What is Miton's VoIP+PBX Internet Telephone System?

VoIP+PBX is a complete PC-Based Internet PBX which runs on Linux and provides all of the features you would expect from a PBX and more. VoIP+PBX does voice over IP in three protocols, and can interoperate with almost all traditional standards-based telephony equipment using relatively inexpensive hardware.

VoIP+PBX provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway). Check the Features section for a more complete list.

VoIP+PBX needs no additional hardware for Voice over IP. For interconnection with digital and analog telephony equipment, VoIP+PBX supports a number of hardware devices, most notably all of the hardware manufactured. For example, single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks as well as a single port FXO card and a one to four-port modular FXS card.

Also supported are the Internet Line Jack and Internet Phone Jack products from Quicknet.

VoIP+PBX supports a wide range of Time Division Multiplex (TDM) protocols for the handling and transmission of voice over traditional telephony interfaces. VoIP+PBX supports US and European standard signaling types used in standard business phone systems, allowing it to bridge between next generation voice-data integrated networks and existing infrastructure. VoIP+PBX not only supports traditional phone equipment, it enhances them with additional capabilities.

Using the IAX Voice over IP protocol, VoIP+PBX merges voice and data traffic seamlessly across disparate networks. While using Packet Voice, it is possible to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

VoIP+PBX provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs[1]. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.

All Miton's Products are built around VoIP+PBX

Miton's products are carefully designed for maximum flexibility. Specific applications are defined around the central VoIP+PBX core system. This advanced VoIP+PBX core handles the internal interconnection of the PBX, cleanly abstracted from the specific protocols, codecs, and hardware interfaces from the telephony applications. This allows VoIP+PBX to use any suitable hardware and technology available now or in the future to perform its essential functions, connecting hardware and applications.

The VoIP+PBX core handles these items internally:

- **PBX Switching** - The essence of VoIP+PBX, of course, is a Private Branch Exchange Switching system, connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces.

- **Application Launcher** - launches applications which perform services for uses, such as voicemail, file playback, and directory listing.

- **Codec Translator** - uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.

- **Scheduler and I/O Manager** - handles low level task scheduling and system management for optimal performance under all load conditions.

Loadable Module APIs:

Four API's are defined for loadable modules, facilitating hardware and protocol abstraction. Using this loadable module system, the VoIP+PBX core does not have to worry with details of how a caller is connecting, what codecs are in use, etc.

- **Channel API** - the channel API handles the type of connection a caller is arriving on, be it a VoIP connection, ISDN, PRI, Robbed bit signaling, or some other technology. Dynamic modules are loaded to handle the lower layer details of these connections.

- **Application API** - the application API allows for various task modules to be run to perform various functions. Conferencing, Paging, Directory Listing. Voicemail, In-line data transmission, and any other task which a PBX system might perform now or in the future are handled by these separate modules.

- **Codec Translator API** - loads codec modules to support various audio encoding and decoding formats such as GSM, Mu-Law, A-law, and even MP3.

- **File Format API** - handles the reading and writing of various file formats for the storage of data in the file system.

Using these API's VoIP+PBX achieves a complete abstraction between its core functions as a PBX server system and the varied technologies existing (or in development) in the telephony arena. The modular form is what allows VoIP+PBX to seamlessly integrate both currently implemented telephony switching hardware and the growing Packet Voice technologies emerging today. The ability to load codec modules allows VoIP+PBX to support both the extremely compact codec necessary for Packet Voice over slow connections such as a telephone modem while still providing high audio quality over less constricted connection types. The application API provides for flexible use of application modules to perform any function flexibly on demand, and allows for open development of new applications to suit unique needs and situations. In addition, loading all applications as modules allows for a flexible system, allowing the administrator to design the best suited path for callers on the PBX system and modify call paths to suit the changing communication needs of a going concern.

VoIP+PBX Features

VoIP+PBX based telephony solutions offer a rich and flexible feature set. VoIP+PBX offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems. VoIP+PBX offers the features one would expect of a large proprietary PBX system such as Voicemail, Conference Bridging, Call Queuing, and Call Detail Records.

Telephony Services:

Voicemail System
Password Protected
Separate Away and Unavailable Messages

Default or Custom Messages
Multiple Mail Folders
Web Interface for Voicemail Checking
E-mail notification of Voicemail
Voicemail Forwarding
Visual Message Waiting Indicator
Message Waiting Stutter Dialtone
Auto Attendant
Interactive Voice Response
Overhead Paging
Flexible Extension Logic
Multiple Line Extensions
Multi-Layered Access Control
Direct Inward System Access
Directory Listing
Conference Bridging
Unlimited Conference Rooms
Access Control
Call Queuing
ADSI Menu System
Support for Advanced Telephony Features
PBX Driven Visual Menu Systems
Visual Notification of Voicemail
Call Detail Records
Local Call Agents
Remote Call Agents
Protocol Bridging
Provides seamless integration of technologies
Offers a unified set of services to users regardless of connection type
Allows interoperability of VoIP systems

Call Features:

Music on Hold
Music on Transfer
Flexible mp3 based system
Volume Control
Random Play
Linear Play
Call Waiting
Caller ID
Caller ID Blocking
Caller ID on Call Waiting
Call Forward on Busy
Call Forward on No Answer
Call Forward Variable
Call Transfer
Call Parking
Call Retrieval
Remote Call Pickup
Do Not Disturb

Scalability:

TDMoE
Allows Direct Connection of VoIP+PBX PBX
Offers Zero Latency
Uses Commodity Ethernet Hardware
Voice over IP
Allows for Integration of Physically Separate Installations
Uses commonly deployed data connections
Allows a unified dialplan across multiple offices

Voice over IP Interoperability:

VoIP+PBX provides transparent bridging between Voice over IP protocols and traditional telephony equipment. In addition, VoIP+PBX can transfer calls from one system to another via the Inter VoIP+PBX Exchange protocol.
Inter-VoIP+PBX Exchange (IAX)
H.323
Session Initiation Protocol (SIP)
Media Gateway Control Protocol (MGCP)

Traditional Telephony Interoperability

Robbed Bit Signaling Types
FXS and FXO
Loopstart
Groundstart
Kewlstart
E&M
E&M Wink
Feature Group D
PRI Protocols
4ESS
Lucent 5E
DMS100
National ISDN2
EuroISDN
BRI (ISDN4Linux)

Codec Support[1]

GSM
G.729 (available through purchase of commercial license(s))
G.723.1 (pass through)
Linear
Mu-Law
A-Law
ADPCM
G.726
ILBC
LPC-10
MP3 (decode only)

Fax Support

The current state of fax support is incomplete.

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[1]. **Codec Definition** = Acronym for COder - DECoder.

The name "codec" is short for "coder-decoder," which is pretty much what a codec does. Most audio and video formats use some sort of compression so that they don't take up a ridiculous amount of disk space or transmission bandwidth. Audio and video files are compressed with a certain codec when they are saved and then decompressed by the codec when they are played back. Common codecs include MPEG and AVI for video files and WAV and AIFF for audio files. Codecs can also be used to compress streaming media (live audio and video) which makes it possible to broadcast a live audio or video clip over a broadband Internet connection

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