

SIPxchange CallManager

Free yourself of proprietary communications technology.

Pingtel's SIPxchange CallManager is a server-based, integrated central call routing engine that also provides directory services across an enterprise VoIP network. CallManager provides a very low-cost centralized call management solution that unifies control of enterprise voice application islands. This standards-based solution interoperates with all legacy systems such as branch PBXs, call center, voice mail and other messaging systems.

Based entirely on Session Initiation Protocol (SIP) IETF standards and operating on all popular Red Hat Linux servers, PSTN/IP gateways and end points, Pingtel SIPxchange CallManager provides very low-cost call management solutions that integrate with legacy TDM and IP networks, meet future scalable requirements for TDM PBX replacement and improve user services and network cost reduction with the features, reliability and flexibility you expect.

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



Key Attributes

Pingtel SIPxchange CallManager offers:

Proxy Forwarding

SIP messages controlling call setup can be forwarded either to other SIP servers or to SIP gateways.

Proxy Authentication

SIP security is insured by authenticating all SIP messages using SIP credentials.

URI Mapping Engine

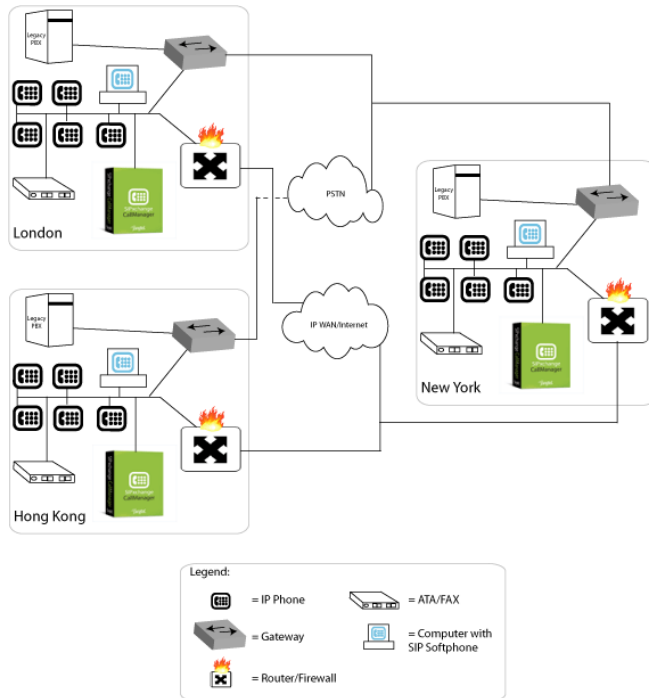
The SIP URI mapping engine maps calls to target destinations based

Rules Based Routing

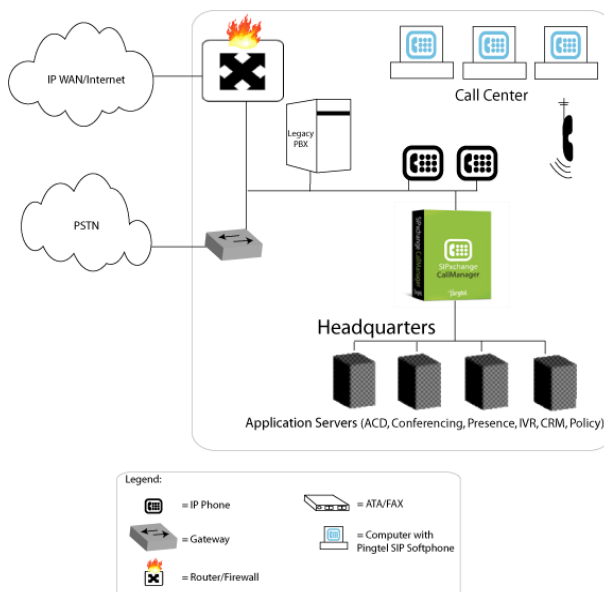
A two-stage matching procedure enables if/then processing on route selection.



Toll Bypass



Application Router



SIP Routing Attributes

Pingtel SIPxchange CallManager offers flexible deployment options

Toll Bypass/WAN Consolidation Solution

Toll Bypass/WAN consolidation converts traditional voice calls between headquarters, branches, and home offices into compressed data packets and carries these calls over the private IP network or the public Internet to dramatically reduce long distance toll charges. SIPxchange CallManager is widely used by Enterprises to route calls from office to office; and in most cases integrates directly with legacy PBXes and the PSTN.

Application Router

SIPxchange CallManager is the perfect solution for centralizing SIP signalling. Centralized signalling and call control allows an enterprise to link all of their SIP applications in to a common administration interface. SIPxchange CallManager enables Enterprises to finally link in their disparate voice systems – conferencing, messaging, call center – by utilizing direct connect SIP or through media gateways.

Benefits

Easy to use

System Administration 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.

Convenient VoIP system migration

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

Immediate savings on calls within the network

Dramatically reduces the costs associated with proprietary solutions.

Future-proofed voice system

Offers scalability options for easy application and technology upgrades in the future.

Accommodation of large call volumes in a VoIP network

SIP protocol usage eliminates long distance toll charges between offices and the need for costly PBX tie lines.

Low-risk/high-reward solution

Open architecture, full SIP implementation and adherence to SIP standards means development and interoperability issues are minimal.

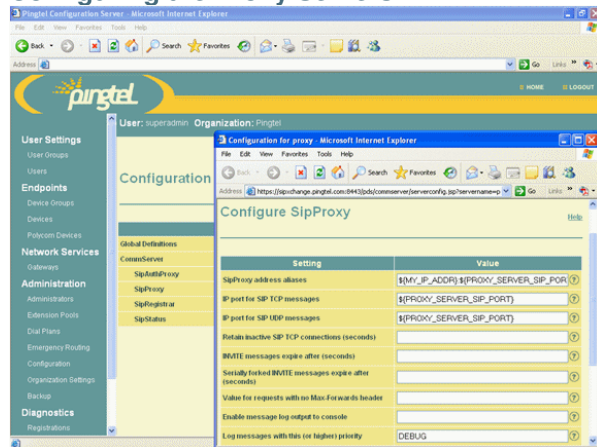
Protection of prior telecommunications investment

VoIP phase-in while continuing to operate legacy voicemail system.

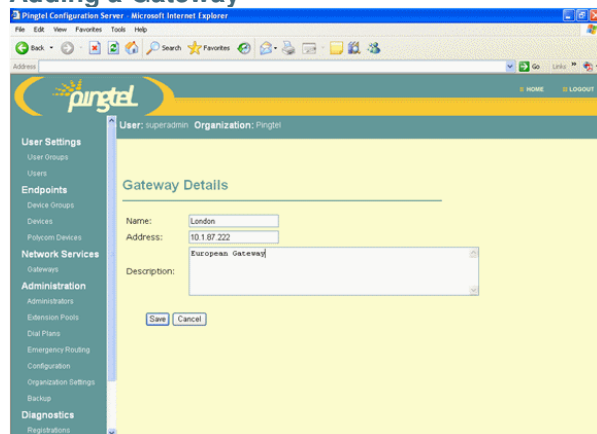
Most flexible method for routing calls

Powerful routing engine provides capabilities that surpass even those of traditional CLASS 4 PSTN equipment.

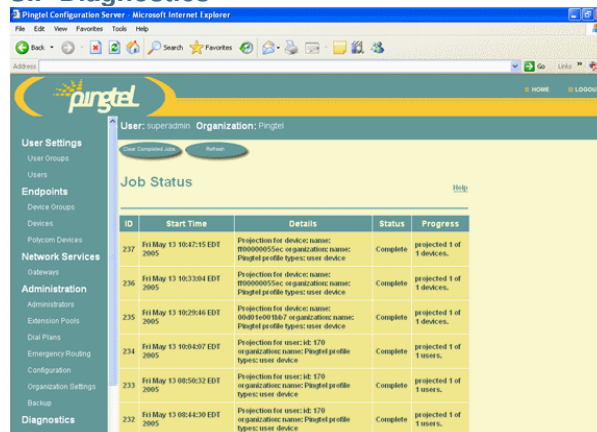
Configuring the Proxy Servers



Adding a Gateway



SIP Diagnostics



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
- Full Hot Standby (using fixed IP addressing schemes)
- Hunt group
- Multi-site / multi-location station and gateway
- Off-premises stations
- URI mapping engine for call routing and inter-company (/domain) SIP calls
- Web services APIs for Config server

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

- 36,000 BHCC per server



SIPxchange PBX Subscription, Maintenance

Pingtel's SIPxchange CallManager 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.

Pingtel is a registered trademark of Pingtel Corp. as is SIPxchange and all Pingtel logos. Other trademarks used herein are the property of their respective owner.

June 8, 2005

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.

Pingtel Corp.
400 West Cummings Park
Suite 2200
Woburn, MA 01801 USA
781-938-5306
800-PINGTEL (US only)
Fax: 781-938-9650