SETU VTEP

VoIP to T1/E1 PRI Gateway

Modern organizations are recognizing VoIP as the cost-effective alternative to the wide-spread ISDN networks. Enterprises are inclined towards availing benefits of IP while protecting their existing investment against changing communication technologies. They seek solutions to avail benefits of both IP and ISDN trunking through a single device to optimize operational expenses and save on long distance calls.

Matrix SETU VTEP is a compact and dedicated gateway for VoIP to T1/E1 PRI network offering high-value communication experience to businesses of all sizes, Service Providers, Call Centers and simple but cost-effective solution for multi-location branch office communication. This intelligently designed gateway incorporates advanced features with multiple connectivity options to connect with a legacy communication system using T1/E1 or PRI signaling. SETU VTEP offers reliable and cost-effective solutions to the changing requirements of the business communication and offer customer value for money.

MATRIX
TELECOM SOLUTIONS

SETU VTEP - OVERVIEW

Matrix SETU VTEP is an out-n-out, compact and feature-rich VoIP to T1/E1 PRI gateway which installs between ISDN PBX and T1/E1 line to connect PBX users to IP network for cost-effective communication. It also provides T1/E1 trunking to an IP-based system. The gateway connects to an existing PBX system using VoIP, T1/E1 or PRI Signaling. SETU VTEP enables multi-locational organizations to avail benefits of VoIP network for making cost-effective inter-office calls.

SETU VTEP is an in-line device with 32 VoIP Channels, FoIP support, Programmable TE/NT mode for T1/E1 and advanced voice codecs support. The dedicated signal processing resources and superior protocol sets ensure multiple call capabilities with toll-grade voice quality.

The gateway is specifically designed for SMBs, Large Enterprises, VoIP Service Providers and System Integrators to offer smooth migration alternative to new-age IP telephony from legacy T1/E1 network. It helps them to control the communication overheads and realize an earlier return on investment through advanced features. Multiple mounting options and remote management through web-based console add to the operating ease of this intelligently designed compact gateway.

SETU VTEP - AT A GLANCE

DEVICE CAPACITY
30 Simultaneous VoIP to T1/E1PRI Calls

DEDICATED IN-LINE CONNECTIVITY
For IP-PBX and ISDN PBX

FAX OVER IP (FoIP)
T.38 and Pass-Through

SUPERIOR VOICE QUALITY
Echo Cancellation
Synchronized Network Clock

REMOTE MANAGEMENT
Web based Device Configuration

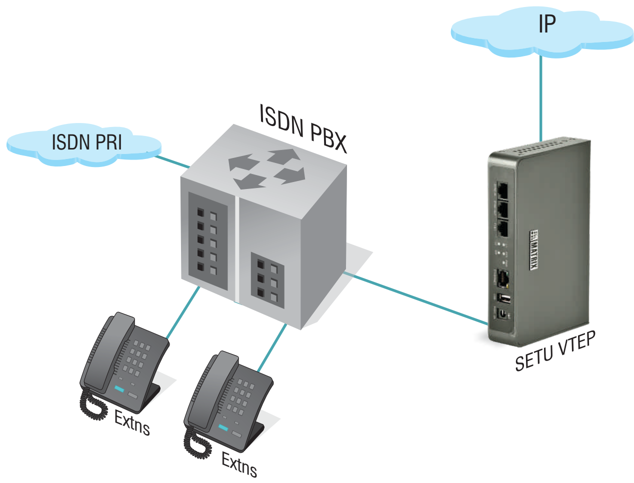
PLUG-N-PLAY DEVICE
Fast and Easy Installation

COMPACT DESIGN
Wall, Table-Top and 19” Rack Mountable

FLEXIBLE INSTALLATION
Programmable TE/NT

SETU VTEP APPLICATION

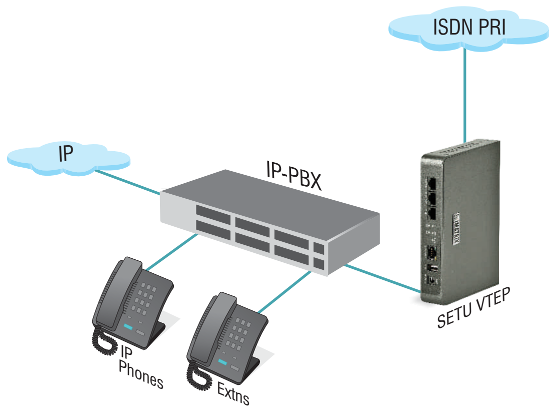
VoIP Access for ISDN PBX



SETU VTEP transparently integrates with existing ISDN PBX using T1/E1 or PRI signaling to provide VoIP connectivity to existing PBX users. Programmable TE/NT modes ensure a hassle-free integration. Dedicated ports for network clock sync avoid synchronization errors and guarantee superior voice quality.

SETU VTEP, connected in-line between the PBX and the IP network, allow the callers to place calls over the cost-effective IP network. Features like Automatic Number Translation and Programmable Access Codes simplify the dialing patterns and enhance user experience. Modern organizations can thereby smoothly migrate to IP telephony without the need to invest in costly VoIP system with the added advantage of lowered operational cost.

Digital Network Access for VoIP Systems

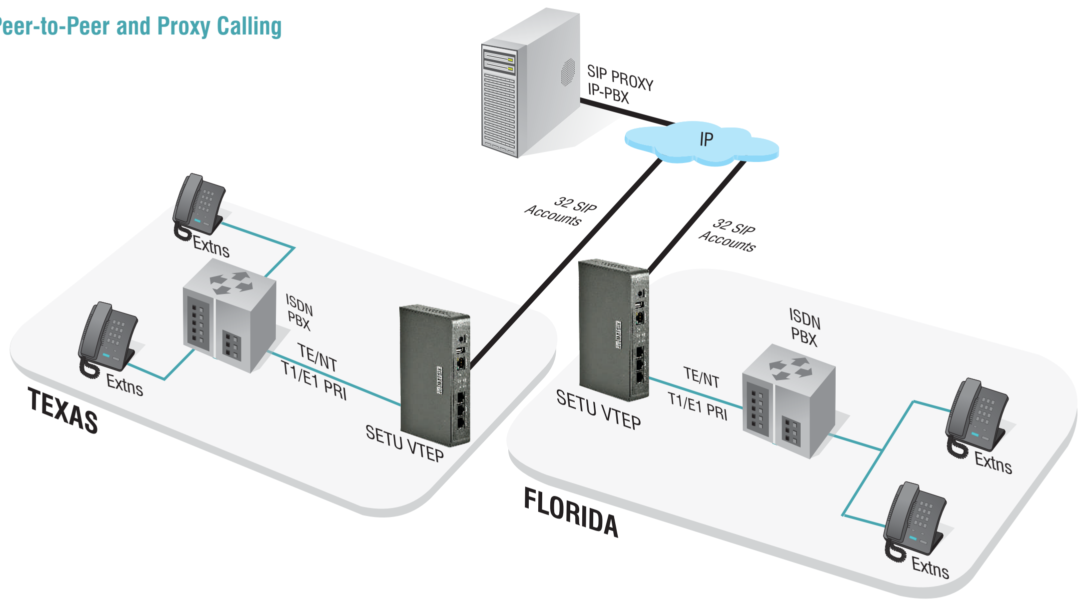


SETU VTEP connects an IP-PBX system to digital T1/E1 or PRI network. It enables the IP-PBX users to place calls over wide-spread ISDN PRI network. SETU VTEP can be connected directly through the LAN or remotely over the public IP network.

MULTI-SITE CONNECTIVITY OVER IP

With continuous expansion of business horizons across geography, organizations tend to have more local footprints. Matrix SETU VTEP facilitates easy and low cost communication between these geographically distant branch offices over cost-effective IP network.

Peer-to-Peer and Proxy Calling



Advantages

- 30 Concurrent Calls
- Register with 32 SIP Accounts
- Simultaneous Proxy and Peer-to-Peer Calling

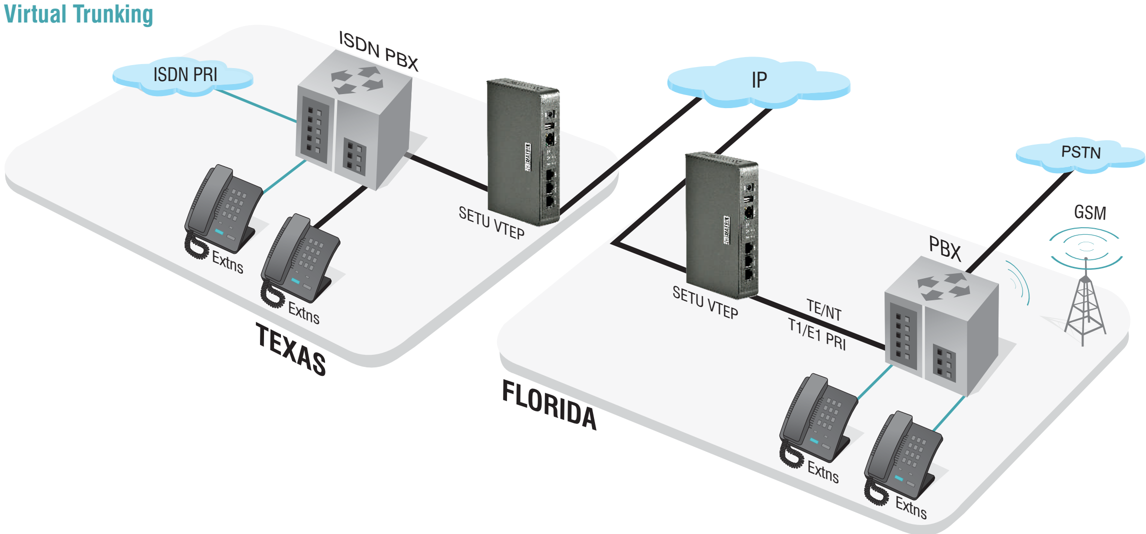
Multisite Connectivity over IP

- Registration with Multiple SIP Proxies
- Connect to Remote Location IP-PBX as a SIP Client
- 500 Direct Dial Access Codes for Peer-to-Peer Calls

To eliminate the enormous communication cost incurred while placing long distance calls between the branch-offices, organizations can utilize cost-effective VoIP network. SETU VTEP supports peer-to-peer calls between the distant locations without going through any proxy server. It establishes easy and direct communication path between the offices, utilizing existing IP connectivity, avoiding charges incurred while routing calls via the ITSP's Proxy. Else, SETU VTEP can also get registered with a SIP proxy and thereby utilize the services offered by an ITSP.

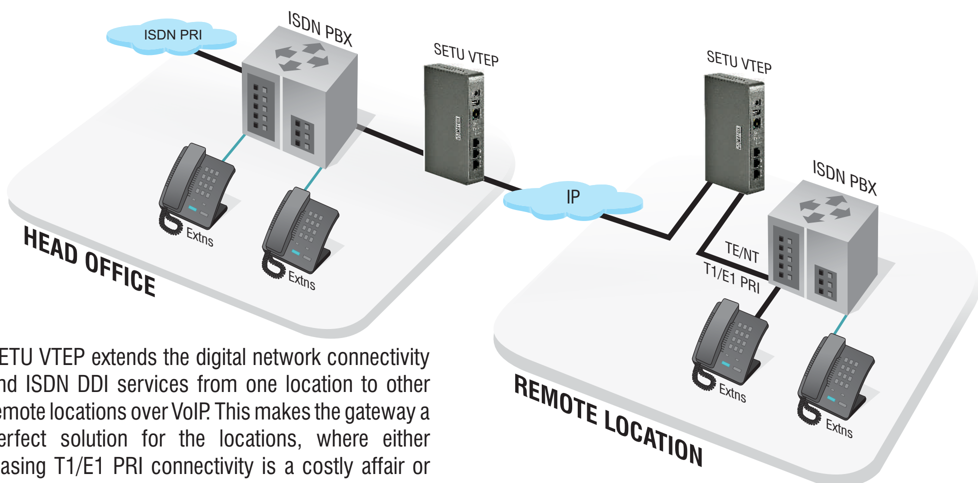
Multi-site organizations can place an IP-PBX at the head office location and simply install SETU VTEP at various branch-offices. SETU VTEP registers as a client to this central IP-PBX. This extends the IP-PBX connectivity among various branch-offices. The remote ISDN terminals can leverage the features and resources of the central IP-PBX system. This also eliminates the cost and complexity of installing IP-PBX at various branch locations.

Virtual Trunking



SETU VTEP enables users of one branch-office to virtually access trunk connectivity of the other branch office. SETU VTEP routes the call over cost-effective IP network till the last mile of termination and facilitates users at one location to access the local ISDN connectivity of remote branch-office to further reduce inter-network call charges.

Extending Digital Connectivity over IP



SETU VTEP extends the digital network connectivity and ISDN DDI services from one location to other remote locations over VoIP. This makes the gateway a perfect solution for the locations, where either leasing T1/E1 PRI connectivity is a costly affair or availability of fixed network is intricate.

SETU VTEP with its built-in intelligence routes an incoming DDI call on the ISDN network at one location to the desired extension of a remote location, extending communication seamlessly over the IP network. The gateway also allows these remote extensions to make outgoing calls using the ISDN trunk. Furthermore, Reverse DDI functionality of SETU VTEP reroutes any callback received on the ISDN trunk, to the specific DDI extension (local or remote), avoiding operator intervention.

KEY FEATURES

Allowed and Denied Number Lists

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

Automatic Number Translation (ANT)

SETU VTEP supports simultaneous registration of multiple SIP accounts. These accounts can be availed from single or multiple ITSPs. While placing a call, a caller is not conscious of the routing logics defined and the SIP account in use. The gateway itself modifies the dialed number or part thereof so that it matches with the numbering plan that is understood by the ITSP whose trunk is used for the specific call.

Call Detail Records (CDR)

SETU VTEP 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.

Caller-ID Based Routing

Based on the Caller-ID details, an incoming call can be routed to a pre-defined port. For example, important business calls may be directed to the higher authorities, calls with specific Caller-ID may be directed to specific departments, while calls from anonymous numbers may be directed to the customer support teams. Users with Digital key phone or IP phone can also have display of the caller's name, if programmed accordingly.

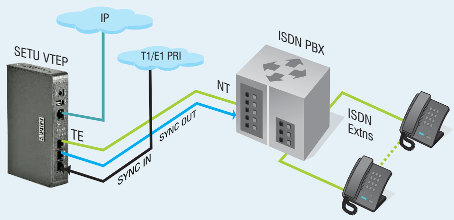
Call Progress Tones and Rings

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.

Network Clock Synchronization

Whenever the PBX system is interfaced between ISDN T1/E1 line and a communication device like VoIP gateway, chances of clock slip exists as PBX and gateway operate on different clock sources. As a consequence of clock slip, many operational and integration problems like poor voice quality and noisy or dropped calls occur during field installations.

Traditional ISDN network provide extremely accurate clock signals. SETU VTEP is equipped with SYNC IN and SYNC OUT ports that acquires stable network clock and uses it to provide clock synchronization with the attached PBX system, eliminating any synchronization issue and noisy or dropped calls.



Digest Authentication

Digest authentication allows an incoming call on the gateway to be screened using encrypted authentication keys. Only on successful authentication, the gateway allows the call to be established. The automated security mechanism allows restricting malicious calls. Calls received are thereby restricted to a number of trusted callers.

Fax over IP (FoIP)

SETU VTEP can be used to send and receive Fax using a SIP account, over the IP network. The gateway supports FoIP using T.38 Vocoder and Pass-Through. Sending Fax over Internet eliminates the need of dedicated analog lines to send fax over long distances.

Dynamic DNS (DDNS)
The compact gateway comes with an embedded Dynamic DNS client that automates the discovery and registration of its IP addresses on the public network. The remote administrator and peer-mode devices can thereby connect to the gateway using the Domain Name associated with the dynamic IP. The gateway can thereby seamlessly establish calls even when allocated a dynamic IP.

Multiple SIP Accounts
32 SIP accounts can be programmed and each call placed can use a specified trunk. Dynamic allocation of SIP account is also possible via dial plans, based on various call routing algorithms.

NAT and STUN
NAT allows multiple devices in a LAN to share a single public IP addresses and automatically creates a firewall between the internal network and the internet. The STUN client allows the IP terminals located behind a NAT to obtain the mapped (public) IP address and port number, allocated for connections to a remote host. The users can thereby connect to the gateway, hidden behind the NAT router/firewall. The STUN support is critical to establish a VoIP call between SIP users, located behind different type of NATs.

Peer-to-Peer Calling
SETU VTEP supports VoIP calls between two locations without going through a proxy server. IP addresses of the locations can be programmed in its Peer-to-Peer table. 500 such entries can be programmed. Short, numeric dialing codes can be defined for calls between these locations. With the embedded Dynamic DNS client, calls can also be established between devices on dynamic Public IP. Organizations having multiple locations like branch offices and factories can use this feature to provide direct dialing between these end-points. Since the Peer-to-Peer calls are placed over the public IP network, the call cost is minimal.

PIN Authentication
Incoming calls on PRI and SIP trunk can be authenticated before the call lands on the gateway. Up to 500 users can be provided with user IDs and passwords for such authentication. The caller has to first prove his authentication. Once authenticated, the caller can then place call using the system ISDN or IP trunk line. Such verification avoids the possibility of malicious calls and misuse of system resources. The feature can be selectively enabled on all or a set of trunks.

Return Call to Original Caller (RCOC)
A call attempt may be unsuccessful if the called party is busy, does not pick up the call or due to any network issues. SETU VTEP logs such unsuccessful calls in a RCOC table with details about the caller, number dialed and time of call. With these details available, if a call back is received from any of the called number, it is possible to route the call to the specific caller who attempted a call to the concerned number. This greatly reduces communication delay and also eliminates the need for operator assistance to redirect the call.

Remote Programming
Matrix SETU VTEP 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.

Syslog Client
Syslog is a protocol used extensively for sending debug messages on IP network. Logging has several benefits which include easier and faster troubleshooting, improved security and a better system administration. Debug messages are sent to remote server on IP network for finding and reducing the number of bugs or defects from a system.

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. With VLAN tagging, the gateway receives packets only tagged for it. The packets sent by the gateway are also given specific tag. VLANs increase overall network performance by grouping users and resources that communicate most frequently with each other.

Quality of Service (QoS)
Layer3 QoS prioritizes voice packets over IP network, as voice traffic is delay sensitive. Toll-quality echo cancellation for a tail length of 128ms can be done on a per channel basis. CNG, in conjunction with VAD algorithms, quickly detects absence of audio and conserves bandwidth by preventing the transmission of these silent packets.

ADVANTAGES

COST SAVING
- Low Cost VoIP Calls
- Peer-to-Peer Calling

ENHANCED SCALABILITY
- 32 SIP Accounts
- Expand Existing PBX Capacity and Capability

EXTENDED REACH
- Multi-site & Multi-branch Connectivity
- Dynamic DNS Support

FLEXIBILITY
- Programmable TE/NT
- ISDN Network Access for IP-PBX
- Simultaneous Proxy & Peer-to-Peer Calling

RELIABILITY
- Echo Cancellation
- Various Voice Codec Support

INVESTMENT PROTECTION
- Maintain Existing Infrastructure

TARGET CUSTOMERS
- Small and Medium Businesses
- Large Enterprises
- VoIP Service Providers
- System Integrators

RESOURCES

Model Name	SETU VTEP
SIP Accounts	32
Main Port	1
Sync-in Port	1
Sync-out Port	1
USB Host Port	1
WAN Port	1

MATRIX GATEWAY RANGE OF PRODUCTS

ETERNITY	The Universal Telephony Gateway
SETU VGFX	Multi-Port SIP based VoIP to GSM-FXO-FXS Gateway
SETU VGB	Multi-Port SIP based VoIP to GSM and BRI Gateway
SETU VTEP	SIP based VoIP to PRI Gateway
SETU VFXTH	Multi-Port SIP based VoIP to FXO-FXS Gateway
SETU VFX	Multi-Port SIP based VoIP to FXS Gateway

TECHNICAL SPECIFICATIONS

VoIP	
VoIP Protocols	SIP v2, SDP, RTP (RFC 2833)
Network Protocol	IPv4,TCP, UDP, DHCP, SNTP, STUN, HTTP
SIP	32 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 ,iLBC
Line Echo Cancellation	G.168 with variable 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 UDPTL and Pass-Through
Quality of Service	Layer 3 DiffServ and ToS
Data Network	1 WAN Port RJ45 Auto MDIX 10/100 BaseT
Security	Password Protected Administration

ISDN PRI	
Channels	T1 – 23B+D, E1 – 30B+2D
Personality	Network (NT) and Terminal (TE)
Line Coding	T1 RBS – AMI/B8ZS, E1 CAS – HDB3
Switch Variant	T1 RBS – AT&T 5ESS, DMS-100, US Ni2 E1 CAS – ETSI NET5, ITU-T Q.921, ITU-T Q.931
Line Signaling	T1 RBS – FXS/FXO Loop Start, FXS/FXO Ground Start, E&M (Immediate Start, Wink Start) E1 CAS – ITU-T Q.400 – Q.490
Framing	T1 RBS – SF-D4/ESF E1 CAS – CEPT1 (with/without CRC) with CAS MF
Protection	Over Voltage and Over Current, Built-in Secondary Protection

ENVIRONMENTAL	
Operating Temperature	: -10ºC to +50ºC (14ºF to 122ºF)
Operating Humidity	: 5-95% RH, Non-Condensing
Storage Temperature	: -40º C to + 85ºC(-40ºF to +185ºF)
Storage Humidity	: 0-95% RH, Non-Condensing

MECHANICAL	
Dimensions (W x H x D)	161.50 x 101.25 x 30.30mm (without Side Clamps) 483 x 101.25 x 30.30mm (with Side Clamps)
Unit Weight	0.55 Kg (without Side Clamps) 0.61 Kg (with Side Clamps)

POWER SUPPLY	
Input	: 5V DC, 2A
Power Consumption	: 5W(Maximum)

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.

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