Product Overview
Software
SIPxchange CallManager

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 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:

Call Routing Engine
Provides the most flexible method for routing calls for long distance cost savings. Capabilities surpass traditional CLASS 4 PSTN equipment

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.

Founding Member SIPfoundry

Benefits

Most flexible method for routing calls
Powerful routing engine provides capabilities that surpass even those of traditional CLASS 4 PSTN equipment.

Easy to install, configure and manage
Browser based system management tool allows you to simply configure your server, managed devices (gateways) and go!

Legacy telecommunications investment protection
Standards-based system supports existing network and meets all requirements for TDM PBX replacement or augmentation.

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.

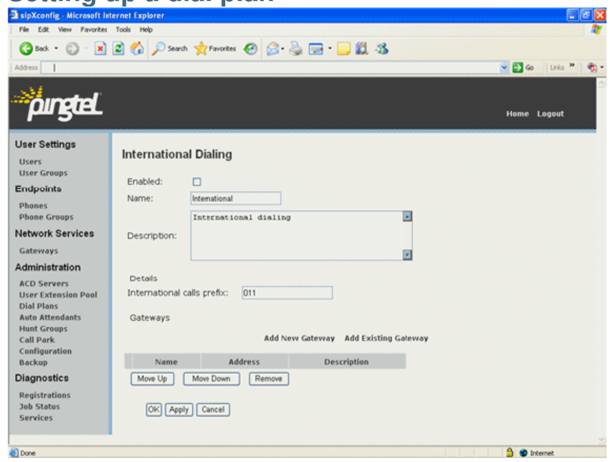
Future-proofed voice system
100 percent SIP, standards-based system, offers scalability options for easy application and technology upgrades in the future.

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.

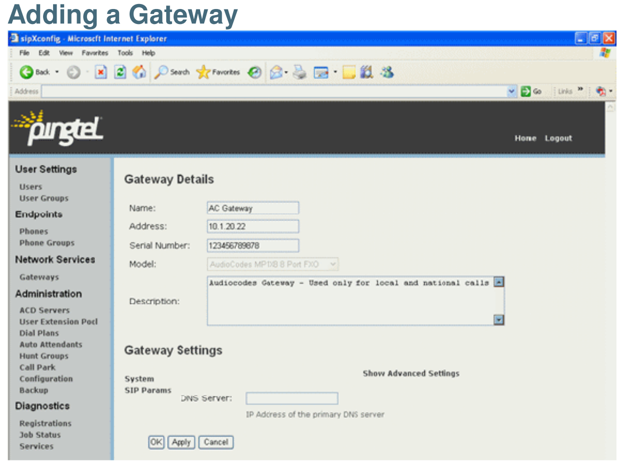
Low-risk/high-reward solution
Open architecture, full SIP implementation and adherence to SIP standards means development and interoperability issues are minimal.

Protection of prior telecommunications investment
VoIP phase-in while continuing to operate legacy voicemail system.

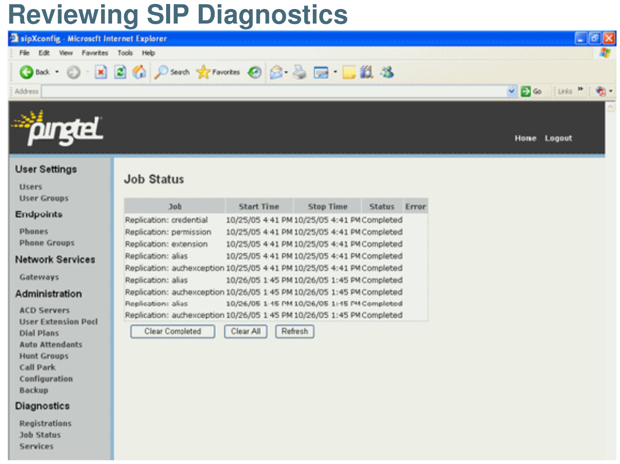
Setting up a dial plan



Adding a Gateway



Reviewing 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 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

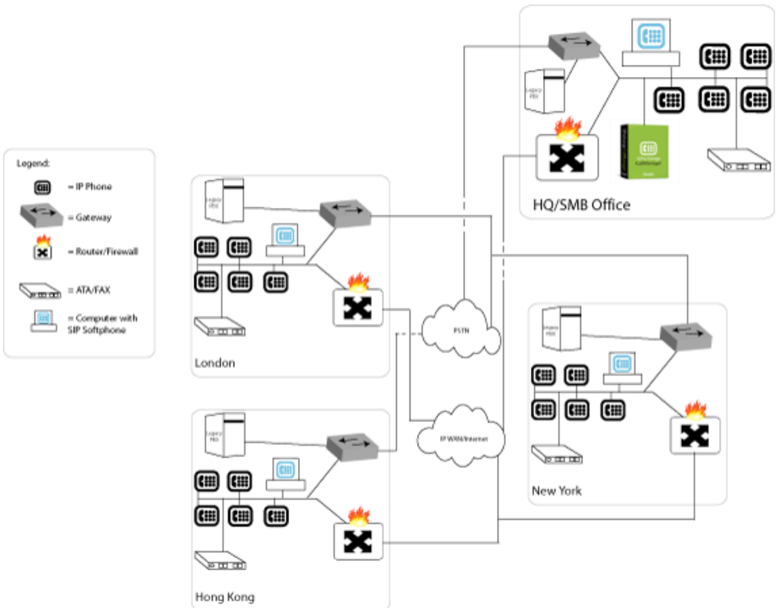
Server requirements

- Pentium 4 2.8 GHz with Hyper threading
- 1 GB of Memory
- 80 GB Hard drive
- Red Hat Enterprise Linux 4.0 ES (Edge Server)

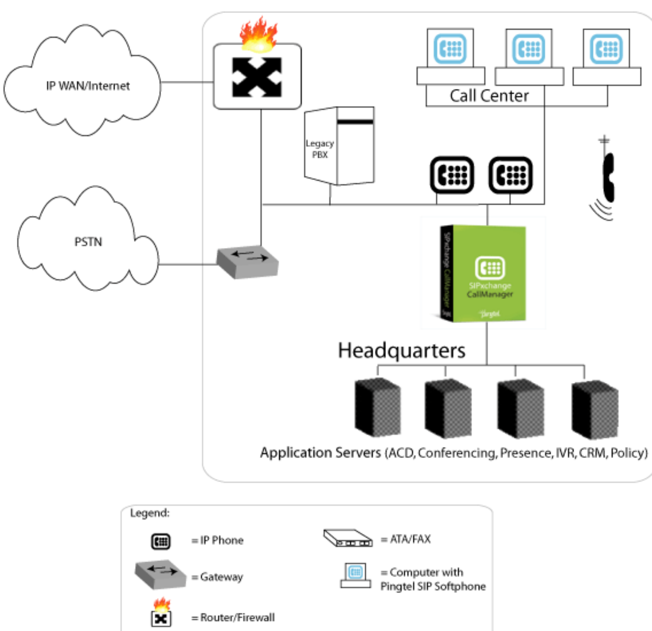
Limits

- Model - 5 CPS (18,000 BHCC)
- Model - 10 CPS (36,000 BHCC)
- Model - 15 CPS (54,000 BHCC)

Toll Bypass



Application Router



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 dispráte voice systems – conferencing, messaging, call center – by utilizing direct connect SIP or through media gateways.

Product Overview

Software

SIPxchange CallManager

SIPxchange CallManager Subscriptions

SIPxchange CallManager is available as either a 1-year or 3-year subscription. See table below for subscription features. Subscriptions are associated with a customer-specified host and licensed number of users.

If you desire installation support, or design and deployment assistance, please consider Pingtel's Jump Start Installation Programs and/or Technical Assistance Center (TAC) Support Bundle service plans. Details on these plans are described in Pingtel's **Subscriptions and Services Overview**.

Subscription Features (1-Year or 3-Year Term)	
Software & updates	√
Documentation & updates	√
Web-based self help	√
Technical Assistance Center (TAC) Access	**Assistance with remedying an operational issue in a previously operational system (Note: Excludes system configuration assistance during initial installation)**
Electronic TAC access	√
Electronic TAC support SLA	**1 business day**
Phone-based TAC support	**Severity One issues only**
Phone-based TAC support SLA	**Fault isolation and resolution for Severity One issues - 24x7 within 15 minutes during normal business hours and within 1 hour outside business hours**

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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 Linux style 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 http://www.pingtel.com.

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