

SIPxchange PBX

A SIP PBX solution that you control ? Yes.

Pingtel's Enterprise SIPxchange PBX - The PBX for Linux, is an enterprise-class, fully featured software IP PBX with integrated voice mail, auto attendant and web-based configuration manager for Voice over IP (VoIP) solutions. The SIPxchange PBX is based entirely on the Session Initiation Protocol (SIP) standard and operates on all popular Red Hat Linux servers, PSTN/IP gateways and end points.

The SIPxchange PBX is the first enterprise grade IP PBX available in open source. It offers proven, robust functionality for up to 200 seats on a single server, and provides low-cost solutions for IP PBX, branch offices, home workers, key systems and call centers. SIPxchange PBX also interoperates seamlessly with legacy components such as TDM PBX, enabling users to choose a phased-in approach to VoIP implementations.

SIPxchange PBX utilizes standards, proven Internet techniques and a distributed architecture to create a highly secure and available IP voice system.

Sold in the same way customers buy supported versions of Linux, SIPxchange PBX is based on a low-cost annual subscription model. The Pingtel SIPxchange PBX is part of a family of SIPxchange products that, in combination, comprise a fully featured, standards-based and software-only communications solution at the lowest total cost of ownership.



Key Attributes

Pingtel SIPxchange PBX offers:

Voice Mail

Integrated voice mail system

Unified Messaging

Voice mail messages can be retrieved by web browser or forwarded to any email client

Multiple Auto Attendants

Auto attendants are easily configured via browser interface.

Configuration Management

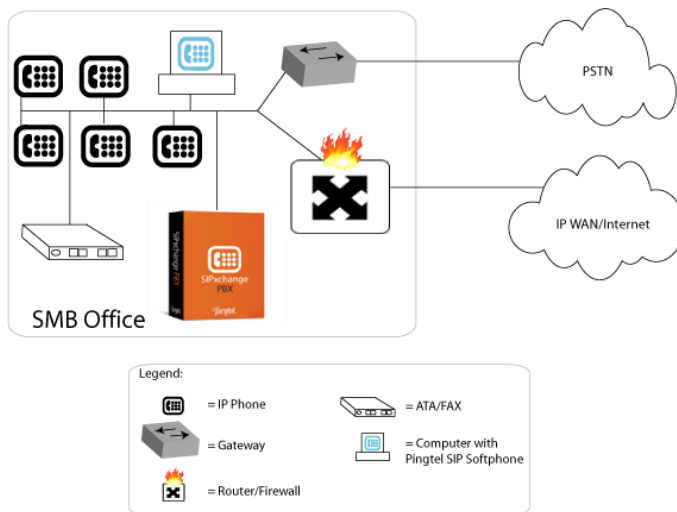
Intuitive browser interface for centralized control and management of dial plans, users and endpoints.

Standard Interfaces for Advanced Admin

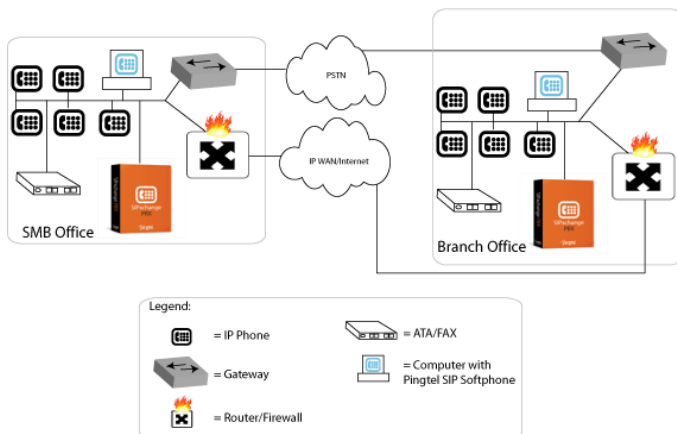
SIPxchange PBX architecture was designed to be robust and flexible.



Single-site PBX



Multi-site PBX



IP PBX Attributes

Pingtel SIPxchange PBX offers flexible deployment options

Single-Site PBX

Pingtel SIPxchange PBX is the ideal Enterprise IP voice solution for any office with 10 to 200 users. It offers an unmatched low purchase price and low operating expenses by leveraging low-cost computer servers, commodity hardware and Linux.

Multi-site PBX

Pingtel SIPxchange PBX provides a complete solution to your enterprise telephony needs. The system's architecture lets you easily distribute servers, gateways and intelligence strategically on your network — within one office or among branch offices — for cost savings, high reliability, backup and load balancing.

Remote Workers

Full PBX Functionality Remotely - The Pingtel remote worker solution is ideal for supporting distributed and mobile professionals with full PBX functionality to any location that has a high-speed or broadband connection. In addition to traditional PBX features, SIPxchange PBX provides these workers with an extensive list of advanced IP telephony features and the ability to easily add new features over time, regardless of their location.

Benefits

Lower Total Cost of Ownership (TCO)

Software subscription model provides support and ongoing updates allowing you to realize a typical return on investment (ROI) of 90 days or less.

Easy to use

System and user interfaces are designed to be simple and easy to use.

Legacy telecommunications investment protection

Standards-based system supports existing network and meets all requirements for TDM PBX replacement.

Unmatched system flexibility and lower-cost solution components

SIP standards compliance ensures interoperability with off-the-shelf solution components and applications from other vendors that conform to SIP standards. Mix and match phones across the enterprise based on individual preferences.

Elimination of expensive second phone lines for remote workers

Leverages existing broadband connections for voice and data.

Enhanced employee productivity

Delivers a wide range of innovative user features.

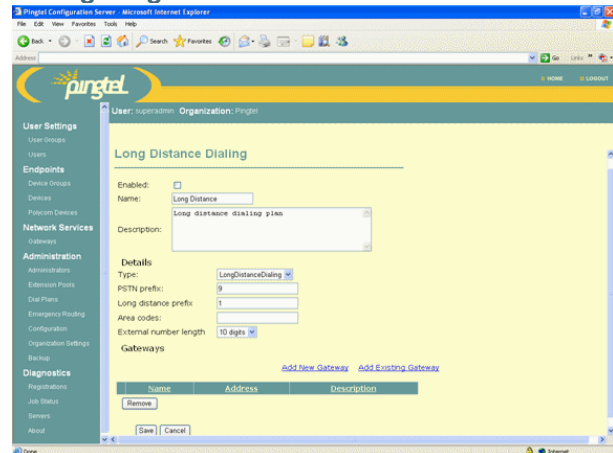
Convenient VoIP system migration

Begin with a single office and extend IP telephony to your remaining organization in a timeframe that suits you.

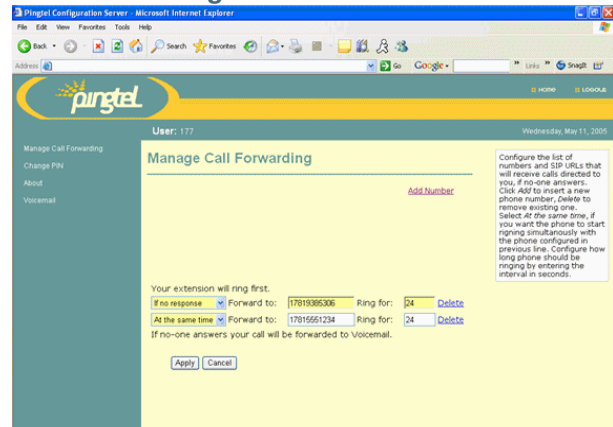
Future-proofed network

Standards-based system enables easy moves, adds and changes, installation of new features and deployment of new applications as they come available.

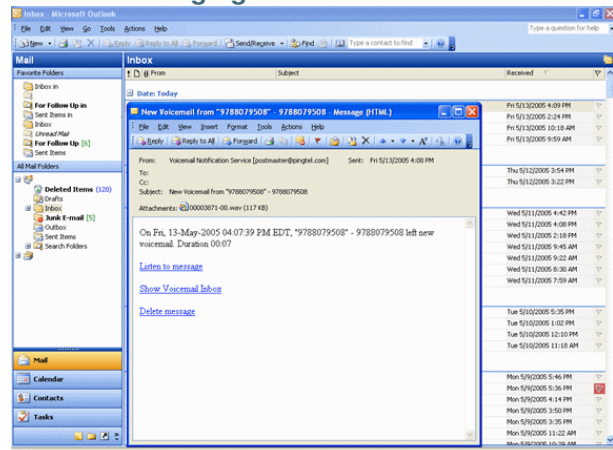
Configuring the Dial Plan



User Forwarding and Follow-me



Unified Messaging with Microsoft Outlook



PBX System Features

- Aliasing facility
- Automatic Route Selection
- Auto-restart services after power failure using watchdog facility
- Browser-based configuration system
- Call Admission Control
- Codec support
- Dynamic call forwarding
- Hunt groups
- Message waiting indication
- Multi-site / multi-location station and gateway
- Multi-station appearance
- Outbound call blocking
- System Security
- URI mapping engine for call routing and inter-company (/domain) SIP calls
- Web services APIs for Config server (SOAP)

Voicemail Features

- wav file messages
- Browser-based interface
- DTMF interface
- Editable message headers
- Email notification of new voicemail messages
- Folders for message organization
- https-based message storage
- Multiple user customizable voicemail greetings
- Operator escape from anywhere

Auto Attendant Features

- Customizable auto attendant message for main greeting
- Dial by extension
- Dial by name
- Operator escape from anywhere
- Customizable IVR menus with VXML

User Features

- Browser-based user interface
- Call coverage
- Call forward
- Call hold / retrieve
- Call waiting / retrieve
- Calling line identification
- Calling party name identification
- Conferencing
- Direct inward dial (DID)
- Message waiting indication
- Multiple call appearance
- Multi-station appearance

SIP Implementation

- RFC 3261 Session Initiation Protocol using both UDP and TCP transports
- Advanced call control using RFCs
 - 3515 Refer Method
 - 3891 Referred-By header
 - 3892 Replaces header
- Provide for consultative and blind transfer and third party call controls
- RFC 3263 Locating SIP Servers - use of DNS SRV records for call routing control and server redundancy.
- RFC 3581 Symmetric Response Routing (rport)
- RFC 3265 SIP Event Notification - for phone configuration and
- RFC 3842 Voice mail message waiting indication (MWI)
- RFC 3262 Reliable Provisional Responses
- RFC 2833 Out-of-band DTMF tones
- RFC 3264 Offer/Answer model for SDP for Codec Negotiation
- Early media (SDP in 180/183)
- Delayed SDP (SDP in ACK)
- Re-INVITE: Codec change, hold, off-hold
- Route/Record-Route header fields
- Configurable RTP/RTCP ports
- Configurable SIP ports

Specifications

Server requirements

- Pentium 4 2.8 GHz with Hyper threading
- 1 GB of Memory
- 80 GB Hard drive
- Red Hat Enterprise Linux 3.0 ES (Edge Server)

Limits

- 200 Users
- Unlimited PSTN Connections with media gateway
- 30 media server ports for Auto Attendant and Voice Mail
- 300 Voice Mail boxes

Note – Some features are dependant upon other SIP components such as Phones and Gateways. See Pingtel's Certified component feature matrix for more details on supported features.



SIPxchange PBX Subscription, Maintenance

Pingtel's SIPxchange PBX is available as a 12-month subscription in two Editions – Standard and Premium. Subscriptions are associated with a customer-specified host. SIPxchange PBX is currently packaged to support processors based on the Intel Pentium IV architecture. Pingtel's subscriptions include:

Updates – Available to Standard and Premium Edition subscribers. Access to all newly released software versions, patches, and documentation is available during the subscription period, in binary and source code format.

Unlimited Web-Based Self-Help - Subscribers receive unlimited access to Pingtel's Internet-based self-help support system. This provides web access to Pingtel's Tech Notes database.

Technical Assistance Center (TAC) Access – Premium Enterprise Edition subscriptions entitle customers to TAC access for the host to which a subscription entitlement is assigned during each one-year subscription term.

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About Pingtel Corp.

Pingtel is reshaping the communications market by delivering the first enterprise class SIP PBXs, SIP call managers/routers and SIP Softphones based on 100% SIP and 100% open source software. Offering enterprise-class communications applications under Linuxstyle subscription licenses, Pingtel combines the best attributes of open source development - low cost, adaptability and flexibility – with the reliable solutions and support enterprises require for voice applications. Pingtel's open source SIP PBX is the linchpin technology that will catalyze the movement of enterprise communications into the data center and away from purpose-built hardware. Like enterprise-grade Linux, this approach will drive commoditization of traditional telephony hardware and software and eliminate vendor lock-ins that keep prices high and limit innovation. For more information, visit www.pingtel.com.

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