SETU VFXTH

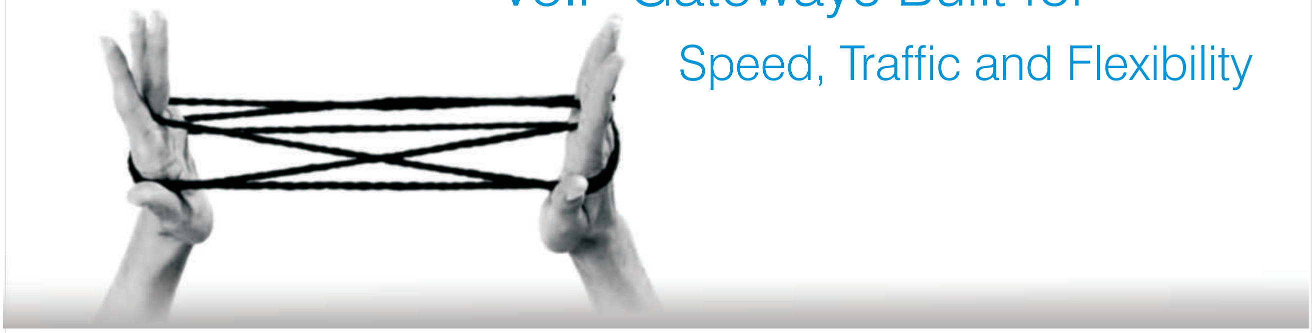
Multi-Port VoIP-FXO-FXS Gateways

Modern organizations need access to omnipresent telecom networks like PSTN and the new-age internet telephony for the benefits of cost, convenience and quality. On the other hand, modern organizations with IP infrastructure cannot remain isolated from the PSTN. All such organizations need a multi-channel, transparent and flexible VoIP to PSTN gateway.

Matrix presents SETU VFXTH - The multi-channel SIP gateway offering seamless connectivity between VoIP and PSTN networks through multiple FXS and FXO ports. Matrix SETU VFXTH offers universal and transparent call routing irrespective of type of ports VoIP-FXS, VoIP-FXO and FXO-FXS. Its superior call and signal processing capabilities ensure unrestricted flow of multiple calls with higher speed and better speech quality.

Let Matrix SETU VFXTH be your high-performance bridge to VoIP and PSTN networks.

MATRIX
TELECOM SOLUTIONS



VoIP Gateways Built for

Speed, Traffic and Flexibility

SETU VFXTH is a family of multi-channel transparent and flexible VoIP to FXS-FXO gateways. On VoIP side, it supports SIP based IP interfaces allowing it to connect to any existing IP network. On the PSTN side, it supports FXS and FXO interfaces. SETU VFXTH supports universal routing facilitating calls between diverse port types. Further, intelligent Least Call Routing selects the most economical call route based on the destination number dialed on per call basis.

SETU VFXTH can register with multiple SIP Servers and IP-PBXs. Single SETU VFXTH can be used to interface with the enterprise IP-PBX for internal calls and public ITSP SIP Servers for international calls. Similarly on the PSTN side, few FXS/FXO ports can be used with the enterprise PBX while the other FXS/FXO ports can be interfaced with the PSTN for public calls. SETU VFXTH offers extensive range of features including Programmable Access Codes, Allowed and Denies Numbers, Automatic Number Translation, CLI Based Routing, Emergency Number Dialing, Least Cost Routing, Peer-Peer Calling and CDR. On the network side it offers NAT/STUN, PPPoE, Fax over IP, Syslog debug, and others.

In summary, Matrix SETU VFXTH is one of the most flexible, feature-rich and dependable VoIP-FXS-FXO Gateways.

SETU VFXTH CONFIGURATIONS

VoIP-FXO-FXS Gateways

Sr. No.	Model Name	VoIP Channels	FXO Ports	FXS Ports
1.	SETU VFXTH0808	16	8	8
2.	SETU VFXTH1212	24	12	12
3.	SETU VFXTH1616	32	16	16

VoIP-FXO Gateways

Sr. No.	Model Name	VoIP Channels	FXO Ports
1.	SETU VFXTH0800	8	8
2.	SETU VFXTH1600	16	16
3.	SETU VFXTH2400	24	24
4.	SETU VFXTH3200	32	32

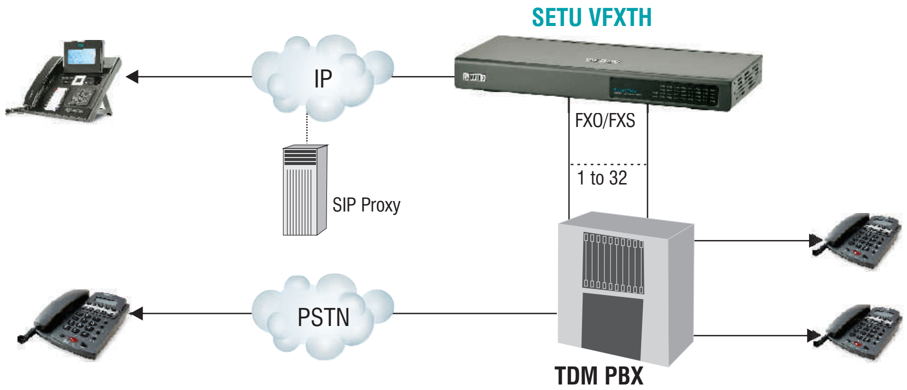
VoIP-FXS Gateways

Sr. No.	Model Name	VoIP Channels	FXS Ports
1.	SETU VFXTH0016	16	16
2.	SETU VFXTH0024	24	24
3.	SETU VFXTH0032	32	32

SETU VFXTH APPLICATIONS

VoIP GATEWAY FOR TRADITIONAL PBX

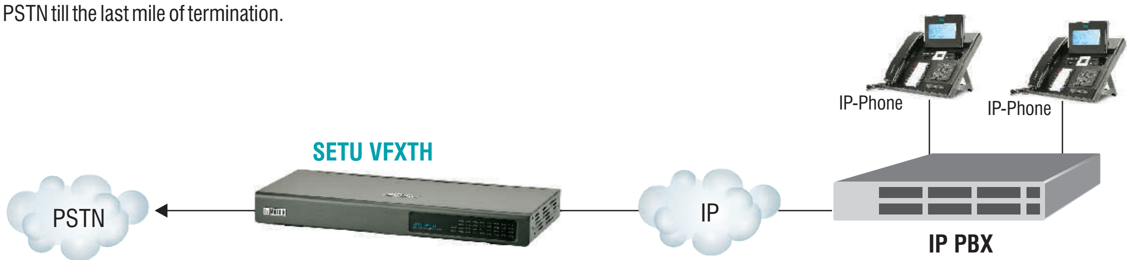
SETU VFXTH helps an organization to integrate new communication technologies to its existing infrastructure. Using SETU VFXTH, a traditional PBX system can connect to VoIP networks. With new networks being integrated, communication happens using the most cost-effective networks. Features like Automatic Number Translation, Programmable Access Codes and host of other advanced functionality, enable users to access new networks without changing their dialing habits.



SETU VFXTH can be connected to a traditional PBX in two ways. Firstly, the FXS ports of SETU VFXTH can be connected to the FXO ports of the PBX. On dialing prefix assigned in the PBX, the user can access these FXS ports to make VoIP calls. Secondly, the FXS port (SLT port) of the PBX can be connected to FXO port of the SETU VFXTH. PBX users can access VoIP network connected to SETU VFXTH by dialing the FXS port number of the PBX.

PSTN GATEWAY FOR IP-PBX

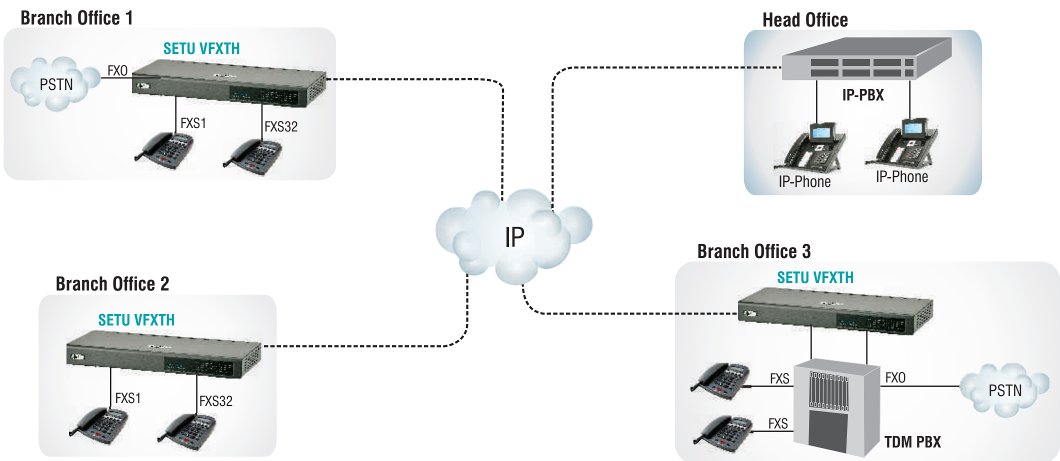
SETU VFXTH enables the IP-PBX users to make calls to the PSTN network. It assures users the most cost effective route of IP for all their calls to PSTN till the last mile of termination.



Multiple Branches can be connected over IP network, reducing overall communication cost in-between the various branches of an organization. SETU VFXTH gets registered to an IP-PBX as a client. Registering multiple gateways to a single IP-PBX, allow the IP-PBX users to relay calls over IP. This also facilitates employees at one branch office to access the local PSTN connectivity of the other branch. This further saves on inter-network toll charges.

MULTI-SITE CONNECTIVITY

Matrix SETU VFXTH facilitates easy and low cost communication between geographically spread, multi-branch offices. A multisite connectivity can be established through the existing IP network. SETU VFXTH connects to existing infrastructure with its varied interfaces. SETU VFXTH located at various branch offices can be registered to the IP-PBX at the head office. This extends the IP-PBX connectivity to the various branch offices. SETU VFXTH can be connected to the local PSTN network via FXO interface. This allows all other registered users of the IP-PBX to access the local PSTN network where the SETU VFXTH is installed, reducing the call cost significantly.



SETU VFXTH alone suffices to provide connectivity to both-the switched (PSTN) and VoIP networks, as in the case of branch office 1 and 2. It also support direct connectivity of up to 32 TDM extensions. On the other hand, SETU VFXTH can be a bridge that connects an existing PBX system to the IP network, as in case of branch office 3.

KEY FEATURES

Allowed and Denied Numbers
Allowed and Denied Lists are used to restrict dialing of long-distance and international numbers. A number is blocked if its prefix matches with any entry in the Denied Lists. Simillarly, a number is allowed to go through if it matches with any entry in the Allowed List. This provides flexibility of allowing only specific numbers, blocking others.

Automatic Number Translation
This feature allows the Gateway to translate the number string dialed by the user to a format compatible with the network through which the call is to be routed. So, the user can dial numbers freely without worrying about the network through which the call will be routed.

Call Progress Tones and Rings
The gateway offers flexibility of programming call progress tones and ring cadence to match the standards of the country of installation. Country Specific Call Progress tones like Dial Tone, Ring Back Tone, and Busy Tone etc. can also be programmed.

Caller Line Identification and Presentation (CLIP)
SETU VFXTH can provide Caller Line Identification Presentation (CLIP) on FXS ports. Analog CLIP protocols such as DTMF, FSK ITU-T V.23 and FSK Bellcore 202A are supported by the gateway.

CLI based Call Routing
A call can be routed to a fixed number from the CLI presented on SETU VFXTH. When a call lands on the network ports of the SETU VFXTH, it detects and verifies the CLI and routes the call to the predefined number.

Call Detail Record (CDR)
SETU VFXTH can store details of 2000 calls in its memory. Call reports can be generated using filters like source port, destination port, calling number, called number, date, time and duration.

Day Light Saving
The Real Time Clock (RTC) of the SETU VFXTH moves forward or backward automatically in tune with the Day Light Saving requirement of the country where it is installed. The options like Week-Day-Month or Date-Month are provided to move the clock forward and backward automatically on the specified day, date and time.

Fax over IP (FoIP)
The Matrix SETU VFXTH user can send and receive Fax over SIP account, when the FAX machine is connected to the FXS port. The SETU VFXTH supports FoIP using T.38 Vocoder and Pass-Through.

FXO
The gateway offers FXO port to connect the PSTN network and route incoming calls to VoIP network or the extensions. It can also be used to connect extension of a PBX to network two PBX over VoIP.

Network Port Parameters
Web Jeeves of SETU VFXTH provides a configuration menu to program its network port parameters to match the LAN addressing scheme of the installation site. Parameters such as IP address, subnet mask, connection type, etc., can also be programmed through a telephone instrument connected to the system.

Peer-to-Peer Calling
SETU VFXTH support VoIP calls, between two locations without going through a proxy server. Fixed IP addresses of various locations can be programmed in the Peer-to-Peer table of the gateway to avail this facility. The gateway supports 500 entries in the Peer-to-Peer table. Numerical dialing codes can be defined to simplify the calling between various locations.

PIN Authentication
The gateway uses PIN Authentication to verify a caller's identity before routing the call from one network to another. It is an important feature that prevents unauthorized usage.

Prefix to Domain Name Conversion
This feature enables the conversion of a numerical code dialled by the user to a domain name. It helps the SIP service provider to understand and route the call to the required destination.

Remote Held/Transfer
The user of SETU VFXTH can keep a called person at remote end on hold or even transfer the call to a third person. This feature can be activated only if the device at the called end supports Call Hold/Call Transfer feature.

SIP Accounts
A maximum of 9 SIP accounts can be programmed and each FXS user can be assigned one of the SIP accounts for outgoing calls. Dynamic allocation of SIP account is also possible using the Dial Plan. These SIP accounts can be from multiple service providers or all from a single service provider.

Universal Routing
Universal Routing allows the gateway to route a call received from one network to another based on the routing mechanism programmed in the system. A source port can be programmed for more than one destination port so that if one port is busy, the call can be routed through another port.

VLAN Tagging
A Virtual Local Area Network (VLAN) may be defined as a group of LANs that have different physical connections, but which communicate as if they are connected on a single network segment. VLANs increase overall network performance by grouping users and resources that communicate most frequently with each other.

Web Based Programming
Matrix SETU VFXTH incorporates Built-in HTTP server and web pages for easy configuration. This Web based programming feature allows a user to configure the gateway from any part of the world, once connected to the IP network

TECHNICAL SPECIFICATIONS

FXS Port Parameters

Signaling	: Loop Start
Connector	: RJ11
Off Hook AC Impedance	: 600Ω/900Ω/Complex
Number of Long Loop Extension	: 4
Loop Limit	: 1800 (Max) Excluding Telephone set
On-Hook Voltage(Tip/Ring)	: -48V
Off Hook Current	: 25mA (Max)
Ringing Voltage	: Trapezoidal 60VRMS/25Hz and Sinusoidal 2VRMS/25Hz
REN	: 3
Dialing	: DTMF and Pulse (10 PPS and 20 PPS)
DTMF Detection	: ITUT Q.24
CLI Presentation	: DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	: Over Voltage Secondary Protection
Return Loss	: >18dB
Longitudinal Balance	: >50dB
Transmission Level Adjust	: Tx Gain: -3dB to +6dB, Rx Gain: -3dB to +6dB

FXO Port Parameters

Signaling	: Loop Start
Connector	: RJ11
Off Hook AC Impedance	: 600Ω/900Ω/Complex
Loop Limit	: 1200Ω
Pulse Dialing	: 10 PPS and 20 PPS
DTMF Dialing and Reception	: ITUT Q.23 and Q.24
CLI Reception	: DTMF, FSK ITU-T V.23 and FSK Bellcore 202
Protection	: Over Voltage and Over Current Secondary Protection
Return Loss	: >18dB
Longitudinal Balance	: >50dB
Transmission Level Adjust	: Tx Gain: -15dB to +10dB : Rx Gain: -15dB to +10dB
Call Maturity	: Delay and Polarity Reversal

VoIP Port Parameters

VoIP Protocols	: SIP v2, SDP, RTP (RFC 2833)
Network Protocol	: IPv4,TCP, UDP, DHCP, SNTP STUN, HTTP
SIP	: 9 SIP Accounts Out Bound Proxy Support
NAT	: STUN and NAT Keep Alive
Voice CODECS	: G.711 A-Law, μ-Law, G.723.1, G.729A, G.729B, GSM-FR, GSM-EFR
Line Echo Cancellation	: G.168 with 128ms Tail Length
Call Progress Tones	: Dial Tone, Ring Back Tone, Busy Tone, Error Tone
Voice	: Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Detection
Fax	: T.38 and Pass-Through
Quality of Service	: Layer 3 DIFFServ and TOS
Data Network	: WAN Port RJ45 Auto MDIX 10/100 BaseT
Security	: Password Protected Administration
Physical Connector	: RJ45

Power Supply

Input	: 24V DC @2.5A
Power Consumption	: 60 Watts
Connector	: DC Power Jack
LED Indications	: 1 GREEN Colour LED for Power, 1 Dual Color LED for Status, 32 Single Colour LEDs for each Port

Mechanical

Dimensions (WxHxD)	: 40.7 X 5.1 X 17.2 cm (16.0" X 2.0" X 6.8")
Unit Weight	: 1.610 Kg (3.54 Lbs) Approx
Shipping Weight	: 2.570 Kg (5.66 Lbs) Approx
Material	: Aluminium 1.5mm Powder Coated
Installation Mounting	: Table-Top, Wall and Rack Mount

Environmental

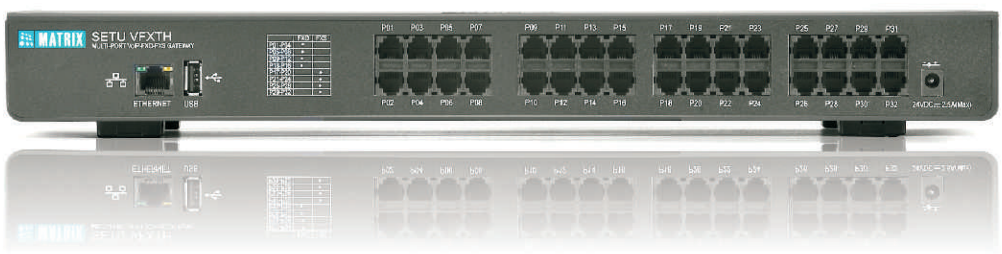
Operating Temperature	: -10°C to +50°C
Storage Temperature	: -40°C to +85°C
Operating Humidity	: 5-95% RH (Non-Condensing)
Storage Humidity	: 0-95% RH (Non-Condensing)

FEATURES LIST

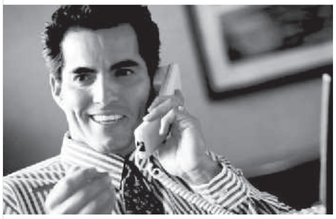
- Allowed and Denied Numbers
- Attended/Blind Call Transfer
- Automatic Number Translation
- Answer and Disconnect Signaling on FXS Port
- Answer and Disconnect Supervision on FXO Port
- Call Detail Record
- Call Progress Tones
- Caller Line Identification and Presentation (CLIP)
- Conference
- CLI on FXS Port
- CLI based Authentication
- Date and Time Settings
- Day Light Saving Mode
- Default the Configuration (System Default)
- Destination Number Determination Method
- Destination Port Determination Method
- Digest Authentication
- Do Not Disturb
- Dynamic DNS
- Emergency Number Dialing
- Fax over IP (T.38 and Pass-Through)
- Hotline
- MAC Cloning
- NAT and STUN Support
- PCAP Trace
- Peer-to-Peer Calling Table
- PIN Authentication
- PPPoE
- Prefix to Domain Name Conversion
- Programmable Access Codes
- Remote Held (SIP Accounts Only)
- Remote Transfer (SIP Accounts Only)
- Speech Gain Setting
- Supplementary Services
  - Call Forward
  - Call Hold
  - Call Waiting
- System Log Client
- VLAN Tagging
- Web based Programming

VoIP PRODUCTS FROM MATRIX

ETERNITY IP-PBX	The IP-PBX with Universal Connectivity and Seamless Mobility
SAPEX	All-in-One Embedded IP-PBX Server
VYOM CCX	High-Density SIP Gateway
ETERNITY	The Universal Telephony Gateway
SETU VGFX	Multi-port SIP based VoIP to GSM-FXO-FXS Gateway
SETU VFXTH	Multi-Port VoIP to FXO-FXS Gateway
SETU VFX	SIP based VoIP Gateway with 4/8 FXS Ports, 1 FXO (PSTN Pass-Through) and 1 Ethernet Port
SETU ATA211G	SIP based Port VoIP to GSM and FXS Gateway
SETU ATA211	SIP based Analog Telephone Adaptor with 1 FXO, 1 FXS Port and 2 Ethernet Ports
SETU ATA2S	SIP based Analog Telephone Adaptor with 2 FXS Ports and 1 Ethernet Port
SETU ATA1S	SIP based Analog Telephone Adaptor with 1 FXS Port and 2 Ethernet Ports
SETU VP248PE	Executive IP-Phone with 6 Lines x 24 Characters LCD Display and PoE
SETU VP248SE	Executive IP-Phone with 2 Lines x 24 Characters LCD Display and PoE
SETU VP236SE	Executive IP-Phone with 2 Lines x 24 Characters LCD Display and PoE
SETU VP236S	Executive IP-Phone with 2 Lines x 24 Characters LCD Display



ABOUT MATRIX



ISO 9001 Company, Matrix is a leader in Telecom and Security solutions for modern businesses and enterprises. An innovative, technology driven and customer focused organization; the company is committed to keep pace with the revolutions in the telecom and security industries. With around 30% of its human resources dedicated to the development of new products, Matrix has launched cutting-edge telecom products like IP-PBX, Universal Gateways, VoIP Gateways and Terminals, GSM Gateways, Access Control and Time-Attendance Systems and Fire Alarm Systems. These solutions are feature-rich, reliable and conform to the international standards. Having global foot-prints in Asia, Europe, North America, South America and Africa through an extensive network of more than 500 channel partners, Matrix ensures that the products serve the needs of its customers faster and longer. Matrix has gained trust and admiration of more than 150,000 customers representing the entire spectrum of industries. Matrix has won many awards for its innovative products.

For further information contact:

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Due to continuous technology upgradations, product specifications are subject to change without notice.