SIPxchange ECS

A complete SIP PBX solution (and more) that <u>you</u> control? Yes!

Pingtel's SIPxchange ECS - The PBX for Linux, is a 100% SIP, enterprise-grade, fully featured software IP PBX with integrated voice mail, ACD, multiple auto attendants and a web-based system configuration and management tool. SIPxchange ECS is based entirely on the Session Initiation Protocol (SIP) standard and operates on popular Red Hat Linux servers, PSTN/IP gateways and end points.

SIPxchange ECS (**E**nterprise **C**ommunications **S**erver) provides dramatic savings for medium and large enterprises with up to 1000 users per location, while delivering the redundancy, interoperability, scalability and "plug and play" benefits enterprises expect. ECS is a SIP PBX "and more"; including a robust real time architecture, integrated presence, and an ACD

SIPxchange ECS is the first enterprise-grade IP PBX available in open source. It offers proven, robust functionality for up to 1000 seats on a single server, and provides low-cost solutions for IP PBX, headquarters, branch offices, home workers, and call centers.

SIPxchange ECS interoperates seamlessly with legacy components such as TDM PBX, enabling users to choose a phased-in approach to VoIP implementations.

SIPxchange ECS 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 ECS 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 fully featured, standards-based and software or appliance based communications solution at the lowest total cost of ownership (TCO).

Key Attributes
SIPxchange ECS offers:

Voice Mail
Integrated voice mail system.

Automated Call Distribution
Automatically distributes calls to multiple users or queues through intelligent routing.

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.

Founding Member
SIPfoundry

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 180 days or less.

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

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

Unmatched system flexibility, interoperability 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 and additional trunk lines between distributed offices
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.
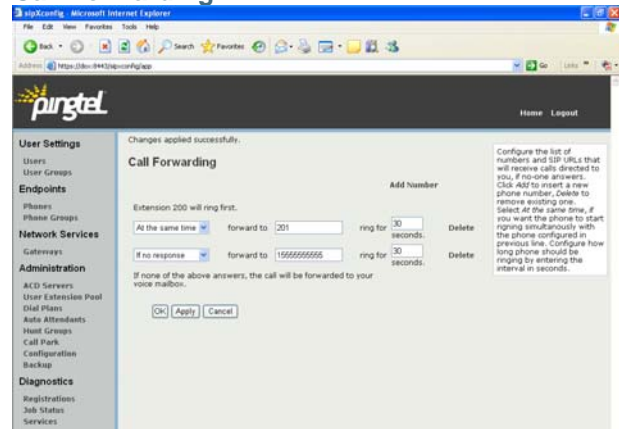
Future-proofed network
100 percent SIP, standards-based system enables easy moves, adds and changes, installation of new features and deployment of new applications as they come available.
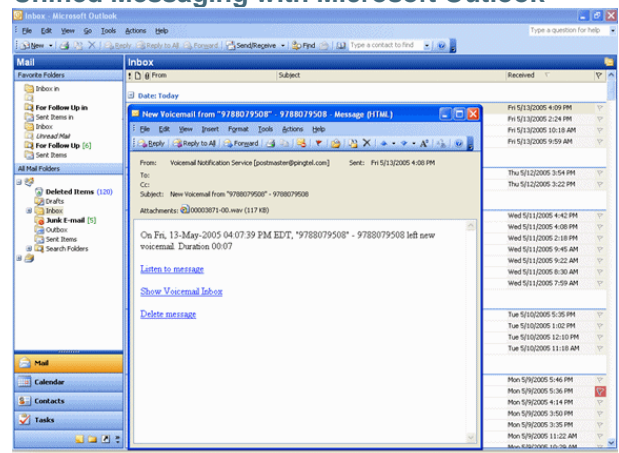
Configuring the Dial Plan
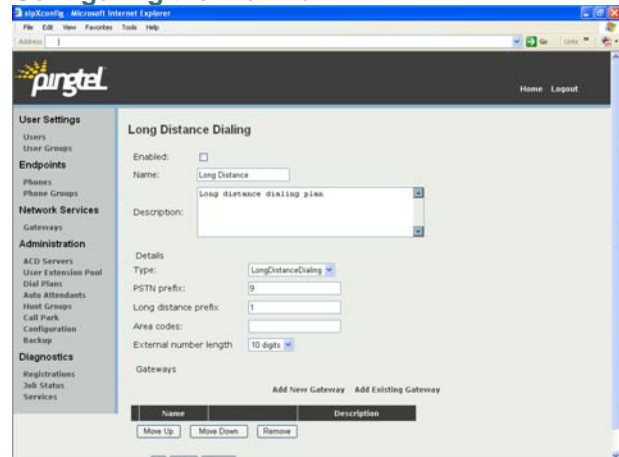
Call Forwarding

Unified Messaging with Microsoft Outlook

ECS/PBX System Features

- Automatic Call Distribution (ACD)
- Aliasing facility
- Automatic Route Selection (ARS)
- Auto-restart services after power failure
- Browser-based configuration and management
- Call Admission Control
- Call Park / Retreive
- Call Pick-up
- CSV Import Tool for Users and SIP
- Dynamic call forwarding
- Hunt groups
- Message waiting indication
- Multiple codec support
- Multi-site / multi-location station and gateway
- Multi-station appearance
- Outbound call blocking
- Search Engine for simple configuration
- SOAP Admin Interface
- Scheduled back-ups
- System Security
- URI mapping engine

User Features

- Browser-based portal
- 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

Voicemail Features

- Browser-based portal
- Distribution Lists
- Email notification of new voicemail messages
- Folders for message organization
- Multiple user customizable voicemail greetings
- Remote voicemail access
- Unified Messaging

Auto Attendant Features

- Customizable auto attendant message for main greeting
- Customizable IVR menus with VXML
- Dial by extension
- Dial by name
- Multiple Auto Attendants
- Multiple Level Auto Attendants
- Operator escape from anywhere
- Time of Day, Day of Week and Holiday Scheduling

Pingtel Managed Devices and Features

- Polycom SoundPoint IP 301, 501, 601
- Polycom Firmware Management
- Polycom SoundStation IP 4000 SIP
- Snom 320, 360
- Grandstream BudgeTone, HandyTone
- Cisco ATA 186/188

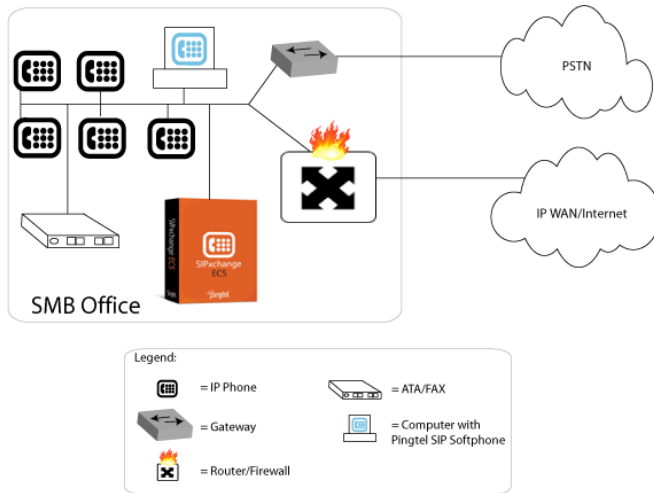## SIP Implementation (Standards-Based)

- 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
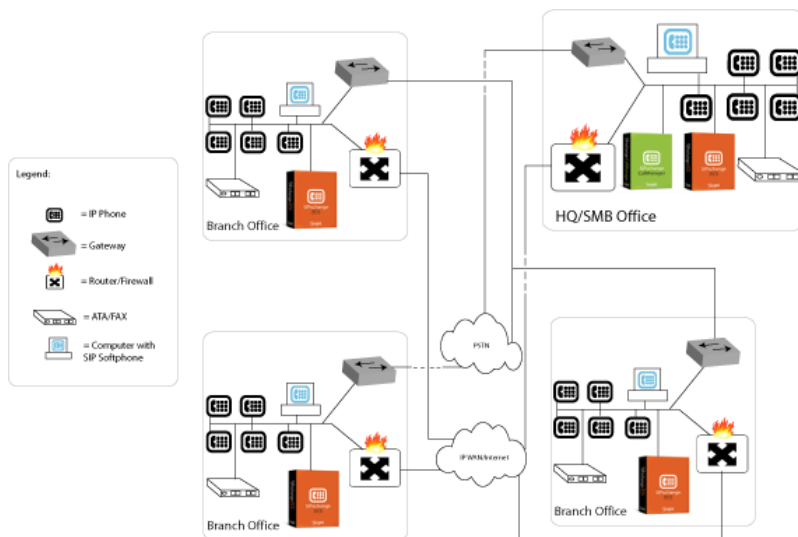- Configurable SIP ports
- TLS

Limits

- Model 35 – 35 users or less
- Model 75 – 75 users or less
- Model 125 – 125 users or less
- Model 250 – 250 users or less
- Model 500 – 500 users or less
- Model 1000 – 1000 users or less
- Unlimited PSTN Connections with media gateway
- 30 media server ports for Auto Attendant, Voice Mail, and ACD Queues
- 300 Voice Mail boxes

Note – Some features are dependant upon other SIP components such as Phones and Gateways.

Software
SIPxchange ECS

pingtel

SIPxchange ECS offers flexible deployment options

Single-site PBX

SMB Office

Legend:

- = IP Phone
- = Gateway
- = Router/Firewall
- = ATA/FAX
- = Computer with Pingtel SIP Softphone

Multi-site PBX

Legend:

- = IP Phone
- = Gateway
- = Router/Firewall
- = ATA/FAX
- = Computer with SIP Softphone

Branch Office

HQ/SMB Office

Branch Office

Branch Office

Single-Site PBX

SIPxchange ECS is the ideal Enterprise IP voice solution for any office with up to 1000 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's SIPxchange ECS 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 ECS 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.

SIPxchange ECS Subscriptions

SIPxchange ECS 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)	
Compiled binaries & electronic documentation	√
Software & 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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