SETU VGFX

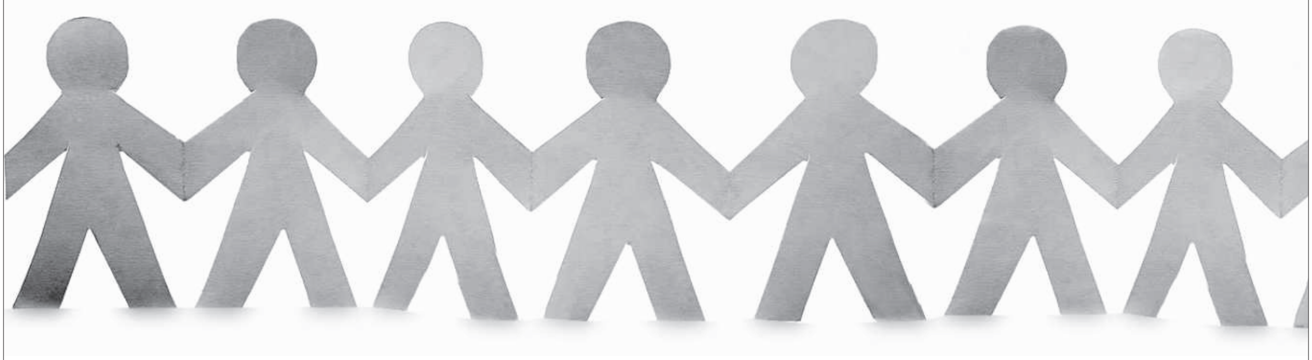
Multi-port SIP based VoIP to GSM, FXO and FXS Gateway

Today, with expanding business horizons, communication has emerged as a life-line for business sustenance. Communication that is seamless, faster and cost effective is what the new-age businesses need. Emergence of new telecom networks and the advantages they offer have encouraged organizations to re-consider their existing telecommunication network usage. They need an access to crucial omnipresent telecom networks mainly VoIP, GSM and POTS to avail the benefit of cost, convenience and quality of service. For instance, an organization with POTS communication backbone, needs connectivity to VoIP and GSM networks or a business communicating through IP based platform, needs to connect to GSM and POTS network. However, not all businesses are equipped with telecom infrastructure flexible enough to access these networks. On the other hand replacing their existing telephony can be a costly affair. Under such circumstances a solution that offers connectivity to these networks using existing communication infrastructure, is the need of the hour for any business.

Matrix presents SETU VGFX- The Single-box Gateway solution, offering seamless connectivity between VoIP, GSM and POTS (FXO and FXS) networks. SETU VGFX supports flexible and intelligent call routing options to ensure that communication always happens through the most cost effective network.

Let Matrix SETU VGFX be your bridge to the new world of diverse telephony!

MATRIX
TELECOM SOLUTIONS



One World. One Telephony.

Your Bridge to VoIP, GSM and FXO-FXS Networks

SETU VGFX allows seamless connection between the VoIP, GSM and POTS network. On VoIP side, it supports (SIP based) IP interface, allowing it to connect to any existing IP network. On the GSM side, it supports Quad-band GSM operation, allowing it to work with any GSM network. On the POTS side, both FXS and/or FXO interfaces are supported. Incoming call from one network can be routed to another destination using an appropriate network, depending on the destination number dialed. Likewise, outgoing calls from FXS port will be routed through an appropriate network, depending on the destination number dialed. It can handle calls on all the ports simultaneously allowing full traffic.

Using the innovative gateway for multi-branch voice communication, an organization avails full benefit of the low-tariff internet telephony, by establishing calls directly between two destinations. Alternately, Matrix SETU VGFX can also act as a SIP client, that too with the flexibility to register with multiple SIP Service Providers.

Programmable Access Codes, Automatic Number Translation, CLI Based Routing, Emergency Number Dialing, Least Cost Routing, Automatic Mobile Network Selection and other advanced functionalities ensures operational ease and convenience.

SETU VGFX CONFIGURATIONS

SETU VGFX8422
Gateway with 8 VoIP Channels, 4 GSM Ports, 2 FXO Ports and 2 FXS Ports

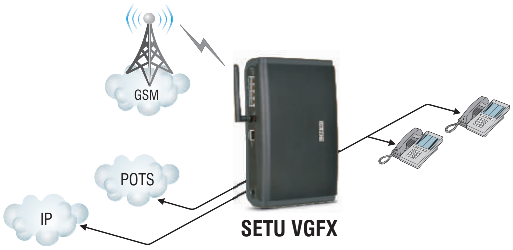
SETU VGFX8440
Gateway with 8 VoIP Channels, 4 GSM Ports and 4 FXO Ports

SETU VGFX8404
Gateway with 8 VoIP Channels, 4 GSM Ports and 4 FXS Ports

PRODUCT APPLICATIONS

SETU VGFX: As a Stand-Alone Gateway

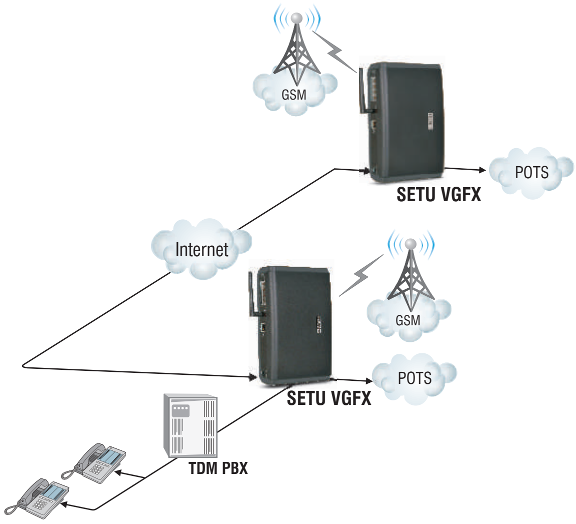
Matrix SETU VGFX is a single-box solution for Small Office/Home Office, which needs to access VoIP, GSM and POTS networks. Users can make calls to these networks using standard telephony instrument connected to SETU VGFX. Additionally, FXS to FXS calls can also be placed. Ideal to be used as a stand-alone gateway, SETU VGFX ensures the most cost-effective route for communication.



SETU VGFX can be configured in a stand-alone mode by connecting standard telephone instruments to its FXS ports and VoIP-GSM-FXO networks to its respective network ports. Calls over IP are made either through peer-to-peer or using the Proxy Server of a SIP service provider.

SETU VGFX: The Gateway using Peer-to-Peer Calls for VoIP

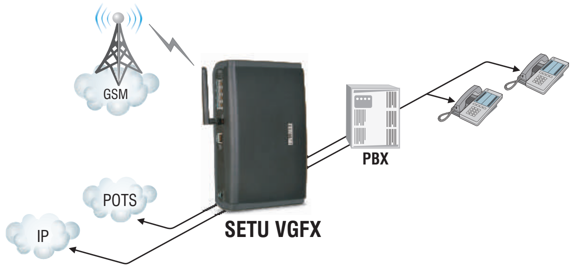
Peer-to-peer calls eliminate the role of IP PBX or SIP proxy server to establish calls over internet. It facilitates easy and low-cost communication between geographically spreaded, multi-branch offices. The user can access the GSM and POTS facilities of the remote branch office to make calls and save long distance call charges borne otherwise.



To establish Peer-to-Peer calls, both communicating ends require internet connectivity with fixed IP address. A Peer-to-Peer table needs to be programmed with the IP addresses of the locations and corresponding numeric dialing codes. Using the FXO and GSM ports, SETU VGFX can be connected to local FXO network and GSM network respectively.

SETU VGFX: GSM-VoIP Gateway for Traditional PBX

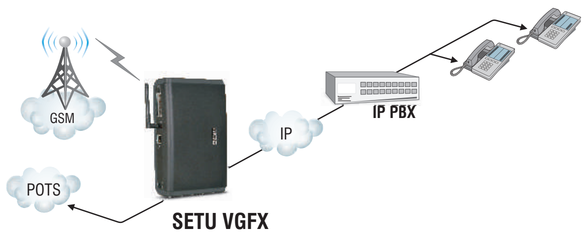
SETU VGFX helps an organization to integrate new communication technologies to its existing infrastructure. Using SETU VGFX, a traditional PBX system can enhance its capability and connect to VoIP and GSM networks. With new networks being integrated, communication happens using the most cost-effective 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 VGFX can be connected to a traditional PBX in two ways. Firstly, the FXS ports of SETU VGFX 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 GSM or VoIP calls. Secondly, the FXS port (SLT port) of the PBX can be connected to FXO port of the SETU VGFX. PBX users can access VoIP and GSM network connected to SETU VGFX by dialing the FXS port number of the PBX.

SETU VGFX: GSM-FXO Gateway for IP PBX

SETU VGFX enables the IP PBX users to make calls to the GSM and POTS network. It assures users the most cost effective route of IP for all their calls to GSM and POTS till the last mile of termination.



SETU VGFX gets registered to an IP PBX as a client. More than one clients can be registered to a single IP PBX, enabling it to connect GSM and POTS network of various geographical locations.

KEY FEATURES

Allowed and Denied Numbers
SETU VGFX offers flexibility to allow and deny the dialing of a particular number or a set of numbers. The denied list restricts a user to dial a number programmed in the denied list.

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 VGFX 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 Authentication
A caller can be authenticated from the CLI presented on SETU VGFX. When a call lands on the network ports of the Gateway, it detects the CLI, verifies its authenticity and routes to the predefined destination.

Call Detail Record (CDR)
SETU VGFX 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 Gateway, adjusts automatically to be in tune with the Day Light Saving requirements of the country of installation.

FXO
The Gateway offers FXO port to connect the PSTN network and route incoming calls to VoIP, GSM or FXS network and vice-versa. It can also be used to connect extension of a PBX to network two PBX over VoIP.

International Mobile Equipment Identity (IMEI)
IMEI number is a unique 15-digit code used by SETU VGFX to identify a GSM device. The Gateway uses IMEI numbers to identify its GSM Ports. This number also helps to associate a GSM port of SETU VGFX with a particular GSM network.

Mobile Network Selection
This feature enables the selection of a GSM network automatically or manually. Automatic network selection allows the Gateway to select the available network every time it is powered-on. While, the Manual network selection mode provides user with the option to select a desired network.

Network Port Parameters
Web Jeeves of SETU VGFX 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 VGFX 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 dialed 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 VGFX 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.

Returned call to Original Caller (RCOC)
RCOC is supported on GSM ports of the Gateway. When an outgoing call is made through the GSM Port and the called party is found busy or not responding, SETU VGFX will store details including the caller's FXS port number. In case the called person returns the call, the Gateway will automatically route this call to the same FXS port from where the call was attempted.

SIP Accounts
Nine SIP accounts can be programmed and each FXS user can be assigned with one SIP account for outgoing calls. Dynamic allocation of SIP accounts is also possible using Dial Plan.

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.

TECHNICAL SPECIFICATIONS

FXS Port Parameters

Signaling	: Loop Start
Connector	: RJ11
Off Hook Line Impedance	: 600Ω/900Ω Complex
Number of Long Loop Extension	: 1
Loop Limit	: 1500 (Max) Excluding Telephone set
Loop Feed	: 39mA (Max)
Ringing Voltage	: Trapezodial-60 Vrms @25Hz 3REN Sinusoidal-52 Vrms
Pulse Dialing	: 10 PPS and 20 PPS @ 1:2, 2:3 and 1:1
DTMF Dialing and Reception	: ITUT Q.23 and Q.24
CLIP	: DTMF, FSK ITU-T V.23 and FSK Bellcore 202A
Flash Timer	: 83-999 ms.(Programmable)
Protection	: Solid state (Over Voltage) Built-in Secondary Protection
Answer Signaling on FXS	: Polarity Reversal
Disconnect Signaling on FXS	: Polarity Reversal and Open Loop Disconnect

FXO Port Parameters

Signaling	: Loop Start
Connector	: RJ11
Off Hook Line Impedance	: 600Ω/900Ω Complex
Loop Limit	: 1200Ω
Pulse Dialing	: 10 PPS and 20 PPS @1:2, 2:3 and 1:1
DTMF Dialing and Reception	: ITUT Q.23 and Q.24
CLI Reception	: DTMF, FSK ITU-T V.23 and FSK Bellcore 202A
Protection	: Solid state (Over Voltage and Over Current) built-in Secondary Protection
Answer Signaling on FXO	: Polarity Reversal
Disconnect Signaling on FXO	: Polarity Reversal and Open Loop Disconnect

GSM Port Parameters

GSM Band	: Quad Band: GSM850, EGSM900 DCS1800, PCS 1900
Compliant	: ETSI GSM Phase2/2+
SIM Card	: One SIM per GSM Port
SIM Interface	: 1.8V, 3V
Transmission Power	: Class 4 (2W) at GSM850 and EGSM900 MHZ band Class 1 (2W) at DCS800 and PCS1900 MHZ band
RF Sensitivity	: Better than -102 dBm
Antenna (Panel Mount)	: 1 Antenna per 4 GSM Ports, 3.0 dBi, 50Ω SMA(Male) Connector

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 Activity 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	: 12VDC @1.25A through External Adaptor (90-265VAC, 47-63Hz)
Power Consumption	: 12W (Maximum)
Connector	: DC Power Jack
LED Indications	: 1 each for Power and Status 1 each for GSM Port 1 each for FXO and FXS Port

Mechanical

Dimensions (WxHxD)	: 23.0 x 5.5 X 16.3cm (9.1” x 2.2” x 6.4”)
Unit Weight	: 1kg (2.2lbs)
Shipping Weight	: 1.4kg(3.08lbs)
Material and Finish	: ABS Plastic
Installation Mounting	: Wall and Table-Top

Environmental

Operating Temperature	: -10°C to +50°C (-14°F to +122°F)
Storage Temperature	: -40°C to +85°C (-40°F to +185°F)
Operating Humidity	: 5-95% RH(Non-Condensing)
Storage Humidity	: 0-95% RH(Non-Condensing) at 40°C

COMPLIANCES

EMI/EMC

Conducted Emission	: CISPR 22
Radiated Emission	: CISPR 22
Harmonic Current Emission	: IEC 61000-3-2
Voltage Flicker	: IEC 61000-3-3
Electro-static Discharge	: IEC 61000-4-2
Radiated Susceptibility	: IEC 61000-4-3
Electrical Fast Transient	: IEC 61000-4-4
Surge	: IEC 61000-4-5
Conducted Immunity	: IEC 61000-4-6
Power Frequency Magnetic Field	: IEC 61000-4-8
Voltage Interruption & Dips	: IEC 61000-4-11

FCC

Conducted Emission	: FCC Part 15
Radiated Emission Field	: FCC Part 15

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
- Day Light Saving
- 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
- Emergency Number Dialing
- Fax over IP (T.38 and Pass-Through)
- Hotline
- International Mobile Equipment Identity (IMEI)
- MAC Cloning
- NAT and STUN Support
- Peer-to-Peer Calling Table
- PIN Authentication
- PPPoE
- Prefix to Domain Name Conversion
- Programmable Access Codes
- Quad-Band Support
- Remote Held (SIP Accounts Only)
- Remote Transfer (SIP Accounts Only)
- Returned Call to Original Caller (RCOC)
- SIM PIN
- Speech Gain Setting
- Supplementary Services
  - Call Forward
  - Call Hold
  - Call Waiting
- System Log Client
- VLAN Tagging
- Web based Programming

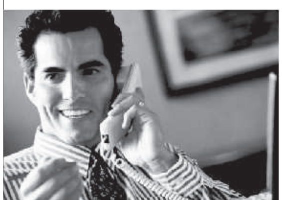
SYSTEM CAPACITY AND RESOURCES

System Resources	Description	SETU VGFX Configurations		
		8422	8440	8404
VoIP Channels	To Make VoIP (SIP) Calls Using Internet or Intranet	8	8	8
GSM Ports	To Connect to GSM Network	4	4	4
FXO Ports	To Connect CO Lines or Analog Extension Port	2	4	-
FXS Ports	To Connect PBX or Analog Telephones	2	-	4

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 SIP based VoIP to FXO-FXS Gateway
SETU VFX	Multi-Port SIP based VoIP to FXS Gateway
SETU ATA1S	SIP based Analog Telephone Adaptor with 1 FXS Port and 2 Ethernet Ports
SETU ATA2S	SIP based Analog Telephone Adaptor with 2 FXS Ports and 2 Ethernet Ports
SETU ATA211	SIP based Analog Telephone Adaptor with 1 FXO, 1 FXS and 2 Ethernet Ports
SETU ATA211G	SIP based Analog Telephone Adaptor with 1 FXS, 1 GSM and 2 Ethernet Ports
SETU VP248PE	The High-Definition IP-Phone with 6 Lines x 24 Characters LCD Display and PoE
SETU VP248SE	The High-Definition IP-Phone with 2 Lines x 24 Characters LCD Display and PoE
SETU VP248P	The High-Definition IP-Phone with 6 Lines x 24 Characters LCD Display
SETU VP248S	The High-Definition 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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