



## SIPxchange ECS 50

A complete SIP PBX solution (and more) that you control? As simple as one, two, three? Yes!

Pingtel's Certified SIPxchange Solutions are 100 percent SIP, enterprise-grade, full-featured, IP PBX solutions with integrated voice mail, ACD, multiple auto attendants and a web-based system configuration and management tool.

The SIPxchange ECS (**E**nterprise **C**ommunications **S**erver) 50 provides dramatic savings for small and medium enterprises with up to 50 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.

The SIPxchange ECS 50 is the first enterprise-grade SIP PBX developed and available in open source. It offers proven, robust functionality up to 50 seats on a single server, and provides low-cost solutions for IP PBX, branch offices, home workers, key systems and call centers.

The SIPxchange ECS 50 interoperates seamlessly with legacy systems such as TDM PBX, enabling users to choose a phased-in approach to VoIP implementations.

Packaged as an appliance (no phones) or a system (with phones), the SIPxchange ECS 50 is a completely integrated solution, incorporating Pingtel's award winning SIPxchange ECS software, the Red Hat Enterprise Linux 4 operating system, an Intel-based server platform, AudioCodes gateways and Polycom phones.

These solutions are part of a family of SIPxchange products that, in combination, comprise fully featured, standards-based, software or appliance based communications solutions at the lowest total cost of ownership (TCO).



### Key Attributes

SIPxchange ECS 50:

#### Voice Mail

Integrated voice mail system.

#### Automated Call Distribution

Automatically distribute 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.





# Product Overview

## Certified SIPxchange Solution

### SIPxchange ECS 50

## Benefits

### Lower Total Cost of Ownership (TCO)

Off-the-shelf appliance, with pre-packaged support options allow you to realize a typical return on investment (ROI) of 180 days or less.

### Easy to configure, install and manage

Pre-packaged appliance with browser based system management tool allows you to simply add your phones, configure 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 and interoperability

SIP standards compliance ensures interoperability with 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.

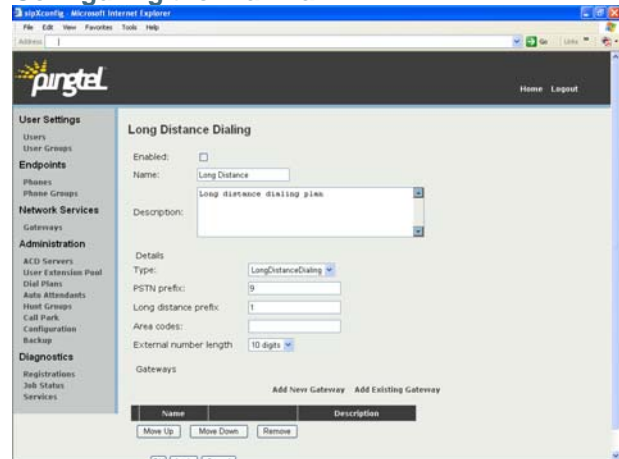
### Convenient VoIP system migration

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

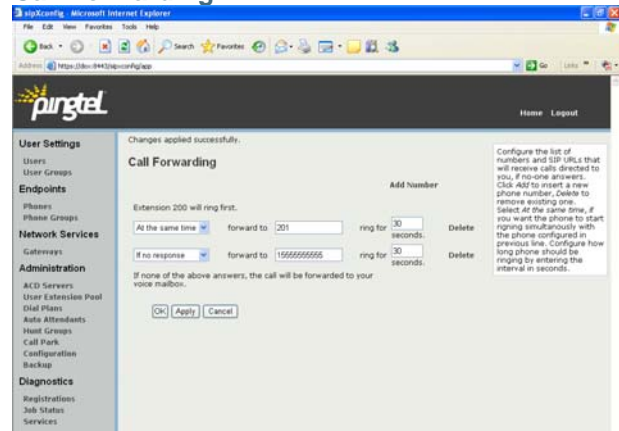
### Future-proofed network

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.

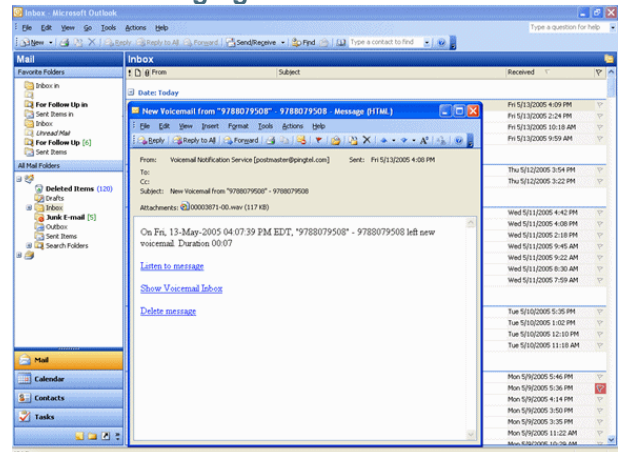
## Configuring the Dial Plan



## 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 / Retrieve
- Dynamic call forwarding
- Hunt groups
- Message waiting indication
- Multiple codec support
- Multi-site / multi-location station and gateway
- Multi-station appearance
- Outbound call blocking
- Scheduled back-ups
- System Security
- URI mapping engine for call routing and inter-company (/domain) SIP calls

## 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
- DTMF interface (touch tone)
- Email notification of new voicemail messages
- Folders for message organization
- Multiple user customizable voicemail greetings
- Operator escape from anywhere
- 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

## Pingtel Managed Devices

- Polycom SoundPoint IP 300, 500, 600
- Polycom SoundPoint IP 301, 501, 601
- Polycom SoundStation IP 4000 SIP
- Snom 320, 360
- Grandstream BudgeTone, HandyTone
- AudioCodes TP-260/SIP, MP-104, MP-108
- 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 50/50 – 50 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.*

**Step One**

Choose Options



- Gateways (one or two external 8-port analog)

**Step Two**

Choose Your Phones (Optional)



Polycom 300 Series



Polycom 500 Series



Polycom 600 Series

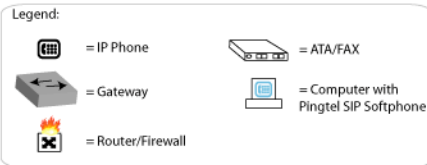
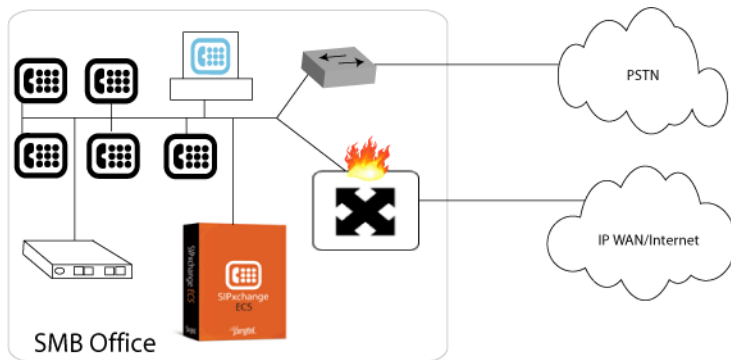
**Step Three**

Choose Your Support Plan

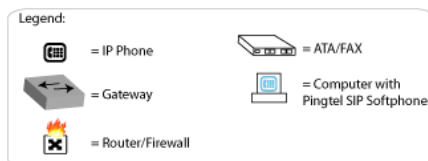
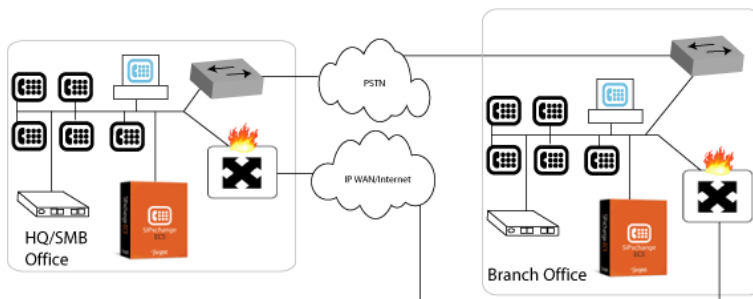
<u>Pingtel Total Solution Support Options</u>							
	<u>Software Support</u>			<u>Hardware Support</u>			
	Duration	SIPxchange	RHEL v4.0	Support Level	Support Window	Support Delivery	Parts Replace
<b>Option 1</b>	1-Year or 3-year	√	√	Priority	7x24x365	Phone Support	Next Business Day
<b>Option 2</b>	1-Year or 3-year	√	√	Priority On-Site - NBD	7x24x365	On-Site Support	Next Business Day
<b>Option 3</b>	1-Year or 3-year	√	√	Priority On-Site	7x24x365	On-Site Support	Same Business Day

Note: Some support options may not be available in all geographies.

## Single-site PBX



## Multi-site PBX



## SIPxchange ECS 50 offers flexible deployment options

### Single-Site PBX

SIPxchange ECS 50 is the ideal Enterprise IP voice solution for any office with up to 50 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

The SIPxchange ECS 50 provides a complete solution to your enterprise telephony needs. The system's architecture lets you easily distribute solutions 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, the SIPxchange ECS 50 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.



Server Hardware Specifications	
Hardware Platform	Pentium 4, 2.8 GHz, 512MB RAM, 80GB HD, CD ROM
Network Interface	10/100 Ethernet RJ-45 Interface
Form Factor	ATX Tower
Weight	30 lbs
AC Power	100-120; 220-240 VAC 47-63 Hz
Operating Environment	10 to 35°C (50 to 95°F) 50% to 90% relative humidity, noncondensing
Regulatory Approval	UL/CUL/CSA; CE Mark, CE Safety; FCC Part 15, Class A; ISTA certified

AudioCodes MP-104 and MP-108 Hardware Specifications	
Network Interface	10/100 Ethernet RJ-45 Interface
Telephony Analog Interface	4-port or 8-port analog – RJ-11
Form Factor	Table Top
AC Power	100-120; 220-240 VAC 47-63 Hz
Operating Environment	5 to 55°C (23 to 131°F) 10% to 90% relative humidity, noncondensing

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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 [www.pingtel.com](http://www.pingtel.com).

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