

SAPEX

The All-in-One Embedded IP-PBX Server

Integrating or migrating to new-age IP telephony is a much crucial decision, especially for small and medium organisations. These organizations need to be more agile and dynamic with limited resources. The right communication solution should not balance features for affordability. Matrix SAPEX is a family of pure IP-PBXs, engineered to bring IP telephony to the SMB and SME premise. The embedded platform integrates a SIP Proxy, Registrar and Presence server in a compact hardware platform. SAPEX users can place voice and video calls over the IP network. Besides centralized call control and media relay services, auto-attendant, voice mail, all PBX telephony features and presence based services are available to SAPEX users. The system employs open-standard SIP protocol and is hence interoperable with SIP proxies, gateways and IP phones. Built-in RADIUS and SMTP client allow advanced services to be integrated.

The multi-functional IP-PBX delivers high performance. Simplified management, reduced communication cost, seamless connectivity with remote users and between geographically dispersed branches, advanced communication means and enhanced productivity are apparent benefits. Communication of small and mid-sized enterprises as well as geographically distributed offices, remote workers and contact centre is much simplified and enhanced with SAPEX.



The All-in-One Embedded IP-PBX Server

SAPEX is a family of fully integrated IP-PBXs, with embedded Registrar, Proxy, Presence and Voice Mail Servers. The otherwise distributed servers, are embedded within the single server platform.

REGISTRAR SERVER

When a user comes online, they get registered with the embedded registrar server. The registrar authenticates the users, stores their information and maintains real-time status of devices used by them for communication.

PROXY SERVER

SAPEX acts as a central proxy server providing various call control features. It can also register as a client to a SIP proxy server. Registration with multiple SIP proxies is supported. This functionality of the IP-PBX allows services offered by the ITSP to be shared among the registered users of the IP-PBX. Flexibility to register with multiple SIP service providers, offers redundancy, allowing switching to the alternate trunk in case the trunk goes down. Calls from system users can be routed through a specific SIP trunk as per predefined dialing plans.

PRESENCE SERVER

The Integrated presence server maintains and distributes presence information of users registered with the IP-PBX. Presence status helps to determine a user's availability and preferred mode of communication, even before a communication session is initiated. Presence notifies that a friend is available to talk and then uses SIP messages to negotiate the means of communication and establish the actual communication session. Further, Instant Messaging (IM) is a popular mode of real-time communication. The capacity to maintain the status of a user at all instant enables IM sessions to be established between the SIP extensions, which leverage presence and instant messaging. Knowing a user's status, it is possible to reach a right contact, in right time and on the right terminal.

VOICE MAIL SERVER

The built-in Voice Mail functionality ensures that a user does not miss upon important calls, by forwarding the calls to his Voice Mail box as and when required.

- **Configurable Voice Mail Size**

The Voice Mail size can be configured individually for all the 500 users. A default 4 GB USB stick supports up to 18 hours of recording. The size of Voice Mail can be increased, replacing the USB stick with one having a higher storage.

- **Voice Mail Retrieval**

Voice Mails can be retrieved by calling to Voice Mail server using individual access codes. A Voice Mail can be stamped for its date and time of arrival and can be notified to the user at the time of retrieval.

- **Event Notifications**

The server can be programmed to delete the voice mails dumped in a user's mail box, at a scheduled time. A user can also receive a notification/indication for any new mail arriving in his mailbox or when a mail is either retrieved or deleted from the mailbox. In this case the SIP terminal should leverage the same.

- **Email Notifications**

The integrated SMTP client functionality, enables notifying the user for any new mail as well as the capacity status of the mailbox via an Email. The Voice Mail can also be delivered as an attachment to the user's Email-ID.

SAPEX VARIANTS

SAPEX SDM - Up to 200 IP Users

SAPEX DDM - Up to 500 IP Users



KEY DIFFERENTIATORS

Back-to-Back User Agent (B2BUA)

SAPEX works as a Back-to-Back User Agent (B2BUA). Basic telephony services like Call Forwarding, Call Transfer necessitates call management and tracking for entire call duration. A SIP server with B2BUA becomes an active participant in a SIP call, enabling many advanced services in addition to these basic telephony services. A B2BUA-enabled SIP server maintains the call state for the entire call duration, enabling support for real-time call monitoring, controlling and accounting services.

DSP Based VoIP Call Processing

SAPEX comes in two variants. SAPEX SDM and SAPEX DDM with single and dual DSP support. The DSP based VoIP call processing ensures maximum throughput with standard voice codecs such as G.711 A-Law, μ -Law, G.729AB, G.723.1L, G.723.1H, GSM-FR, GSM-EFR, iLBC-20 and iLBC-30. Whether it is a high-bandwidth codec such as G.711 or a LBR (high compression) codec such as G.729, number of simultaneous transcoded calls vary minimally between the range of 25 to 30 (SAPEXDDM).

Extended Connectivity

Unlike, a traditional telephony system, SAPEX does not bind a user to a fixed location. Instead of a phone number, an IP-PBX user is identified by his SIP URI. In IP telephony, a VoIP call is established over the IP network. An IP user is free to carry his extension anywhere in the world. Wherever a user is, provided with an adequate Internet link, users can establish calls retaining the same contact credentials (user name, password). NAT and STUN support enables the VoIP call to be established, when the communicating devices are hidden behind NAT/firewalls. Dynamic DNS (DDNS) support maintains SAPEX connectivity with remote users even with a dynamic addressing scheme.

Feature Transparency

SAPEX, with B2BUA functionality and voice transcoding support, provides DTMF access to standard telephony features. Features such as Call Forward, Call Hold, Call Toggle, Call Park, Call Pickup, Call Transfer, 3-Party Conferencing, Do-Not-Disturb (DND) and others, can be accessed from a SIP terminal, with the same ease as in case of traditional telephony.

Localization

SAPEX can be configured as per the telecom standards of the country where it is installed. It comes along with built-in web server functionality which supports English, French, Spanish, German, Portuguese and Italian language. Time Zones, Day Light Saving, Call Progress Tones can be set specific to a country's telecom requirements.

Open-Standard SIP

SAPEX supports open-standard SIP protocol to establish calls over the IP network. Devices such as IP phones, ATAs and Gateways, with support for SIP, can be easily registered as clients to the IP-PBX. Matrix offers a range of SIP supported VoIP products including ATAs, VoIP to FXS Gateway, VoIP to GSM, FXO and FXS Gateway and IP Phones to fulfill varied communication requirements. Following the route of open-standard, SAPEX is ensured for full interoperability with an entity with SIP support. Thus, moving towards the next generation IP telephony with SAPEX, an organization can also integrate to existing telephony network through the gateway devices and thereby plan for a smooth migration to a full IP infrastructure.

Simplified Configuration and Management

SAPEX accounts for a highly simplified communication architecture, as it is possible to route calls also over the IP network, besides data. With the embedded web server, an administrator can remotely program, configure the system using the multi-lingual GUI. A SAPEX extension can be located anywhere on the global IP network. An administrator can monitor and manage user registration and feature access in real-time. A new user can be added, granted calling rights, defined under a user group, granted voice mail access with a few clicks from the intuitive GUI itself. An Administrator can easily monitor the network and SIP trunk status from the GUI itself. SAPEX also supports debugs at various levels, over IP network, via the syslog client.

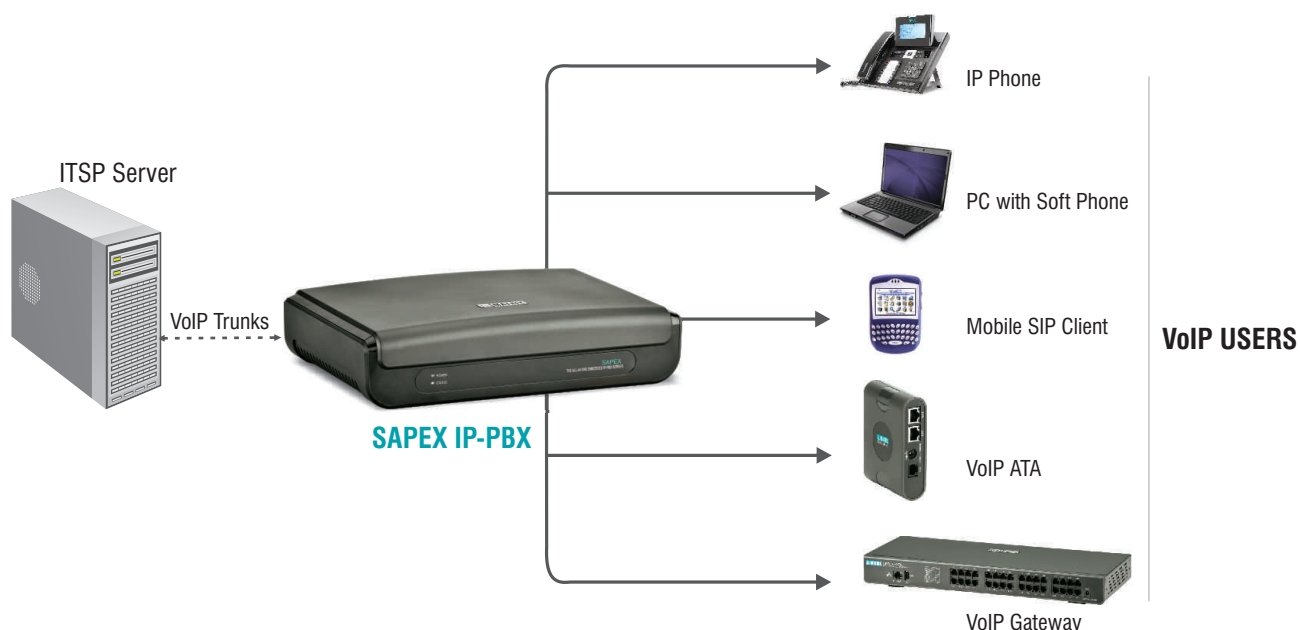
Voice Transcoding

With Matrix IP-PBX it is possible to establish communication between SIP devices with diverse audio specifications. This is established with the voice transcoding functionality, which allows for negotiation of audio codecs between the SIP devices engaged in the SIP call. Being B2BUA-enabled, SAPEX gains the power to negotiate and bind the communicating devices to common terms for a successful call establishment, thereby minimizing the ratio of dropped calls. The administrator can set preference of vocoders and ensure optimal bandwidth utilization. The transcoding feature can be selectively enabled for a set of users. If transcoding is not required, turning off the transcoding feature, system can support up to 44 concurrent calls. An administrator can thereby make a tradeoff between the number of transcoded calls and the maximum calls to be supported by the system.

SAPEX IP Connectivity

Business today has extended beyond the geographic boundaries. The requirement of modern age businesses and professionals have changed and grown. The requirement today is of a communication technology that is cost-effective, boundless, decentralized, flexible and efficient.

That's the reason the world is advancing towards IP telephony. IP telephony is highly cost-effective. The IP network is wide-existent, making communication boundless. An IP user is free to carry his extension anywhere on the global IP network. This facilitates a decentralized communication setup. IP telephony offers wider flexibility in choosing the mode (call, IM) and the terminal (IP phone, a soft phone, mobile with SIP client, PDA) for communication. Presence and IM functionalities have completely changed the way organizations work and account for improved productivity.



Centralized IP Server For Distributed Users

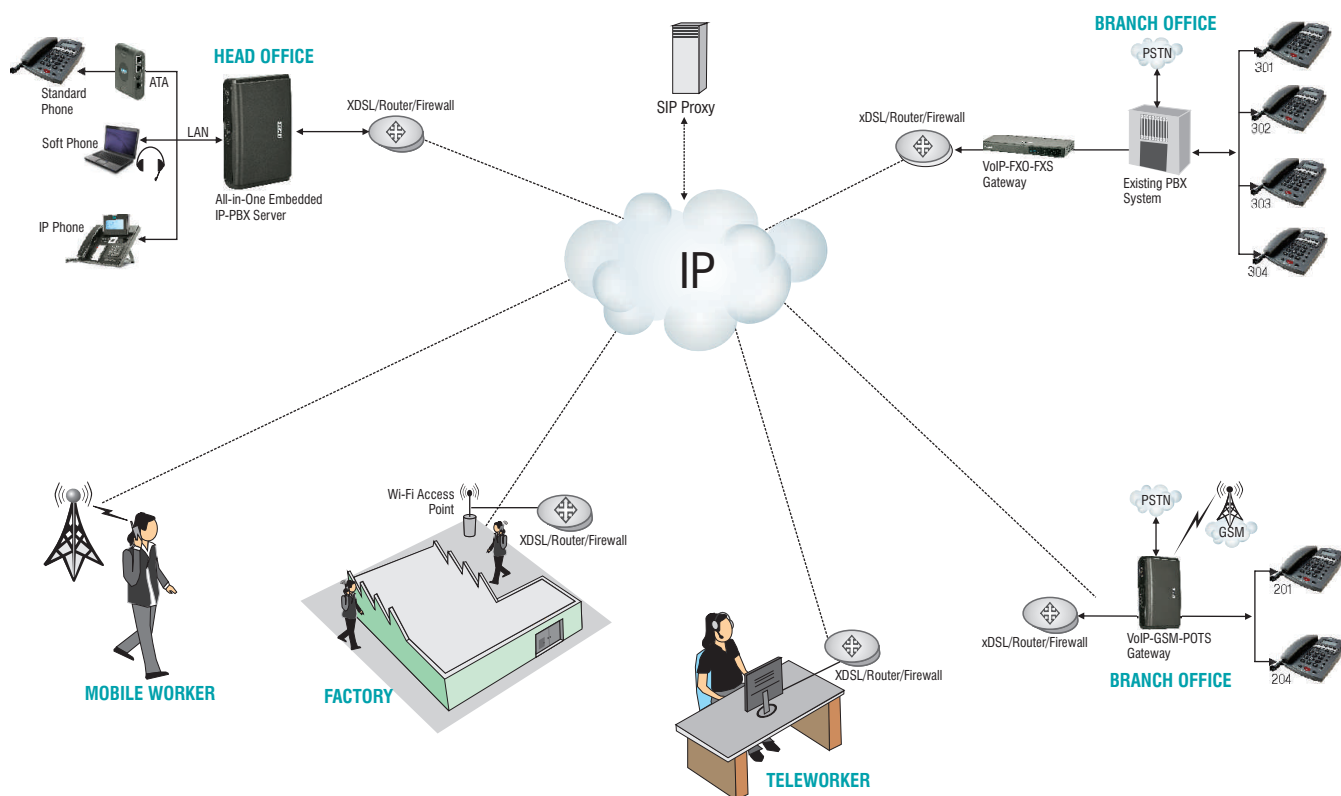
In the world of IP, an end user terminal can be an IP phone, a softphone, mobile with SIP client, PDA, etc. A user can have multiple contact points mapped to a common user identity. A user can be reached on any of the active terminal at a given instant, with the flexibility to register from any geographic location on the IP network.

A remote user's connectivity is maintained even when behind a NAT or firewall. A SIP user located over a public/private network can be registered easily with SAPEX without the need to use VPN or other tunneling technologies. The embedded Dynamic DNS client ensures a remote user can register to the IP-PBX configured for a dynamic IP without much trouble.

Presence further determines the availability of a user (such as online, offline), his willingness to participate in a communication session

(busy, available on phone, out of office and others) and his preferred mode of communication (call or instant messaging), before an actual conversation begins. A user has more choices with him now. A user now has a right to alter his presence status at his will and intimate the same to others, instantly, through a Presence and IM client.

The All-in-One IP-PBX embeds the distributed SIP Registrar, Proxy, Presence and Voice Mail Servers to a single element. The comprehensive, full-featured and compact IP-PBX is capable of handling up to 500 users. There are two models – SAPEX SDM with 200 users and SAPEX DDM with 500 users. Devices such as IP phones, ATAs and Gateways, with support for SIP, can be easily registered as clients to the IP server.



SAPEX proves to be a complete solution for inter-branch office communications. Dispersed branches can be tied together over the IP network, with SAPEX located at central branch office. Low-tariff internet telephony between geographically spread locations helps to reduce the communication cost to a great extent. SAPEX supports peer-to-peer calls between locations without going through a proxy server. The multi-site connectivity over IP also facilitates usage of common dial plans and numbering across the geographically distant branches.

A remotely registered user can establish calls with other SAPEX users, as simply as calling a user connected within its own LAN network. The IP-PBX can register to as many as 10 SIP accounts, providing least cost routing through a dial plan set up.

Multiple Extensions can be grouped under a single user group. This facilitates call distribution between pre-defined users, as per the assigned priorities. Time Tables defined in the IP-PBX further delineate the way for calls to be routed as per the day timing. The IP-PBX also supports call routing based on the CLI of an incoming call. SAPEX supports various telephony features such as Call Forward, Call Hold, Call Pickup, Call Park, Call Transfer and 3-Party Conferencing.

The seamlessly connected branches can share a common auto-attendant and voice mail system also. The SMTP client paves the way for the voicemails to be picked-up from a user's voice mail box and efficiently delivered to a user's Email-ID as attachments. A user has the flexibility to receive a scheduled delivery of such voice mails. A built-in RADIUS client facilitates efficient call logging to a remote server.

■ KEY FEATURES

Allowed and Denied Numbers

SAPEX offers flexibility to allow and deny dialing of particular number or a set of numbers. The denied list restricts a user from dialing a number programmed in the denied list.

Automatic Number Translation

SAPEX supports 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. SAPEX itself modifies the dialed number or part thereof so that it matches with the numbering plan that is understood by the ITSP. For example, if a user has dialed the number 223344 to call www.abc.com, then SAPEX adds the appropriate access code "*777" specified by the ITSP and dials out the number "*777223344" instead of 223344.

Auto-Attendant

The Auto-Attendant informs a caller of the way to reach his ultimate destination. Customized welcome and guiding prompts as per time of the day and Music-on-Hold can be played to a caller. With the help of Automated Attendant, a caller can find-his-way to either reach to a desired extension, retrieve information or to leave back a message for the concerned user in his mail box.

Busy Lamp Field (BLF)

A Busy Lamp field is an array of line status lamps. An extension user can view the status of other extensions e.g. busy, ringing or idle, if a user's Class of Service (CoS) is provisioned for it. The busy lamp indication forms the umpire's verdict on an extension status, for the operator to either transfer a call or else pick up the call himself.

Call Forking

IP based communications offer wider terminal options such as an IP phone, a softphone, mobile with SIP client or a PDA. SIP provides a mechanism called Uniform Resource Identifier (URI), mapping a user's identity to multiple devices. Up to three such terminals can be programmed for a single user. So, when a call is initiated, the same is attempted to all (3) user terminals in parallel, known as call forking. A user now experiences extended connectivity, no matter whether he uses an office IP phone or his cell phone (with SIP client) while on tour or a soft phone to communicate. This also eliminates the need to keep a track of users multiple contact addresses.

Call Progress Tones

SAPEX IP-PBX provides users, the flexibility to match the Call Progress Tones and Ring Cadences to the standard ones used in a country. Country Specific Call Progress Tones like Dial Tone, Ring Back Tone, Busy Tone, Error Tone and others can be programmed.

Caller-ID Based Routing

Based on the Caller-ID details, an incoming call can be routed to a pre-defined extension. For example, calls important to business may be directed to the higher authorities, calls with specific CLI may be directed to specific departments, while calls from anonymous numbers may be directed to the customer support teams.

Call Forking

IP based communications offer wider terminal options such as an IP phone, a softphone, mobile with SIP client or a PDA. SIP provides a mechanism called Uniform Resource Identifier (URI), mapping a user's identity to multiple devices. Up to three such terminals can be programmed for a single user. So, when a call is initiated to those extensions, it is attempted to all (3) user terminals in parallel, known as call forking. The user now experiences extended connectivity; no matter whether he uses an office IP phone, cell phone (with SIP client) or an IP softphone to communicate. Moreover, it also eliminates the need to keep a track of user's multiple contact addresses.

Direct-Dialing-In (DDI Routing)

A call landing on SIP trunk can be directly routed to an extension as per the DDI numbering. The DDI facility should be activated on the SIP trunk by the SIP service provider. Unlike traditional telephony services, IP telephony does not bind a number to its geographical location. Here, calls are placed over internet and numbers are mapped to IP addresses, which may be anywhere on the internet. An IP extension can always be called irrespective of its current location.

Dial Plan

SAPEX supports multiple SIP trunk registrations. Registration with maximum of 10 SIP servers is supported. Calling rates differ on the basis of area of call, service provider, call time, etc. A Dial Plan allows a user to place a call through the most cost-effective SIP trunk, as per a defined call routing logic. Each user can be assigned multiple Dial Plans, either of which gets activated for a specified timing. The Dial Plans may be same for all users or may differ individually.

Do-Not-Disturb (DND)

This feature is useful when a user does not want to receive any incoming calls without logging off from the IP-PBX or switching off the phone in use.

Daylight Saving Time Adjustment

The Real Time Clock (RTC) of the IP-PBX adjusts automatically to be in tune with the Day Light Saving requirements of the country where it is installed.

Dynamic DNS (DDNS)

Matrix SAPEX supports Dynamic DNS Client which automates the discovery and registration of its IP addresses on the public network. The remote administrator and the IP clients can thereby connect to the IP-PBX using Domain Name associated with the dynamic IP, preventing reconfiguring of systems, whenever a network infrastructure changes.

FAX Homing

Fax Homing allows a user to utilize a common SIP Trunk for both-voice calls and for receiving a fax. With FAX Homing enabled on a SIP trunk, an auto-attendant can be employed to answer incoming calls. Once a fax tone gets detected, call can be directly routed to an extension where fax machine is connected. This obviates any kind of operator intervention. Such optimal usage of a common SIP trunk for both-voice and fax adds to the cost benefit and saves time.

Instant Messaging (IM)

Instant Messaging is a much popular tool of communication. Ability to communicate via text messages, adds an additional and easy means to communicate with colleagues. Further, with most IM clients, it is possible to alter one's availability status (Online, Offline, Busy, etc) and intimate the same to others, instantly. SAPEX identifies the users as Presentities (whose status is to be viewed) and Watchers (one who needs to know the status of another user). A Watcher SUBSCRIBES (requests) the presence server for the status of presentity. If the presenter has PUBLISHED (intimated) his status, the watcher can be NOTIFIED (informed) about the status of presentity. An administrator can grant certain users the right to not PUBLISH their status, yet avail the presence and IM functionality.

NAT and STUN Support

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 an IP terminal 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 easily register to the IP-PBX, 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

SAPEX supports VoIP calls between two locations without going through a proxy server. Fixed IP addresses of the locations can be programmed in its Peer-to-Peer table. 500 such entries can be programmed for SAPEX. Short, numeric dialing codes can be defined for calls between these locations. Since the Peer-to-Peer calls are placed over the public IP network, the call cost is minimal.

RADIUS Client

A built-in RADIUS client facilitates efficient call logging to a remote server. SAPEX logs the details of calls in CDR (Call Detail Records) files. These CDR files contain the essential information to monitor and account a user for the services utilized. These CDR files are therefore requiring a safe and longer storage. Any storage internal to a system gets overwritten in case the system memory fills to its maximum capacity (500 calls). If a user needs to refer older call details, user will have to take CDR print out, very often. In such cases, the embedded RADIUS (Remote Authentication Dial-In User Service) client enables the IP-PBX to send these CDR files to a remote server called RADIUS/Database server. Further integration with a billing server, can benefit the service providers in accurate billing of the clients and thereby offer advanced value-added services to their subscribers.

SMTP Client

System supports SMTP client which enables it to send the voice mail and call logs as attachment to a predefined Email -ID.

Time Table

SAPEX offers flexibility to divide a day into time zones defined as working hours, non-working hours and break-hours. As per the time zone, incoming calls can be routed to different extensions. Such efficient call routing delivers a sound and effective communication setup, boosting the overall productivity.

User Groups

Multiple extensions can be grouped under a User Group. This facilitates call reception between pre-defined users. On reception of a call, the extensions will ring according to the assigned priorities. Thus ensuring no call remain unanswered. Maximum of 16 user groups can be defined, with 8 members in each group.

Video Calling

SAPEX family of IP-PBX also support video calls to be established over the IP network. With two end terminals, capable of leveraging a video call, SAPEX acts as a relay unit between them. SAPEX supports basic video calling, with no supplementary services.

VoIP Silence Disconnect Timer

The VoIP Silence Disconnect Timer parameter defines the time limit after which a call is to be disconnected, if no voice packets are received. This leads to better utilization of available bandwidth.



IP PHONES

Matrix SETU VP248 is a range of feature-rich executive IP Phones. They provide intuitive operation for the call management functions. It supports a host of additional features providing the user fast access to the functions of SAPEX IP-PBX at a single touch of a button. SETU VP IP Phones are available in following four variants:

SETU VP248PE	6 Lines x 24 Characters LCD Display with PoE
SETU VP248SE	2 Lines x 24 Characters LCD Display with PoE
SETU VP248P	6 Lines x 24 Characters LCD Display
SETU VP248S	2 Lines x 24 Characters LCD Display



SETUVP248SE



SETUVP248PE

SETU VP248 Key Features

- Open Standard SIP Support
- 3 SIP Accounts
- LAN and WAN Ports
- Programmable Keys
- Anonymous Call Rejection
- Auto Configuration
- Auto Answer with Headset Interface
- Conference
- DHCP, PPPoE, NAT and STUN
- Peer-to-Peer Calling
- Dialed, Received, Missed and Rejected Call Logs
- G.711, G.722, G.723, G.726 and G.729AB
- Phone Book with 100 Entries
- Ringer, Speech and LCD Controls
- Voice Mail Key
- Web Configuration

FEATURES LIST

SIP Server

- Embedded Registrar, Proxy, Presence Servers
- Back-to-Back User Agent (B2BUA)
- Embedded Dynamic DNS (DDNS) Client
- Embedded RADIUS and SMTP Client
- Supports SIP v2
- Registration of Multiple SIP Trunks
- Support for 500 User Registrations
- NAT and STUN Support

Calling and Routing Features

- Access Codes
- Allowed and Denied Numbers
- Automatic Number Translation
- Anonymous Call Rejection
- Caller Line Identification and Restriction (CLIR)
- Call Forward
- Call Forking
- Call Hold
- Call Pickup (Group and Selective)
- Call Park
- Call Release Timer
- Call Transfer (Blind and Attended)
- Conference (3-Party)
- Configurable Time Zones for Call Routing
- Caller-ID Based Routing
- Do-Not-Disturb (DND)
- Direct-Dialing-In (DDI Routing)
- Dial Plan (Multiple)
- Emergency Number Dialing
- FAX Homing
- Peer-to-Peer Calling
- Selective SIP Trunk Access
- Time Table
- User Group

Management Features

- Web Based Configuration/Firmware Management
- Multi-Lingual Web Based Programming Tool (Jeeves)
- Network, User, SIP Trunk and Voice Mail Status Display (Jeeves)
- Status LED Indication for SIP Trunk Status
- Region based default setting of Language, Time Zone, DST, CPTG
- Syslog Client
- PCAP Trace
- Busy Lamp Field (BLF)
- Call Detail Records (CDR)
- User Class of Service (CoS)
- Soft Restart
- SNMP V1, V2C, V3

Voice Mail

- Individual Voice Mail for each User with Access Codes
- Configurable Mailbox Size
- Customizable Greetings
- Voice Mail to Email

Auto-Attendant

- Call Routing to Operator, System User or Voice Mail Server
- Music-on-Hold (MoH)
- Customizable Greetings and Voice Prompts

Voice Functionalities

- Country Specific Call Progress Tones
- DTMF (Inband, Outband, SIP INFO)
- SIP QoS and RTP QoS (DiffServ)
- Voice Transcoding

Date and Time Settings

- Day Light Saving Time Adjustment
- Real Time Clock (RTC)
- SNTP

TECHNICAL SPECIFICATIONS

VoIP

VoIP Protocols	: SIP v2, SIP over TCP, Symmetric RTP, RTCP, 100rel/PRACK SIMPLE: Session Initiation Protocol for Instant Messaging and Presence Leveraging Extension
Network Protocol	: IPv4, TCP, UDP, SNTP, STUN, ARP, ICMP, PPP, DNS, SMT, RADIUS, SNMP (V1,V2C,V3)
SIP	: 10 SIP Trunks Outbound Proxy Support
NAT	: STUN and NAT Keep Alive
Voice CODECS	: G.711 A-Law, μ -Law, G.729AB, G.723.1L, G.723.1H, GSM-FR, GSM-EFR, iLBC-20, iLBC-30
Call Progress Tones	: Dial Tone, Ring Back Tone, Busy Tone, Error Tone
Fax	: T.38 Relay and Pass-Through
Quality of Service	: Layer 3 DiffServ
Security	: MD5 Authentication for SIP, Password Protected Configuration by Admin and User

POWER SUPPLY

Input	: 5VDC@2A through External Adaptor
Power Consumption	: 10W (Typical)
Connector	: DC Power Jack
LED Indications	: LED for Power Status LED for SIP Trunk Status

MECHANICAL

Dimensions (WxHxD)	: 230mmX55mmX163mm (9.06"X2.17"X6.42")
Unit Weight	: 0.520 Kg (1.14 lbs)
Shipping Weight	: 1.360 Kg (2.99 lbs)
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)



■ SYSTEM CONFIGURATION

System Resources	SAPEX SDM	SAPEX DDM
Users	200 (50 IP Users Pre-activated)	500 (50 IP Users Pre-activated)
SIP Trunks	10	10
Ports		
WAN Port	1 (RJ45, 10/100 Base T)	
LAN Port	1 (RJ45, 10/100 Base T)	
USB Port	1 (Internal)	
DC Jack	1 (DC Power Jack)	
CODEC	Concurrent Calls (Transcoding)	
G.723L/H	15	30
G.729	16	31
GSM EFR	13	26
GSM FR	21	30
iLBC 20/30	13	26

■ VoIP PRODUCTS FROM MATRIX

ETERNITY	The Universal Telephony Gateway
SETU VTEP	Single Span SIP based VoIP to ISDN PRI Gateway
SETU VGB	Multi-Port SIP based VoIP to GSM-ISDN BRI 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 ATA	VoIP Adaptors with FXO and GSM/3G Ports
SETU VP248	The High-Definition IP Phone with PoE



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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