sipXecs Enterprise Communications System

A complete IP PBX for today and tomorrow that is easy to use, fully featured and robust

sipXecs Enterprise Communications System (ECS) is a highly robust and scalable, enterprise-grade, and fully featured unified communications solution with integrated voicemail, unified messaging, presence, call center, multiple auto attendants, paging and intercom services and a powerful plug & play web-based configuration and management system. sipXecs is based entirely on the Session Initiation Protocol (SIP) standard and operates on standard Linux operating system. It interoperates with a large number of third party phones, PSTN/IP gateways and applications such as Microsoft Exchange 2007 without compromising ease of use.

sipXecs is a 2nd generation voice over IP (VoIP) open source system not built for geeks but enterprise users requiring easy to use, stable and scalable solutions for mission critical applications. Thousands of small and large enterprises rely on sipXecs for their communications needs ranging from 10 people offices with 4 analog trunk lines to companies such as Amazon.com with thousands of users.

sipXecs provides dramatic cost savings while delivering the system and trunk redundancy, interoperability, scalability and plug–and-play benefits that SMB customers expect. sipXecs is a unified communications solution based on a robust real-time architecture that grows with your business.

sipXecs is the first business-grade IP PBX built on standards based open source. It offers proven, robust functionality for thousands of seats on an optionally redundant system, and provides low-cost communications solutions for main and remote offices, home workers, and call centers.

sipXecs interoperates with TDM PBXs via Voice over IP (VoIP) gateways, enabling users to choose a phased-in approach to VoIP implementations. sipXecs relies on standards and utilizes proven Internet techniques and a distributed architecture to create a highly secure and available IP voice system.

Switching to sipXecs has proven to be the right choice for many enterprises already. Open source offers lowest total cost of ownership (TCO) and it does not lock you into a vendor specific and proprietary solution.

SIPfoundry

Key Attributes

sipXecs offers:

Voice Mail
Integrated voice mail system with personal auto-attendant per user

Automated Call Distribution
Integrated Call Center solution that distributes calls to multiple agents and 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
True Plug & Play. Intuitive browser interface for centralized control and management of dial plans, users and endpoints.

User Self-control
Powerful Web user portal puts the user in control to individually manage key features like time-based find-me / follow-me.

Really easy to install and use
sipXecs puts the user in control. It was designed so that everybody can be the administrator. It installs in hours and all adds, moves and changes are done by the user using a Web interface.

Used Asterisk before?
If you know Asterisk you will find sipXecs easy to use, much more stable, and scalable. No more guessing about how things work, no more asking the experts. The sipXecs project is adding new capabilities fast following a disciplined roadmap.

http://www.sipfoundry.org

Overview
sipXecs ECS

Benefits

Commercial support available as an option
Pingtel's Support and Maintenance services provide support and ongoing updates allowing you to realize return on investment (ROI) quickly.

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! True intuitive management with click-and-go.

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

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.

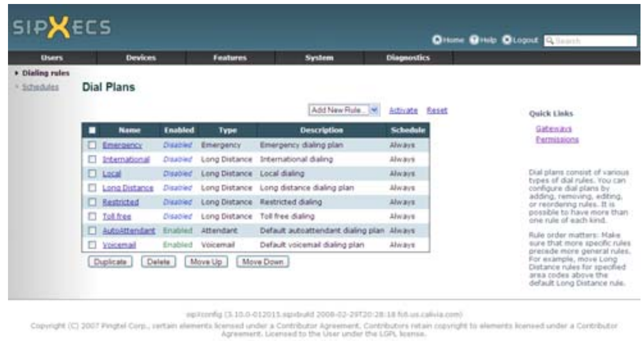
Elimination of expensive second phone lines for remote workers and additional trunk lines between distributed offices
Leverages existing broadband connections for voice and data.

Enhanced employee productivity
Delivers a wide range of innovative user features including Microsoft desktop and Exchange 2007 integration.

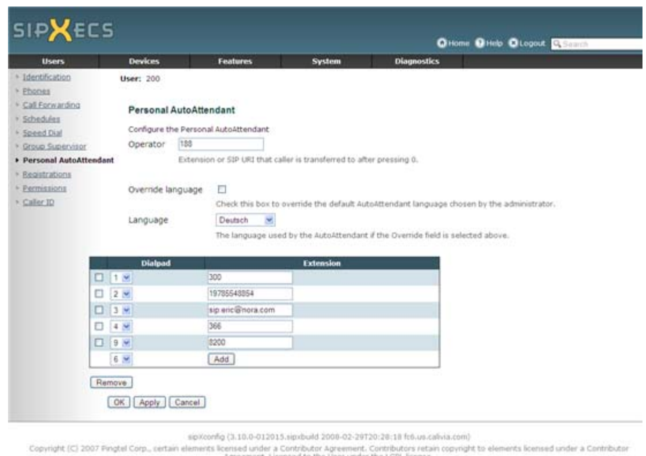
Convenient VoIP system migration
Begin with a single office and extend IP telephony to your remaining organization in a timeframe that suits you.

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 become available.

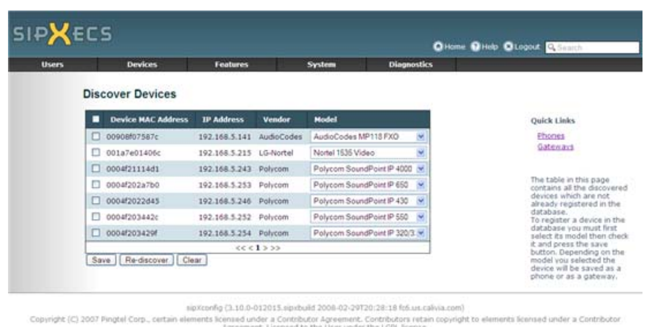
Configuring the Dial Plan is easy



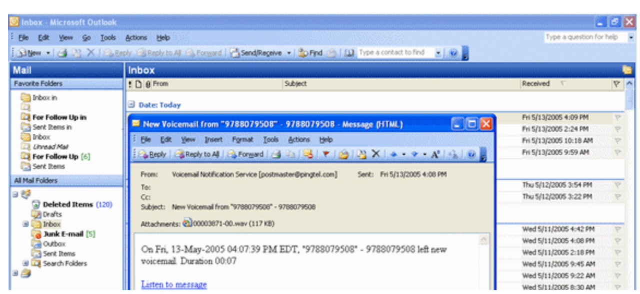
Personal Auto-Attendant per user



One – two – three clicks to configuring users, phones and lines



Unified Messaging with Microsoft Outlook



sipXecs ECS

sipXecs Telephony System Features

- Transfer (consultative & blind)
- Call coverage
- Call hold / retrieve
- Consultation hold
- Music on Hold for IETF standards compliant phones
- Uploadable music file
- 3-way conference
- Call pickup (global and directed call pickup)
- Call park & retrieve
- SIP URI dialing
- CLID (Calling Line Identification)
- CNIP (Calling party Name Identification Presentation)
- CLIP (Call Line Identification Presentation)
- CLIR (Call Line Identification Restriction)
- Per gateway CLIP manipulation
- Call waiting / retrieve
- Do not Disturb (DnD)
- Forward on busy, no answer, do not disturb
- Multiple line appearances
- Multiple calls per line
- Multiple station appearance
- Outbound call blocking
- Click-to-dial (Windows)
- Redial
- Call history (dialed, missed, received)
- Auto off-hook / ring down
- Incoming only
- Configuration of individual Speed Dial softkeys
- Auto-generation of Directory information

User Self-Control

- Every user on the system gets access to a personal Web user portal for self-management and control
- Management of voicemail
- Configuration of unified messaging preferences
- Time based find-me / follow-me
- Flexible configuration of call forwarding
- Personal call history
- Personal phone book, speed dial and presence management
- ACD presence and supervision capabilities
- Individual phone management

Superior Voice Quality

- Peer-to-peer media routing with quality optimization
- Lower delay and jitter
- Support for any codec supported by the phone
- Codec negotiation directly between phones

Dial Plan

- Easy to use GUI based dial plan manipulation
- Time-based dialing rules with different admin defined schedules
- Rules based least cost routing
- Automatic gateway redundancy and failover
- Specific E911 routing
- Permission based rules
- Prefix manipulation
- Dial plan templating for international dial plans

. User Management

- Create a user, provision a phone and assign a line in only three clicks – easy!
- Numeric or alpha-numeric User ID
- User PIN management (UI or TUI)
- Aliasing facility (numeric and alpha-numeric aliases)
- Extension and alias uniqueness assurance
- Granular per user permissions
- Call permissions:
    - 900 Dialing
    - International Dialing
    - Long Distance Dialing
    - Mobile Dialing
    - Local Dialing
    - Toll Free Dialing
    - Forward Calls External
- System permissions:
    - User has voicemail inbox
    - User listed in auto-attendant directory
    - User can record system prompts
    - User has superuser access
    - User allowed to change PIN from TUI
- Custom permissions added by the administrator
- Supervisor permission for groups (e.g. Call Center supervisor)
- SIP password management for security
- User groups with group properties
- Per user call forwarding (find me / follow me)
    - To local extension, PSTN number, or SIP address
    - Scheduled forwarding based on user defined individual schedules
    - Parallel or serial ring
    - Allows definition of ring time before trying next number
    - Allows several forwarding destinations
    - Follow-me configuration using user portal
- Extension pool with automatic assignment
- Per user Caller ID (CLID) assignment
- Per user Caller ID blocking

PSTN Trunking

- Unlimited number of PSTN gateways and trunk lines
- Direct Inward Dialing (DID)
- Local DID per gateway
- DNIS
- CLIP Management
    - User CLIP
    - Gateway default CLIP
    - Prefix stripping / appending
- Per gateway CLIR
- Automatic Route Selection (ARS)
- Least-cost routing (LCR)
- Automatic failover if unavailable
- Automatic failover if busy
- FAX support (pass-through)
- Mixing of PSTN trunks with SIP trunks

sipXecs ECS

Performance

- Unlimited number of simultaneous calls
- Unlimited number of trunk lines
- 54,000 BHCC, 120,000 BHCC redundant(dependant on server platform)
- Up to 10,000 users per dual-server HA system
- Automatic time distribution of re-registrations

High Availability

- Optionally fully redundant call control system
- Based on DNS SRV (no cluster required)
- Load balance under normal operating conditions
- Geographic dispersion of redundant systems
- Real-time synchronization of state information
- Reports on load distribution

Security

- All outbound calls authenticated
- DoS attack prevention
- HTTPS secure Web access
- Secure user SIP password management
- TLS based signaling for SIP trunks (requires session border controller)

System Administration Features

- Browser based configuration and management
- LDAP integration (OpenLDAP)
- Integration with Microsoft Exchange 2007 for voicemail and Active Directory
- SOAP Web Services interface
- CSV import and export of user and device data
- Integrated backup & restore
- Scheduled backups
- Diagnostics
  - Display active registrations
  - Display job status
  - Status of services
  - Snapshot logs for debugging
  - Logging (customizable log levels, message log per service)
- Domain Aliasing
- Support for DNS SRV
- Automatic restart after power failure
- Server statistics (integrated graphs and SNMP)
- Login history report (successful and unsuccessful)

Plug & Play Device Management

- Plug & play management of phones (see the list of plug & play managed devices)
- Plug & play management of PSTN gateways (see the list of plug & play managed devices)
- Auto-generation of phone / gateway config profiles
- Auto-pickup of profile by the phones / gateways
- Centralized management of all the parameters
- Centralized backup and restore of all configurations
- Auto-generation of lines by assigning users to devices
- Device group management & properties
- Firmware upgrade management
- Auto-discovery of phones and gateways

Voicemail Subsystem

- Integrated voicemail system at no extra cost
- Browser based user portal for voicemail management
- Message Waiting Indication (MWI)
- User configurable distribution lists
- Manage Notifications:
  - Email notification of new voicemail messages
  - Forwarding of message as .wav file
- Manage folders: Folders for message organization
- Manage greetings: Multiple customizable greetings
- Operator escape from anywhere
- Remote voicemail access
- Unlimited number of inboxes
- Up to 60 virtual media server ports per server
- Message store only limited by disk size
- Auto-removal of deleted messages
- Daily report on disk usage sent to admin

Personal Auto Attendant

- User configurable personal auto-attendant for every user on the system
- Individual zero-out to a personal assistant or receptionist
- Individual selection of language
- Personal greeting

Auto Attendants

- Unlimited number of auto-attendants
- Customizable IVR menus
- Dial by extension and name
- Night and holiday service
- Special auto-attendant
- Transfer on invalid response
- Nested auto-attendants (multi-level)
- Fully customizable actions: Operator, Dial by Name, Repeat Prompt, Voicemail login, Disconnect, Auto-Attendant, Goto Extension, Deposit Voicemail
- Uploadable custom prompts
- Configurable DTMF handling

Hunt Groups

- Unlimited number of hunt groups
- Serial and parallel forking (rings sequentially or at the same time)
- Configurable ring time per attempt
- Enable / disable user call forwarding rules while hunting
- Flexible configuration of destinations if no answer

Call Park Server

- Unlimited number of park orbits
- Visual indication on the phone of the state of the park orbit using the presence server
- Music on park
- Configurable call retrieve code
- Configurable call retrieve timeout
- Automatic park timeout
- Configurable park escape key
- Allow multiple calls on one orbit

Call Center Server (ACD)

- Supports several ACD servers, optional on separate server hardware
- Several (unlimited) queues per server
- Several lines per queue
- Support trunk lines (many calls per line) or single call per line
- Dedicated overflow queues or overflow to hunt group, extension or voicemail
- Configurable call routing scheme per queue:
  - Ring All
  - Circular
  - Linear
  - Longest idle
- Agent barge in (early termination of welcome message if agent becomes available)
- Agent presence monitor using presence server
- Separate welcome and queue audio
- Call termination tone or audio
- Configurable answer mode
- Agent wrap-up time configurable per queue
- Auto sign-out of agents if calls are not answered
- Configurable maximum ring delay
- Configurable maximum queue length
- Configurable maximum wait time until overflow condition
- Unlimited number of agents per queue
- Real-time Statistics:
  - Agent statistics
  - Call statistics
  - Queue statistics
- Supervisor authorization for agent monitoring per group
- ACD historic reports for agents, calls, queues
- All reporting stored in database for post-processing if needed

Group Paging

- Integrated group paging server
- Unlimited number of paging groups
- Supports regular SIP phones using auto-answer
- Supports dedicated in-ceiling devices (SIP)

Analog Lines (FXS)

- Supports any SIP compliant FXS gateway
- FAX support (pass-through)
- Analog cordless phone support
- Plug & play management of FXS gateways from AudioCodes

SIP Trunking

- SIP call origination & termination
- Branch office routing
- Internet Calling rules define call routing
- Proxy to proxy interconnect using ACLs
- Least-cost-routing (LCR)
- NAT and firewall traversal requires a session border controller (SBC)
- Registration with an ITSP requires an SBC

Call Detail Records collection

- Comprehensive Call Detail Records collection
- All data stored in a database accessible by third party reporting packages
- Display of CDRs in the administration user interface in real-time
- Easy export to Excel of CDR data
- Fully supports redundant call control
- Real-time display of currently active calls in the system
- Individual call history in user portal

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
- RFC 3327 Path header (pending)
- 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
- Several newer IETF drafts
- No proprietary add-ons

Supported Devices plug & play

- **Phones and gateways with commercial support:**
- Polycom SoundPoint and SoundStation
- LG-Nortel LIP-6804, 6812, 6830, 11xx, video phone
- Snom phones (not including DECT)
- Audiocodes gateways MP-112, MP-114, MP-118, MP-124, Mediant 1000, TP-260

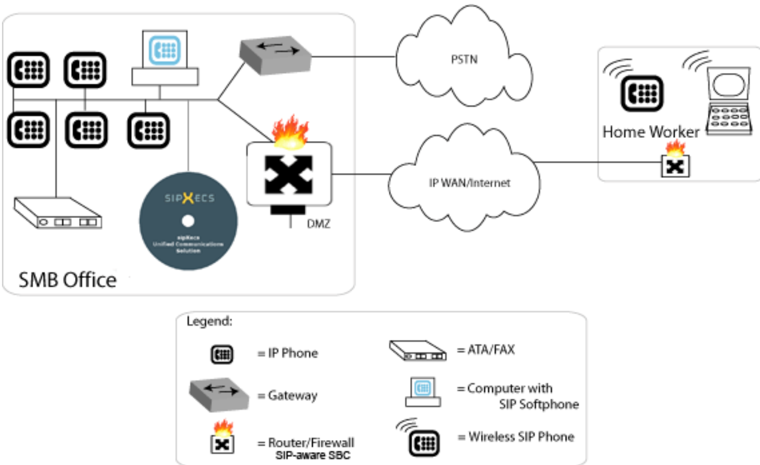
- **Phones and gateways with community support:**
- Grandstream, Linksys, Clearone, IpDialog SIPTone, Cisco, Hitachi, Aastra (pending), Mitel

Required Hardware

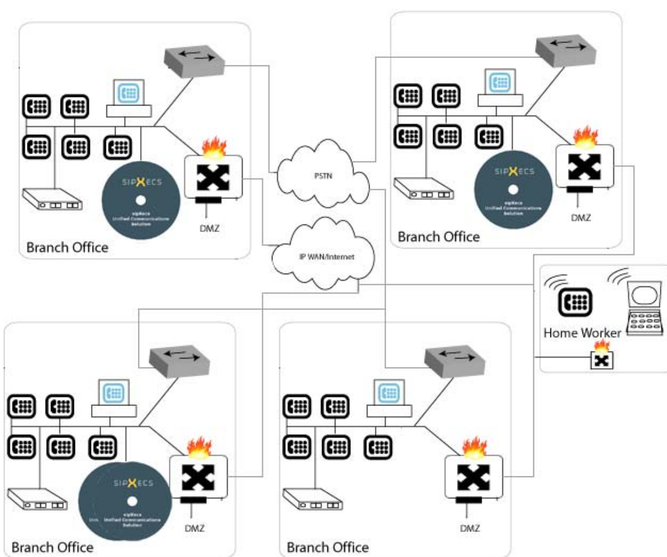
- sipXecs runs on standard Intel servers with no additional required hardware. It runs on Intel/AMD or PowerPC processors. A 64bit version is pending.

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

Single-site IP PBX with Remote Workers



Redundant Multi-site IP PBX with Remote Workers



sipXecs offers flexible deployment options

SMB Single-Site IP PBX

sipXecs is the ideal enterprise IP voice solution for any office, starting at 4 to 12 users with 4 analog trunk lines up to several hundred users in a single location using one or several digital trunk lines (T1 / PRI). It offers an unmatched low purchase price and low operating expenses by leveraging low-cost computer servers, commodity hardware, and Linux.

Large Single-Site IP PBX

sipXecs can scale up easily using more powerful servers or by distributing its components over more than one server. Such configurations can serve up to several thousand users per location.

Multi-site IP PBX

sipXecs 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. Capabilities such as automatic trunk fail-over and redundancy, high-availability, and least cost routing are must-have features in today's environment.

Remote Workers

Full PBX Functionality Remotely - The sipXecs 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, sipXecs 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.

open source

Frequently Asked Questions (FAQ)

Why does sipXecs scale better than other solutions?
sipXecs is architected as a distributed system where the flow of media is separate from signaling. Calls, therefore, do not go through the sipXecs server. Scaling can easily be done adding server hardware to run redundant call control, separate ACD or media servers, etc.

How comes sipXecs can easily support load-sharing redundancy?
sipXecs is a SIP proxy based system and not a back-to-back user agent (B2BUA) design. Like a regular router, it routes SIP messages through a network from hop to hop where every hop is a SIP proxy server. Each proxy can be configured to load-share with one or several other proxies offering true high-availability.

What codecs does sipXecs support?
Since media does not flow through the sipXecs server, phones and gateways negotiate the best codec among themselves on a per call basis. Any codec, including video, that is supported by all end-points that participate in a call will work.

Why does sipXecs use external gateways?
Internal (PCI card based) gateways do not scale and are a single point of failure. There are only that many T1s that can be connected to a single server and there is only that much processing available to process all the media. External gateways offer unlimited scalability, trunk redundancy and failover, and a choice of gateway vendor.

Voice quality – what is the story?
Systems that route media peer-to-peer along the most optimal route and without any additional server in-between offer lower delay and less jitter, which results in better voice quality.

How about video?
The sipXecs server does not sit in the media stream and therefore is not the bottleneck for video. The only limit is the capacity of the LAN.

I need SIP trunking to connect to an ITSP
sipXecs offers secure SIP trunking using a Session Border Controller. For most enterprises this is the best possible solution. For those who need less security we are working on a native SIP trunking gateway that will be part of sipXecs starting with release 4.0.

Why is sipXecs more robust?
sipXecs is engineered with testability in mind. Throughout the code there is very high unit test coverage. In addition, we use an elaborate partly automated, partly manual test framework.

Why is sipXecs more standards compliant than other solutions?
sipXecs uses a real SIP stack developed by the sipXecs team. Many of the sipXecs engineers are part of the IETF and continue to contribute to the SIP standard. Standards compliance is critical to us.

NETWORKWORLD® CLEAR CHOICE

sipXecs wins Clear Choice Award

"sipXecs earns the Clear Choice award for triumphing over the field"

sipXecs won over three different commercial Asterisk based solutions in a test conducted by Miercom Labs.

License
sipXecs is licensed under the L-GPL license. Contributors sign the SIPfoundry contributor agreement, which establishes a shared copyright.

GNU LGPL SOME RIGHTS RESERVED

Plug and Play Management
Zero-touch management is a key feature of sipXecs. Phones and gateways are auto-discovered and auto-configured. All configurations are centrally backed up. There is no need to ever use the device's Web interface.

AudioCodes LG-NORTEL snom NORTEL POLYCOM

Devices from the above manufacturers are fully supported. Additional manufacturers, like Linksys, Cisco, Grandstream, Aastra (pending), ClearOne, Mitel, and IpDialog are community supported. Creating a new plugin for a new phone or device is easy if your favorite one is not yet supported.

http://www.sipfoundry.org