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Warm Welcome

Matrix SETU VFXTH

Multi-port VoIP-FXO-FXS Gateways

Introduction

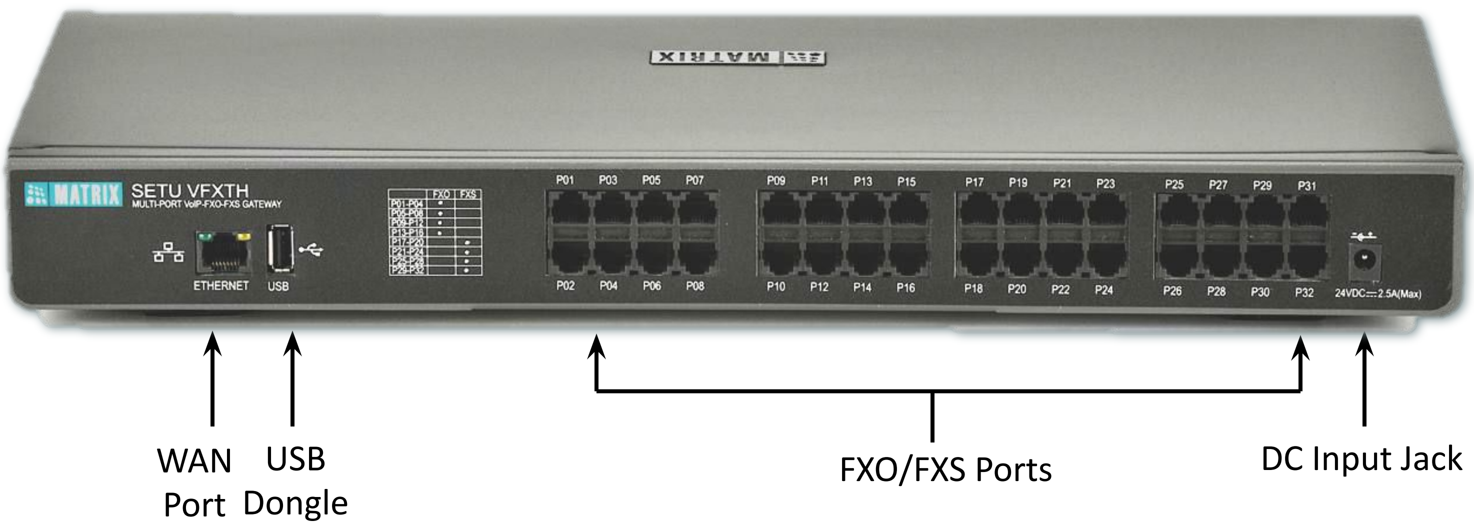
SETU VFXTH

Multi-port VoIP-FXO-FXS Gateway

- Configurations up to 32 FXO/FXS ports
- Seamless connectivity between VoIP and PSTN network
- Gateway for interfacing with existing IP-PBX or TDM PBX

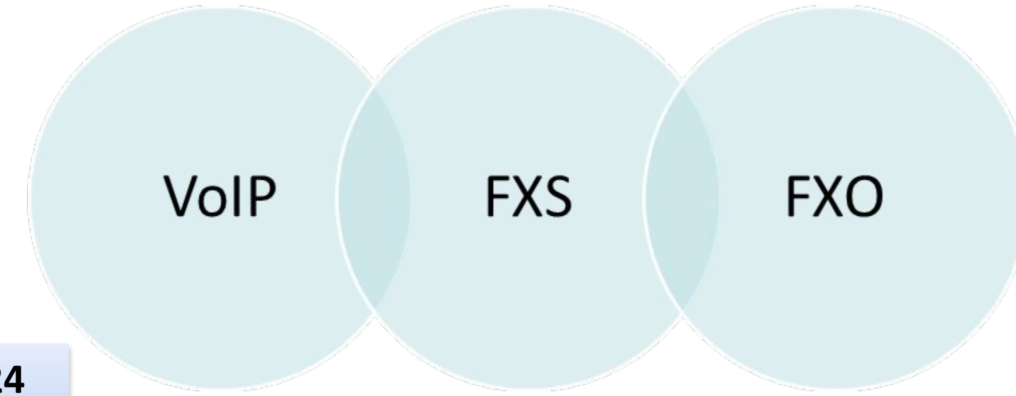
Multi-port VoIP-FXO-FXS Gateway

Interfaces



Multi-port VoIP-FXO-FXS Gateway

Product Variants



SETU VFXTH0016
16 VoIP Channels
16 FXS

SETU VFXTH0024
24 VoIP Channels
24 FXS

SETU VFXTH0032
32 VoIP Channels
32 FXS

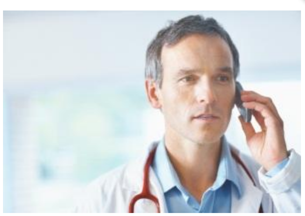
SETU VFX3200
32 VoIP Channels
32 FXO

SETU VFXTH0808
16 VoIP Channels
8 FXO 8 FXS

Target Customers

Corporate Branch Offices

Organizations with Field Staff

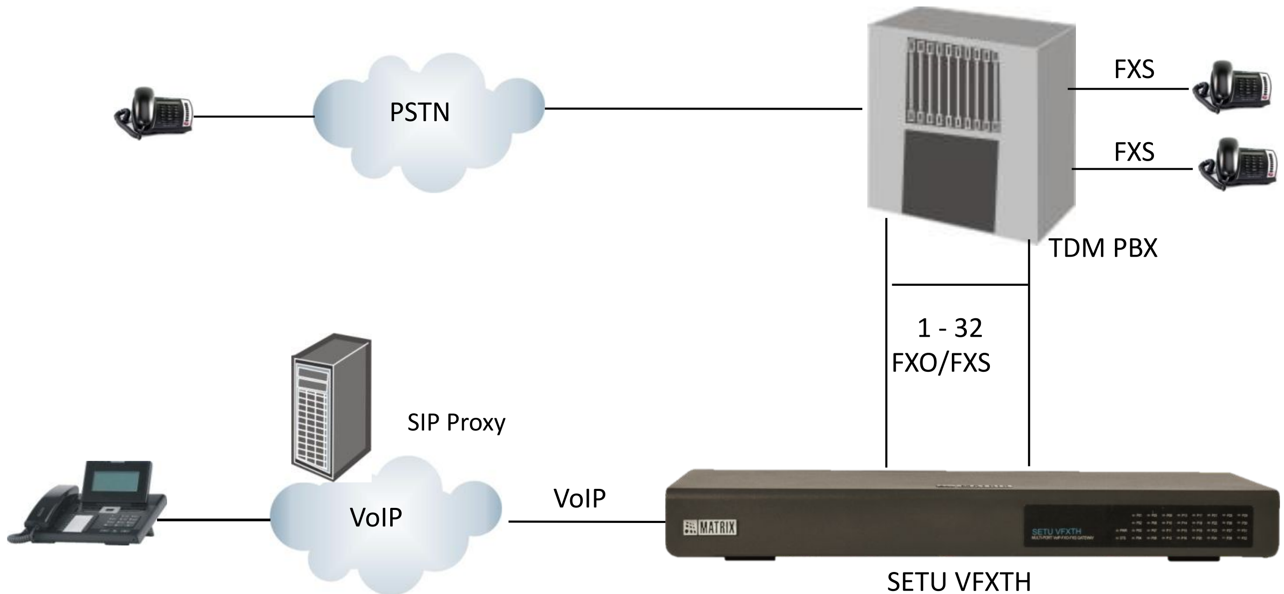


Remote Project Sites

Call Centers

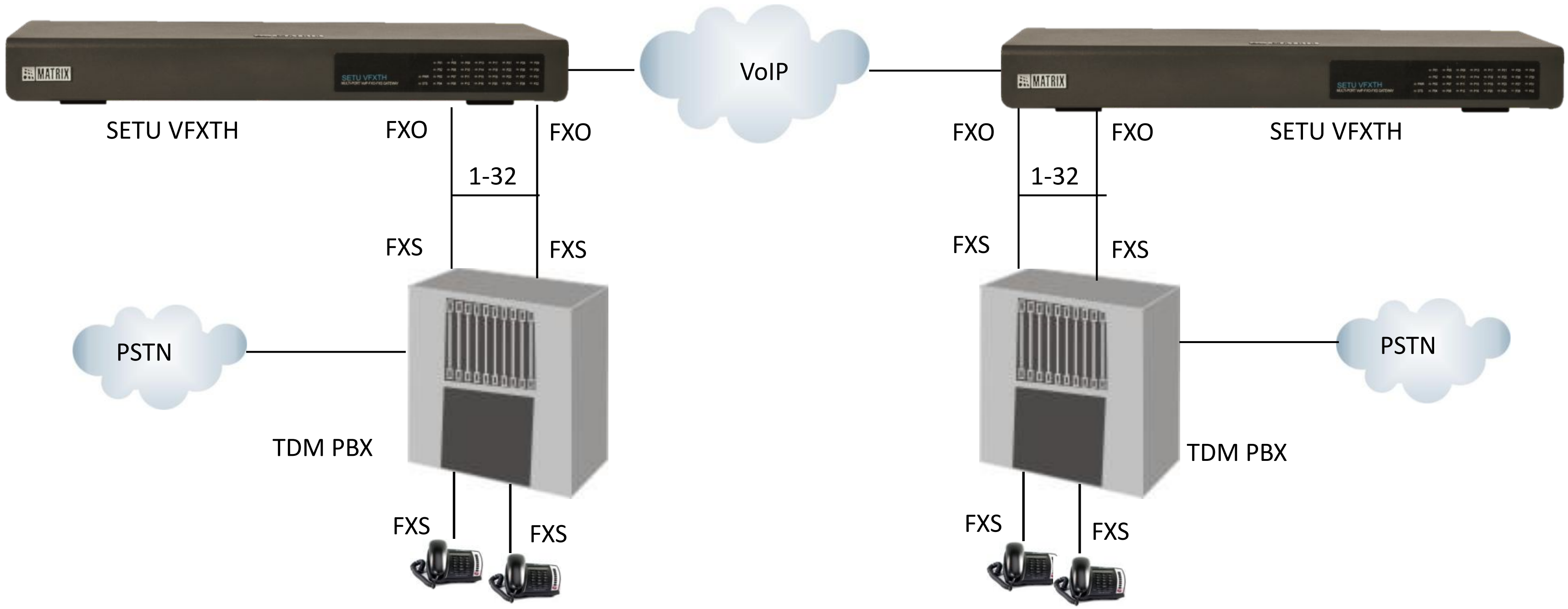
Applications

VoIP Gateway for TDM PBX



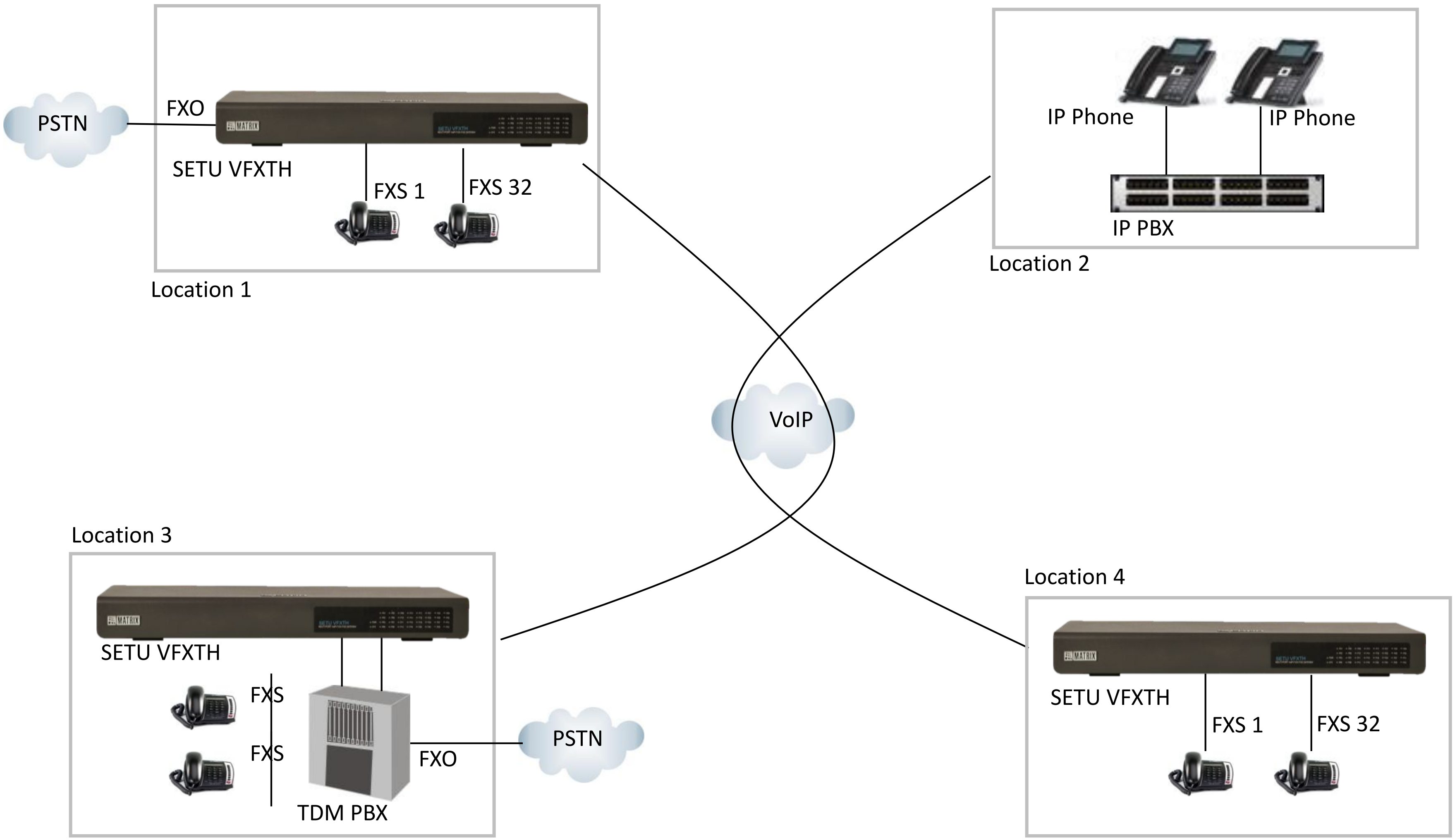
PSTN Gateway for IP-PBX

Peer-to-peer Application



- Allows making and receiving calls over IP network without the help of a proxy server (Internet Telephony Service Provider)
- Program destination number and destination address in the peer-to-peer table
- Number can be of minimum 1 and maximum 24 digits
- Useful for inter-branch or intra-office communication
- Require internet connection with fixed IP Addresses

Multi-Site Connectivity



Key Features

Allowed and Denied List

- Can be programmed to allow or deny numbers from being dialed
- Avoid misuse and restrict unproductive calls
- Very useful feature to control cost
- Can be programmed separately for VoIP, FXO and FXS ports
- 24 Number lists can be programmed
- 64 Entries per list are supported

Automatic Number Translation

- Translates full number or part of a dialed number to match with numbering plan of the destination network
- Automatic Number Translation (ANT) is supported on VoIP, FXO and FXS ports
  - ✔ User dials 001-xxxx to reach a number in the USA, but as the ITSP understands 1-xxxx so ANT replaces the number 001-xxxx with 1-xxxx to let the ITSP understand the dialed number string

Call Detail Record

- Call details can be generated using various filters
  - ✔ Calls originated from FXO, FXS and SIP ports
  - ✔ Calls terminated on FXO, FXS and SIP ports
  - ✔ Calls made between with date, day and time
  - ✔ Calls with and without PIN authentication number
  - ✔ Called and calling party numbers matching with numbers list
  - ✔ Call duration with hour, minutes and seconds
- Call details of 2000 calls can be stored in the buffer
- Call details can be downloaded to a computer

Caller Line Identification

- Display calling party's name and number on FXS port
- Supports CLI on FXO and SIP ports
- CLIP on call transfer
- Supported CLI
  - ✔ DTMF
  - ✔ V.23 FSK
  - ✔ Bellcore FSK

Call Progress Tones and Rings

- Different countries have different tones to indicate the process of call activities such as Dial Tone, Ring Back Tone, Error Tone, Busy Tone etc.
- Specific cadence can be programmed to match the tone used in each country
- Call Progress Tones can be programmed country wise or can be customised as per the user requirement

Conference

- Built-in feature
- Total Eight 3-Party conference
- Applicable if the call is originated from and terminated on FXS port
- External and internal parties are possible

Day Light Saving

- Daylight saving is the procedure of setting clocks ahead at a particular time of year to make way for additional hour of daylight in the evening
- Real Time Clock (RTC) moves backward or forward automatically in tune with the daylight saving requirement of the country
- DST can be forwarded or set backward according to day-month wise or date-month wise as per the requirement

Do Not Disturb

- It is applicable on FXS ports
- Do Not Disturb (DND) feature enables user to have privacy of not receiving the calls for particular time period
- Outgoing calls can be made when Do Not Disturb (DND) is enabled

Digest Authentication

- An industry standard for web authentication
- Used to authenticate a caller in peer-to-peer network
- Allows a group of devices to make and receive calls among them
- Authentication password is to be pre configured in all the devices
- Program such 500 entries in digest authentication table

Emergency Number Dialing

- Allows the caller to contact local emergency services like

✔ Police

✔ Fire Station

✔ Hospital

- Maximum 5 emergency numbers can be programmed
- Dialed out using pre-assigned port

Fax over IP

- Send and receive FAX using VoIP and PSTN network
- VoIP to PSTN Fax is possible if supported by ITSP
- Send and receive Fax over IP using
  - ✔ T.38 (RTP)
  - ✔ T.38 (UDPTL)
  - ✔ Pass-Through

Hotline

- Get connected to a pre-defined destination on picking up the receiver
- Hotline is supported on FXS ports
- Each FXS port can have different hotline number
- Provision to impose delay before hotline activation
- Delay timer can be minimum 1 and maximum 9 seconds

Least Cost Routing

- Different service providers have different tariffs for different regions and countries
- System automatically select the most economical route to place the call
- Route selected depends on the destination number dialed
- Ensure each call at least possible cost

PIN Authentication

- PIN Authentication is to authenticate a caller to prove identity to proceed the call from one network to another
- Supported on FXO and VoIP ports
- Support 500 PIN authentication entries
- Very useful feature to avoid misuse of the services

Real Time Clock

- Date and time is very important parameter for some features like call detail record, daylight saving time etc.
- Real Time Clock uses the Simple Network Time Protocol (SNTP) to get time from the time server
- Flexibility to choose one from three pre-configured free reliable time servers
- Flexibility to program time server address of their preference

System Log

- System Log protocol is used for sending debug messages on IP network
- It is a Client/Server protocol
- Uses UDP as transport protocol for debugging process
- Logging has several benefits
  - ✔ Easier and faster troubleshooting
  - ✔ Security enhancement
  - ✔ Better system administration

Universal Call Routing

- SETU VFXTH offers network connectivity like VoIP, FXO and FXS ports
- Calls can be originated and terminated on any port type
- Calls originated are routed to the destination as per programmed routing mechanism
- More than one destination port can be programmed for each source port
- Three types of routing groups are supported

TR-069 Auto-Configuration

- TR-069 is a Secure Remote Management Protocol
- It is used for Auto-configuration and Remote management of the device at customer premises from an Auto-Configuration Server
- It supports a variety of functionalities:

✔ Dynamic Service Provisioning

✔ Firmware Management

✔ Status and Performance Monitoring

✔ Remote Diagnostics

Feature List

- Allowed and Denied Numbers List
- TR-069 Auto-Configuration
- Automatic Number Translation
- Auto Provisioning for Mass Deployments
- Call Detail Records
- Call Progress Tone and Rings
- CLI Based Call Routing
- Dynamic DNS
- Digest Authentication
- Do Not Disturb
- Emergency Number Dialing
- Hotline
- PCAP Trace
- Message Wait Indication
- Block ICMP and PING
- SNMP Monitoring
- SRTP/TLS over SIP
- System Log Client
- VLAN Tagging
- Web based Management
- PBX Functionality
  - ✔ Call Wait
  - ✔ Call Hold
  - ✔ Call Transfer
  - ✔ Call Forward
  - ✔ Call Pickup
  - ✔ Conference

Specifications

Maximum Number of VoIP Channels	32 Channels
Maximum Number of FXO Ports (RJ11)	32 Ports
Maximum Number of FXS Ports (RJ11)	32 Ports
WAN Port (Ethernet Port)	1 Ethernet Port
Power Supply	External Adaptor 24V DC/2.5A
Power Consumption	60 W (Typical)
LED Indications	1 GREEN colour LED for Power, 1 Dual Colour LED for Status, 32 Single Colour LEDs for each Port
Dimensions (W x H x D)	40.7 X 5.1 X 17.2 Cm (16.0” X 2.0” X 6.8”)
Installation Mounting	Table-Top, Wall and Rack Mount

Matrix VoIP Product Range

PRODUCT	DESCRIPTION
SETU ATA	VoIP Adaptor with GSM, FXO and FXS Ports
SETU VGFX	Multi-port SIP based VoIP to GSM/3G-FXO-FXS Gateway
SETU VGB	Multi-port SIP based VoIP to ISDN BRI Gateway
SETU VTEP	SIP based VoIP to T1/E1 PRI Gateway
SETU VFX	Low-Density Multi-port SIP based VoIP to FXO-FXS Gateway
SIMADO GFX	Multi-port GSM to FXS Gateway
SIMADO GBR	Multi-port GSM to BRI Gateway

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We put
more
in the box

so your
business
can think
more
out of
the box.

Thank You.

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