Warm Welcome

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

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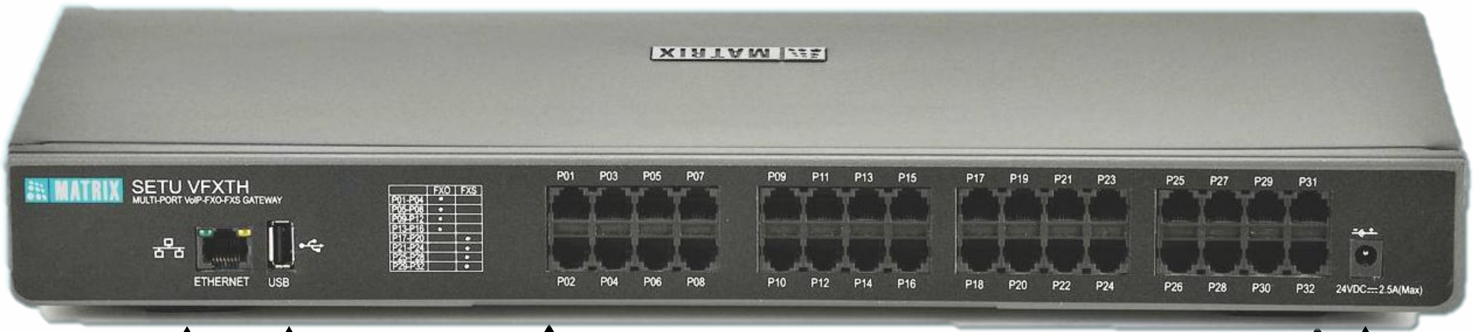
Multi-port VoIP-FXO-FXS Gateway

Interfaces



WAN Port

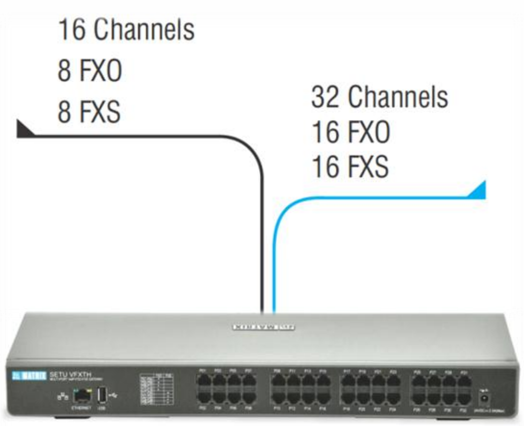
USB Dongle

FXO/FXS Ports

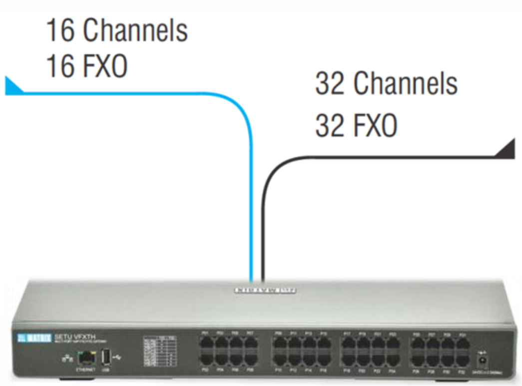
DC Input Jack

Multi-port VoIP-FXO-FXS Gateway

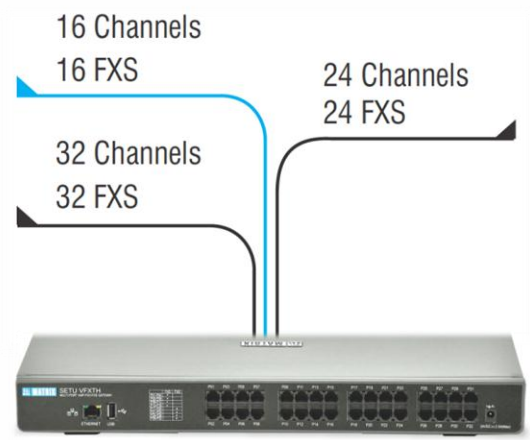
Variants



FXO & FXS Ports



FXO Ports



FXS Ports

Target Customers

Corporate Branch Offices

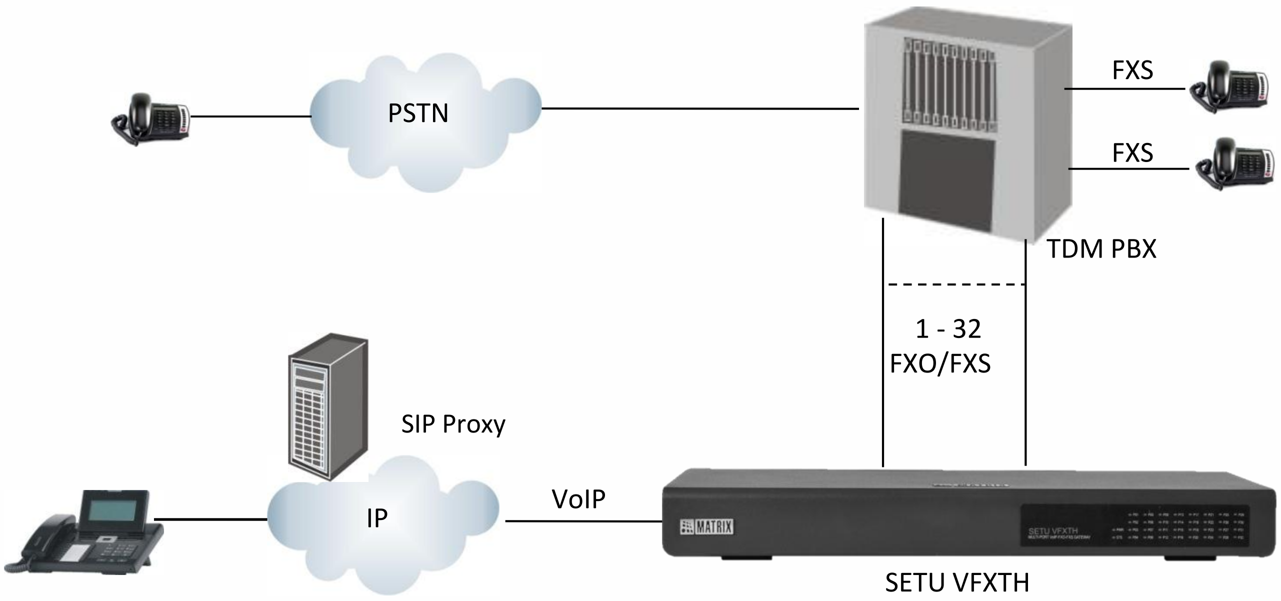
Organizations with Field Staff

Remote Project Sites

Call Centers

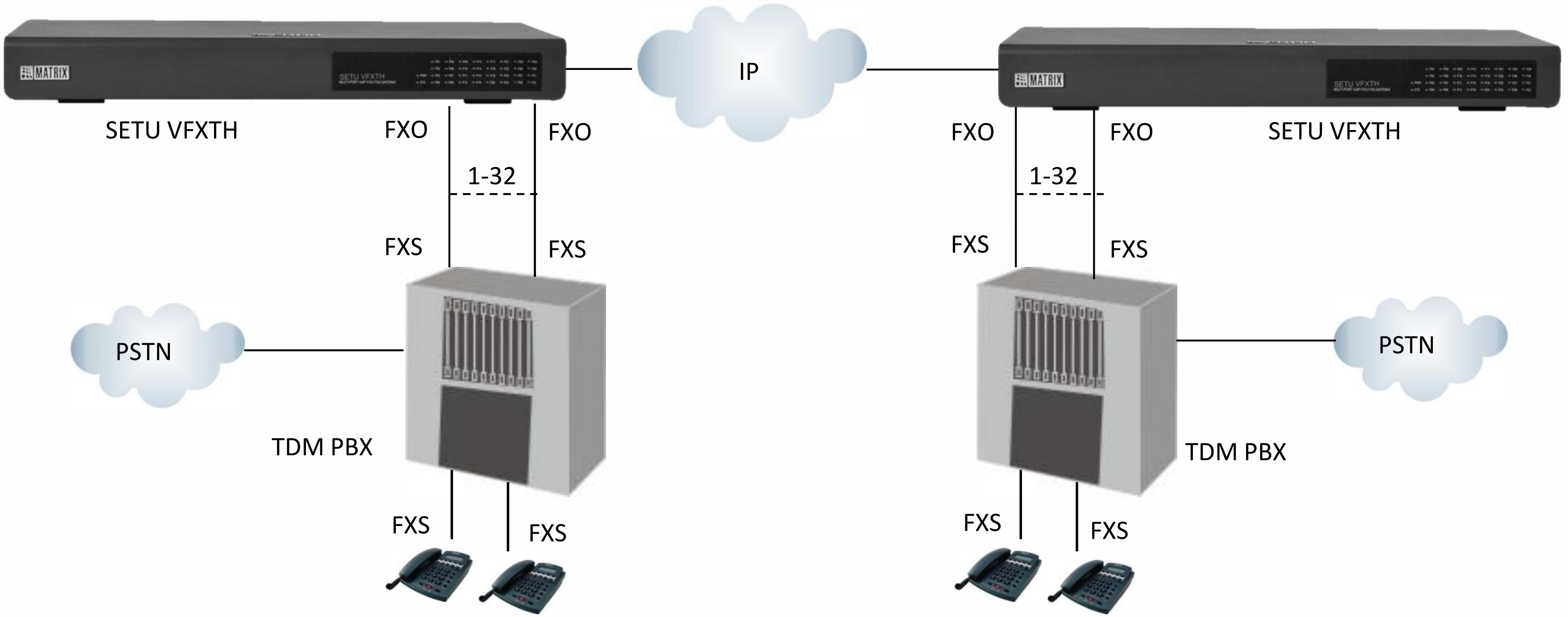
Applications

VoIP Gateway for TDM PBX

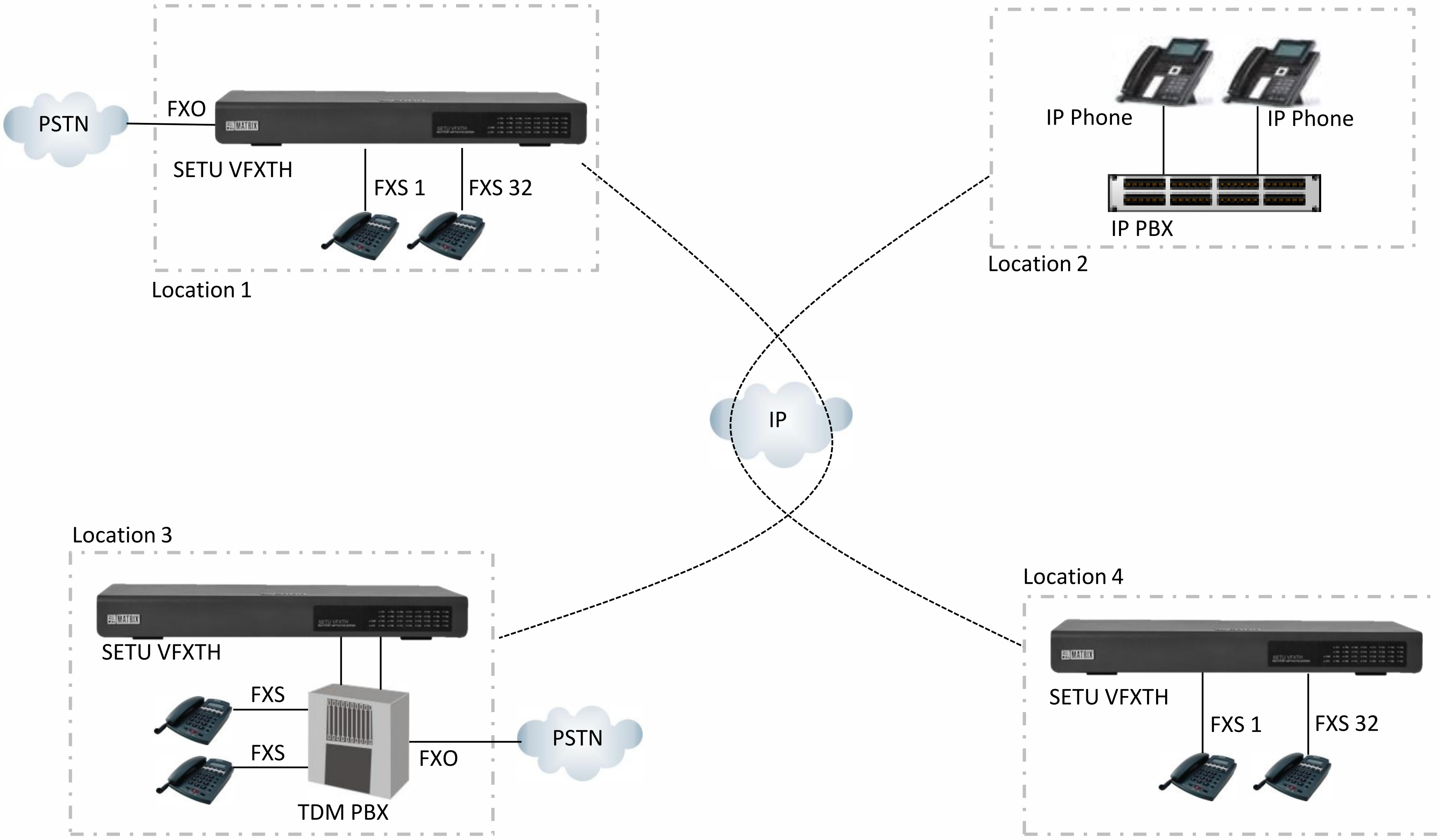


PSTN Gateway for IP-PBX

Peer-to-peer Application



Multi-Site Connectivity



Key Features

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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
- 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 IP 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

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- 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

Peer-to-peer Calling

- 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

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

Feature List

- Allowed and Denied Numbers List
- 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
- SNMP Monitoring
- SRTP/TLS over SIP
- System Log Client
- VLAN Tagging
- Web based Programming
- 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 Watt (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 GSM/3G-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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Thank You

- Type of Presentation: Product Presentation
- Number of Slides: 34
- Revised On: 1st March 2013
- Version-Release Number: V1R2

For Further Information Please Contact:

Email ID: Info.Telecom@MatrixComSec.com

Visit us at www.MatrixComSec.com