

EIV-Small Office IP Phone System



Features

- An superior cost-saving phone system without a KTS/PBX box
- Powerful Plug-and-play design offers easy installation
- Support multiple SIP Trunks with excellent interoperability with Soft switch, IP-PBX and general SIP Server
- Intelligent technologies with P2P and SIP protocol convergence
- Quality voice and high performance data through the phones
- Auto-provisioning and Web management for ISP/ITSP carrier
- Rich supplementary call services and phone features
- Supports multiple call/line appearance
- Supports Expansion EDM Module for additional keys (optional)

The EIV-Small Office IP Phones supports a low-cost but very powerful VoIP phone system for small offices. There is no need of any iPBX or IP-KTS central control box to be installed, up to 16 x EIV Small Office IP phones can work together with most supplementary call features. All the IP phones can share the same SIP truck lines to connect to the ISP/ITSP VoIP service network to communicate with outside/PSTN world.

With its sophisticated design, the proprietary Peer-to-Peer protocol is implemented to build a virtual IP environment and offer the switching mechanism with SIP protocol in the same phone. This approach reduces the outbound traffic and demand of more SIP accounts. Of course, as well it generates a higher communication security that all internal calls will go thru P2P channels and not be exposed to the Internet public network.

With this sophisticated design and their versatile features, the EIV-Small Office phone system will increase productivity and gain better investment return. Since it supports both Plug-and-play and Auto-provisioning, users can install the phone easily without the assistance of technician or system installer. For connecting to outside/PSTN, the EIV-Small Office can work with different IP-PBX, soft-switch, IP-Centrex (like SIP-B), or SIP server. In addition, by adding the Extended Dial Module (EDM) to display all extensions status and one touch feature enhances the EIV-Smart Office solution and functionality.

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Specifications

System

- Maximal up to 16 client for one EIV-Small Office IP phone system
- Maximum up to 4 SIP trunk support

Standards

- Peer-to-Peer Proprietary protocol and Peer Discovery
- IETE SIP V2 (RFC3261) standards
- RTP, RTCP
- Static IP assignment, DHCP
- DNS, TFTP/FTP, HTTP
- Telnet
- SNTP, STUN NAT, SNMP V2

Call Features

- Answering Position (VAA/Operator/All Ring / DID)
- Auto Answer
- Call Blocking
- Call Conferencing (3-way)
- Call Forwarding
- Call Hold & Hold Reminder
- Call Transfer/Recall
- Calling Line Identification (Name and Number)
- Distinctive Ringing
- Message Waiting Indicator
- Multiple Line Appearances
- Paging
- Redial, Call log, Phone book, Web, Speed Dial, and On-hook dialing
- WAA/VM

Voice Handling

- Codec G..711 a/μ law, G.729
- Supports VAD, CNG, AGC and Acoustic Echo Cancellation
- Jitter buffering and packet loss concealment
- Full-duplex Speakerphone
- Enhanced voice quality for handset, headset and hands-free

Phone Functions

- Call Record (Answered, missed, dialed)
- Call Timer & Duration
- Hands free/headset Support
- Last Number Redial
- Mute
- Phone book
- Phone lock/unlock
- Speed Dial
- Volume Control

Voice Mail Sub-System

- Message Play, Stop, Next, Previous, Delete
- Message Waiting Indication
- Personal greeting Recording and Playback
- Greeting for Auto-Attendant
- Password change
- Audio storage in IP phone

Auto-Attendant Sub-System

- Call Answer Control
- Day/Night Service – Manual switch
- Dial by extension
- Escape to Attendant
- Greetings & prompt user-configurable

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Network & Security

- QoS: DSCP
- Admin/User Password = Reset Admin Password
- P2P Signal Encryption

Configuration and Management

- SIP Trunk control/status on Line 1 to 4 keys
- Programmable call/phone features on EDM module
- Security and encryption support
- Remote and local configuration
- Web management
- Plug & Play for easy user installation and configuration
- Reset to factory default
- System clock setting by manual or automatic thru SNTP server
- Engineering trace log (syslog)
- Software Upgrade thru TFTP/HTTP server
- Auto-provisioning
- Backup/Restore configuration to/from PC
- Reset Phonebook and Configuration
- Customer Differentiation/Protection support

Display

- 128 x 64 graphic LCD display
- LCD: MWI, Line 1~4, Speaker, Mute, Headset
- Multi languages support (Option)

Keys

- Context-sensitive soft-keys & navigator keys
- Speaker, Mute, Headset Hold, Transfer, Conference, Message, Phonebook, Redial
- Line keys (1~4) for SIP Trunk
- Optional EDM module supported for more function keys (with LED)

I/O Ports

- Dual Base-T/100 Base-TX FJ-45 ports
- Headset port with FJ-9 connector
- EDM module connector

Power Supply

- PoE 802.3af built-in
- 5V DC/1A switching power adapter

