law and u-law), iLBC, G.729, G.723, GSM.

Telephony features:
- Voice Calls using SIP proxy and Voice calls without using SIP proxy (Peer-to-Peer Calling)
- Call Waiting and Cancel Call Waiting
- Call Forwarding
- Call Transfer
- Caller ID
- Answer Signaling
- Disconnect Signaling

PSTN Life Line Support:
- Send Emergency Call to PSTN
- Make PSTN Call by Dialing access code and a number programmed

Time Settings: Synchronizing with specific Time Server, programmed

Provisioning, Administration and Maintenance:
- Programming Using Web Page
- Phone Programming of some parameters like:
  - WAN IP Address
  - DND
  - Hotline

LED Indication:
1 LED for Power
1 LED for each FXS Port
1 LED for System

Security **:** Password Protected Administrator

Dimension (W x H x D) : 155 x 220 x 35 mm (6.10” x 8.66” x 1.38”)

Power Supply:

External Adaptor : 12VDC@2.0Amp.
Power Consumption : 20W (Typical)

Environmental:

Operating Temperature : -10°C to 50°C (140° to 122°F)
Operating Humidity : 5-95% RH, Non-Condensing
Storage Temperature : -40°C to 85°C (-40°F to 185°F)
Storage Humidity : 0-95% RH, Non-Condensing

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Appendix B: Glossary

CELP

It is the Code Excited Linear Prediction compression. It is the compression algorithm used in low bit-rate voice encoding. Used in ITU-T Recommendations G.728, G.729, G.723.

DHCP

It is the Dynamic Host Configuration Protocol. It provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.

DNS

It is the Domain Name System. System used on the Internet for translating names of network nodes into addresses.

FTP

File Transfer Protocol - A protocol used to transfer files over a TCP/IP network.

FXS

It is Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Matrix-Gateway's FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment and PBXs.

G.711

Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs. Described in the ITU-T standard in its G-series recommendations.

G.723

It is an ITU standard for speech codecs that uses the ADPCM method and provides toll quality audio of 20 and 40 Kbps.

G.729

Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. It is described in the ITU-T standard in its G-series recommendations.

GSM Codec:
This codec uses the information from previous samples in order to predict the current sample. The speech signal is divided into blocks of 20 ms. These blocks are then passed to the speech codec, which has a rate of 13 kbps, in order to obtain blocks of 260 bits.

HTTP
Hyper Text Transport Protocol-The communications protocol used to connect to servers on the World Wide Web.

iLBC Codec:
internet Low Bit Rate Codec (iLBC) is a royalty free narrowband speech codec. It is suitable for VoIP applications, streaming audio, archival and messaging. iLBC handles the case of lost frames through graceful speech quality degradation. It is defined in RFC 3951. This codec works on sampling frequency of 8KHz/16 bit.

IP
It is the ‘Internet Protocol’, Network layer protocol in the TCP/IP stack offering a connectionless internetwork-service. IP provides features for addressing, type-of-service specification, fragmentation and reassembly and security. It is defined in RFC 791.

MAC
Media Access Control Address-The unique address that a manufacturer assigns to each networking device.

- MAC is Media Access Control address in the form of 48-bit number, which is unique to the LAN NIC (Network Interface Card).
- It is programmed into the card at the time of manufacture. IEEE registration authority administers MAC address scheme for all LANs which conform to IEEE, 802stds. Including both Ethernet and token ring.
- Consists of two parts: 24-bit company id (manufacturer ID0 and 24-bit Extension ID (board ID).

- Destination and source MAC names are contained in the header of the LAN packet and are used by various devices like hubs, bridges.
- A VoIP Service Provider will typically have its subscribers register with the Service Provider's VoIP service server before starting a subscriber's service. Often, an ITSP will require the registration of the MAC addresses of any devices directly connected to their network.

NTP

It is the Network Time Protocol. Protocol built on top of TCP that assures accurate local time-keeping with reference to radio and atomic clocks located on the Internet. This protocol is capable of synchronizing distributed clocks within milliseconds over long time periods.

SDP

It is the Session Definition Protocol, an IETF protocol for the definition of Multimedia Services. SDP messages can be part of SGCP and MGCP messages.

TCP

It is the Transmission Control Protocol, a connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.

UAS

It is the User agent server (or user agent), a server application that contacts the user when a SIP request is received and then returns a response on behalf of the user. The response accepts, rejects or redirects the request.

UDP

User Datagram Protocol-A network protocol for transmitting data that does not require acknowledgment from the recipient of the data that is sent.

VAD

It is the Voice activity detection, when enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

VoIP

It is the Voice over IP, the capability to carry normal telephony-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP enables a gateway or router to carry voice traffic (for example, telephone calls and faxes) over an IP network. In VoIP, the DSP segments the voice signal into frames, which then are coupled in groups of two and stored in voice packets.

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Appendix C: Features at a Glance

Features	**Access Code**
Enter SE Programming	***#19***
Access Lifeline Port	*##*
Set Hotline	***41***
Cancel Hotline	***42***
Set Call Forward-Unconditional	***51***
Cancel Call Forward-Unconditional	***52***
Set Call Forward-Busy	***53***
Cancel Call Forward-Busy	***54***
Set Call Forward-No Reply	***55***
Cancel Call Forward-No Reply	***56***
Set Do Not Disturb (DND)	***61***
Cancel Do Not Disturb (DND)	***62***
Set Call Waiting	***71***
Cancel Call Waiting	***72***
Retrieve Hold Call	***81***
Hold Call	***Flash-1***
Blind Transfer	***Flash-2***
Release Call Waiting Call	***Flash-3***
Accept Call Waiting Call	***Flash-4***
Reject Remote Held Call	***Flash-5***

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Appendix D: Programming Commands

Description	Commands
Exit Programming Mode	***00#****
To program Network Port IP Address	***11-XXXX XXXX XXXX XXXX-#****
To program Subnet Mask	***12-XXXX XXXX XXXX XXXX-#****
To display the Network IP Address on the Phone	***21-#****
To display Subnet Address	***22-#****
To display Gateway Address	***23-#****
To display DNS Address	***24-#****

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Notes

Notes

Programming Register

S.N.	Date	Major Programming Changes made

Programming Register

S.N.	Date	Major Programming Changes made