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Administrator's Guide for the Polycom® SoundPoint® IP/SoundStation® IP/ VVX™ Family

SIP 3.2.3 | March 2010 | 1725-11530-323 Rev. A

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About This Guide

The Administrator's Guide for the SoundPoint IP/SoundStation IP/VVX family is for administrators who need to configure, customize, manage, and troubleshoot SoundPoint IP/SoundStation IP/VVX phone systems. This guide covers the SoundPoint IP 320, 321, 330, 331, 335, 430, 450, 550, 560, 650, and 670 desktop phones, the SoundStation IP 6000 and 7000 conference phones, and the Polycom VVX 1500 business media phone.

The following related documents for SoundPoint IP/SoundStation IP/VVX family are available:

- Quick Start Guides, which describe how to assemble the phones
- Quick User Guides, which describe the most basic features available on the phones
- User Guides, which describe the basic and advanced features available on the phones
- Developer's Guide, which assists in the development of applications that run on the SoundPoint IP/SoundStation IP/VVX phone's Microbrowser
- Technical Bulletins, which describe workarounds to existing issues and provide expanded descriptions and examples
- Release Notes, which describe the new and changed features and fixed problems in the latest version of the software

For support or service, please contact your Polycom® reseller or go to Polycom Technical Support at http://www.polycom.com/voicedocumentation/.

Polycom recommends that you record the phone model numbers, software (both the BootROM and SIP), and partner platform for future reference.

SoundPoint IP/SoundStation IP/VVX models: _____

BootROM version: _____

SIP Software version: _____

Partner Platform: _____

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1

Introducing the SoundPoint IP / SoundStation IP / VVX Family

This chapter introduces the SoundPoint IP / SoundStation IP / VVX family, which is supported by the software described in this guide.

The SoundPoint IP / SoundStation IP / VVX family provides a powerful, yet flexible IP communications solution for Ethernet TCP/IP networks, delivering excellent voice quality. The high-resolution graphic display supplies content for call information, multiple languages, directory access, and system status. The SoundPoint IP / SoundStation IP / VVX family supports advanced functionality, including multiple call and flexible line appearances, HTTPS secure provisioning, presence, custom ring tones, and local conferencing.

The SoundPoint IP / SoundStation IP / VVX phones are end points in the overall network topology designed to interoperate with other compatible equipment including application servers, media servers, internet-working gateways, voice bridges, and other end points

The following models are described:

- SoundPoint IP Desktop Phones
- SoundStation IP Conference Phones
- Polycom VVX 1500 Business Media Phone

For a list of key features available on the SoundPoint IP / SoundStation IP / VVX phones running the latest software, refer to Key Features of Your SoundPoint IP / SoundStation IP / VVX Phones on page 1-6.

SoundPoint IP Desktop Phones

This section describes the current SoundPoint IP desktop phones. For individual guides, refer to the product literature available at http://www.polycom.com/voicedocumentation/. Additional options are also available. For more information, contact your Polycom distributor.

Note Documentation for the SoundPoint IP 300, 301, 500, 501, 600, and 601 desktop phones and the SoundStation IP 4000 conference phone is available at http://www.polycom.com/voicedocumentation/ .

The currently supported desktop phones are:

- SoundPoint IP 320/321/330/331/335

- SoundPoint IP 430

- SoundPoint IP 450

- SoundPoint IP 550/560

- SoundPoint IP 650

- SoundPoint IP 670

SoundStation IP Conference Phones

This section describes the current SoundPoint IP conference phones. For individual guides, refer to the product literature available at http://www.polycom.com/voicedocumentation/. Additional options are also available. For more information, contact your Polycom distributor.

The currently supported conference phones are:

- SoundStation IP 6000

- SoundStation IP 7000

Polycom VVX 1500 Business Media Phone

This section describes the current Polycom VVX 1500 business media phone. For the individual guide, refer to the product literature available at http://www.polycom.com/voicedocumentation/. Additional options are also available. For more information, contact your Polycom distributor.

Key Features of Your SoundPoint IP / SoundStation IP / VVX Phones

The key features of the SoundPoint IP / SoundStation IP / VVX phones are:

- Award winning sound quality and full-duplex speakerphone or conference phone
  - Permits natural, high-quality, two-way conversations
  - Uses Polycom's industry leading Acoustic Clarity Technology
- Easy-to-use
  - An easy transition from traditional PBX systems into the world of IP
  - Up to 18 dedicated hard keys for access to commonly used features
  - Up to four context-sensitive soft keys for further menu-driven activities

- Platform independent
  - Supports multiple protocols and platforms enabling standardization on one phone for multiple locations, systems and vendors
  - Polycom's support of the leading protocols and industry partners makes it a future-proof choice
- Field upgradeable
  - Upgrade SoundPoint IP / SoundStation IP / VVX as standards develop and protocols evolve
  - Extends the life of the phone to protect your investment
  - Application flexibility for call management and new telephony applications
- Large LCD
  - Easy-to-use, easily readable and intuitive interface
  - Support of rich application content, including multiple call appearances, presence and instant messaging, and XML services
  - 102 x 23 pixel graphical LCD for the SoundPoint IP 320/321/330/331/335
  - 256 x 116 pixel graphical grayscale LCD for the SoundPoint IP 450
  - 320 x 160 pixel graphical grayscale LCD for the SoundPoint IP 550/560/650 (supports Asian characters)
  - 320 x 160 pixel graphical color LCD for the SoundPoint IP 670 (supports Asian characters)
  - 248 x 68 pixel graphical LCD for the SoundStation IP 6000
  - 256 x 128 pixel graphical grayscale LCD for the SoundStation IP 7000
  - 800 x 480 pixel graphical color LCD for the Polycom VVX 1500 (touch screen)
- Dual auto-sensing 10/100/1000baseT Ethernet ports
  - Leverages existing infrastructure investment
  - No re-wiring with existing CAT 5 cabling
  - Simplifies installation
  - 1000baseT is supported by the SoundPoint IP 560 and 670 and Polycom VVX 1500 only
- Power over Ethernet (PoE) port or Power pack option
  - Built-in IEEE 802.3af PoE port on the SoundPoint IP 320/321/330/331/335, 450, 550, 560, 650, and 670, the SoundStation IP 6000 and 7000, and Polycom VVX 1500 (auto-sensing)

  - Unused pairs on Ethernet port are used to deliver power to the phone via a wall adapter allowing fewer wires to desktop (for the SoundStation IP 6000 and 7000 conference phones)

- Multiple language support on most phones
  - Set on-screen language to your preference. Select from Chinese (Simplified), Danish, Dutch, English (Canada, United Kingdom, and United States), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese (Brazilian), Russian, Slovenian, Spanish (International), and Swedish.
  - Chinese (Simplified), Japanese, and Korean are not supported on the SoundPoint IP 320/321/330/331/335 phones.
- Microbrowser
  - Supports a subset of XHTML constructs; otherwise runs like any other Web browser.
- Polycom Browser on the Polycom VVX 1500
  - Supports XHTML 1.1 constructs, HTML 4.01, JavaScript, CCS 2.1, and SVG 1.1 (partial support).
- XML status/control API
  - Ability to poll phones for call status and device information.
  - Ability to receive telephony notification events.

2

Overview

This chapter provides an overview of the Session Initiation Protocol (SIP) application and how the phones fit into the network configuration.

SIP is the Internet Engineering Task Force (IETF) standard for multimedia communications over IP. It is an ASCII-based, application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints. Like other voice over IP (VoIP) protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

For the SoundPoint IP / SoundStation IP / VVX phones to successfully operate as a SIP endpoint in your network, it must meet the following requirements:

- A working IP network is established.
- Routers are configured for VoIP.
- VoIP gateways are configured for SIP.
- The latest (or compatible) SoundPoint IP / SoundStation IP / VVX phone SIP application image is available.
- A call server is active and configured to receive and send SIP messages.

  For more information on IP PBX and softswitch vendors, go to http://www.polycom.com/techpartners1/ .

This chapter contains information on:

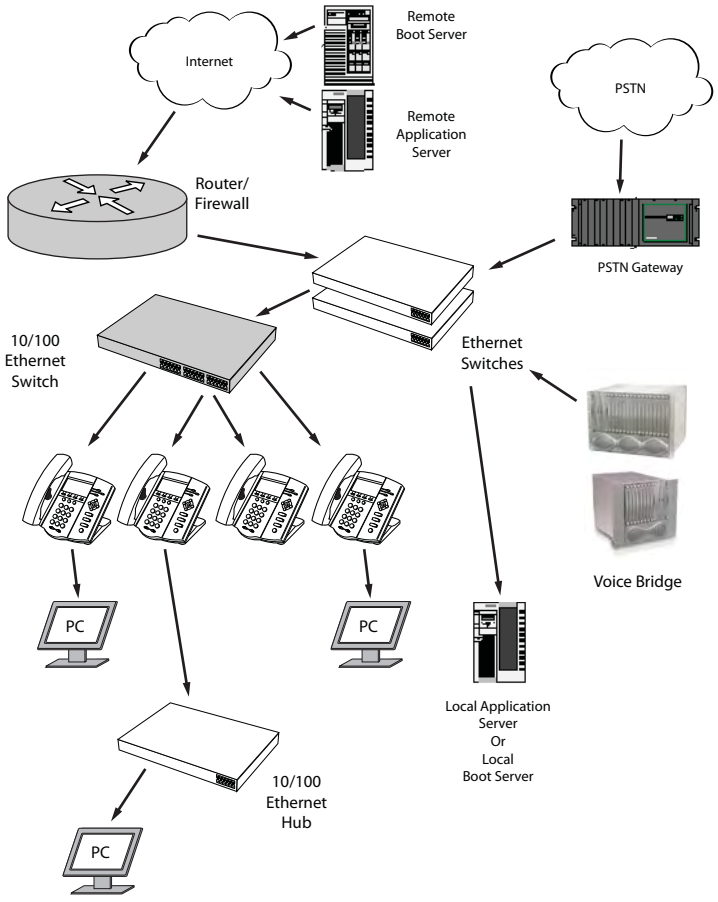
- Where SoundPoint IP / SoundStation IP / VVX Phones Fit
- Session Initiation Protocol Application Architecture
- Available Features
- New Features in SIP 3.2

To install your SoundPoint IP / SoundStation IP / VVX phones on the network, refer to Setting up Your System on page 3-1. To configure your SoundPoint IP / SoundStation IP / VVX phones with the desired features,

refer to Configuring Your System on page 4-1. To troubleshoot any problems with your SoundPoint IP / SoundStation IP / VVX phones on the network, refer to Troubleshooting Your SoundPoint IP / SoundStation IP / VVX Phones on page 5-1.

Where SoundPoint IP / SoundStation IP / VVX Phones Fit

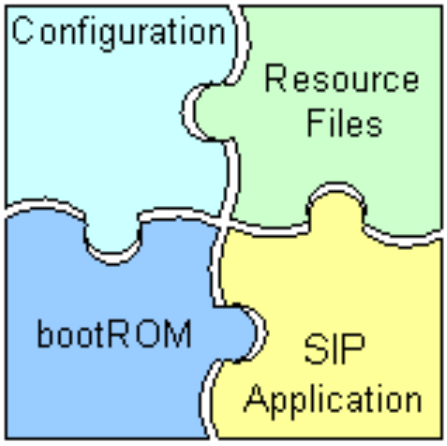
The phones connect physically to a standard office twisted-pair (IEEE 802.3) 10/100 megabytes per second Ethernet LAN and send and receive all data using the same packet-based technology. Since the phone is a data terminal, digitized audio being just another type of data from its perspective, the phone is capable of vastly more than traditional business phones. As SoundPoint IP / SoundStation IP / VVX phones run the same protocols as your office personal computer, many innovative applications can be developed without resorting to specialized technology.



Session Initiation Protocol Application Architecture

The software architecture of the SIP application is made of 4 basic components:

- BootROM—loads first when the phone is powered on
- SIP Application—software that makes the device a phone
- Configuration—configuration parameters stored in separate files
- Resource Files—optional, needed by some of the advanced features



BootROM

The BootROM is a small application that resides in the flash memory on the phone. All phones come from the factory with a BootROM pre-loaded.

The BootROM performs the following tasks in order:

1. Performs a power on self test (POST).
2. (Optional) Allows you to enter the setup menu where various network and provisioning options can be set.

   The BootROM software controls the user interface when the setup menu is accessed.
3. Requests IP settings and accesses the provisioning server (or boot server) to look for any updates to the BootROM application.

   If updates are found, they are downloaded and saved to flash memory, eventually overwriting itself after verifying the integrity of the download.
4. If a new BootROM is downloaded, formats the file system clearing out any application software and configuration files that may have been present.

5. Downloads the master configuration file.

   This file is either called **<MAC-address>.cfg** or **000000000000.cfg** . This file is used by the BootROM and the application for a list of other files that are needed for the operation of the phone.

6. Examines the master configuration file for the name of the application file, and then looks for this file on the provisioning server.

   If the copy on the provisioning server is different than the one stored in flash memory or there is no file stored in flash memory, the application file is downloaded.

7. Extracts the application from flash memory.

8. Installs the application into RAM, then uploads a log file with events from the boot cycle.

The BootROM will then terminate, and the application takes over.

SIP Application

The SIP application manages the VoIP stack, the digital signal processor (DSP), the user interface, and the network interaction. The SIP application manages everything to do with the phone's operation.

The application is a single file binary image and contains a digital signature to prevent tampering or loading rogue software images.

There is a new image file in each release of software.

The application performs the following tasks in order:

1. Downloads system, per-phone configuration, and resource files.

   These files are called **sip.cfg** and **phone1.cfg** by default. You can customize the filenames.

2. Controls all aspects of the phone.

3. Uploads log files.

BootROM and SIP Application Wrapper

Both the BootROM and the application run on multiple platforms (meaning all previously released versions of hardware that are still supported).

Current build archives have both split and combined images, so it is up to the administrator which model(s) to support. Using split files saves a lot of internal network traffic during reboots and updates.

Configuration

The SoundPoint IP / SoundStation IP / VVX phones can be configured automatically through files stored on a central provisioning server, manually through the phone's local UI or web interface, or by using a combination of the automatic and manual methods.

The recommended method for configuring phones is automatically through a central provisioning server, but if one is not available, the manual method will allow changes to most of the key settings.

Warning

Configuration files should only be modified by a knowledgeable system administrator. Applying incorrect parameters may render the phone unusable. The configuration files which accompany a specific release of the SIP software must be used together with that software. Failure to do this may render the phone unusable.

Note

You can make changes to the configuration files through the web interface to the phone. Using your chosen browser, enter the phone's IP address as the browser address. For more information, refer to Modifying Phone's Configuration Using the Web Interface on page C-27.

Changes made through the web interface are written to the override file (highest priority). These changes remain active and will take precedence over the configuration files stored on the provisioning server until **Reset Local Config** is performed.

The phone configuration files consist of:

- Master Configuration Files
- Application Configuration Files
- Override Files

This section also contains information on:

- Central Provisioning
- Manual Configuration

Master Configuration Files

The master configuration files can be one of:

- Specified master configuration file
- Per-phone master configuration file
- Default master configuration file

For more information, refer to Master Configuration Files on page A-2.

Application Configuration Files

Typically, the files are arranged in the following manner although parameters may be moved around within the files and the filenames themselves can be changed as needed. These files dictate the behavior of the phone once it is running the executable specified in the master configuration file.

The application files are:

- Application—It contains parameters that affect the basic operation of the phone such as voice codecs, gains, and tones and the IP address of an application server. All phones in an installation usually share this category of files. Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first. By default, **sip.cfg** is included.
- Per-phone—It contains parameters unique to a particular phone user. Typical parameters include:
  - display name
  - unique addresses

  Each phone in an installation usually has its own customized version of user files derived from Polycom templates. By default, **phone1.cfg** is included.

Override Files

This file contains all changes that are made by a user through the their phone (for example, time/date formats, ring types, and backlight intensity). The file allows the phone to keep user preferences through reboots and upgrades.

There is an option to clear the override file available to the system administrator—press the **Menu** key, and then select **Settings > Advanced > Admin Settings > Reset to Default > Reset Local Config**. You will be prompted to enter the administrative password.

Central Provisioning

The phones can be centrally provisioned from a provisioning server through a system of global and per-phone configuration files. The provisioning server also facilitates automated application upgrades, logging, and a measure of fault tolerance. Multiple redundant provisioning servers can be configured to improve reliability.

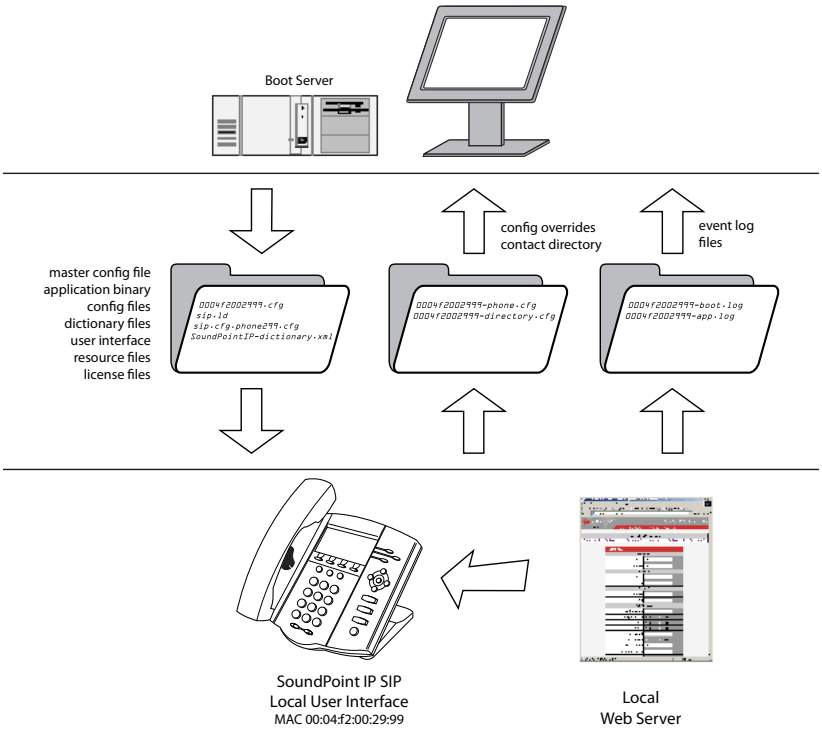
In the central provisioning method, there are two major classifications of configuration files:

- System configuration files
- Per-phone configuration files

Parameters can be stored in the files in any order and can be placed in any number of files. The default is to have 2 files, one for per-phone setting and one for system settings. The per-phone file is typically loaded first, and could contain system level parameters, letting you override that parameter for a given user. For example, it might be desirable to set the default CODEC for a remote user differently than for all the users who reside in the head office. By adding the CODEC settings to a particular user's per-phone file, the values in the system file are ignored.

Note Verify the order of the configuration files. Parameters in the configuration file loaded first will overwrite those in later configuration files.

The following figure shows one possible layout of the central provisioning method.



Manual Configuration

When the manual configuration method is employed, any changes made are stored in a configuration override file. This file is stored on the phone, but a copy will also be uploaded to the central provisioning server if one is being used. When the phone boots, this file is loaded by the application after any centrally provisioned files have been read, and its settings will override those in the centrally provisioned files.

This can create a lot of confusion about where parameters are being set, and so it is best to avoid using the manual method unless you have good reason to do so.

Resource Files

In addition to the application and the configuration files, the phones may require resource files that are used by some of the advanced features. These files are optional, but if the particular feature is being employed, these files are required.

Some examples of resource files include:

- Language dictionaries
- Custom fonts
- Ring tones
- Synthesized tones
- Contact directories

Note If you need to remove the resource files from a phone at some later date—for example, you are giving the phone to a new user—instructions on how to put the phone into the factory default state can be found in "Quick Tip 18298: Resetting and Rebooting SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Available Features

This section provides information about the features available on the SoundPoint IP / SoundStation IP / VVX phones:

- Basic User Features
  - Automatic Off-Hook Call Placement—Supports an optional automatic off-hook call placement feature for each registration.
  - Call Forward—Provides a flexible call forwarding feature to forward calls to another destination.
  - Call Hold—Pauses activity on one call so that the user may use the phone for another task, such as making or receiving another call.
  - Call Log—Contains call information such as remote party identification, time and date, and call duration in three separate lists, missed calls, received calls, and placed calls on most platforms.
  - Call Park/Retrieve—An active call can be parked. A parked call can be retrieved by any phone.

- Call Timer—A separate call timer, in hours, minutes, and seconds, is maintained for each distinct call in progress.
- Call Transfer—Call transfer allows the user to transfer a call in progress to some other destination.
- Call Waiting—When an incoming call arrives while the user is active on another call, the incoming call is presented to the user visually on the display and a configurable sound effect will be mixed with the active call audio.
- Called Party Identification—The phone displays and logs the identity of the party specified for outgoing calls.
- Calling Party Identification—The phone displays the caller identity, derived from the network signaling, when an incoming call is presented, if information is provided by the call server.
- Connected Party Identification—The identity of the party to which the user has connected is displayed and logged, if the name is provided by the call server.
- Context Sensitive Volume Control—The volume of user interface sound effects, such as the ringer, and the receive volume of call audio is adjustable.
- Customizable Audio Sound Effects—Audio sound effects used for incoming call alerting and other indications are customizable.
- Directed Call Pick-Up and Group Call Pick-Up—Calls to another phone can be picked up by dialing the extension of the other phone. Calls to another phone within a pre-defined group can be picked up without dialing the extension of the other phone.
- Distinctive Call Waiting—Calls can be mapped to distinct call waiting types.
- Distinctive Incoming Call Treatment—The phone can automatically apply distinctive treatment to calls containing specific attributes.
- Distinctive Ringing—The user can select the ring type for each line and the ring type for specific callers can be assigned in the contact directory.
- Do Not Disturb—A do-not-disturb feature is available to temporarily stop all incoming call alerting.
- Graphic Display Backgrounds—A picture or design displayed on the background of the graphic display.
- Handset, Headset, and Speakerphone—SoundPoint IP phones come standard with a handset and a dedicated headset connection (headset not supplied). All SoundPoint IP, SoundStation IP, and Polycom VVX phones have full-duplex speakerphones.
- Idle Display Animation—All phones can display a customized animation on the idle display in addition to the time and date.

  - Last Call Return—The phone allows call server-based last call return.
  - Local / Centralized Conferencing—The phone can conference together the local user with the remote parties of two independent calls and can support centralized conferences for which external resources are used such as a conference bridge. The advanced aspects of conferencing are part of the Productivity Suite.
  - Local Contact Directory—The phone maintains a local contact directory that can be downloaded from the provisioning server and edited locally. Any edits to the Contact Directory made on the phone are saved to the provisioning server as a backup.
  - Local Digit Map—The phone has a local digit map to automate the setup phase of number-only calls.
  - Message Waiting Indication—The phone will flash a message-waiting indicator (MWI) LED when instant messages and voice messages are waiting.
  - Microphone Mute—When the microphone mute feature is activated, visual feedback is provided.
  - Missed Call Notification—The phone can display the number of calls missed since the user last looked at the Missed Calls list.
  - Soft Key Activated User Interface—The user interface makes extensive use of intuitive, context-sensitive soft key menus.
  - Speed Dial—The speed dial system allows calls to be placed quickly from dedicated keys as well as from a speed dial menu.
  - Time and Date Display—Time and date can be displayed in certain operating modes such as when the phone is idle and during a call.

- Advanced Features
  - Access URL in SIP Message—Ability for the SoundPoint IP phones to be able to receive a URL inside a SIP message (for example, as a SIP header extension in a SIP INVITE) and subsequently access that given URL in the Microbrowser.
  - SIP-B Automatic Call Distribution—Supports ACD agent available and unavailable and allows ACD login and logout. Requires call server support.
  - Bridged Line Appearance—Calls and lines on multiple phones can be logically related to each other. Requires call server support.
  - Busy Lamp Field—Allows monitoring the hook status and remote party information of users through the busy lamp field (BLF) LEDs and displays on an attendant console phone. This feature may require call server support.
  - Configurable Feature Keys—Certain key functions can be changed from the factory defaults.

- Configurable Soft Keys—Allows users to create their own soft keys and have them displayed with or without the standard SoundPoint IP and SoundStation IP soft keys.
- Corporate Directory—The phone can be configured to access your corporate directory if it has a standard LDAP interface. This feature is part of the Productivity Suite.
- Customizable Fonts and Indicators—The phone's user interface can be customized by changing the fonts and graphic icons used on the display and the LED indicator patterns.
- Display of Warnings from SIP Headers—Displays a "pop-up" to user that is found in the Warning Field from a SIP header.
- Downloadable Fonts—New fonts can be loaded onto the phone.
- Enhanced Busy Lamp Field—Allows an attendant to see a remote line that is Ringing and answer a remote ringing call using a single key-press. Also allows the attendant to view the caller-id of remote active and ringing calls. This feature may require call server support.
- Enhanced Feature Keys—Allows users to redefine soft keys to suit their needs. In SIP 3.0, this feature required a license key.
- Instant Messaging—Supports sending and receiving instant text messages.
- Browser and Microbrowser—The SoundPoint IP 430, 450, 550, 560, 600, 601, 650, and 670 desktop phones, the SoundStation IP 6000, and 7000 conference phones, and the Polycom VVX 1500 phones (pre-SIP 3.2.2) support an XHTML microbrowser.
- Microsoft Live Communications Server 2005 Integration—SoundPoint IP and SoundStation IP phones can be used with Microsoft Live Communications Server 2005 and Microsoft Office Communicator to help improve business efficiency and increase productivity and to share ideas and information immediately with business contacts. Requires call server support.
- Multilingual User Interface—All phones have multilingual user interfaces.
- Multiple Call Appearances—The phone supports multiple concurrent calls. The hold feature can be used to pause activity on one call and switch to another call.
- Multiple Line Keys per Registration—More than one line key can be allocated to a single registration.
- Multiple Registrations—SoundPoint IP desktop phones and Polycom VVX 1500 phones support multiple registrations per phone. However, SoundStation IP conference phones support a single registration.
- Network Address Translation—The phones can work with certain types of network address translation (NAT).

  - Presence—Allows the phone to monitor the status of other users/devices and allows other users to monitor it. Requires call server support.
  - Real-Time Transport Protocol Ports—The phone treats all real- time transport protocol (RTP) streams as bi-directional from a control perspective and expects that both RTP end points will negotiate the respective destination IP addresses and ports.
  - Recording and Playback of Audio Calls — Recording and playback allows the user to record any active conversation using the phone on a USB device. The files are date and time stamped for easy archiving and can be played back on the phone or on any computer with a media playback program that supports the **.wav** format. This feature is part of the Productivity Suite.
  - Server Redundancy—Server redundancy is often required in VoIP deployments to ensure continuity of phone service for events where the call server needs to be taken offline for maintenance, the server fails, or the connection from the phone to the server fails.
  - Shared Call Appearances—Calls and lines on multiple phones can be logically related to each other. Requires call server support.
  - Static DNS Cache—Set up a static DNS cache and provide for negative caching.
  - Synthesized Call Progress Tones—In order to emulate the familiar and efficient audible call progress feedback generated by the PSTN and traditional PBX equipment, call progress tones are synthesized during the life cycle of a call. Customizable for certain regions, for example, Europe has different tones from North America.
  - Voice Mail Integration—Compatible with voice mail servers.

- Audio Features
  - Acoustic Echo Cancellation—Employs advanced acoustic echo cancellation for hands-free operation.
  - Audio Codecs—Supports a wide range of industry standard audio codecs.
  - Automatic Gain Control—Designed for hands-free operation, boosts the transmit gain of the local user in certain circumstances.
  - Background Noise Suppression—Designed primarily for hands-free operation, reduces background noise to enhance communication in noisy environments.
  - Comfort Noise Fill—Designed to help provide a consistent noise level to the remote user of a hands-free call.
  - DTMF Event RTP Payload—Conforms to RFC 2833, which describes a standard RTP-compatible technique for conveying DTMF dialing and other telephony events over an RTP media stream.

- DTMF Tone Generation—Generates dual tone multi-frequency (DTMF) tones in response to user dialing on the dial pad.
- Dynamic Noise Reduction— Provides maximum microphone sensitivity, while automatically reducing background noise on SoundStation IP 7000 conference phones.
- IEEE 802.1p/Q—The phone will tag all Ethernet packets it transmits with an 802.1Q VLAN header.
- IP Type-of-Service—Allows for the setting of TOS settings.
- Jitter Buffer and Packet Error Concealment—Employs a high-performance jitter buffer and packet error concealment system designed to mitigate packet inter-arrival jitter and out-of-order or lost (lost or excessively delayed by the network) packets.
- Low-Delay Audio Packet Transmission—Designed to minimize latency for audio packet transmission.
- Treble/Bass Controls—Equalizes the tone of the high and low frequency sound from the speakers on SoundStation IP 7000 conference phones.
- Voice Activity Detection—Conserves network bandwidth by detecting periods of relative "silence" in the transmit data path and replacing that silence efficiently with special packets that indicate silence is occurring.
- Voice Quality Monitoring—Generates various quality metrics including MOS and R-factor for listening and conversational quality. This feature is part of the Productivity Suite.

• Security Features

- Local User and Administrator Privilege Levels—Several local settings menus are protected with two privilege levels, user and administrator, each with its own password.
- Configuration File Encryption—Confidential information stored in configuration files must be protected (encrypted). The phone can recognize encrypted files, which it downloads from the provisioning server and it can encrypt files before uploading them to the provisioning server.
- Custom Certificates—When trying to establish a connection to a provisioning server for application provisioning, the phone trusts certificates issued by widely recognized certificate authorities (CAs).
- Incoming Signaling Validation—Levels of security are provided for validating incoming network signaling.
- Secure Real-Time Transport Protocol—Encrypting audio streams to avoid interception and eavesdropping.

For more information on each feature and its associated configuration parameters, see the appropriate section in Configuring Your System on page 4-1.

New Features in SIP 3.2

Note

The SoundPoint IP 300 and 500 phones will be supported on the latest maintenance patch release of the SIP 2.1 software stream—currently SIP 2.1.4 . Any new features introduced after SIP 2.1.4 are not supported. Refer to the *SIP 2.1 Administrator Guide*, which is available at http://www.polycom.com/global/documents/support/setup_maintenance/products/voice/sip_2.1_addendum_to_sip_2.0_administrator%27s_guide.pdf .

The SoundPoint IP 301, 501, 600, and 601 and the SoundStation IP 4000 phones will be supported on the latest maintenance patch release of the SIP 3.1 software stream—currently SIP 3.1.3 . Any new features introduced after 3.1.3 are not supported. Configuration parameters related to these phones will be removed from the **sip.cfg** and **phone1.cfg** files in the next major release. To administer these phones, refer to the *SIP 3.1 Administrator's Guide*, which is available at http://www.polycom.com/voicedocumentation/ .

The following new features were introduced in SIP 3.1.2:

- Feature Synchronized Automatic Call Distribution—Supports ACD agent available and unavailable and allows ACD sign in and sign out. Requires call server support.
- Quick Setup of SoundPoint IP / SoundStation IP / VVX Phones—Simplifies the process of entering provisioning server parameters.

The following new feature enhancement was introduced in SIP 3.1.3:

- Corporate Directory—The phone's user interface to access your corporate directory has changed. Also Microsoft ADAM and SunLDAP are also supported in addition to Active Directory and OpenLDAP.

The following new features were introduced in SIP 3.2:

- LLDP and Supported TLVs—Support for Link Layer Discovery Protocol (LLDP) and media extensions (LLDP-MED) such as VLAN configuration. For provisioning information, refer to Ethernet Menu on page 3-12.
- iLBC added to Audio Codecs—Support for Internet Low Bitrate Codec (iLBC) added for the SoundPoint IP 32x/33x, 450, 550, 560, 650, and 670, d SoundStation IP 6000 and 700, and Polycom VVX 1500.
- Video Codecs—Support the standard video codecs on the Polycom VVX 1500 phones.

- Mutual TLS Authentication—Support for phone authentication of the server and server authentication of the phone.
- Digital Certificates— Support for digital certificates and associated private keys on certain models of SoundPoint IP, SoundStation IP, and Polycom VVX phones.
- Capturing Phone's Current Screen—Allows the phone's current display to be displayed in a web browser.

The following existing features were changed in SIP 3.2:

- Busy Lamp Field— The BLF feature has been enhanced as follows:
  - To provide individual subscription-based BLF monitoring (without requiring a centralized resource list to be maintained by the call server.
  - To allow the single button 'remote pick-up' feature to be implemented using Directed Call Pick-Up using SIP signaling as well as the star code method supported in SIP 3.1 .
- Secure Real-Time Transport Protocol—Information has been transferred from the "Technical Bulletin 25751: Secure Real-Time Transport Protocol on SoundPoint IP Phones" to this guide.

The following new features were introduced in SIP 3.2.2:

- H.323 Protocol—Support for the H.323 protocol for the Polycom VVX 1500 phone.
- Browser and Microbrowser—Support for a Webkit-style browser for the Polycom VVX 1500 phone only.

Documentation of the newly released SoundPoint IP 321/331/335 and 450 desktop phones and Polycom VVX 1500 business media phone has also been added.

Note When SoundPoint IP 32x/33x is used in this guide, it includes the SoundPoint IP 320, 321, 330, 331, and 335 phones.

3

Setting up Your System

Your SoundPoint IP / SoundStation IP / VVX SIP phone is designed to be used like a regular phone on a public switched telephone network (PSTN).

This chapter provides basic instructions for setting up your SoundPoint IP / SoundStation IP / VVX phones. This chapter contains information on:

- Setting Up the Network
- Setting Up the Provisioning Server
- Deploying Phones From the Provisioning Server
- Upgrading SIP Application

Because of the large number of optional installations and configurations that are available, this chapter focuses on one particular way that the SIP application and the required external systems might initially be installed and configured in your network.

For more information on configuring your system, refer to Configuring Your System on page 4-1. For more information on the configuration files required for setting up your system, refer to Configuration Files on page A-1.

For installation and maintenance of SoundPoint IP / SoundStation IP / VVX phones, the use of a provisioning server is strongly recommended. This allows for flexibility in installing, upgrading, maintaining, and configuring the phone. Configuration, log, and directory files are normally located on this server. Allowing the phone write access to the server is encouraged.

The phone is designed such that, if it cannot locate a provisioning server when it boots up, it will operate with internally saved parameters. This is useful for occasions when the provisioning server is not available, but is not intended to be used for long-term operation of the phones.

However, if you want to register a single SoundPoint IP / SoundStation IP / VVX phone, refer to “Quick Tip 44011: Register Standalone SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones“ at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Setting Up the Network

Regardless of whether or not you will be installing a centrally provisioned system, you must perform basic TCP/IP network setup, such as IP address and subnet mask configuration, to get your organization's phones up and running.

The SIP application uses the network to query the provisioning server for upgrades, which is an optional process that will happen automatically when properly deployed. For more information on the basic network settings, refer to DHCP or Manual TCP/IP Setup on page 3-2.

The BootROM on the phone performs the provisioning functions of downloading the BootROM, the **<MACaddress>.cfg** file, and the SIP application, and uploading log files. For more information, refer to Supported Provisioning Protocols on page 3-4.

Basic network settings can be changed during BootROM download using the BootROM's setup menu. A similar menu system is present in the application for changing the same network parameters. For more information, refer to Modifying the Network Configuration on page 3-6.

DHCP or Manual TCP/IP Setup

Basic network settings can be derived from DHCP, or entered manually using the phone's LCD-based user interface, or downloaded from configuration files.

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Polycom recommends using DHCP where possible to eliminate repetitive manual data entry.

The following table shows the manually entered networking parameters that may be overridden by parameters obtained from a DHCP server, an alternate DHCP server, or configuration file:

Parameter	DHCP Option	DHCP	Alternate DHCP	Configuration File (application only)	Local FLASH
		⇨ priority when more than one source exists ⇨			
		1	2	3	4
IP address	1	•	-	-	•
subnet mask	1	•	-	-	•
IP gateway	3	•	-	-	•

Parameter	DHCP Option	DHCP	Alternate DHCP	Configuration File (application only)	Local FLASH
boot server address	Refer to DHCP Menu on page 3-8	•	•	-	•
SIP server address	151 ***Note:*** *This value is configurable*.	•	-	-	•
SNTP server address	42 then 4	•	-	•	•
SNTP GMT offset	2	•	-	•	•
DNS server IP address	6	•	-	-	•
alternate DNS server IP address	6	•	-	-	•
DNS domain	15	•	-	-	•
VLAN ID	Refer to DHCP Menu on page 3-8	***Warning:*** *Link Layer Discovery Protocol (LLDP) overrides Cisco Discovery Protocol (CDP). CDP overrides Local FLASH which overrides DHCP VLAN Discovery.*			

For more information on DHCP options, go to http://www.ietf.org/rfc/rfc2131.txt?number=2131 or http://www.ietf.org/rfc/rfc2132.txt?number=2132.

Note

The configuration file value for ***SNTP server address*** and ***SNTP GMT offset*** can be configured to override the DHCP value. Refer to `tcpIpApp.sntp.address.overrideDHCP` in Time Synchronization <sntp/> on page A-75.

The CDP Compatibility value can be obtained from a connected Ethernet switch if the switch supports CDP.

In the case where you do not have control of your DHCP server or do not have the ability to set the DHCP options, an alternate method of automatically discovering the provisioning server address is required. Connecting to a secondary DHCP server that responds to DHCP INFORM queries with a requested provisioning server value is one possibility. For more information, refer to http://www.ietf.org/rfc/rfc3361.txt?number=3361 and http://www.ietf.org/rfc/rfc3925.txt?number=3925.

Supported Provisioning Protocols

The BootROM performs the provisioning functions of downloading configuration files, uploading and downloading the configuration override file and user directory, and downloading the dictionary and uploading log files.

The protocol that will be used to transfer files from the provisioning server depends on several factors including the phone model and whether the BootROM or SIP application stage of provisioning is in progress. By default, the phones are shipped with FTP enabled as the provisioning protocol. If an unsupported protocol is specified, this may result in a defined behavior (see the table below for details of which protocol the phone will use). The Specified Protocol listed in the table can be selected in the ***Server Type*** field or the ***Server Address*** can include a transfer protocol, for example http://usr:pwd@server (refer to Server Menu on page 3-10). The boot server address can be an IP address, domain string name, or URL. The boot server address can also be obtained through DHCP. Configuration file names in the **<MACaddress>.cfg** file can include a transfer protocol, for example https://usr:pwd@server/dir/file.cfg. If a user name and password are specified as part of the server address or file name, they will be used only if the server supports them.

Note A URL should contain forward slashes instead of back slashes and should not contain spaces. Escape characters are not supported. If a user name and password are not specified, the Server User and Server Password will be used (refer to Server Menu on page 3-10).

Specified Protocol	Protocol used by BootROM IP 32x, 33x, 430, 450, 550, 560, 650, 670, 6000, 7000 VVX 1500	Protocol used by SIP Software IP 32x, 33x, 430, 450, 550, 560, 650, 670, 6000, 7000 VVX 1500
FTP	FTP	FTP
TFTP	TFTP	TFTP
HTTP	HTTP	HTTP
HTTPS	HTTP	HTTPS

Note There are two types of FTP methods—active and passive. The SIP application is not compatible with active FTP. Secure provisioning was implemented in a previous release.

Note

Setting Option 66 to tftp://192.168.9.10 has the effect of forcing a TFTP download. Using a TFTP URL (for example, tftp://provserver.polycom.com) has the same effect.

Note

Both digest and basic authentication are supported when using HTTP/S for the SIP application. Only digest authentication is supported when using HTTP by the BootROM. If the **Server Type** is configured as HTTPS, the BootROM will contact the same address and apply the same username and password to authentication challenges only the protocol used will be HTTP. No SSL negotiation will take place, so servers that do not allow unsecured HTTP connections will not be able to provision files.

For downloading the BootROM and application images to the phone, the secure HTTPS protocol is not available. To guarantee software integrity, the BootROM will only download cryptographically signed BootROM or application images. For HTTPS, widely recognized certificate authorities are trusted by the phone (refer to Trusted Certificate Authority List on page C-1) and custom certificates can be added to the phone (refer to "Technical Bulletin 17877: Using Custom Certificates With SoundPoint IP, SoundStation IP, and É Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

As of SIP 3.2, Mutual Transport Layer Security (TLS) authentication is available. For more information, refer to Mutual TLS Authentication on page 4-97.

Note

If you want to use digest authentication against the Microsoft Internet Information Services server:

- Use Microsoft Internet Information Server 6.0 or later.
- Digest authentication needs the user name and password to be saved in reversible encryption.
- The user account on the server must have administrative privileges.
- The wildcard must be set as MIME type; otherwise the phone will not download *.cfg, *.ld and other required files. This is due to the fact that the Microsoft Internet Information Server cannot recognize these extensions and will return a "File not found" error. To configure wildcard for MIME type, refer to http://support.microsoft.com/kb/326965 .

For more information, refer to http://www.microsoft.com/technet/prodtechnol/WindowsServer2003/Library/IIS/809552a3-3473-48a7-9683-c6df0cdfda21.mspx?mfr=true .

Modifying the Network Configuration

You can access the network configuration menu:

- **During BootROM Phase**. The network configuration menu is accessible during the auto-boot countdown of the BootROM phase of operation. Press the **Setup** soft key to launch the main menu.
- **During Application Phase**. The network configuration menu is accessible from the phone's main menu. Select **Menu>Settings>Advanced>Admin Settings>Network Configuration**. Advanced Settings are locked by default. Enter the administrator password to unlock. The factory default password is 456. Polycom recommends that you change the administrative password from the default value.

Phone network configuration parameters may be modified by means of:

- Main Menu
- DHCP Menu
- Server Menu
- Ethernet Menu
- Syslog Menu

Use the soft keys, the arrow keys, the **Select** and **Delete** keys to make changes.

Certain parameters are read-only due to the value of other parameters. For example, if the DHCP Client parameter is enabled, the ***Phone IP Addr*** and ***Subnet Mask*** parameters are dimmed or not visible since these are guaranteed to be supplied by the DHCP server (mandatory DHCP parameters) and the statically assigned IP address and subnet mask will never be used in this configuration.

Resetting to Factory Defaults

The basic network configuration referred to in the subsequent sections can be reset to factory defaults using a menu selection from the **Advanced Settings** menu or using a multiple key combination described in Multiple Key Combinations on page C-11.

Main Menu

The following configuration parameters can be modified on the main setup menu:

Name	Possible Values	Description
DHCP Client	Enabled, Disabled	If enabled, DHCP will be used to obtain the parameters discussed in DHCP or Manual TCP/IP Setup on page 3-2.
DHCP Menu		Refer to DHCP Menu on page 3-8. ***Note:*** *Disabled when DHCP client is disabled.*
Phone IP Address	dotted-decimal IP address	Phone's IP address. ***Note:*** *Disabled when DHCP client is enabled.*
Subnet Mask	dotted-decimal subnet mask	Phone's subnet mask. ***Note:*** *Disabled when DHCP client is enabled*.
IP Gateway	dotted-decimal IP address	Phone's default router.
Server Menu		Refer to Server Menu on page 3-10.
SNTP Address	dotted-decimal IP address OR domain name string	Simple Network Time Protocol (SNTP) server from which the phone will obtain the current time.
GMT Offset	-13 through +12	Offset of the local time zone from Greenwich Mean Time (GMT) in half hour increments.
DNS Server	dotted-decimal IP address	Primary server to which the phone directs Domain Name System (DNS) queries.
DNS Alternate Server	dotted-decimal IP address	Secondary server to which the phone directs Domain Name System queries.
DNS Domain	domain name string	Phone's DNS domain.
Ethernet		Refer to Ethernet Menu on page 3-12.
EM Power	Enabled, Disabled	This parameter is relevant if the phone gets Power over Ethernet (PoE). If enabled, the phone will set power requirements in CDP to 12W so that up to three Expansion Modules (EM) can be powered. If disabled, the phone will set power requirements in CDP to 5W which means no Expansion Modules can be powered (it will not work).
Syslog		Refer to Syslog Menu on page 3-13.

Note A parameter value of "???" indicates that the parameter has not yet been set and saved in the phone's configuration. Any such parameter should have its value set before continuing.

The ***EM Power*** parameter is only available on SoundPoint IP 650 and 670 phones.

Note To switch the text entry mode on the SoundPoint IP 32x/33x, press the #. You may want to use URL or IP address modes when entering server addresses.

DHCP Menu

The DHCP menu is accessible only when the DHCP client is enabled. The following DHCP configuration parameters can be modified on the DHCP menu:

Name	Possible Values	Description
Boot Server	0=Option 66	The phone will look for option number 66 (string type) in the response received from the DHCP server. The DHCP server should send address information in option 66 that matches one of the formats described for ***Server Address*** in the next section, Server Menu. If the DHCP server sends nothing, the following scenarios are possible: • If a boot server value is stored in flash memory and the value is not "0.0.0.0", then the value stored in flash is used. • Otherwise the phone sends out a DHCP INFORM query. - If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value. - If no alternate DHCP server responds, the INFORM query process will retry and eventually time out.

Name	Possible Values	Description
Boot Server (continued)	1=Custom	The phone will look for the option number specified by the ***Boot Server Option*** parameter (below), and the type specified by the ***Boot Server Option Type*** parameter (below) in the response received from the DHCP server. If the DHCP server sends nothing, the following scenarios are possible: • If a boot server value is stored in flash memory and the value is not “0.0.0.0”, then the value stored in flash is used. • Otherwise the phone sends out a DHCP INFORM query. - If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value. - If no alternate DHCP server responds, the INFORM query process will retry and eventually time out.
	2=Static	The phone will use the boot server configured through the Server Menu. For more information, refer to the next section, Server Menu.
	3=Custom+Option 66	The phone will first use the custom option if present or use Option 66 if the custom option is not present. If the DHCP server sends nothing, the following scenarios are possible: • If a boot server value is stored in flash memory and the value is not “0.0.0.0”, then the value stored in flash is used. • Otherwise the phone sends out a DHCP INFORM query. - If a single alternate DHCP server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value. The phone prefers the custom option value over the Option 66 value, but if no custom option is given, the phone will use the Option 66 value. - If no alternate DHCP server responds, the INFORM query process will retry and eventually time out.
Boot Server Option	128 through 254 (Cannot be the same as VLAN ID Option)	When the boot server parameter is set to Custom, this parameter specifies the DHCP option number in which the phone will look for its boot server.
Boot Server Option Type	0=IP Address, 1=String	When the ***Boot Server*** parameter is set to Custom, this parameter specifies the type of the DHCP option in which the phone will look for its boot server. The ***IP Address*** must specify the boot server. The String must match one of the formats described for ***Server Address*** in the next section, Server Menu.

Name	Possible Values	Description
VLAN Discovery	0=Disabled (default)	No VLAN discovery through DHCP.
	1=Fixed	Use predefined DHCP vendor-specific option values of 128, 144, 157 and 191. If this is used, the ***VLAN ID Option*** field will be ignored
	2=Custom	Use the number specified in the ***VLAN ID Option*** field as the DHCP private option value.
VLAN ID Option	128 through 254 (Cannot be the same as ***Boot Server Option***) (default is 129)	The DHCP private option value (when ***VLAN Discovery*** is set to Custom). For more information, refer to Assigning a VLAN ID Using DHCP on page C-23.

Note

If multiple alternate DHCP servers respond:

- The phone should gather the responses from alternate DHCP servers.
- If configured for `Custom+Option66`, the phone will select the first response that contains a valid "custom" option value.
- If none of the responses contain a "custom" option value, the phone will select the first response that contains a valid “option66” value.

Server Menu

The following server configuration parameters can be modified on the Server menu:

Name	Possible Values	Description
Server Type	0=FTP, 1=TFTP, 2=HTTP, 3=HTTPS, 4=FTPS, 5=Invalid	The protocol that the phone will use to obtain configuration and phone application files from the provisioning server. Refer to Supported Provisioning Protocols on page 3-4. ***Note****: Active FTP is not supported for BootROM version 3.0 or later. Passive FTP is still supported.* ***Note:*** *Only implicit FTPS is supported.*

Name	Possible Values	Description
Server Address	dotted-decimal IP address OR domain name string OR URL All addresses can be followed by an optional directory and optional file name.	The provisioning server to use if the DHCP client is disabled, the DHCP server does not send a boot server option, or the ***Boot Server*** parameter is set to Static. The phone can contact multiple IP addresses per DNS name. These redundant provisioning servers must all use the same protocol. If a URL is used it can include a user name and password. Refer to Supported Provisioning Protocols on page 3-4. A directory and the master configuration file can be specified. ***Note:*** *":", "@", or "/" can be used in the user name or password these characters if they are correctly escaped using the method specified in RFC 1738.*
Server User	any string	The user name used when the phone logs into the server (if required) for the selected ***Server Type***. ***Note:*** *If the Server Address is a URL with a user name, this will be ignored.*
Server Password	any string	The password used when the phone logs in to the server if required for the selected ***Server Type***. ***Note:*** *If the Server Address is a URL with user name and password, this will be ignored.*
File Transmit Tries	1 to 10 Default 3	The number of attempts to transfer a file. (An attempt is defined as trying to download the file from all IP addresses that map to a particular domain name.)
Retry Wait	0 to 300 Default 1	The minimum amount of time that must elapse before retrying a file transfer, in seconds. The time is measured from the start of a transfer attempt which is defined as the set of upload/download transactions made with the IP addresses that map to a given provisioning server's DNS host name. If the set of transactions in an attempt is equal to or greater than the Retry Wait value, then there will be no further delay before the next attempt is started. For more information, refer to Deploying Phones From the Provisioning Server on page 3-17.
Tag SN to UA	Disabled, Enabled	If enabled, the phone's serial number (MAC address) is included in the User-Agent header of the Microbrowser. The default value is Disabled.

Note The ***Server User*** and ***Server Password*** parameters should be changed from the default values. Note that for insecure protocols the user chosen should have very few privileges on the server.

Ethernet Menu

The following Ethernet configuration parameters can be modified on the Ethernet menu:

Name	Possible Values	Description
LLDP	Enabled, Disabled	If enabled, the phone will use the LLDP protocol to communicate with the network switch for certain network parameters. Most often this will be used to set the VLAN that the phone should use for voice traffic. It also reports power management to the switch. The default value is Enabled. If the switch does not support it, VLAN Discovery is used. Refer to DHCP Menu on page 3-8. There are four ways to get VLAN on the phone and they can all be turned on, but the VLAN used is chosen by priority of each method. The priority is: 1. LLDP; 2. CDP; 3. DVD (VLAN Via DHCP); 4. Static (VLAN ID entered in config menu). For more information, refer to LLDP and Supported TLVs on page C-30.
CDP Compatibility	Enabled, Disabled	If enabled, the phone will use CDP compatible signaling to communicate with the network switch for certain network parameters. Most often this will be used to set the VLAN that the phone should use for Voice Traffic, and for the phone to communicate its PoE power requirements to the switch. The default value is Enabled.
VLAN ID	Null, 0 through 4094	Phone's 802.1Q VLAN identifier. The default value is Null. ***Note:*** *Null = no VLAN tagging*
VLAN Filtering	Enabled, Disabled	Filter received Ethernet packets so that the TCP/IP stack does not process bad data or too much data. Enable/disable the VLAN filtering state. The default value is Disabled.

Name	Possible Values	Description
Storm Filtering	Enabled, Disabled	Filter received Ethernet packets so that the TCP/IP stack does not process bad data or too much data. Enable/disable the DoS storm prevention state. The default value is Enabled.
LAN Port Mode	0 = Auto 1 = 10HD 2 = 10FD 3 = 100HD 4 = 100FD 5 = 1000FD	The network speed over the Ethernet. The default value is Auto. HD means half duplex and FD means full duplex. ***Note**: Polycom recommends that you do not change this setting.*
PC Port Mode	0 = Auto 1 = 10HD 2 = 10FD 3 = 100HD 4 = 100FD 5 = 1000FD -1 = Disabled	The network speed over the Ethernet. The default value is Auto. HD means half duplex and FD means full duplex. ***Note**: Polycom recommends that you do not change this setting unless you want to disable the PC port.*

Note

The LAN Port Mode applies to all phones supported by SIP 3.0 . The PC Port Mode parameters are only available on phones with a second Ethernet port.

Only the SoundPoint IP 560 and 670 and Polycom VVX 1500 phones supports the LAN Port Mode and PC Port Mode setting of 1000FD.The 1000BT LAN Clock and 1000BT PC Clock parameters are only available on SoundPoint IP 560 and 670 phones

Syslog Menu

Syslog is a standard for forwarding log messages in an IP network. The term “syslog” is often used for both the actual syslog protocol, as well as the application or library sending syslog messages.

The syslog protocol is a very simplistic protocol: the syslog sender sends a small textual message (less than 1024 bytes) to the syslog receiver. The receiver is commonly called “syslogd”, “syslog daemon” or “syslog server”. Syslog messages can be sent through UDP, TCP, or TLS. The data is sent in cleartext.

Syslog is supported by a wide variety of devices and receivers. Because of this, syslog can be used to integrate log data from many different types of systems into a central repository.

The syslog protocol is defined in RFC 3164. For more information on syslog, go to http://www.ietf.org/rfc/rfc3164.txt?number=3164 .

The following syslog configuration parameters can be modified on the Syslog menu:

Name	Possible Values	Description
Server Address	dotted-decimal IP address OR domain name string	The syslog server IP address or host name. The default value is NULL.
Server Type	None=0, UDP=1, TCP=2, TLS=3	The protocol that the phone will use to write to the syslog server. If set to "None", transmission is turned off, but the server address is preserved.
Facility	0 to 23	A description of what generated the log message. For more information, refer to section 4.1.1 of RFC 3164. The default value is 16, which maps to "local 0".
Render Level	0 to 6	Specifies the lowest class of event that will be rendered to syslog. It is based on `log.render.level` and can be a lower value. Refer to Basic Logging <level/><change/> and <render/> on page A-106. ***Note:*** *Use left and right arrow keys to change values.*
Prepend MAC Address	Enabled, Disabled	If enabled, the phone's MAC address is prepended to the log message sent to the syslog server.

Setting Up the Provisioning Server

The provisioning server can be on the local LAN or anywhere on the Internet.

Multiple provisioning servers can be configured by having the provisioning server DNS name map to multiple IP addresses. The default number of provisioning servers is one and the maximum number is eight. The following protocols are supported for redundant provisioning servers: HTTPS, HTTP, and FTP. For more information on the protocol used on each platform, refer to Supported Provisioning Protocols on page 3-4.

All of the provisioning servers must be reachable by the same protocol and the content available on them must be identical. The parameters described in section Server Menu on page 3-10 can be used to configure the number of times each server will be tried for a file transfer and also how long to wait between each attempt. The maximum number of servers to be tried is configurable. For more information, contact your Certified Polycom Reseller.

Note Be aware of how logs, overrides and directories are uploaded to servers that map to multiple IP addresses. The server that these files are uploaded to may change over time.

If you want to use redundancy for uploads, synchronize the files between servers in the background.

However, you may want to disable the redundancy for uploads by specifying specific IP addresses instead of URLs for logs, overrides, and directory in the **<MAC-address>.cfg** .

To set up the provisioning server:

Note Use this procedure as a recommendation if this is your first provisioning server setup.

1. Install a provisioning server application or locate suitable existing server(s).

Polycom recommends that you use RFC-compliant servers.

2. Create an account and home directory.

Note If the provisioning protocol requires an account name and password, the server account name and password must match those configured in the phones. Defaults are: provisioning protocol: FTP, name: PlcmSpIp, password: PlcmSpIp.

Each phone may open multiple connections to the server.

The phone will attempt to upload log files, a configuration override file, and a directory file to the server. This requires that the phone's account has delete, write, and read permissions. The phone will still function without these permissions, but will not be able to upload files.

The files downloaded from the server by the phone should be made read-only.

Note Typically all phones are configured with the same server account, but the server account provides a means of conveniently partitioning the configuration. Give each account an unique home directory on the server and change the configuration on an account-by-account basis.

3. Copy all files from the distribution zip file to the phone home directory.

   Maintain the same folder hierarchy.

   There are two distribution zip files. The combined image file contains:

   - **sip.ld**
   - **sip.cfg**
   - **phone1.cfg**
   - **000000000000.cfg**
   - **000000000000-directory~.xml**
   - **SoundPointIP-dictionary.xml** (one of each supported language)
   - **SoundPointIPWelcome.wav**

   The split image file contains individual **sip.ld** files for each model as well as the configuration files and dictionary files.

   Refer to the latest *Release Notes* for a detailed description of each file in the distribution and further information on determining which distribution to use.

Provisioning Server Security Policy

You must decide on a provisioning server security policy.

Polycom recommends allowing file uploads to the provisioning server where the security environment permits. This allows event log files to be uploaded and changes made by the phone user to the configuration (through the web server and local user interface) and changes made to the directory to be backed up. This greatly eases our ability to provide customer support in diagnosing issues that may occur with the phone operation.

For organizational purposes, configuring a separate log file directory, override directory, contact directory, and license directory is recommended, but not required. The different directories can have different access permissions. For example, for LOG, CONTACTS, and OVERRIDES, allow full access (read and write) and for all others, read-only access. For more information on LOG_FILE_DIRECTORY, OVERRIDES, CONTACTS, and LICENSE, refer to Master Configuration Files on page A-2.

File permissions should give the minimum access required and the account used should have no other rights on the server.

The phone's server account needs to be able to add files to which it can write in the log file directory and the root directory. It must also be able to list files in all directories mentioned in the **<MAC-address>.cfg** file. All other files that the phone needs to read, such as the application executable and the standard configuration files, should be made read-only through file server file permissions.

Deploying Phones From the Provisioning Server

You can successfully deploy SoundPoint IP / SoundStation IP / VVX phones from one or more provisioning servers.

For all SoundPoint IP / SoundStation IP / VVX phones, follow the normal provisioning process in the next section, Provisioning Phones. However, if you have decided to daisy-chain two SoundStation IP 7000 conference phones together, read the information in Provisioning SoundStation IP 7000 Phones Using C-Link on page 3-20 to understand the different provisioning options available.

Provisioning Phones

The default configuration files will work without any changes; however, if you change any configuration file, then the others will have to adjusted accordingly.

For more information on why to create another configuration file, refer to the "Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

For more information on phone configuration and provisioning, refer to the appropriate Technical Bulletins and Quick Tips at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

For more information on encrypting configuration files, refer to Encrypting Configuration Files on page C-4.

To deploy phones from the provisioning server:

1. Create per-phone configuration files by performing the following steps:
   a. Obtain a list of phone Ethernet addresses (barcoded label on underside of phone and on the outside of the box).
   b. Create per-phone **phone[MACaddress].cfg** file by using the **phone1.cfg** file from the distribution as templates.

      For more information on the **phone1.cfg** file, refer to Per-Phone Configuration on page A-133.

Note

Throughout this guide, the terms Ethernet address and MAC address are used interchangeable.

Do not use **[MACaddress]-phone.cfg** as the per-phone filename. This filename is used by the phone itself to store user preferences (overrides).

c Edit contents of **phone[MACaddress].cfg** if desired.

For example, edit the parameters.

2. Create new configuration file(s) in the style of **sip.cfg** by performing the following steps:

a Create application **sipXXXX.cfg** file by using the **sip.cfg** file from the distribution as templates.

For more information on the **sip.cfg** file, refer to Application Configuration on page A-5.

b Edit contents of **sipXXXX.cfg** if desired.

For example, edit the parameters.

Most of the default settings are typically adequate, however, if SNTP settings are not available through DHCP, the SNTP GMT offset and (possibly) the SNTP server address will need to be edited for the correct local conditions. Changing the default daylight savings parameters will likely be necessary outside of North American locations. (Optional) Disable the local web (HTTP) server or change its signaling port if local security policy dictates (refer to Web Server <httpd/> on page A-79). Change the default location settings for user interface language and time and date format (refer to Localization <lcl/> on page A-27).

3. Create a master configuration file by performing the following steps:

a Create per-phone or per-platform **<MACaddress>.cfg** files by using the **00000000000.cfg** and files from the distribution as templates.

For more information, refer to Master Configuration Files on page A-2.

b Edit the CONFIG_FILES attribute of the **<MACaddress>.cfg** files so that it references the appropriate **phone[MACaddress].cfg** file.

For example, replace the reference to **phone1.cfg** with **phone[MACaddress].cfg**.

c Edit the CONFIG_FILES attribute of the **<MACaddress>.cfg** files so that it references the appropriate **sipXXXX.cfg** file.

For example, replace the reference to **sip.cfg** with **sip650.cfg**.

d Edit the LOG_FILE_DIRECTORY attribute of the **<MACaddress>.cfg** files so that it points to the log file directory.

e Edit the CONTACT_DIRECTORY attribute of the **<MACaddress>.cfg** files so that it points to the organization's contact directory.

4. Reboot the phones by pressing the reboot multiple key combination.

   For more information, refer to Multiple Key Combinations on page C-11.

   The BootROM and SIP application modify the APPLICATION APP_FILE_PATH attribute of the **<MACaddress>.cfg** files so that it references the appropriate **sip.ld** files.

   For example, the reference to **sip.ld** is changed to **2345-11670-001.sip.ld** to boot the SoundPoint IP 670 image.

Note

At this point, the phone sends a DHCP Discover packet to the DHCP server. This is found in the Bootstrap Protocol/option "Vendor Class Identifier" section of the packet and includes the phone's part number and the BootROM version.

For example, a SoundPoint IP 650 might send the following information:

5EL@ DC?5cSc52*46*(9N7*<u6=pPolycomSoundPointIP-SPIP_6502345-12600-001,1BR/4.0.0.0155/23-May-07 13:35BR/4.0.0.0155/23-May-07 13:35

For more information, refer to Parsing Vendor ID Information on page C-24.

5. Ensure that the configuration process completed correctly.

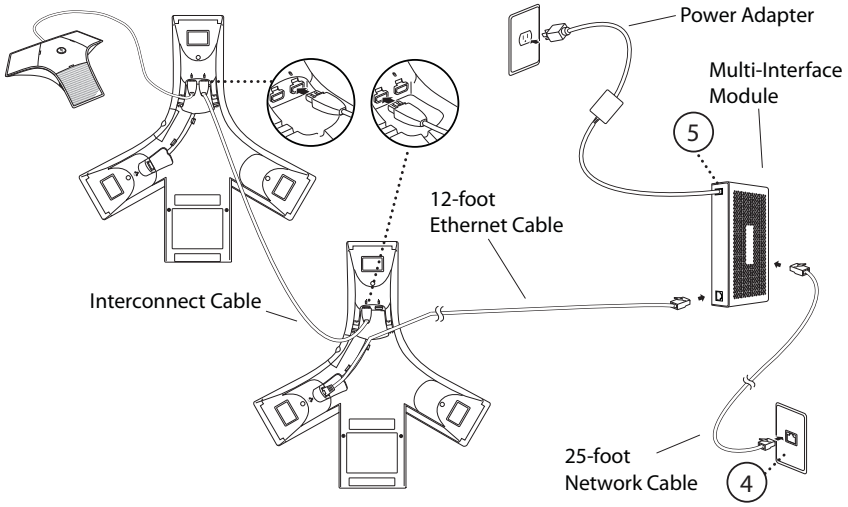
   For example, on the phone, press the **Menu** key, and then select **Status > Platform > Application** to see the SIP application version and **Status > Platform > Configuration** to see the configuration files downloaded to the phone.

   Monitor the provisioning server event log and the uploaded event log files (if permitted). All configuration files used by the provisioning server are logged.

You can now instruct your users to start making calls.

Provisioning SoundStation IP 7000 Phones Using C-Link

Normally the SoundStation IP 7000 conference phone is provisioned over the Ethernet by the provisioning server. However, when two SoundStation IP 7000 phones are daisy-chained together, the one that is not directly connected to the Ethernet can still be provisioned (known as the secondary).



The provisioning over C-Link feature is automatically enabled when a SoundStation IP 7000 phone is not connected to the Ethernet. Both SoundStation IP 7000 phones must be running the same version of the SIP application.

The steps for provisioning the secondary SoundStation IP 7000 phone are the same as for the primary SoundStation IP 7000 phone. You can reboot the primary without rebooting the secondary. However, the primary and secondary should be rebooted together for the primary/secondary relationship to be recognized. If you power up both SoundStation IP 7000 phones, the primary will power up first.

Currently, provisioning over C-Link is supported for the following configurations of SoundStation IP 7000 conference phones:

- Two SoundStation IP 7000 conference phone daisy-chained together
- Two SoundStation IP 7000 conference phone daisy-chained together with one external microphone, specifically designed for the SoundStation IP 7000 conference phone

The provisioning server (or proxy) for the secondary is determined by the following criteria:

- The primary phone must be powered up using Multi-Interface Module. PoE will not provide enough power for both phones.
- If the secondary is configured for DHCP, use the primary's provisioning server if the primary is configured for DHCP.

- If the secondary is not configured for DHCP, use the secondary's static provisioning server if it exists.
- If the secondary's static provisioning server does not exists, use the primary's provisioning server (ignoring the source).

For more information on daisy-chaining and setting up the SoundStation IP 7000 conference phone, refer to the *Setup Guide for the Polycom SoundStation IP 7000 Phone*, which is available at http://www.polycom.com/voicedocumentation/.

Upgrading SIP Application

You can upgrade the SIP application that is running on the SoundPoint IP, SoundStation IP, and VVX phones in your organization. The exact steps that you perform are dependent on the version of the SIP application that is currently running on the phones and the version that you want to upgrade to.

The BootROM, application executable, and configuration files can be updated automatically through the centralized provisioning model. These files are read-only by default.

Most organization can use the instructions shown in the next section, Supporting SoundPoint IP, SoundStation IP, and Polycom VVX Phones.

However, if your organization has a mixture of SoundPoint IP 300, 301, 500, 501, 600, 601 and/or SoundStation IP 4000 phones deployed along with other models, you will need to change the phone configuration files to continue to support the SoundPoint IP 300, 301, 500, 501, 600, and 601 and SoundStation IP 4000 phones when software releases SIP 3.2.0 or later are deployed. These models were discontinued as follows:

- The SoundPoint IP 300 and 500 phones as of May 2006.
- The SoundPoint IP 301, 600, and 601 phones as March 2008.
- The SoundPoint IP 501 phone as of August 2009.
- The SoundStation IP 4000 phone as of May 2009.

In all cases, refer to Supporting SoundPoint IP 300, 301, 500, 501, 600 and 601 and SoundStation IP 4000 Phones on page 3-23.

Warning

The SoundPoint IP 300 and 500 phones will be supported on the latest maintenance patch release of the SIP 2.1 software stream—currently SIP 2.1.4. Any critical issues that affect SoundPoint IP 300 and 500 phones will be addressed by a maintenance patch on this stream until the End of Life date for these products. Phones should be upgraded to BootROM 4.0.0 for these changes to be effective.

The SoundPoint IP 301, 501, 600, and 601 and the SoundStation IP 4000 phones will be supported on the latest maintenance patch release of the SIP 3.1 software stream—currently SIP 3.1.3 . Any critical issues that affect SoundPoint IP 300 and 500 phones will be addressed by a maintenance patch on this stream until the End of Life date for these products. Phones should be upgraded to BootROM 4.0.0 or later for these changes to be effective.

Supporting SoundPoint IP, SoundStation IP, and Polycom VVX Phones

Warning

If you need to upgrade any Polycom VVX 1500 phones running SIP 3.1.3 or earlier to SIP 3.2.2, you must perform additional steps before rebooting the phone to download the new SIP software. Refer to "Technical Bulletin 53522: Upgrading the Polycom VVX 1500 Phone to SIP 3.2.2" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

To automatically update:

1. Back up old application and configuration files.

   The old configuration can be easily restored by reverting to the backup files.

2. Customize new configuration files or apply new or changed parameters to the old configuration files.

   Differences between old and new versions of configuration files are explained in the *Release Notes* that accompany the software. Both mandatory and optional changes may present. Changes to site-wide configuration files such as **sip.cfg** can be done manually, but a scripting tool is useful to change per-phone configuration files.

Warning

The configuration files listed in CONFIG_FILES attribute of the master configuration file must be updated when the software is updated. Any new configuration files must be added to the CONFIG_FILES attribute in the appropriate order.

Mandatory changes must be made or the software may not behave as expected.

For more information, refer to the "Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

3. Save the new configuration files and images (such as **sip.ld**) on the provisioning server.

4. Reboot the phones using automatic methods such as polling or check-sync.

   Using the reboot multiple key combination should be done as a backup option only. For more information, refer to Multiple Key Combinations on page C-11.

   Since the APPLICATION APP_FILE_PATH attribute of the **<MACaddress>.cfg** files references the individual **sip.ld** files, it is possible to verify that an update is applied to phones of a particular model.

   For example, the reference to **sip.ld** is changed to **2345-11670-001.sip.ld** to boot the SoundPoint IP 670 image.

   The phones can be rebooted remotely through the SIP signaling protocol. Refer to Special Events <specialEvent/> on page A-20.

The phones can be configured to periodically poll the provisioning server to check for changed configuration files or application executable. If a change is detected, the phone will reboot to download the change. Refer to Provisioning <prov/> on page A-114.

Supporting SoundPoint IP 300, 301, 500, 501, 600 and 601 and SoundStation IP 4000 Phones

With enhancements available since BootROM 4.0.0 and SIP 2.1.2, you can modify the **000000000000.cfg** or **<MACaddress>.cfg** configuration file to direct phones to load the software image and configuration files based on the phone model number. Refer to Master Configuration Files on page A-2.

The SIP 3.2.0 or later software distributions contain only the new distribution files for the new release. You must rename the **sip.ld**, **sip.cfg**, and **phone1.cfg** from a previous 2.1.x distribution that is compatible with SoundPoint IP 300 and 500 phones or a previous 3.1.y distribution that is compatible with SoundPoint IP 301, 501, 600, and 601 SoundStation IP 4000 phones.

The following procedure must be used for upgrading to SIP 3.2.0 or later for installations that have SoundPoint IP 300, 301, 500, 501, 600, 601 and SoundStation IP 4000 phones deployed. It is also recommended that this same approach be followed even if these phones are not part of the deployment as it will simplify management of phone systems with future software releases.

To upgrade your SIP application:

1. Do one of the following steps:

   a. Place all **bootrom.ld** files corresponding to BootROM release zip file onto the provisioning server.

   b. Ensure that all phones are running BootROM 4.0.0 or later code.

2. Copy **sip.ld**, **sip.cfg** and **phone1.cfg** from the SIP 3.2.0 or later release distribution onto the provisioning server.

   These are the relevant files for all phones except the SoundPoint IP 300, 301, 500, 501, 600, 601 and SoundStation IP 4000 phones.

3. Rename **sip.ld**, **sip.cfg**, and **phone1.cfg** from the previous distribution to **sip_21x.ld**, **sip_21x.cfg**, and **phone1_21x.cfg** respectively on the provisioning server.

   These are the relevant files for supporting the SoundPoint IP 300 and 500 phones.

4. Rename **sip.ld**, **sip.cfg**, and **phone1.cfg** from the previous distribution to **sip_31y.ld**, **sip_31y.cfg**, and **phone1_31y.cfg** respectively on the provisioning server.

   These are the relevant files for supporting the SoundPoint IP 301, 501, 600, 601 and SoundStation IP 4000 phones.

5. Modify the **000000000000.cfg** file, if required, to match your configuration file structure.

   For example:

```
    <APPLICATION
    APP_FILE_PATH="sip.ld"
    APP_FILE_PATH_SPIP500="sip_214.ld"
    APP_FILE_PATH_SPIP300="sip_214.ld"
    APP_FILE_PATH_SPIP601="sip_313.ld"
    APP_FILE_PATH_SPIP600="sip_313.ld"
    APP_FILE_PATH_SPIP501="sip_313.ld"
    APP_FILE_PATH_SPIP301="sip_313.ld"
    APP_FILE_PATH_SSIP4000="sip_313.ld"
    CONFIG_FILES="[PHONE_MAC_ADDRESS]-user.cfg, phone1.cfg, sip.cfg"
    CONFIG_FILES_SPIP500="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_214.cfg, sip_214.cfg"
    CONFIG_FILES_SPIP300="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_214.cfg, sip_214.cfg"
    CONFIG_FILES_SPIP601="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_313.cfg, sip_313.cfg"
    CONFIG_FILES_SPIP600="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_313.cfg, sip_313.cfg"
    CONFIG_FILES_SPIP501="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_313.cfg, sip_313.cfg"
    CONFIG_FILES_SPIP301="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_313.cfg, sip_313.cfg"
    CONFIG_FILES_SSIP4000="[PHONE_MAC_ADDRESS]-user.cfg,
phone1_313.cfg, sip_313.cfg"
    MISC_FILES=""
    LOG_FILE_DIRECTORY=""
    OVERRIDES_DIRECTORY=""
    CONTACTS_DIRECTORY=""
    />
```

6. Remove any **<MACaddress>.cfg** files that may have been used with earlier releases from the provisioning server.

Note

This approach takes advantage of an enhancement that was added in SIP2.0.1/BootROM 3.2.1 that allows for the substitution of the phone specific [MACADDRESS] inside configuration files. This avoids the need to create unique **<MACaddress>.cfg** files for each phone such that the default **000000000000.cfg** file can be used for all phones in a deployment.

If this approach is not used, then changes will need to be made to all the **<MACaddress>.cfg** files for SoundPoint IP 300, 301, 500, 501, 600, and 601 and SoundStation IP 4000 phones or all of the **<MACaddress>.cfg** files if it is not explicitly known which phones are SoundPoint IP 300 and 500 phones.

For more information, refer to "Technical Bulletin 35311: Supporting SoundPoint IP 300, 301, 500, 501, 600, and 601 and SoundStation IP 4000 Phones with SIP 2.2.0 or SIP 3.2.0 and Later Releases"at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

4

Configuring Your System

After you set up your SoundPoint IP / SoundStation IP / VVX phones on the network, you can allow users to place and answer calls using the default configuration, however, you may require some basic changes to optimize your system for best results.

This chapter provides information for making configuration changes for:

- Setting Up Basic Features
- Setting Up Advanced Features
- Setting Up Audio Features
- Setting Up Video Features
- Setting Up Security Features

This chapter also provides instructions on:

- Configuring SoundPoint IP / SoundStation IP / VVX Phones Locally

To troubleshoot any problems with your SoundPoint IP / SoundStation IP / VVX phones on the network, refer to Troubleshooting Your SoundPoint IP / SoundStation IP / VVX Phones on page 5-1. For more information on the configuration files, refer to Configuration Files on page A-1.

Setting Up Basic Features

This section provides information for making configuration changes for the following basic features:

- Call Log
- Call Timer
- Call Waiting
- Called Party Identification
- Calling Party Identification

- Missed Call Notification
- Connected Party Identification
- Context Sensitive Volume Control
- Customizable Audio Sound Effects
- Message Waiting Indication
- Distinctive Incoming Call Treatment
- Distinctive Ringing
- Distinctive Call Waiting
- Do Not Disturb
- Handset, Headset, and Speakerphone
- Local Contact Directory
- Local Digit Map
- Microphone Mute
- Soft Key Activated User Interface
- Speed Dial
- Time and Date Display
- Idle Display Animation
- Ethernet Switch
- Graphic Display Backgrounds

This section also provides information for making configuration changes for the following basic call management features:

- Automatic Off-Hook Call Placement
- Call Hold
- Call Transfer
- Local / Centralized Conferencing
- Call Forward
- Directed Call Pick-Up
- Group Call Pick-Up
- Call Park/Retrieve
- Last Call Return

Call Log

The phone maintains a call log. The log contains call information such as remote party identification, time and date, and call duration. It can be used to redial previous outgoing calls, return incoming calls, and save contact information from call log entries to the contact directory.

The call log is stored in volatile memory and is maintained automatically by the phone in three separate lists: Missed Calls, Received Calls and Placed Calls. The call lists can be cleared manually by the user and will be erased when the phone is restarted.

Note

On some SoundPoint IP platforms, missed calls and received calls appear in one list. Missed calls appear as and received calls appear as .

The “call list” feature can be disabled on all SoundPoint IP and SoundStation IP platforms except the SoundPoint IP 32x/33x and SoundStation IP 7000.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration File: **sip.cfg**	Enable or disable all call lists or individual call lists. • For more information, refer to Feature <feature/> on page A-116.

Call Timer

A call timer is provided on the display. A separate call timer is maintained for each distinct call in progress. The call duration appears in hours, minutes, and seconds.

There are no related configuration changes.

Call Waiting

When an incoming call arrives while the user is active on another call, the incoming call is presented to the user visually on the LCD display. A configurable sound effect such as the familiar call-waiting beep will be mixed with the active call audio as well.

Configuration changes can performed centrally at the provisioning server:

Central (provisioning server)	Configuration File: **phone1.cfg**	Specify the ring tone heard on an incoming call when another call is active. • For more information, refer to Call Waiting <callWaiting/> on page A-142. Disable call waiting. • For more information, refer to Registration <reg/> on page A-134.

For related configuration changes, refer to Customizable Audio Sound Effects on page 4-6.

Called Party Identification

The phone displays and logs the identity of the remote party specified for outgoing calls. This is the party that the user intends to connect with.

The identity displayed is based on the number of the placed call and information obtained from the network signaling.

Note The phone does not match the number of the placed call to any entries in the Local Contact Directory or Corporate Directory.

There are no related configuration changes.

Calling Party Identification

The phone displays the caller identity, derived from the network signaling, when an incoming call is presented, if the information is provided by the call server. For calls from parties for which a directory entry exists, the local name assigned to the Contact Directory entry may optionally be substituted.

Note The phone does not match the received number to any entries in the Corporate Directory.

Configuration changes can performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration File: **sip.cfg**	Specify whether or not to use directory name substitution. • For more information, refer to User Preferences <up/> on page A-31.
Local	Web Server (if enabled)	Specify whether or not to use directory name substitution. Navigate to: http://<phoneIPAddress>/coreConf.htm#us Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Missed Call Notification

The phone can display the number of calls missed since the user last looked at the Missed Calls list. The phone can be configured to use a built-in missed call counter or to display information provided by a Session Initiation Protocol (SIP) server.

Note On some SoundPoint IP platforms, missed calls and received calls appear in one list.

Configuration changes can performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.
	Configuration file: **phone1.cfg**	Specify per-registration whether all missed-call events or only remote/server-generated missed-call events will be displayed. • For more information, refer to Missed Call Configuration <serverMissedCall/> on page A-141.

Connected Party Identification

The identity of the remote party to which the user has connected is displayed and logged, if the name and ID is provided by the call server. The connected party identity is derived from the network signaling. In some cases the remote party will be different from the called party identity due to network call diversion. For example, Bob places a call to Alice, but he ends up connected to Fred.

There are no related configuration changes.

Context Sensitive Volume Control

The volume of user interface sound effects, such as the ringer, and the receive volume of call audio is adjustable for speakerphone, handset, and headset separately. While transmit levels are fixed according to the TIA/EIA-810-A standard, receive volume is adjustable. For SoundPoint IP and VVX phones, if using the default configuration parameters, the receive handset/headset volume resets to nominal after each call to comply with regulatory requirements. Handsfree volume persists with subsequent calls.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Adjust receive and handset/headset volume. • For more information, refer to Volume Persistence <volume/> on page A-50.

Customizable Audio Sound Effects

Audio sound effects used for incoming call alerting and other indications are customizable. Sound effects can be composed of patterns of synthesized tones or sample audio files. The default sample audio files may be replaced with alternates in **.wav** file format. Supported **.wav** formats include:

- mono G.711 (13-bit dynamic range, 8-khz sample rate)
- mono L16/16000 (16-bit dynamic range, 16-kHz sample rate)
- mono L16/32000 (16-bit dynamic range, 32-kHz sample rate)
- mono L16/48000 (16-bit dynamic range, 48-kHz sample rate)

Note L16/32000 and L16/48000 are only supported on SoundPoint IP 7000 phones.

Note The alternate sampled audio sound effect files must be present on the provisioning server or the Internet for downloading at boot time.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration File: **sip.cfg**	Specify patterns used for sound effects and the individual tones or sampled audio files used within them. • For more information, refer to Sampled Audio for Sound Effects <saf/> on page A-37 or Sound Effects <se/> on page A-38.
Local	Web Server (if enabled)	Specify sampled audio wave files to replace the built-in defaults. Navigate to http://<phoneIPAddress>/coreConf.htm#sa Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Message Waiting Indication

The phone will flash a message-waiting indicator (MWI) LED when instant messages and voice messages are waiting.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **phone1.cfg**	Specify per-registration whether the MWI LED is enabled or disabled. • For more information, refer to Message Waiting Indicator <mwi/> on page A-149. Specify whether MWI notification is displayed for registration x (pre-SIP 2.1 behavior is enabled). • For more information, refer to User Preferences <up/> on page A-31.

Distinctive Incoming Call Treatment

The phone can automatically apply distinctive treatment to calls containing specific attributes. The distinctive treatment that can be applied includes customizable alerting sound effects and automatic call diversion or rejection. Call attributes that can trigger distinctive treatment include the calling party name or SIP contact (number or URL format).

For related configuration changes, refer to Local Contact Directory on page 4-10.

Distinctive Ringing

There are three options for distinctive ringing:

1. The user can select the ring type for each line by pressing the **Menu** key, and then selecting **Settings > Basic > Ring Type**. This option has the third (lowest) priority.
2. The ring type for specific callers can be assigned in the contact directory. For more information, refer to Distinctive Incoming Call Treatment, the previous section. This option is second in priority.
3. The `voIpProt.SIP.alertInfo.x.value` and `voIpProt.SIP.alertInfo.x.class` fields can be used to map calls to specific ring types. This option requires server support and is first (highest) in priority.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the mapping of Alert-Info strings to ring types. • For more information, refer to Alert Information <alertInfo/> on page A-19.
	Configuration file: **phone1.cfg**	Specify the ring type to be used for each line. • For more information, refer to Registration <reg/> on page A-134.
	XML File: **<Ethernet address>-directory.xml**	This file can be created manually using an XML editor. • For more information, refer to Local Contact Directory on page 4-10.
Local	Local Phone User Interface	The user can edit the ring types selected for each line under the Settings menu. The user can also edit the directory contents. Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.

Distinctive Call Waiting

The `voIpProt.SIP.alertInfo.x.value` and `voIpProt.SIP.alertInfo.x.class` fields can be used to map calls to distinct call waiting types, currently limited to two styles. This feature requires server support.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the mapping of Alert-Info strings to call waiting types. • For more information, refer to Alert Information <alertInfo/> on page A-19.

Do Not Disturb

A Do Not Disturb (DND) feature is available to temporarily stop all incoming call alerting. Calls can optionally be treated as though the phone is busy while DND is enabled. DND can be configured as a per-registration feature. Incoming calls received while DND is enabled are logged as missed. For more information on forwarding calls while DND is enabled, refer to Call Forward on page 4-22.

Server-based DND is active if the feature is enabled on both the phone and the server and the phone is registered. The server-based DND feature is applicable for all registrations on the phone (no per-registration mode) and it disables local Call Forward and DND features unless configured otherwise.

Server-based DND will behave the same as per-SIP 2.1 per-registration feature with the following exceptions:

- Server based DND cannot be used if the phone is configured as a shared line.
- If server-based DND is enabled, but inactive, and the user presses the DND key or selects the DND option on the Feature menu, the "Do Not Disturb" message does not appear on the user's phone (incoming call alerting will continue).

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Enable or disable server-based DND. • For more information, refer to SIP <SIP/> on page A-11 Enable or disable local DND behavior when server-based enabled. • For more information, refer to SIP <SIP/> on page A-11. Specify whether or not DND results in incoming calls being given busy treatment. • For more information, refer to Call Handling Configuration <call/> on page A-80.
	Configuration file: **phone1.cfg**	Enable or disable server-based DND as a per-registration feature. • For more information, refer to Registration <reg/>on page A-134. Specify whether DND is treated as a per-registration feature or a global feature on the phone. • For more information, refer to Do Not Disturb <dnd/> on page A-144.
Local	Local Phone User Interface	Enable or disable DND using the **Do Not Disturb** key on the SoundPoint IP 550, 560, 650, and 670 and the Polycom VVX 1500 or the "Do Not Disturb" option on the Features menu on the SoundPoint IP 32x/33x, 430, and 450 and SoundStation IP 5000, 6000 and 7000. ***Note:*** *The LED on the* ***Do Not Disturb*** *key on the Polycom VVX 1500 is red when pressed or when server-based DND is enabled.*

Handset, Headset, and Speakerphone

SoundPoint IP phones come standard with a handset and a dedicated connector is provided for a headset (not supplied). All Polycom phones are full-duplex speakerphones. The SoundPoint IP phones provide dedicated keys for convenient selection of either the speakerphone or headset.

All Polycom desktop phones can be configured to use the electronic hookswitch. For more information, refer to "Technical Bulletin 35150: Using an Electronic Hookswitch with SoundPoint IP and Polycom VVX 1500 Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Enable or disable persistent headset mode. For more information, refer to User Preferences <up/> on page A-31. Enable or disable hands-free speakerphone mode. • For more information, refer to User Preferences <up/> on page A-31.
	Configuration file: **phone1.cfg**	Specify whether or not the electronic hookswitch is enabled and what type of headset is attached. • For more information, refer to User Preferences <up/>on page A-134.
Local	Web Server (if enabled)	Enable or disable persistent headset mode. Navigate to: http://<phoneIPAddress>/coreConf.htm#us
	Local Phone User Interface	Enable or disable persistent headset mode through the Settings menu (***Settings > Basic > Preferences > Headset > Headset Memory Mode***). Enable or disable hands-free speakerphone mode through the Settings menu (***Settings > Advanced > Admin Settings > Phone Settings***). Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Local Contact Directory

The phone maintains a local contact directory. The directory can be downloaded from the provisioning server and edited locally (if configured in that way). Contact information from previous calls may be easily added to the directory for convenient future access.

The directory is the central database for several other features including speed-dial, distinctive incoming call treatment, presence, and instant messaging. The maximum number of entries in the local contact directory is phone-dependent.

Note

If a user makes a change to the local contact directory, there is a five second timeout before it is uploaded to the provisioning server as **<mac-address>-directory.cfg**.

If so configured, the first and last name fields of the local contact directory entries which match incoming calls will be used for caller identification display and in the call lists (instead of the name provided through network signaling).

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Set whether the directory uses volatile storage on the phone. • For more information, refer to Local Directory <local/> on page A-86. Specify whether or not the local contact directory is read only. • For more information, refer to Local Directory <local/> on page A-86.
	XML file**:** **000000000000-directory.xml**	A sample file named **000000000000-directory~.xml** (Note the extra "~" in the filename) is included with the application file distribution. This file can be used as a template for the per-phone ***<Ethernet address>*-directory.xml** directories (edit contents, then rename to ***<Ethernet address>*-directory.xml**). It also can be used to seed new phones with an initial directory (edit contents, then remove "~" from file name). Telephones without a local directory, such as new units from the factory, will download the **00000000000-directory.xml** directory and base their initial directory on it. These files should be edited with an XML editor. These files can be downloaded once per reflash. For information on file format, refer to the next section, Local Contact Directory File Format.
	XML file: **<Ethernet address>-directory.xml**	This file can be created manually using an XML editor. For information on file format, refer to the next section, Local Contact Directory File Format.
Local	Local Phone User Interface	The user can edit the directory contents if configured in that way. Changes will be stored in the phone's flash file system and backed up to the provisioning server copy of ***<Ethernet address>*-directory.xml** if this is configured. When the phone boots, the provisioning server copy of the directory, if present, will overwrite the local copy.

Local Contact Directory File Format

An example of a local contact directory is shown below. The subsequent table provides an explanation of each element. Elements can appear in any order.

```
<?xml version="1.0" encoding="UTF-8" standalone="yes" ?>
<directory>
   <item_list>
      <item>
         <ln>Doe</ln>
         <fn>John</fn>
         <ct>1001</ct>
         <sd>1</sd>
         <lb>Mr</lb>
         <pt>H323</pt>
         <rt>1</rt>
```

```
            <dc/>
            <ad>0</ad>
            <ar>0</ar>
            <bw>0</bw>
            <bb>0</bb>
         </item>
          ...
          <item>
            <ln>Smith</ln>
            <fn>Bill</fn>
            <ct>1003</ct>
            <sd>3</sd>
            <lb>Dr</lb>
            <pt>SIP</pt>
            <rt>3</rt>
            <dc/>
            <ad>0</ad>
            <ar>0</ar>
            <bw>0</bw>
            <bb>0</bb>
         </item>
      </item_list>
   </directory>
```

Element	Permitted Values	Interpretation
fn	UTF-8 encoded string of up to 40 bytes	first name ***Note**: In some cases, this will be less than 40 characters due to UTF-8's variable length encoding.*
ln	UTF-8 encoded string of up to 40 bytes	last name ***Note**: In some cases, this will be less than 40 characters due to UTF-8's variable length encoding.*
ct	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL	contact Used by the phone to address a remote party in the same way that a string of digits or a SIP URL are dialed manually by the user. This element is also used to associate incoming callers with a particular directory entry. For Polycom VVX 1500 phones, the maximum field length is 128 characters; for all other phones, the maximum is 32 characters. ***Note**: This field cannot be null or duplicated.*
sd	Null, 1 to 9999	speed-dial index Associates a particular entry with a speed dial bin for one-touch dialing or dialing from the speed dial menu. ***Note**: On the SoundPoint IP 32x/33x and the SoundStation IP 6000 and 7000, the maximum speed-dial index is 99.*

Element	Permitted Values	Interpretation
lb	UTF-8 encoded string of up to 40 bytes	label ***Note**: In some cases, this will be less than 40 characters due to UTF-8's variable length encoding.* ***Note**: The label of a contact directory item is by default the label attribute of the item. If the label attribute does not exist or is Null, then the concatenation of first name and last name will be used as label. A space is added between first and last names.*
pt	"SIP", "H323", or "Unspecified"	protocol The protocol to use when placing a call to this contact.
rt	Null, 1 to 21	ring type When incoming calls can be associated with a directory entry by matching the address fields, this field is used to specify ring type to be used.
dc	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL	divert contact The forward-to address for the autodivert feature.
ad	0,1	auto divert If set to 1, automatically diverts callers that match the directory entry to the address specified in divert contact. ***Note**: If auto-divert is enabled, it has precedence over auto-reject.*
ar	0,1	auto-reject If set to 1, automatically rejects callers that match the directory entry. ***Note**: If auto-divert is also enabled, it has precedence over auto-reject.*
bw	0,1	buddy watching If set to 1, add this contact to the list of watched phones.
bb	0,1	buddy block If set to 1, block this contact from watching this phone.

Local Digit Map

The phone has a local digit map feature to automate the setup phase of number-only calls. When properly configured, this feature eliminates the need for using the **Dial** or **Send** soft key when making outgoing calls. As soon as a digit pattern matching the digit map is found, the call setup process will complete automatically. The configuration syntax is based on recommendations in 2.1.5 of RFC 3435. The phone behavior when the user dials digits that do not match the digit map is configurable. It is possible to strip a trailing # from the digits sent or to replace certain matched digits (with

the introduction of "R" to the digit map). It is also possible to direct the protocol used to place a call (with the introduction of "S" and "H" to the digit map).

For more information digit maps, refer to "Technical Bulletin 11572: Changes to Local Digit Maps on SoundPoint IP / SoundStation IP / VVX Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Note Digit maps do not apply to on-hook dialing. The parameter settings described in Dial Plan <dialplan/> on page A-23 are ignored during on-hook dialing.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify impossible match behavior, trailing # behavior, digit map matching strings, and time out value. • For more information, refer to Dial Plan <dialplan/> on page A-23.
	Configuration file: **phone1.cfg**	Specify per-registration impossible match behavior, trailing # behavior, digit map matching strings, and time out values that override those in **sip.cfg**. • For more information, refer to Dial Plan <dialplan/> on page A-145.
Local	Web Server (if enabled)	Specify impossible match behavior, trailing # behavior, digit map matching strings, and time out value. Navigate to: http://<phoneIPAddress>/appConf.htm#ls Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.

Microphone Mute

A microphone mute feature is provided. When activated, visual feedback is provided. This is a local function and cannot be overridden by the network.

There are no related configuration changes.

Soft Key Activated User Interface

The user interface makes extensive use of intuitive, context-sensitive soft key menus. The soft key function is shown above the key on the graphic display.

Using the Configurable Soft Key configuration parameters, an administrator can modify the default soft keys by removing them at different call stages and/or adding specific single or multiple functions. Refer to Enhanced Feature Keys on page 4-38 and Configurable Soft Keys on page 4-48.

Speed Dial

Entries in the local directory can be linked to the speed dial system. The speed dial system allows calls to be placed quickly from dedicated keys as well as from a speed dial menu.

For SoundPoint IP 32x/33x desktop phones and SoundStation IP 6000 and 7000 conference phones, the speed dial index range is 1 to 99. For all other SoundPoint IP and Polycom VVX phones, the range is 1 to 9999.

If Presence watching is enabled for speed dial entries, their status will be shown on the idle display (if the SIP server supports this feature). For more information, refer to Presence on page 4-64.

Configuration changes can performed centrally at the provisioning server or locally:

Central (provisioning server)	XML file: **<Ethernet address>-directory.xml**	The `<sd>x</sd>` element in the ***<Ethernet address>*-directory.xml** file links a directory entry to a speed dial resource within the phone. Speed dial entries are mapped automatically to unused line keys (line keys are not available on the SoundStation IP 6000 and 7000) and are available for selection within the speed dial menu. (Press the **Up** arrow key from the idle display to jump to the Speed Dial list). • For more information, refer to Local Contact Directory on page 4-10.
Local	Local Phone User Interface	The next available Speed Dial Index is assigned to new directory entries. Key pad short cuts are available to facilitate assigning and modifying the Speed Dial Index value for entries in the directory. The Speed Dial Index field is used to link directory entries to speed dial operations. Changes will be stored in the phone's flash file system and backed up to the provisioning server copy of ***<Ethernet address>*-directory.xml** if this is configured. When the phone boots, the provisioning server copy of the directory, if present, will overwrite the local copy.

Time and Date Display

The phone maintains a local clock and calendar. Time and date can be displayed in certain operating modes such as when the phone is idle and during a call. The clock and calendar must be synchronized to a remote Simple Network Time Protocol (SNTP) timeserver. The time and date displayed on the phone will flash continuously to indicate that they are not accurate until a successful SNTP response is received. The time and date display can use one

of several different formats and can be turned off. The SoundPoint IP 32x/33x and IP 4xx phones have a limited selection of date formats due to a smaller display size.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn time and date display on or off. • For more information, refer to User Preferences <up/> on page A-31. Set the time and date display formats. • For more information, refer to Date and Time <datetime/> on page A-31. Set the basic SNTP settings and daylight savings parameters. • For more information, refer to Time Synchronization <sntp/> on page A-75.
Local	Web Server (if enabled)	Set the basic SNTP and daylight savings settings. Navigate to: http://<phoneIPAddress>/coreConf.htm#ti Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	The basic SNTP settings can be made in the Network Configuration menu. For more information, refer to DHCP or Manual TCP/IP Setup on page 3-2. The user can edit the time and date format and enable or disable the time and date display under the Settings menu. Changes are saved to local flash and backed up to <*Ethernet address*>-phone.cfg on the provisioning server. They will permanently override global settings unless deleted through the Reset Local Config menu selection.

Idle Display Animation

All phones can display a customized animation on the idle display in addition to the time and date. For example, a company logo could be displayed (refer to Adding a Background Logo on page C-6).

Note Currently customized animations are not supported on the Polycom VVX 1500.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	To turn idle display animation on or off. • For more information, refer to Indicators <ind/> on page A-100. To replace the animation used for the idle display. • For more information, refer to Animations <anim/> <IP_330/>, <IP_335/>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/> on page A-101. To change the position of the idle display animation. • For more information, refer to Graphic Icons <gi/> <IP_330>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/> on page A-103.

Ethernet Switch

The SoundPoint IP phones (except the SoundPoint IP 32x) and the Polycom VVX 1500 contain two Ethernet ports, labeled LAN and PC, and an embedded Ethernet switch that runs at full line-rate. The SoundStation IP phones contain only one Ethernet port, labeled LAN. The Ethernet switch allows a personal computer and other Ethernet devices to connect to the office LAN by daisy chaining through the phone, eliminating the need for a stand-alone hub. The SoundPoint IP switch gives higher transmit priority to packets originating in the phone. The phone can be powered through a local AC power adapter or can be line-powered (power supplied through the signaling or idle pairs of the LAN Ethernet cable). Line powering typically requires that the phone plugs directly into a dedicated LAN jack. Devices that do not require LAN power can then plug into the SoundPoint IP PC Ethernet port. To disable the PC Ethernet port, refer to Disabling PC Ethernet Port on page C-27.

SoundPoint IP Switch - Port Priorities

To help ensure good voice quality, the Ethernet switch embedded in the SoundPoint IP phones should be configured to give voice traffic emanating from the phone higher transmit priority than those from a device connected to the PC port. If not using a VLAN (VLAN set to blank in the setup menu), this will automatically be the case. If using a VLAN, ensure that the 802.1p priorities for both default and real-time transport protocol (RTP) packet types are set to 2 or greater. Otherwise, these packets will compete equally with those from the PC port. For more information, refer toVoice Settings <voice/> on page A-44 and Video Settings <video/> on page A-64.

Graphic Display Backgrounds

You can set up a picture or design to be displayed on the background of the graphic display of all SoundPoint IP 450, 550, 560, 650, and 670 and Polycom VVX 1500 phones.

Note When installing a background of your choice, care needs to be taken to ensure that the background does not adversely affect the visibility of the text on the phone display. As a general rule, backgrounds should be light in shading for better usability.

For SoundPoint IP 450, 550, 560, 650, and 670 phones:

- There are a number of default backgrounds, both solid color and pictures. Both BMP and JPEG files are supported. You can also select the label color for soft key and line key labels. Users can select which background and label color appears on their phone.

  You can modify the supported solid color and pictures backgrounds. For example, you can add a gray solid color background or modify a picture to one of your choice.

For Polycom VVX 1500 phones:

- You can select the pictures or designs displayed on the background. The supported formats include JPEG, BMP, and PNG and the maximum size is 800x480. A default picture is displayed when the phone starts up the first time.

  Users can select which background appears on their individual phones. Users can also select a background from an image displayed by the digital picture frame feature (refer to Digital Picture Frame on page 4-37).

Note Support for resolutions greater than 800x480 is inconsistent. Content may be truncated or nor displayed. Progressive/multiscan JPEG images are not supported at this time.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **phone1.cfg**	Specify which background will be displayed. • For more information, refer to Backgrounds <bg/> on page A-95.
Local	Local Phone User Interface	On the Polycom VVX 1500, the user can save one of the images as the background by selecting **Save as Background** on the touch screen.

To modify the backgrounds displayed on the supported SoundPoint IP phones:

1. Modify the **sip.cfg** configuration file as follows:
   a. Open **sip.cfg** in an XML editor.
   b. Locate the background parameter.

c For the solid backgrounds, set the name and RGB values. For example:

```
bg.hiRes.gray.pat.solid.3.name="Gray"
bg.hiRes.gray.pat.solid.3.red="128"
bg.hiRes.gray.pat.solid.3.green="128"
bg.hiRes.gray.pat.solid.3.blue="128"
```

d For images, select a filename. For example:

```
bg.hiRes.gray.bm.3.name="polycom.jpg"
bg.hiRes.gray.bm.3.em.name="polycomEM.jpg"
bg.hiRes.gray.bm.3.adj="0"
```

The default size for images on a phone is 320 x 160. The default size for images on an Expansion Module is 160 x 320. Use a photo editor on a computer to adjust the image you want to display. (Edit the image so the main subject is centered in the upper right corner of the display.)

Download the file to the provisioning server.

e Save the modified **sip.cfg** configuration file.

Automatic Off-Hook Call Placement

The phone supports an optional automatic off-hook call placement feature for each registration. This feature is sometimes referred to as 'hot-dialing'.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **phone1.cfg**	Specify which registrations have the feature and what contact to call when going off hook. • For more information, refer to Automatic Off-Hook Call Placement <autoOffHook/> on page A-140.

Call Hold

The purpose of hold is to pause activity on one call so that the user may use the phone for another task, such as to make or receive another call. Network signaling is employed to request that the remote party stop sending media and to inform them that they are being held. A configurable local hold reminder feature can be used to remind the user that they have placed calls on hold. The call hold reminder is always played through the speakerphone.

As of SIP 3.1, you can supply a Music on Hold URI if supported by the call server. For more information, refer to draft RFC *draft-worley-service-example*.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify whether RFC 2543 (c=0.0.0.0) or RFC 3264 (a=sendonly or a=inactive) outgoing hold signaling is used. • For more information, refer to SIP <SIP/> on page A-11. Specify local hold reminder options. • For more information, refer to Hold, Local Reminder <hold/><localReminder/> on page A-85. Specify the Music on Hold URI. • For more information, refer to Music on Hold <musicOnHold/> on page A-21.
	Configuration file: **phone1.cfg**	Specify the Music on Hold URI. • For more information, refer to Music on Hold <musicOnHold/> on page A-21.
Local	Web Server (if enabled)	Specify whether or not to use RFC 2543 (c=0.0.0.0) outgoing hold signaling. The alternative is RFC 3264 (a=sendonly or a=inactive). Navigate to: http://<phoneIPAddress>/appConf.htm#ls Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	Use the **Call Server Configuration** menu to specify whether or not to use RFC 2543 (c=0.0.0.0) outgoing hold signaling. The alternative is RFC 3264 (a=sendonly or a=inactive).

Call Transfer

Call transfer enables the user (party A) to move an existing call (party B) into a new call between party B and another user (party C) selected by party A. The phone offers three types of transfers:

- Blind transfers—The call is transferred immediately to party C after party A has finished dialing party C's number. Party A does not hear ring-back.
- Attended transfers—Party A dials party C's number and hears ring-back and decides to complete the transfer before party C answers. This option can be disabled.
- Consultative transfers—Party A dials party C's number and talks privately with party C after the call is answered, and then completes the transfer or hangs up.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify whether to allow a transfer during the proceeding state of a consultation call. • For more information, refer to SIP <SIP/> on page A-11. Specify whether a transfer is blind or not. • For more information, refer to Call Handling Configuration <call/> on page A-80.

Local / Centralized Conferencing

The phone can conference together the local user with the remote parties of a configurable number of independent calls by using the phone's local audio processing resources for the audio bridging. There is no dependency on network signaling for local conferences.

All phones support three-party local conferencing. The SoundPoint IP 450, 550, 560, 650, and 670 and SoundStation IP 7000 phones may support four-way local conferencing.

Note

Four-party conferencing requires a license key for activation. For more information, refer to Manage Conferences on page 4-22.

If the initiator of a three-party local conference ends the call, the other members of the call may still communicate. If the initiator of a four-party local conference ends the call, the conference ends.

The phone also supports centralized conferences for which external resources are used such as a conference bridge. This relies on network signaling.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the conference hold behavior (all parties on hold or only host is on hold). • For more information, refer to Call Handling Configuration <call/> on page A-80. Specify whether or not all parties hear sound effects while setting up a conference. • For more information, refer to Call Handling Configuration <call/> on page A-80. Specify which type of conference to establish and the address of the centralized conference resource. • For more information, refer to Conference Setup <conference/> on page A-20.

Manage Conferences

Note

This feature is supported on the SoundPoint IP 450, 550, 560, 650, and 670 desktop phones, the SoundStation IP 7000 conference phone, and the Polycom VVX business media phone.

This feature requires a license key for activation on all phones except the SoundStation IP 7000 and the Polycom VVX 1500. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The individual parties within a conference can be managed. New parties can be added and information about the conference participants can be viewed (for example, names, phone numbers, send/receive status or media flow, receive and transmit codecs, and hold status).

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.

Call Forward

The phone provides a flexible call forwarding feature to forward calls to another destination. Call forwarding can be applied in the following cases:

- Automatically to all calls
- Calls from a specific caller (extension)
- When the phone is busy
- When Do Not Disturb is active
- After an extended period of alerting

The user can elect to manually forward calls while they are in the alerting state to a predefined or manually specified destination. The call forwarding feature works in conjunction with the distinctive incoming call treatment feature (refer to Distinctive Incoming Call Treatment on page 4-7). The user's ability to originate calls is unaffected by all call forwarding options. Each registration has its own forwarding properties.

Server-based call forwarding is active if the feature is enabled on both the phone and the server and the phone is registered. If server-based call forwarding is enabled on any of the phone's registrations, the other registrations are not affected. Server-based call forwarding disables local Call Forward and DND features unless configured otherwise.

Server-based call forwarding will behave the same as per-SIP 2.1 feature with the following exception:

- If server-based call forwarding is enabled, but inactive, and the user selects the call forward soft key, the "moving arrow" icon does not appear on the user's phone (incoming calls are not forwarded).

Note Server-based and local call forwarding are disabled if Shared Call Appearance or Bridged Line Appearance is enabled.

The Diversion field with a SIP header is often used by the call server to inform the phone of a call's history. For example, when a phone has been set to enable call forwarding, the Diversion header allows the receiving phone to indicate who the call was from, and from which phone number it was forwarded. (For more information, refer to Header Support on page B-4.) .

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Enable or disable server-based call forwarding. • For more information, refer to SIP <SIP/> on page A-11. Enable or disable local call forwarding behavior when server-based enabled. • For more information, refer to SIP <SIP/> on page A-11. Enable or disable display of Diversion header and the order in which to display the caller ID and number. • For more information, refer to SIP <SIP/> on page A-11.
	Configuration file: **phone1.cfg**	Enable or disable server-based call forwarding as a per-registration feature. • For more information, refer to Registration <reg/>on page A-134. Set all call diversion settings including a global forward-to contact and individual settings for call forward all, call forward busy, call forward no-answer, and call forward do-not-disturb. • For more information, refer to Diversion <divert/> on page A-142.
Local	Web Server (if enabled)	Set all call diversion settings. Navigate to: http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	The user can set the call-forward-all setting from the idle display (enable/disable and specify the forward-to contact) as well as divert callers while the call is alerting. Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.

Directed Call Pick-Up

Calls to another phone can be picked up by dialing the extension of the other phone. This feature depends on support from a SIP server. With many SIP servers, directed call pick-up is implemented using a particular star code sequence. With some SIP servers, specific network signaling is used to implement this feature.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116. Determine the type of directed call pickup. • For more information, refer to Call Handling Configuration <call/> on page A-80. Determine the type of SIP header to include. • For more information, refer to Protocol <voIpProt/> on page A-7.

Group Call Pick-Up

Calls to another phone within a pre-defined group can be picked up without dialing the extension of the other phone. This feature depends on support from a SIP server. With many SIP servers, group call pick-up is implemented using a particular star code sequence. With some SIP servers, specific network signaling is used to implement this feature.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.

Call Park/Retrieve

An active call can be parked, and the parked call can be retrieved by another phone. This feature depends on support from a SIP server. With many SIP servers, this feature is implemented using a particular star code sequence. With some SIP servers, specific network signaling is used to implement this feature.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116. Determine the type of call park and retrieval string. • For more information, refer to Call Handling Configuration <call/> on page A-80.

Last Call Return

The phone allows server-based last call return. This feature depends on support from a SIP server. With many SIP servers, this feature is implemented using a particular star code sequence. With some SIP servers, specific network signaling is used to implement this feature.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116. Specify the string sent to the server for last-call-return. • For more information, refer to Call Handling Configuration <call/> on page A-80.

Setting Up Advanced Features

This section provides information for making configuration changes for the following advanced features:

- Configurable Feature Keys
- Multiple Line Keys per Registration
- Multiple Call Appearances
- Customizable Fonts and Indicators
- Instant Messaging
- Multilingual User Interface
- Downloadable Fonts
- Synthesized Call Progress Tones
- Browser and Microbrowser
- Real-Time Transport Protocol Ports
- Network Address Translation
- Corporate Directory
- Recording and Playback of Audio Calls
- Digital Picture Frame
- Enhanced Feature Keys
- Configurable Soft Keys
- LCD Power Saving

This section also provides information for making configuration changes for the following advanced call server features:

- Shared Call Appearances
- Bridged Line Appearance
- Busy Lamp Field
- Voice Mail Integration
- Multiple Registrations
- SIP-B Automatic Call Distribution
- Feature Synchronized Automatic Call Distribution
- Server Redundancy
- Presence
- Microsoft Live Communications Server 2005 Integration
- Access URL in SIP Message
- Static DNS Cache
- Display of Warnings from SIP Headers
- Quick Setup of SoundPoint IP / SoundStation IP / VVX Phones

Configurable Feature Keys

All key functions can be changed from the factory defaults. The scrolling timeout for specific keys can be configured.

Note Since there is no Redial key on the SoundPoint IP 32x/33x phone, the redial function cannot be remapped.

The rules for remapping of key functions are:

- The phone keys that have removable key caps can be mapped to the following:
  - Any function that is implemented as a removable key cap on any of the phones (Directories, Applications, Conference, Transfer, Redial, Menu, Messages, Do Not Disturb, Call Lists)
  - A speed-dial
  - An enhanced feature key operation
  - Null

- The phone keys without removable key caps cannot be remapped. These include:
  - Any keys on the dial pad
  - Volume control
  - Handsfree, Mute, Headset
  - Hold
  - Navigation Cluster

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration File: **sip.cfg**	Set the key scrolling timeout, key functions, and sub-pointers for each key (usually not necessary). • For more information, refer to Keys <key/> on page A-94.

For more information on the default feature key layouts, refer to Default Feature Key Layouts on page C-13.

Multiple Line Keys per Registration

More than one Line Key can be allocated to a single registration (phone number or line) on SoundPoint IP and Polycom VVX 1500 phones. The number of Line Keys allocated per registration is configurable.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **phone1.cfg**	Specify the number of line keys to assign per registration. • For more information, refer to Registration <reg/> on page A-134.
Local	Web Server (if enabled)	Specify the number of line keys to assign per registration. Navigate to http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	Specify the number of line keys to assign per registration using the **Line Configuration** menu. Either the Web Server or the provisioning server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the Line Configuration menu is used, it is assumed that all registrations use the same server.

Multiple Call Appearances

The phone supports multiple concurrent calls. The hold feature can be used to pause activity on one call and switch to another call. The number of concurrent calls per line key is configurable. Each registration can have more than one line key assigned to it (refer to the previous section, Multiple Line Keys per Registration).

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the default number of calls that can be active or on hold per line key. • For more information, refer to Call Handling Configuration <call/> on page A-80.
	Configuration file: **phone1.cfg**	Specify per-registration the number of calls that can be active or on hold per line key assigned to that registration. This will override the default value specified in **sip.cfg**. • For more information, refer to Registration <reg/> on page A-134.
Local	Web Server (if enabled)	Specify the default number of calls that can be active or on hold per line key and the number of calls per registration that can be active or on hold per line key assigned to that registration. Navigate to http://<phoneIPAddress>/appConf.htm#ls and http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	Specify per-registration the number of calls that can be active or on hold per line key assigned to that registration using the **Line Configuration** menu. Either the Web Server or the provisioning server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the Line Configuration menu is used, it is assumed that all registrations use the same server.

Customizable Fonts and Indicators

The phone's user interface can be customized by changing the fonts and graphic icons used on the display and the LED indicator patterns. Pre-existing fonts embedded in the software can be overwritten or new fonts can be downloaded. The bitmaps and bitmap animations used for graphic icons on the display can be changed and repositioned. LED flashing sequences and colors can be changed.

Note Customizable fonts and indicators are not supported on the Polycom VVX 1500.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration File: **sip.cfg**	Specify fonts to overwrite existing ones or specify new fonts. • For more information, refer to Fonts <font/> on page A-91. Specify which bitmaps to use. • For more information, refer to Bitmaps <bitmap/>on page A-100. Specify how to create animations and LED indicator patterns. • For more information, refer to Indicators <ind/> on page A-100.

Instant Messaging

The phone supports sending and receiving instant text messages. The user is alerted to incoming messages visually and audibly. The user can view the messages immediately or when it is convenient. For sending messages, the user can either select a message from a preset list of short messages or an alphanumeric text entry mode allows the typing of custom messages using the dial pad. Message sending can be initiated by replying to an incoming message or by initiating a new dialog. The destination for new dialog messages can be entered manually or selected from the contact directory, the preferred method.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.

Multilingual User Interface

The system administrator or the user can select the language. Support for major western European languages is included and additional languages can be easily added. Support for Asian languages (Chinese, Japanese, and Korean) is also included, but will display only on the higher resolution displays of the SoundPoint IP 450, 550, 560, 650, and 670, SoundStation IP 6000, and 7000, and Polycom VVX 1500. A WGL4 character set is displayed the SoundStation IP 7000. For more information, refer to http://www.microsoft.com/OpenType/otspec/WGL4E.HTM.

For basic character support and extended character support (available on SoundPoint IP 450, 550, 560, 650 and 670 and SoundStation IP platforms), refer to Multilingual <ml/> on page A-28. (Note that within a Unicode range, some characters may not be supported due to their infrequent usage.)

The SoundPoint IP and SoundStation IP user interface is available in the following languages by default: Simplified Chinese (if displayable), Danish, Dutch, English, French, German, Italian, Japanese (if displayable), Korean (if displayable), Norwegian, Polish, Brazilian Portuguese, Russian, Slovenian, International Spanish, and Swedish.

Note The multilingual feature relies on dictionary files resident on the provisioning server. The dictionary files are downloaded from the provisioning server whenever the language is changed or at boot time when a language other than the internal US English language has been configured. If the dictionary files are inaccessible, the language will revert to the internal language.

Note Currently, the multilingual feature is only available in the SIP application. The BootROM application is available in English only.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the boot-up language and the selection of language choices to be made available to the user. • For more information, refer to Multilingual <ml/> on page A-28. For instructions on adding new languages, refer to To add new languages to those included with the distribution: on page A-29.
Local	Local Phone User Interface	The user can select the preferred language under the Settings menu. The languages appears in the list in the language itself. For example, German appears in the list as "Deutsch" and Swedish appears as "Svenska". For administrator convenience, the ISO representation of each language is also included in the language selection menu. Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.

Downloadable Fonts

New fonts can be loaded onto the phone. For guidelines on downloading fonts, refer to Fonts <font/> on page A-91.

Note Downloadable fonts are not supported on the SoundStation IP 6000 and 7000 and the Polycom VVX 1500.

Synthesized Call Progress Tones

In order to emulate the familiar and efficient audible call progress feedback generated by the PSTN and traditional PBX equipment, call progress tones are synthesized during the life cycle of a call. These call progress tones are easily configurable for compatibility with worldwide telephony standards or local preferences.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the basic tone frequencies, levels, and basic repetitive cadences. • For more information, refer to Chord-Sets <chord/> on page A-36. Specify downloaded sampled audio files for advanced call progress tones. • For more information, refer to Sampled Audio for Sound Effects <saf/> on page A-37. Specify patterns. • For more information, refer to Patterns <pat/> on page A-39 and Call Progress Patterns on page A-40.

Browser and Microbrowser

The SoundPoint IP 430, 450, 550, 560, 650, and 670 phones, the SoundStation IP 6000, and 7000 phones, and the Polycom VVX 1500 phones (running releases before SIP 3.2.2) support an XHTML Microbrowser. This can be launched by pressing the **Applications** key or it can be accessed through the **Menu** key by selecting Applications.

Note On some older phones, the **Applications** key is labeled **Services**.

The Polycom VVX 1500 phones running SIP 3.2.2 or later support a full browser. This can be launched by pressing the **App** key or it can accessed through the **Menu** key by selecting **Applications**.

Note If Browser uses over 30MB of memory and either the amount of free memory on the phone is below 6MB or the real time is between 1am to 5am, then the browser will restart. Once the browser has restarted, the last displayed web page is restored.

Two instances of the Microbrowser or Browser may run concurrently:

- An instance with standard interactive user interface
- An instance that does not support user input, but appears in a window on the idle display

For more information, refer to the *Web Application Developer's Guide,* which can be found at http://www.polycom.com/voicedocumentation/.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the Application browser home page, a proxy to use, and size limits. • For more information, refer to Microbrowser <mb/> on page A-119. Specify the telephone notification and state polling events to be recorded and location of the push server. • For more information, refer to Applications <apps/> on page A-122.
Local	Web Server (if enabled)	Specify the Applications browser home page and proxy to use. Navigate to http://<phoneIPAddress>/coreConf.htm#mb Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.

Real-Time Transport Protocol Ports

The phone is compatible with RFC 1889 - RTP: A Transport Protocol for Real-Time Applications - and the updated RFCs 3550 and 3551. Consistent with RFC 1889, the phone treats all RTP streams as bi-directional from a control perspective and expects that both RTP end points will negotiate the respective destination IP addresses and ports. This allows real-time transport control protocol (RTCP) to operate correctly even with RTP media flowing in only a single direction, or not at all. It also allows greater security: packets from unauthorized sources can be rejected.

The phone can filter incoming RTP packets arriving on a particular port by IP address. Packets arriving from a non-negotiated IP address can be discarded.

The phone can also enforce symmetric port operation for RTP packets: packets arriving with the source port set to other than the negotiated remote sink port can be rejected.

The phone can also fix the destination transport port to a specified value regardless of the negotiated port. This can be useful for communicating through firewalls. When this is enabled, all RTP traffic will be sent to the specified port and will be expected to arrive on that port as well. Incoming packets are sorted by the source IP address and port, allowing multiple RTP streams to be multiplexed.

The RTP port range used by the phone can be specified. Since conferencing and multiple RTP streams are supported, several ports can be used concurrently. Consistent with RFC 1889, the next higher odd port is used to send and receive RTCP.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify whether to filter incoming RTP packets by IP address, whether to require symmetric port usage or whether to jam the destination port and specify the local RTP port range start. • For more information, refer to RTP <rtp/> on page A-73.
Local	Web Server (if enabled)	Specify whether to filter incoming RTP packets by IP address, whether to require symmetric port usage, whether to jam the destination port and specify the local RTP port range start. Navigate to: http://<phoneIPAddress>/netConf.htm#rt Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Network Address Translation

The phone can work with certain types of network address translation (NAT). The phone's signaling and RTP traffic use symmetric ports (the source port in transmitted packets is the same as the associated listening port used to receive packets) and the external IP address and ports used by the NAT on the phone's behalf can be configured on a per-phone basis.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the external NAT IP address and the ports to be used for signaling and RTP traffic. • For more information, refer to Network Address Translation <nat/> on page A-150.
Local	Web Server (if enabled)	Specify the external NAT IP address and the ports to be used for signaling and the RTP traffic. Navigate to: http://<phoneIPAddress>/netConf.htm#na Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.

Corporate Directory

Note This feature requires a license key for activation except on the SoundStation IP 7000 and Polycom VVX 1500. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The SoundPoint IP, SoundStation IP, and Polycom VVX phones can be configured to interface with a corporate directory server that supports the Lightweight Directory Access Protocol (LDAP) version 3. Currently the following LDAP servers are supported:

- Microsoft Active Directory 2003
- Sun ONE Directory Server 5.2 p6
- Open LDAP Directory Server 2.4.12
- Microsoft Active Directory Application Mode (ADAM) 1.0 SP1

Both corporate directories that support server-side sorting and those that do not are supported. In the latter case, the sorting is performed on the phone.

POLYCOM

Polycom recommends using corporate directories that have server-side sorting.

Polycom recommends that you consult your LDAP Administrator when making any configuration changes for this feature.

The corporate directory can be browsed or searched. Entries retrieved from the LDAP server can be saved to the local contact directory on the phone. Phone calls can be placed based on the phone number contained in the LDAP entry.

The corporate directory interface is read only, so that editing or deleting existing directory entries as well as adding new directory entries from the phone is not be possible. (There is no matching of first and last names in the corporate directory to incoming calls, caller identification display, and in the call lists.)

All attributes are considered to be Unicode text. Validity checking will be performed when a call is placed or the entry is saved to the local contact directory.

The corporate directory LDAP server status can be reviewed through the Status menu (**Status > CD Server Status**).

For detailed examples for all currently supported LDAP directories, refer to "Technical Bulletin 41137: Best Practices When Using Corporate Directory on SoundPoint IP / SoundStation IP / VVX Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the location of the corporate directory's LDAP server, the LDAP attributes, how often to refresh the local cache from the LDAP server, and other miscellaneous parameters. • For more information, refer to Corporate Directory <corp/> on page A-87.
Local	Local Phone User Interface	Enable or disable persistent viewing through the Settings menu (***Settings > Basic > Preferences > Corporate Directory > View Persistency***). Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

This section contains the following information:

- Corporate Directory LDAP Attributes
- Browsing the Corporate Directory

Corporate Directory LDAP Attributes

The entry attributes in the corporate directory are mapped through **sip.cfg** configuration file attributes to the LDAP attributes first_name, last_name, phone_number, and others so the SIP application knows how to use them for searching, dialing, or saving to the local contact directory. Multiple attributes of the same type are allowed.

Note The maximum of eight attributes can be configured in **sip.cfg** .

The configuration order dictates how the attributes are displayed and sorted. The first attribute is the primary sort index and the second attribute is the secondary sort index. The other attributes are not used in sorting.

To limit the amount of data displayed in the corporate directory, filtering of the entries can be configured for all attribute types. Filtering can be configured to be retained if the phone reboots.

For more information on LDAP attributes, refer to *RFC 4510 - Lightweight Directory Access Protocol (LDAP): Technical Specification Road Map*.

Browsing the Corporate Directory

The SoundPoint IP or SoundStation IP phone will establish a session with the corporate directory and download enough entries to fill its cache:

- when the corporate directory is first accessed

- when the phone boots up if the background synchronization parameter is enabled

The requested entries are based on the configured attributes (see previous section).

If the background synchronization parameter is enabled, a timer is initiated to permit a periodic download from the corporate directory.

Entries are sorted according to the order in which the first two attributes are configured (for example, last name, then first name).

The browse position within the corporate directory as well as the attribute filters are maintained for subsequent corporate directory access can be saved (if so configured).

Recording and Playback of Audio Calls

Note This feature requires a license key for activation except for the Polycom VVX 1500. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The SoundPoint IP 650 and 670 and the Polycom VVX 1500 phones can be configured to allow recording of audio calls on a supported USB device.

The filenames of the recorded **.wav** files will include a date/time stamp (for example, **20Apr2007_190012.wav** was created on April 20, 2007 at 19:00:12). An indication of the recording time remaining—the space available of the attached USB storage media—appears on the graphic display. The user can browse through all recorded files through the menu shown on the graphic display.

Note Notify your users that they may be required by federal, state, and/or local laws to notify some or all called parties when they are recording.

Playback of recorded files can occur on the phone as well as on other devices, such as a Windows® or Apple® based computer using an application like Windows Media Player® or iTunes®.

The user controls which calls are recorded and played back.

For a list of supported USB devices, refer to "Technical Bulletin 38084: Supported USB Devices for SoundPoint IP 650 and 670 and Polycom VVX 1500 Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.

Digital Picture Frame

Note This feature is only supported on the Polycom VVX 1500.

A slide show of multiple personal images stored on a USB flash drive can be displayed on the Polycom VVX 1500 phone during the idle mode. The supported formats include JPEG, BMP, and PNG. The maximum image size is 9999x9999. A maximum of 1000 images can be displayed and these must be stored in a directory of the USB flash drive that you create.

Note Although 9999x9999 images and progressive/multiscan JPEG images are supported, the maximum image size that can be downloaded is restricted by the available memory in the phone.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off and configure how it appears. • For more information, refer to Feature <feature/> on page A-116 • For more information, refer to User Preferences <up/> on page A-155.
	Configuration file: **phone1.cfg**	Configure how the feature appears. • For more information, refer to User Preferences <up/> on page A-155.

Note The digital picture frame can be accessed through the PicFrame:// URL.

Enhanced Feature Keys

Note The Enhanced Feature Key feature from SIP 3.0 is compatible with Enhanced Feature Key feature from SIP 3.0 . However, improvements have been made, and Polycom recommends that existing configuration files be reviewed and updated.

Customers replacing legacy telephony PBX or key system would like to get equivalent functionality from their new VoIP telephony system. The enhanced feature key capability is designed to allow system administrators to program the speed-dials and soft keys on their phones to interact with the phone user to implement commonly used functions such as "Call Park" in an intuitive fashion.

This capability applies to the SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670 desktop phones and Polycom VVX 1500 business media phones. The enhanced feature key functionality is implemented using Star Code sequences and SIP messaging.

The enhanced feature key macro language was designed to follow current configuration file standards and to be extensible. It is described in more detail in Enhanced Feature Key Definition Language.

The particular Star Code sequence and the associated prompts displayed on the SoundPoint IP phone for the enhanced feature are defined by macros. These macros are case sensitive.

The enhanced feature key capability can be used to provide a customized, interactive user interface by mapping functions from speed-dial keys, soft keys and re-mapped hard function keys.

This section provides detailed information on:

- Enhanced Feature Key Definition Language
- Macro Definition
- Configuration File Changes
- Useful Tips
- Examples

For more examples including sample configuration files, refer to "Technical Bulletin 42250: Using Enhanced Feature Keys and Configurable Soft Keys on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Enhanced Feature Key Definition Language

This section defines the additional fields to be entered into a configuration file for controlling the enhanced feature key behavior. The definition language follows the XML style notation. The following elements are part of the definition language:

- <efk/>
- <efklist/>
- <efkprompt/>
- <version/>
- Special Characters

<efk/>

This element indicates the start of enhanced feature key definition section. The `efk` element has the following format:

```
<efk> ... </efk>
```

<efklist/>

This element describes behavior of enhanced feature key.

The different blocks of the enhanced feature key definitions are uniquely identified by number following `efk.efklist` prefix (for example, `efk.efklist.1.<suffix>`).

Note In SIP 3.0, a maximum of 50 element groups is supported, however, the exact number is dependent on available RAM and processing speed. The disabled elements are included in the total count.

This element contains the following parameters:

Name	Interpretation
mname	This is the unique identifier that is used for the speed-dial configuration to reference the enhanced feature key entry. It cannot start with a digit. This parameter must have a value and it cannot be Null.
status	This parameter has the following values: • If set to 1, this key is enabled. • If set to 0 or Null, this key is disabled. If this parameter is omitted, the value 0 is used.

Name	Interpretation
label	This field defines the text string that will be used as a label on any user text entry screens during enhanced feature key operation. The value can be any string including the null string (in this case, no label appears). If this parameter is omitted, the Null string is used. ***Note**: If you exceed the phone physical layout text limits, the text will be shortened and "..." will be appended.*
type	The SIP method to be performed once the macro starts executing. This parameter has the following values: • If set to "invite ", the action required is performed using the SIP INVITE method. ***Note**: This parameter is included for backwards compatability only. Do not use if at all possible. If the action.string contains types, this parameter is ignored. If this parameter is omitted, the default is INVITE.*
action.string	The action string contains a macro definition of the action to be performed. For more information, refer to Macro Definition on page 4-42. This parameter must have a value and it cannot be Null.

<efkprompt/>

This element describes the behavior of the user prompts.

The different blocks are uniquely identified by number following `efk.efkprompt` prefix (for example, `efk.efkprompt.1.<suffix>`).

Note

In SIP 3.0, a maximum of four user prompts were supported. In SIP 3.0, a maximum of ten user prompts are supported.

This element contains the following parameters:

Name	Interpretation
status	This parameter has the following values: • If set to 1, this key is enabled. • If set to 0, this key is disabled. This parameter must have a value and it cannot be Null. ***Note**: If a macro attempts to use a prompt that is disabled or invalid, the macro execution fails.*

Name	Interpretation
label	This parameter sets the prompt text that will be presented to the user on the user prompt screen. The value can be any string including the null string (in this case, no label appears). If this parameter is omitted, the Null string is used. ***Note**: If you exceed the phone physical layout text limits, the text will be shortened and "..." will be appended.*
userfeedback	This parameter specifies the user input feedback method. It has the following values: • If set to “visible”, the text appears as clear text. • If set to “masked”, the text appears as “*” characters. For example, if a password is entered. If this parameter is omitted, the value “visible” is used. If this parameter has an invalid value (including Null), this prompt is invalid and all parameters depending on this prompt are invalid.
type	The type of characters entered by the user. This parameter has the following values: • If set to “numeric “, the characters are interpreted as numbers. • If set to “text”, the characters are interpreted as letters. If this parameter is omitted, the value “numeric” is used. If this parameter has an invalid value (including Null), this prompt is invalid and all parameters depending on this prompt are invalid. ***Note**: A mix of numeric and text is not supported.*

<version/>

This element contains the version of the enhanced feature key elements. The `version` element has the following format:

```
<version efk.version=”2”/>
```

If this parameter is omitted or has an invalid value (including Null), the enhanced feature key is disabled. This parameter is not required if there are no `efk.efklist` entries.

Note In SIP 3.0, “1” is the only supported version. In SIP 3.1 or later, “2” is the only supported version.

Special Characters

The following special characters are used to implement the enhanced feature key functionality:

- ! — The characters following it are a macro name.
- ' or ASCII (0x27) — This character delimits the commands within the macro.
- $ — This character delimits the parts of the macro string. This character must exist in pairs, where the delimits the characters to be expanded.
- ^ — This character indicates that the following characters represent the expanded macro (as in the action string).

Macro names and action strings cannot contain these characters. If they do, unpredictable results may occur.

Macro Definition

The `action.string` in the `efklist` element can be defined by either:

- Macro Action
- Prompt Macro Substitution
- Expanded Macros

Macro Action

The action string is executed in the order it appears. User input is collected before any action is taken.

The action string contains the following fields:

Name	Interpretation
$L<label>$	This is the label for the entire operation. The value can be any string including the null string (in this case, no label appears). This label will be used if no other operation label collection method worked (up to the point where this field is introduced). Make this the first entry in action string to be sure this label is used; otherwise another label may be used and this one ignored.
digits	The digits to be sent. The appearance of this this parameter depends on the action string.

Name	Interpretation
$C<command>$	This is the command. It can appear anywhere in the action string. Supported commands (or shortcuts) include: • hangup (hu) • hold (h) • waitconnect (wc) • pause <number of seconds> (p <num sec>) where the maximum value is 10
$T<type>$	The embedded action type. Multiple actions can be defined. Supported action types include: • invite • dtmf • refer ***Note**: Polycom recommends that you always define this field. If it is not defined, the supplied digits will be dialed using INVITE (if no active call) or DTMF (if an active call). The use of refer method is call server dependent and may require the addition of star codes.*
$M<macro>$	The embedded macro. The <macro> string must begin with a letter. If the macro name is not defined, the execution of the action string fails.
$P<prompt num>N<num digits>$	The user input prompt string. Refer to Prompt Macro Substitution on this page.
$S<speed dial index>$	The speed dial index. Only digits are valid. The action is found in the `contact` field of the local directory entry pointed to by the index.
$F<internal function>$	An internal function. For more information, refer to Internal Key Functions on page C-19.
URL	A URL. Only one per action string is supported.

Prompt Macro Substitution

The `action.string` in the `efklist` element can be defined by a macro substitution string, "PnNn" where:

- Pn is the prompt x as defined in the `efk.efkprompt.x`
- Nn is the number of digits or letters that the user can enter. The maximum number is 32. The user needs to press the **Enter** soft key to complete data entry.

Note If the maximum number of characters is greater than 32 or less than one, macro execution fails.

The macros provide a generic and easy to manage way to define the prompt to be displayed to the user, the maximum number of characters that the user can input, and action that the phone performs once all user input has been collected. The macros are case sensitive.

If a macro attempts to use a prompt that is disabled, the macro execution fails. A prompt is not required for every macro.

Expanded Macros

Expanded macros are prefixed with the "^" character and are inserted directly into the local directory `contact` field. For more information, refer to Local Contact Directory File Format on page 4-11.

Configuration File Changes

Note The configuration file changes and the enhanced feature key definitions can be included together in one configuration file.

A sample configuration for this feature—including the enhanced feature keys definitions shown in the following section, Examples— may be included with the SIP 3.1 release.

Create a new configuration file in the style of **sip.cfg** in order to make configuration changes. For more information on why to create another configuration file,refer to the "Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.
	Configuration file: **phone1.cfg**	Specify two calls per line key. • For more information, refer to Registration <reg/> on page A-134.
	XML file: **<Ethernet address>-directory.xml**	This file holds the macro names which correspond to the `mname` fields in the configuration file where the enhanced feature keys are defined. Macro names must be embedded into the contact (`cn`) fields with the "!" prefix. You can also add labels in the first name (`fn`) fields. For information on file format, refer to Local Contact Directory File Format on page 4-11.

Useful Tips

The following information should be noted:

- Activation of the enhanced feature key will fail if configured values are invalid except where noted in previous sections.
- All failures are logged at level 4 (minor).
- If two macros have the same name, the first one will be used and the subsequent ones will be ignored.
- "'!" and "^" macro prefixes cannot be mixed in the same macro line.
- A sequence of characters prefixed with "!" are parsed as a macro name. The exception is the speed dial reference, which starts with "!" and contains digits only.
- A sequence of characters prefixed with "^" is the action string.
- The sequence of characters accessed from speed dial keys must be prefixed by either "!" or "^" so it will be processed as an enhanced feature key. All macro references and action strings added to the local directory `contact` field must be prefixed by either "!" or "^".
- Action strings used in soft key definitions do not need to be prefixed by "^". However, the "!" prefix must be used if macros or speed dials are referenced.

  For more information, refer to Configurable Soft Keys on page 4-48.
- A sequence of macro names in the same macro is supported (for example, "!m1!m2").
- A sequence of speed dial references is supported (for example, "!1!2").
- A sequence of macro names and speed dial references is supported (for example, "!m1!2!m2").
- Macro names that appear in the local contact directory must follow the format "!<macro name>" , where <macro name> must match an `<elklist> mname` entry. The maximum macro length is 100 characters.
- A sequence of macros is supported, but cannot be mixed with other action types.
- Action strings that appear in the local contact directory must follow the format "^<action string>". Action strings can reference other macros or speed dial indexes. Protection against recursive macro calls exists (the enhanced feature keys fails once 50 macro substitutions is reached).

Examples

Configuration File Changes

You must make the the following changes to the `<feature/>` parameter that is defined in the **sip.cfg** configuration file:

```
<feature feature.18.name="enhanced-feature-keys"
feature.18.enabled="1"/>
```

Action String Example

The action string "`$Changup$*444*$P1N4$$Tinvite$$Cwaitconnect$$P2N3$$Cpause2$$Tdtmf$$Changup$`" is executed as follows:

1. The user is prompted for 4 digits. For example, "1234".
2. The user is prompted for 3 digits. For example, "567".
3. The user's active call is disconnected.
4. The string "*444*1234" is sent using the INVITE method.
5. Once connected, there is a 2 second pause, and then the string "567" is sent using DTMF dialing on the active call.
6. The active call is disconnected.

Speed Dial Example

Your organization voice mail system is accessible through 7700 and your voice mail password is 2154. You could use a speed dial key to access your voice mail if you entered "7700$Cpause3$2154" as the contact number.

Enhanced Feature Key XML Files

You must ensure that the following XML code exists for the definition of "Call Park":

```
...
<efklist
...
    efk.efklist.2.mname="callpark"
    efk.efklist.2.status="1"
    efk.efklist.2.label="Call Park"
    efk.efklist.2.use.idle="1"
    efk.efklist.2.use.active="1"
    efk.efklist.2.use.alerting="1"
    efk.efklist.2.use.dialtone="1"
    efk.efklist.2.use.proceeding="1"
    efk.efklist.2.use.setup="1"
    efk.efklist.2.type="invite"
    efk.efklist.2.action.string="*68*$P1N10$"
...
/>
<efkprompt
    efk.efkprompt.1.status="1"
```

```
    efk.efkprompt.1.label="Enter Number: "
    efk.efkprompt.1.userfeedback="visible"
    efk.efkprompt.1.type="numeric"
    efk.efkprompt.1.digitmatching="style1"
...
/>
...
```

Contact Directory Changes

You must make the following contact directory changes for the definition of "Call Park":

```
<directory>
    <item_list>
        <item>
            <fn>Call Park</fn>
            <ct>!callpark</ct>
            <sd>2</sd>
            <rt>4</rt>
            <ad>0</ad>
            <ar>0</ar>
            <bw>0</bw>
            <bb>0</bb>
        </item>
    </item_list>
</directory>
```

Note To avoid users accidently deleting the definitions in the contact directory, make the contact directory read only. For more information, refer to Local Directory <local/> on page A-86.

Using Call Park Key

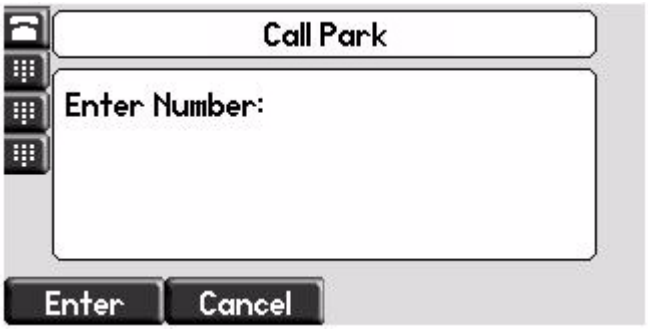
The following figure shows the second speed dial key mapped to Call Park (as well as others mapped to Park Return and Call Pickup).

To use the Call Park key during an active call:

1. When there is an active call on line 2233:

   a Select the **Call Park** soft key.

   The Call Park screen appears.



   b Enter the number where you want to park the active call, then select the **Enter** soft key.

   The Call Park * code (*68) is prepended to the number you entered and the call is parked at that location by the call server. The active call is put on hold during this operation.

Configurable Soft Keys

This feature enables phone system administrators to "program" certain frequently used functions onto the soft keys at the bottom of the phone display. This programming can be controlled based on call state. For example a Call Park function can be presented to the user when in an active call state.

If certain hard keys are missing, you may want to create a soft key. For example, if there is no **Do Not Disturb** key on a phone, you could create a **Do Not Disturb** soft key.

New soft keys can be mapped into:

- An Enhanced Feature Key sequence
- A speed dial contact directory entry

- Directly into the Enhanced Feature Key macro
- Directly into a URL
- A chained list of actions

It is possible to disable the display of specific standard keys—the soft keys that are displayed on SoundStation IP, SoundStation IP, and Polycom VVX 1500 phones—to make room for other soft keys that your organization wants displayed. To ensure that the usability of features is not compromised, the disabling of certain soft keys in certain circumstances may be restricted. When a standard soft key is disabled, the space where it was remains empty. The standard keys that can be disabled include:

- **New Call**
- **End Call**
- **Split**
- **Join**
- **Forward**
- **Directories** (or **Dir** as it is called on the SoundPoint IP 32x/33x)
- **Callers** (appears on the SoundPoint IP 32x/33x)
- **MyStatus** and **Buddies**
- **Hold**, **Transfer**, and **Conference**

Note The **Hold**, **Transfer,** and **Conference** are grouped together to avoid usability issues.

Custom soft keys can be added in the following call states:

- Idle—There are no active calls.
- Active—This state starts when a call is connected. It stops when the call stops or changes to another state (like hold or dial tone).
- Alerting (or ringing or incoming proceeding)—The phone is ringing.
- Dial tone—You can hear the dial tone.
- Proceeding (or outgoing proceeding)—This state starts when the phone sends a request to the network. It stops when the call is connected.
- Setup—This state starts when the user starts keying in a phone number. This state ends when the Proceeding state starts.
- Hold—The call is put on hold locally.

Custom soft keys can be configured to precede the standard soft keys that are still displayed. The order of the custom soft keys follows the configuration order. The standard soft keys are shifted to the right and any empty spaces are removed.

If the custom soft keys are configured to not precede the standard soft keys, then the standard soft keys do not move. The order of the custom soft keys—starting from the leftmost empty space—follows the empty spaces. Any extra custom soft keys that are left after all empty spaces are used are appended at the end.

Up to 10 soft keys can be configured. Any additional soft keys are ignored. If more soft keys are defined than fit on the graphic display at one time, a **More** soft key is displayed followed by the remainder of the soft keys that you have defined.

This capability applies to the SoundPoint IP 32x, 33x, 430, 450, 550, 560, 650, and 670, and Polycom VVX 1500 phones. This capability is linked to the Enhanced Feature Key feature (refer to Enhanced Feature Keys on page 4-38.)

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116. Specify the soft key label, in what states it should be displayed, and prompt for input if required. • For more information, refer to Soft Keys <softkey/> on page A-128.

Configuration File Examples

For more examples, refer to "Technical Bulletin 42250: Using Enhanced Feature Keys and Configurable Soft Keys on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

To disable the New Call soft key:

1. Update the **sip.cfg** configuration as follows:

```
softkey.feature.newcall = 0
```

2. Reboot the phone.

   The **New Call** soft key is not displayed and the space where it usually appears is empty.

To map a chained list of actions to a soft key:

1. Configure speed dial index 2 in contact directory with a regular phone number. For example, enter "2900" in the contact field.
2. Configure speed dial index 1 in contact directory with "!2" in contact field.
3. Update the **sip.cfg** configuration as follows:

```
softkey.1.label = ChainAct
softkey.1.action = $S1$$Tinvite$
softkey.1.use.idle = 1
```

4. Reboot the phone.

   If you press the soft key **ChainAct**, the phone dials number 2900.

To map the Do Not Disturb Enhanced Feature Key sequence to a soft key:

1. Update **sip.cfg** as follows:

```
softkey.1.label = DND
softkey.1.action = $FDoNotDisturb$
softkey.1.use.idle = 1
```

2. Reboot the phone.

   A **DND** soft key is displayed on the phone when it is in the idle state.

   When the **DND** soft key is pressed, the Do Not Disturb icon is displayed.

To map a Send to Voice Mail Enhanced Feature Key sequence to a soft key:

Note The exact star code to transfer the active call to Voice Mail depends on your call server.

1. Update **sip.cfg** as follows:

```
softkey.2.label = ToVMail
softkey.2.action = ^*55$P1N10$$Tinvite$
softkey.2.use.alerting = 1
```

2. Reboot the phone.

   When another party calls, the **ToVMail** soft key is displayed. When the user presses **ToVMail** soft key, the other party is transferred to voice mail.

LCD Power Saving

Note This feature is only supported on the Polycom VVX 1500.

This feature applies during configured non-working hours and when the phone is idle. Working hours are defined in the configuration files and users can change the default values through the phone's menu to accommodate their individual schedules. The Polycom VVX 1500 phone enters power-saving mode after it has been idle for a certain period of time and its camera doesn't detect motion. The phone's ability to detect the users' presence is biased for easy detection during office hours and for difficult detection during off hours.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off and configure how it works. • For more information, refer to LCD Power Saving <powerSaving/> on page A-132.

Shared Call Appearances

Calls and lines on multiple phones can be logically related to each other. A call that is active on one phone will be presented visually to phones that share that call appearance. Mutual exclusion features emulate traditional PBX or key system privacy for shared calls. Incoming calls can be presented to multiple phones simultaneously. Users at the different locations have the ability to interrupt remote active calls.

This feature is dependent on support from a SIP server that binds the appearances together logically and looks after the necessary state notifications and performs an access control function. For more information, refer to Shared Call Appearance Signaling on page B-10.

Note Shared call appearances and bridged line appearances are two different signaling methods of implementing a feature whereby more than one phone can share the same line or registration. These implementations are dependent on the SIP server. The methods are mutually exclusive and you should confirm with the call server vendor which (if any) method is supported.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify whether diversion should be disabled on shared lines. • For more information, refer to Shared Calls <shared/> on page A-84. Specify line-seize subscription period. • For more information, refer to Server <server/> on page A-8. Specify standard or non-standard behavior for processing line-seize subscription for mutual exclusion feature. • For more information, refer to Special Events <specialEvent/> on page A-20.
	Configuration file: **phone1.cfg**	Specify per-registration line type (private or shared), barge-in capabilities, and line-seize subscription period if using per-registration servers. A shared line will subscribe to a server providing call state information. • For more information, refer to Registration <reg/> on page A-134. Specify per-registration whether diversion should be disabled on shared lines. • For more information, refer to Diversion <divert/> on page A-142.
Local	Web Server (if enabled)	Specify line-seize subscription period. Navigate to http://<phoneIPAddress>/appConf.htm#se Specify standard or non-standard behavior for processing line-seize subscription for mutual exclusion feature. Navigate to http://<phoneIPAddress>/appConf.htm#ls Specify per-registration line type (private or shared) and line-seize subscription period if using per-registration servers, and whether diversion should be disabled on shared lines. Navigate to http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	Specify per-registration line type (private or shared) using the **Line Configuration** menu. Either the Web Server or the provisioning server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the Line Configuration menu is used, it is assumed that all registrations use the same server.

Bridged Line Appearance

Calls and lines on multiple phones can be logically related to each other. A call that is active on one phone will be presented visually to phones that share that line. Incoming calls can be presented to multiple phones simultaneously. This feature is dependent on support from a SIP server that binds the appearances together logically and looks after the necessary state notifications and performs an access control function. For more information, refer to Bridged Line Appearance Signaling on page B-10.

Note

Bridged line appearances and shared call appearances are two different signaling methods of implementing a feature whereby more than one phone can share the same line or registration. These implementations are dependent on the SIP server. The methods are mutually exclusive and you should confirm with the call server vendor which (if any) method is supported.

In the configuration files, bridged lines are configured by "shared line" parameters.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify whether diversion should be disabled on shared lines. • For more information, refer to Call Handling Configuration <call/> on page A-80.
	Configuration file: **phone1.cfg**	Specify per-registration line type (private or shared) and the shared line third party name. A shared line will subscribe to a server providing call state information. • For more information, refer to Registration <reg/> on page A-134. Specify per-registration whether diversion should be disabled on shared lines. • For more information, refer to Diversion <divert/> on page A-142.
Local	Web Server (if enabled)	Specify per-registration line type (private or shared) and third party name, and whether diversion should be disabled on shared lines. Navigate to http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection and the **<Ethernet address>-phone.cfg** is removed from the provisioning server.
	Local Phone User Interface	Specify per-registration line type (private or shared) and the shared line third party name using the **Line Configuration** menu. Either the Web Server or the provisioning server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the Line Configuration menu is used, it is assumed that all registrations use the same server.

Busy Lamp Field

Note This feature is available only on SoundPoint IP 430, 450, 550, 560, 600, 601, 650, and 670 phones. Other SoundPoint IP phone models may be monitored, but cannot be configured to monitor other phones.

Some aspects of this feature are dependent on the SIP server signaling. You should consult your SIP server partner or Polycom Channel partner for information as needed.

The Busy Lamp Field (BLF) feature enhances support for a phone-based attendant console. It allows monitoring the hook status and remote party information of users through the busy lamp fields and displays on an attendant console phone.

In the SIP 3.1 release, the BLF feature was updated for the following:

- Visual and audible indication when a remote line is in an alerting state
- Display of the caller ID of calls on remotely monitored lines
- Single button "Directed Call Pickup" on a remote line

In the SIP 3.2 release, the BLF feature is updated for the following:

- Configurable list of remote parties to a maximum of 47 with configurable line key labels
- The introduction of configurable default key press actions
- The ability to remove spontaneous call appearances from incoming calls on monitored lines

Note The SIP 3.2 update to the BLF feature is not supported on the SoundPoint IP 430.

For more information, refer to "Quick Tip 37381: Enhanced BLF"at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Polycom recommends that the BLF not be used in conjunction with the Microsoft Live Communications Server 2005 feature. For more information, refer to Microsoft Live Communications Server 2005 Integration on page 4-64.

Note Use this feature with TCPpreferred transport (refer to Server <server/> on page A-8). You can also use UDP transport on SoundPoint IP 650 and 670 phones.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **phone1.cfg**	Specify the list SIP URI and index of the registration which will be used to send a SUBSCRIBE to the list SIP URI specified in `attendant.uri`. • For more information, refer to Attendant <attendant/> on page A-151. Specify the list of monitored resources. • For more information, refer to Resource List <resourceList/> on page A-152 and Behaviors <behaviors/> on page A-153.

Voice Mail Integration

The phone is compatible with voice mail servers. The subscribe contact and callback mode can be configured per user/registration on the phone. The phone can be configured with a SIP URL to be called automatically by the phone when the user elects to retrieve messages. Voice mail access can be configured to be through a single key press (for example, the **Messages** key on the SoundPoint IP 430, 450, 550, 560, 650, and 670, and the **MSG** key on the Polycom VVX 1500). A message-waiting signal from a voice mail server triggers the message-waiting indicator to flash and the call waiting audio tone is played through the active audio path.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	For one-touch voice mail access, enable the "one-touch voice mail" user preference. • For more information, refer to User Preferences <up/> on page A-31.
	Configuration file: **phone1.cfg**	For one-touch voice mail access, bypass instant messages to remove the step of selecting between instant messages and voice mail after pressing the **Messages** key on the SoundPoint IP 430, 450, 550, 560, 650, and 670 and the **MSG** key on the Polycom VVX 1500 (Instant messages are still accessible from the Main Menu). On a per-registration basis, specify a subscribe contact for solicited NOTIFY applications, a callback mode (self call-back or another contact), and the contact to call when the user accesses voice mail. • For more information, refer to Messaging <msg/> on page A-149.

Local	Web Server (if enabled)	For one-touch voice mail access, enable the “one-touch voice mail” user preference and bypass instant messages to remove the step of selecting between instant messages and voice mail after pressing the **Messages** key on the SoundPoint IP 430, 450, 550, 560, 650, and 670 and the **MSG** key on the Polycom VVX 1500 (Instant messages are still accessible from the Main Menu). Navigate to http://<phoneIPAddress>/coreConf.htm#us On a per-registration basis, specify a subscribe contact for solicited NOTIFY applications, a callback mode (self call-back or another contact) to call when the user accesses voice mail. Navigate to http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Multiple Registrations

The SoundPoint IP 32x/33x and 430 support a maximum of two registrations, the SoundPoint IP 450 supports three registrations, the SoundPoint IP 550 and 560 support four, and the SoundPoint IP 650 and 670 and the Polycom VVX 1500 support 6. Up to three SoundPoint IP Expansion Modules can be added to a single host SoundPoint IP 650 and 670 phone increasing the total number of registrations to 34. The SoundStation IP 6000 and 7000 supports a single registration.

Each registration can be mapped to one or more line keys (a line key can be used for only one registration). The user can select which registration to use for outgoing calls or which to use when initiating new instant message dialogs.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the local SIP signaling port and an array of SIP servers to register to. For each server specify the registration period and the signaling failure behavior. • For more information, refer to Server <server/> on page A-8 and Server <server/> on page A-8.
	Configuration file: **phone1.cfg**	For up to maximum number of registrations, specify a display name, a SIP address, an optional display label, an authentication user ID and password, the number of line keys to use, and an optional array of registration servers. The authentication user ID and password are optional and for security reasons can be omitted from the configuration files. The local flash parameters will be used instead. The optional array of servers and their associated parameters will override the servers specified in **sip.cfg** if non-Null. • For more information, refer to Registration <reg/> on page A-134.

Local	Web Server (if enabled)	Specify the local SIP signaling port and an array of SIP servers to register to. Navigate to http://<phoneIPAddress>/appConf.htm#se For up to six registrations (depending on the phone model, in this case the maximum is six even for the IP 650 and 670), specify a display name, a SIP address, an optional display label, an authentication user ID and password, the number of line keys to use, and an optional array of registration servers. The authentication user ID and password are optional and for security reasons can be omitted from the configuration files. The local flash parameters will be used instead. The optional array of servers will override the servers specified in **sip.cfg** in non-Null. This will also override the servers on the appConf.htm web page. Navigate to http://<phoneIPAddress>/reg.htm Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.
Local (continued)	Local Phone User Interface	Use the **Call Server Configuration** and **Line Configuration** menu to specify the local SIP signaling port, a default SIP server to register to and registration information for up to twelve registrations (depending on the phone model). These configuration menus contains a sub-set of all the parameters available in the configuration files. Either the Web Server or the provisioning server configuration files or the local phone user interface should be used to configure registrations, not a mixture of these options. When the Line Configuration menu is used, it is assumed that all registrations use the same server. Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection. For more information, refer to Server <server/> on page A-8, Server <server/> on page A-8, and Registration <reg/> on page A-134.

SIP-B Automatic Call Distribution

Note For more information on SIP-B and supported features on SoundPoint IP, SoundStation IP, and Polycom VVX phones, contact Polycom Product Management.

The phone allows Automatic Call Distribution (ACD) login and logout. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.
	Configuration file: **phone1.cfg**	Enable this feature per registration. • For more information, refer to Registration <reg/> on page A-134.

The phone also supports ACD agent availability. This feature depends on support from a SIP server.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.
	Configuration file: **phone1.cfg**	Enable this feature per registration. • For more information, refer to Registration <reg/> on page A-134.

Feature Synchronized Automatic Call Distribution

As of SIP 3.1.2, you can use your SoundPoint IP phones in a call center agent/supervisor role on a supported call server.

When this feature is enabled, the phone will indicate the ACD Call Center Agent state as directed by the call server. The call center agent is provided with an entry method to initiate Sign In/Sign Out and other ACD states through soft keys, however, the phone state will only change once the server has acknowledged that the phone can move into that new state—in this way, the ACD state is maintained in synchronization with the call server and any ACD computer-based soft-clients. The SIP signaling used for this implementation is described in the Device Key Synchronization Requirements Document; Release R14 sp2; Document version 1.6. Contact Polycom Product Management for more information.

The Feature Synchronized ACD feature is supported on SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670, SoundStation IP 6000 and 7000, and Polycom VVX 1500 phones.

Note The Feature Synchronized ACD feature is distinct from the existing SIP-B Automatic Call Distribution functionality, which was added in SIP 1.6 .

For details on how to configure SoundPoint IP, SoundStation IP, and VVX phones for Feature Synchronized ACD, refer to “Technical Bulletin 34787: Using Feature Synchronized Automatic Call Distribution with SoundPoint IP and Polycom VVX 1500 Phones” at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Enable or disable Feature Synchronized ACD. • For more information, refer to SIP <SIP/> on page A-11. Turn this feature on or off. • For more information, refer to Feature <feature/> on page A-116.
	Configuration file: **phone1.cfg**	Set the registration to be used for Feature Synchronized ACD and the users sign-in state. • For more information, refer to Automatic Call Distribution <acd/>on page A-156.

Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of phone service for events where the call server needs to be taken offline for maintenance, the server fails, or the connection between the phone and the server fails.

Two types of redundancy are possible:

- **Fail-over**: In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using DNS mechanisms or "IP Address Moving" from the primary to the back-up server.
- **Fallback**: In this mode, a second less featured call server (router or gateway device) with SIP capability takes over call control to provide basic calling capability, but without some of the richer features offered by the primary call server (for example, shared lines, presence, and Message Waiting Indicator). Polycom phones support configuration of multiple servers per SIP registration for this purpose.

In some cases, a combination of the two may be deployed.

Note Your SIP server provider should be consulted for recommended methods of configuring phones and servers for fail-over configuration.

Warning Prior to SIP 2.1, the `reg.x.server.y` parameters (refer to Registration <reg/> on page A-134) could be used for fail-over configuration. The older behavior is no longer supported. Customers that are using the `reg.x.server.y`. configuration parameters where y>=2 should take care to ensure that their current deployments are not adversely affected. For example the phone will only support advanced SIP features such as shared lines, missed calls, presence with the primary server (y=1).

For more information, refer to "Technical Bulletin 5844: SIP Server Fallback Enhancements on SoundPoint IP Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify global primary and fallback server configuration parameters. • For more information, refer to Protocol <voIpProt/> on page A-7.
	Configuration file: **phone1.cfg**	Specify per registration primary and fallback server configuration parameters values that override those in **sip.cfg**. • For more information, refer to Registration <reg/> on page A-134.

DNS SIP Server Name Resolution

If a DNS name is given for a proxy/registrar address, the IP address(es) associated with that name will be discovered as specified in RFC 3263. If a port is given, the only lookup will be an A record. If no port is given, NAPTR and SRV records will be tried, before falling back on A records if NAPTR and SRV records return no results. If no port is given, and none is found through DNS, 5060 will be used.

Refer to http://www.ietf.org/rfc/rfc3263.txt for an example.

Note Failure to resolve a DNS name is treated as signaling failure that will cause a failover.

Behavior When the Primary Server Connection Fails

For Outgoing Calls (INVITE Fallback)

When the user initiates a call, the phone will go through the following steps to connect the call:

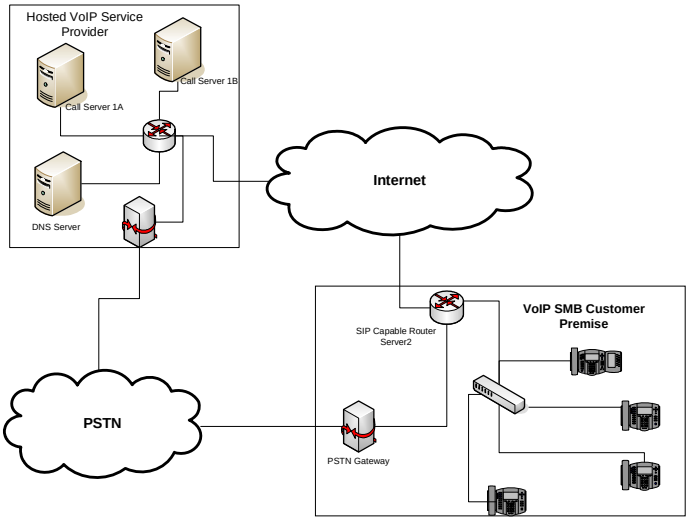
1. Try to make the call using the working server.
2. If the working server does not respond correctly to the INVITE, then try and make a call using the next server in the list (even if there is no current registration with these servers). This could be the case if the Internet connection has gone down, but the registration to the working server has not yet expired.
3. If the second server is also unavailable, the phone will try all possible servers (even those not currently registered) until it either succeeds in making a call or exhausts the list at which point the call will fail.

At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used as described below:

- If TCP is used, then the signaling fails if the connection fails or the Send fails.
- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted through all servers in the list and this is the last server, then the signaling fails after the complete UDP timeout defined in RFC 3261. If it is not the last server in the list, the maximum number of retries using the configurable retry timeout is used. For more information, refer to Server <server/> on page A-8 and Registration <reg/> on page A-134.

Warning

If DNS is used to resolve the address for Servers, the DNS server is unavailable, and the TTL for the DNS records has expired, the phone will attempt to contact the DNS server to resolve the address of all servers in its list *before* initiating a call. These attempts will timeout, but the timeout mechanism can cause long delays (for example, two minutes) before the phone call proceeds "using the working server". To mitigate this issue, long TTLs should be used. It is strongly recommended that an on-site DNS server is deployed as part of the redundancy solution.



Phone Configuration

The phones at the customer site are configured as follows:

- Server 1 (the primary server) will be configured with the address of the service provider call server. The IP address of the server(s) to be used will be provided by the DNS server. For example:

  ```
  reg.1.server.1.address="voipserver.serviceprovider.com"
  ```

- Server 2 (the fallback server) will be configured to the address of the router/gateway that provides the fallback telephony support and is on-site. For example:

  ```
  reg.1.server.2.address=172.23.0.1
  ```

Note It is possible to configure the phone for more than two servers per registration, but you need to exercise caution when doing this to ensure that the phone and network load generated by registration refresh of multiple registrations do not become excessive. This would be of particularly concern if a phone had multiple registrations with multiple servers per registration and it is expected that some of these servers will be unavailable.

Phone Operation for Registration

After the phone has booted up, it will register to all the servers that are configured.

Server 1 is the primary server and supports greater SIP functionality than any of servers. For example, SUBSCRIBE/NOTIFY services (used for features such as shared lines, presence, and BLF) will only be established with Server 1.

Upon registration timer expiry of each server registration, the phone will attempt to re-register. If this is unsuccessful, normal SIP re-registration behavior (typically at intervals of 30 to 60 seconds) will proceed and continue until the registration is successful (for example, when the Internet link is once again operational). While the primary server registration is unavailable, the next highest priority server in the list will serve as the working server. As soon as the primary server registration succeeds, it will return to being the working server.

Note If `reg.x.server.y.register` is set to 0, then phone will not register to that server. However, the INVITE will fail over to that server if all higher priority servers are down.

Recommended Practices for Fallback Deployments

In situations where server redundancy for fall-back purpose is used, the following measures should be taken to optimize the effectiveness of the solution:

1. Deploy an on-site DNS server to avoid long call initiation delays that can result if the DNS server records expire.
2. Do not use OutBoundProxy configurations on the phone if the OutBoundProxy could be unreachable when the fallback occurs. SoundPoint IP phones can only be configured with one OutBoundProxy per registration and all traffic for that registration will be routed through this proxy for all servers attached to that registration. If Server 2 is not accessible through the configured proxy, call signaling with Server 2 will fail.
3. Avoid using too many servers as part of the redundancy configuration as each registration will generate more traffic.
4. Educate users as to the features that will not be available when in “fallback” operating mode.

Presence

The Presence feature allows the phone to monitor the status of other users/devices and allows other users to monitor it. The status of monitored users is displayed visually and is updated in real time in the Buddies display screen or, for speed dial entries, on the phone's idle display. Users can block others from monitoring their phones and are notified when a change in monitored status occurs. Phone status changes are broadcast automatically to monitoring phones when the user engages in calls or invokes do-not-disturb. The user can also manually specify a state to convey, overriding, and perhaps masking, the automatic behavior.

Note Notification when a change in monitored status occurs will be available in a subsequent release.

The presence feature works differently when Microsoft Live Communications Server 2005 is used as the call server. For more information, refer to the next section, Microsoft Live Communications Server 2005 Integration.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	XML file: **<Ethernet address>-directory. xml**	The <bw>0</bw> (buddy watching) and <bb>0</bb> (buddy blocking) elements in the ***<Ethernet address>*-directory.xml** file dictate the Presence aspects of directory entries. • For more information, refer to Local Contact Directory on page 4-10.
Local	Local Phone User Interface	The user can edit the directory contents. The *Watch Buddy* and *Block Buddy* fields control the buddy behavior of contacts. Changes will be stored in the phone's flash file system and backed up to the provisioning server copy of ***<Ethernet address>*-directory.xml** if this is configured. When the phone boots, the provisioning server copy of the directory, if present, will overwrite the local copy.

Microsoft Live Communications Server 2005 Integration

SoundPoint IP, SoundStation IP, and VVX phones can used with Microsoft Live Communications Server 2005 and Microsoft Office Communicator to help improve business efficiencies and increase productivity and to share ideas and information immediately with business contacts.

For instructions on changing the configuration files, refer to Configuration File Examples on page 4-65.

Note Any contacts added through the SoundPoint IP, SoundStation IP, and VVX phone's buddy list will appear as a contact in Microsoft Office Communicator and Windows Messenger.

Polycom recommends that the BLF not be used in conjunction with the Microsoft Live Communications Server 2005 feature. For more information, refer to Busy Lamp Field on page 4-55.

Configuration changes can performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify that support for Microsoft Live Communications Server 2005 is enabled. • For more information, refer to SIP <SIP/> on page A-11. Specify the line/registration number used to send SUBSCRIBE for presence. • For more information, refer to Presence <pres/> on page A-91. Turn the presence and messaging features on or off. • For more information, refer to Feature <feature/> on page A-116.
	Configuration file: **phone1.cfg**	Specify the number of line keys to assign per registration. • For more information, refer to Registration <reg/> on page A-134. Specify the line/registration number which has roaming buddies support enabled. • For more information, refer to Roaming Buddies <roaming_buddies/> on page A-154. Specify the line/registration number which has roaming privacy support enabled. • For more information, refer to Roaming Privacy <roaming_privacy/> on page A-154.

Configuration File Examples

SoundPoint IP, SoundStation IP, and VVX phones can be deployed in two basic methods. In the first method, Microsoft Live Communications Server 2005 serves as the call server and the phones have a single registration. In the second method, the phone has a primary registration to call server—that is not Microsoft Live Communications Server (LCS)—and a secondary registration to LCS for presence purposes.

To set up a single registration with Microsoft Live Communications Server 2005 as the call server:

1. Modify the **sip.cfg** configuration file as follows:
   a. Open **sip.cfg** in an XML editor.
   b. Locate the feature parameter.
   c. For the `feature.1.name = presence` attribute, set `feature.1.enabled` to 1.

d For the `feature.2.name = messaging` attribute, set `feature.2.enabled` to 1.

e Locate the `voIpProt` parameter.

Set the `voIpProt.server.x.transport` attribute to TCPpreferred or TLS.

Your selection depends on the LCS configuration.

Note The TLS protocol is not supported on SoundPoint IP 300 and 500 phones.

f Set the `voIpProt.server.x.address` to the LCS address.

For example, `voIpProt.server.1.address = "lcs2005.local"`

g Set the `voIpProt.SIP.lcs` attribute to 1.

h (Optional) If SIP forking is desired, set `voIpProt.SIP.ms-forking` attribute to 1.

Refer to SIP <SIP/> on page A-11.

i Save the modified **sip.cfg** configuration file.

2. Modify the **phone1.cfg** configuration file as follows:

a Open **phone1.cfg** in an XML editor.

b Locate the registration parameter.

c Set the `reg.1.address` to the LCS address.

For example, `reg.1.address = "7778"`

d Set the `reg.1.server.y.address` to the LCS server name.

e (Optional) Set the `reg.1.server.y.transport` attribute to TCPpreferred or TLS.

Your selection depends on the LCS configuration.

f Set `reg.1.auth.userId` to the phone's LCS username.

For example, `reg.1.auth.userId = "jbloggs"`

g Set `reg.1.auth.password` to the LCS password.

For example, `reg.1.auth.password = "Password2"`

h Locate the `roaming_buddies` attribute.

i Set the `roaming_buddies.reg` element to 1.

Refer to Roaming Buddies <roaming_buddies/> on page A-154.

j Locate the `roaming_privacy` attribute.

k Set the `roaming_privacy.reg` element to 1.

Refer to Roaming Privacy <roaming_privacy/> on page A-154.

l Save the modified **phone1.cfg** configuration file.

To set up a dual registration with Microsoft Live Communications Server 2005 as the presence server:

1. (Optional) Modify the **sip.cfg** configuration file as follows:
   a Open **sip.cfg** in an XML editor.
   b Locate the feature parameter.
   c For the `feature.1.name = presence` attribute, set `feature.1.enabled` to 1.
   d For the `feature.2.name = messaging` attribute, set `feature.2.enabled` to 1.
   e Locate the `voIpProt` parameter.
   f If SIP forking is desired, set `voIpProt.SIP.ms-forking` attribute to 1.

   Refer to SIP <SIP/> on page A-11.
   g Save the modified **sip.cfg** configuration file.
2. Modify the **phone1.cfg** configuration file as follows:
   a Open **phone1.cfg** in an XML editor.
   b Locate the registration parameter.
   c Select a registration to be used for the Microsoft Live Communications Server 2005.

   Typically, this would be 2.
   d Set the `reg.x.address` to the LCS address.

   For example, `reg.2.address = "7778"`
   e Set the `reg.x.server.y.address` to the LCS server name.
   f (Optional) Set the `reg.2.server.y.transport` attribute to TCPpreferred or TLS.

   Your selection depends on the LCS configuration.
   g Set `reg.x.auth.userId` to the phone's LCS username.

   For example, `reg.2.auth.userId = "jbloggs"`
   h Set `reg.x.auth.password` to the LCS password.

   For example, `reg.2.auth.password = "Password2"`
   i Locate the `roaming_buddies` attribute.

j Set the `roaming_buddies.reg` element to the number corresponding to the LCS registration.

For example, `roaming_buddies.reg = 2`

Refer to Roaming Buddies <roaming_buddies/> on page A-154.

k Locate the `roaming_privacy` attribute.

l Set the `roaming_privacy.reg` element to the number corresponding to the LCS registration.

For example, `roaming_privacy.reg = 2`

Refer to Roaming Privacy <roaming_privacy/> on page A-154.

m Save the modified **phone1.cfg** configuration file.

Access URL in SIP Message

Introduced in SIP 2.2, this feature that allows information contained in incoming SIP signaling to refer to XHTML web content that can be rendered by the SoundPoint IP and SoundStation IP phone's Microbrowser and the Polycom VVX 1500 phone's Browser.

Supporting this feature allows use of the phone's display to provide information before someone takes a call and while they are on a call (for example, a SIP re-INVITE). The information accessible at the URL can be anything that you want to have displayed.

Configuration changes can performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Turn this feature on or off. • For more information, refer to Microbrowser <mb/> on page A-119.

This section provides detailed information on:

- Web Content Examples
- User Interface
- Signaling Changes

Web Content Examples

This feature can be used in the following circumstances:

- Call Center—Customer information

  The URL provided allows the phone to access information about a customer and display it before the agent takes the call.

- Call Center—Scripts for different call center groups

  The phone can access a script of questions for an agent to ask a caller when a call comes in. The script can be different for each agent group.

- Restaurant menu on a hotel phone

  A guest dials a number for the restaurant and a voice indicates that the menu is now available for viewing on the phone.

User Interface

There are three user interface aspects to this feature:

- Web content status indication
- Web content retrieval (spontaneous and on-demand)
- Settings menu item to control active versus passive behavior

Web Content Status Indication

When valid web content (validity is determined through a SIP header parameter) is available for a SIP call, it is indicated by an icon that appears after the call appearance status text, regardless of the call state. In the examples shown below, a lightning bolt symbol is used to indicate that web content is available for the displayed call appearance and the user is encouraged to press the **Select** key to retrieve and display the content through the Microbrowser.

SoundPoint IP 330 Graphic Display

SoundPoint IP 550 Graphic Display

Web Content Retrieval

Web content is retrieved either spontaneously (active mode) or at the request of the user (passive mode).

- **Active Mode.** Two methods can be used to achieve spontaneous web content retrieval: static configuration parameters or parameters received as part of the SIP signaling. If parameters received in the SIP signaling conflict with the static configuration, the parameters in the SIP signaling will take precedence.

  If the phone is configured to spontaneously retrieve web content, the phone will launch the interactive Microbrowser and have it fetch the appropriate URL upon arrival of the appropriate SIP signaling, subject to some conditions described below.

  Since new web content URLs can be received at any time—as the first URL for a call or a replacement URL—rules are needed to match displayed web content with automatic phone behavior, which are valid actions from within the Microbrowser context.

  Spontaneous web content will only be retrieved and displayed for a call if that call occupies, or will occupy, the UI focus at the time of the event.

- **Passive Mode.** Web content can also be retrieved when the user chooses to do so. The fact that web content is available for viewing is shown through the call appearance-based web content icon described in Web Content Status Indication on page 4-69. The Select key can be used to fetch the associated web content for the call that is in focus. If the web content has expired, the icon will be removed and the Select key will perform no function.

  Passive mode is recommended for applications where the Microbrowser is used for other applications. In the SIP 2.2 feature, interactive microbrowser sessions will be interrupted by the arrival of active-mode web content URLs, which may cause annoyance, although the Back navigation function will work in this context.

Settings Menu

If enabled, a new SIP web content entry is added to the **Setting > Basic > Preferences** menu to allow the user to change the current content retrieval mode. Two options are provided: passive mode and active mode.

Signaling Changes

A new SIP header must be used to report web content associated with SIP phone calls (the SSAWC header follow the BNF for the standard SIP header Alert-Info):

```
Alert-Info = "Alert-Info" HCOLON alert-param *(COMMA alert-param)
alert-param = LAQUOT absoluteURI RAQUOT *( SEMI generic-param )
```

The web content must be located with an absolute URI, which begins with the scheme identifier. Currently only the HTTP scheme is supported.

So an example header might look like:

```
Access-URL: <http://server.polycom.com/content23456.xhtml>
```

This header may be placed in SIP requests and responses, as appropriate so long as the messages are part of an INVITE-initiated dialog and the phone can associate them with an existing phone call.

This feature also requires the definition of two optional parameters:

- An *expires* parameter is defined to indicate the lifespan of the URL itself, or, assuming that the URL is permanent, the time span for which the content is expected to have relevance to the call with which it is associated. If the parameter is absent or invalid, this will be interpreted to mean that the content or the URL itself will be persistent in nature. A value, if it is present, will indicate the lifespan of the content in seconds (zero has special significance—see example below). When the lifespan expires, the phone will remove both the indication of the URL and the ability of the user to retrieve it.

  For example:

  ```
  Access-URL:
  <http://server.polycom.com/content23456.xhtml>;expires=60
  ```

  If the server wishes to invalidate a previous URL, it can send a new header (through UPDATE) with expires=0. The *expires* parameter is ignored when determining whether to spontaneously retrieve the web content unless expires=0.

- A *mode* parameter is defined to indicate whether the web content should be displayed spontaneously or retrieved on-demand. Two values are allowed: active and passive. If the parameter is absent or invalid, this will be interpreted the same as passive, meaning that the web content will be retrievable on-demand but will not be spontaneously displayed. If the value is set to active, the web content will be spontaneously displayed, subject to the rules discussed under **Active Mode** in Web Content Retrieval on page 4-70.

  For example:

  ```
  Access-URL:
  <http://server.polycom.com/content23456.xhtml>;expires=60;mode
  =passive
  ```

  In this case, the phone will indicate in the call appearance user interface that web content is available for a period of 60 seconds and will retrieve the web content at the request of the user for a period of up to 60 seconds but the phone will not spontaneously switch to the microbrowser application and download the content.

Static DNS Cache

Starting with SIP 2.1.0, failover redundancy can only be utilized when the configured IP server hostname resolves (through SRV or A record) to multiple IP addresses. Unfortunately, some customer's are unable to configure the DNS to take advantage of failover redundancy.

The solution in SIP 3.1 is to provide the ability to statically configure a set of DNS NAPTR SRV and/or A records into the phone.

When a phone is configured with a DNS server, it will behave as follows by default:

- An initial attempt to resolve a hostname that is within the static DNS cache, for example to register with its SIP registrar, results in a query to the DNS.
- If the initial DNS query returns no results for the hostname or cannot be contacted, then the values in the static cache are used for their configured time interval.
- After the configured time interval has elapsed, a resolution attempt of the hostname will again result in a query to the DNS.
- If a DNS query for a hostname that is in the static cache returns a result, the values from the DNS are used and the statically cached values are ignored.

When a phone is not configured with a DNS server, it will behave as follows

- An attempt to resolve a hostname that is within the static DNS cache will always return the results from the static cache.

Support for negative DNS caching as described in RFC 2308 is also provided to allow faster failover when prior DNS queries have returned no results from the DNS server. For more information, go to http://tools.ietf.org/html/rfc2308 .

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Specify DNS NAPTR, SRV, and A records for use when the phone is not configured to use a DNS server. • For more information, refer to DNS Cache <dns/> on page A-126.

Configuration File Examples

Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first.

For more information, refer to the "Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

Example 1

This example shows how to configure static DNS cache using A records IP addresses in SIP server address fields.

When the static DNS cache is not used, the **sip.cfg** configuration would look as follows:

```
reg.1.address="1001"
reg.1.server.1.address="172.23.0.140"
reg.1.server.1.port="5075"
reg.1.server.1.transport="UDPOnly"
reg.1.server.2.address="172.23.0.150"
reg.1.server.2.port="5075"
reg.1.server.2.transport="UDPOnly"
```

When the static DNS cache is used, the **sip.cfg** configuration would look as follows:

```
reg.1.address="1001"
reg.1.server.1.address="sipserver.example.com"
reg.1.server.1.port="5075"
reg.1.server.1.transport="UDPOnly"
reg.1.server.2.address=""
reg.1.server.2.port=""
reg.1.server.2.transport=""

dns.cache.A.1.name="sipserver.example.com"
dns.cache.A.1.ttl="3600"
dns.cache.A.1.address="172.23.0.140"
dns.cache.A.2.name="sipserver.example.com"
dns.cache.A.2.ttl="3600"
dns.cache.A.2.address="172.23.0.150"
```

Note Above addresses are presented to SIP application in order, for example, `dns.cache.A.1`, `dns.cache.A.2`, and so on.

Example 2

This example shows how to configure static DNS cache where your DNS provides A records for `server.X.address` but not SRV. In this case, the static DNS cache on the phone provides SRV records. For more information, go to http://tools.ietf.org/html/rfc3263 .

When the static DNS cache is not used, the **sip.cfg** configuration would look as follows:

```
reg.1.address="1002@sipserver.example.com"
reg.1.server.1.address="primary.sipserver.example.com"
reg.1.server.1.port="5075"
reg.1.server.1.transport="UDPOnly"
reg.1.server.2.address="secondary.sipserver.example.com"
```

```
reg.1.server.2.port="5075"
reg.1.server.2.transport="UDPOnly"
```

When the static DNS cache is used, the **sip.cfg** configuration would look as follows:

```
reg.1.address="1002"
reg.1.server.1.address="sipserver.example.com"
reg.1.server.1.port=""
reg.1.server.1.transport="UDPOnly"
reg.1.server.2.address=""
reg.1.server.2.port=""
reg.1.server.2.transport=""

dns.cache.SRV.1.name="_sip._udp.sipserver.example.com "
dns.cache.SRV.1.ttl= "3600"
dns.cache.SRV.1.priority="1"
dns.cache.SRV.1.weight="1"
dns.cache.SRV.1.port="5075"
dns.cache.SRV.1.target="primary.sipserver.example.com"

dns.cache.SRV.2.name="_sip._udp.sipserver.example.com "
dns.cache.SRV.2.ttl= "3600"
dns.cache.SRV.2.priority="2"
dns.cache.SRV.2.weight="1"
dns.cache.SRV.2.port="5075"
dns.cache.SRV.2.target="secondary.sipserver.example.com
```

Note The `reg.1.server.1.port` and `reg.1.server.2.port` values in this example are set to null to force SRV lookups.

Example 3

This example shows how to configure static DNS cache where your DNS provides NAPTR and SRV records for `server.X.address` .

When the static DNS cache is not used, the **sip.cfg** configuration would look as follows:

```
reg.1.address="1002@sipserver.example.com
reg.1.server.1.address="172.23.0.140"
reg.1.server.1.port="5075"
reg.1.server.1.transport="UDPOnly"
reg.1.server.2.address="172.23.0.150"
reg.1.server.2.port="5075"
reg.1.server.2.transport="UDPOnly"
```

When the static DNS cache is used, the **sip.cfg** configuration would look as follows:

```
reg.1.address="1002"
```

```
reg.1.server.1.address="sipserver.example.com"
reg.1.server.1.port=""
reg.1.server.1.transport=""
reg.1.server.2.address=""
reg.1.server.2.port=""
reg.1.server.2.transport=""

dns.cache.NAPTR.1.name="sipserver.example.com"
dns.cache.NAPTR.1.ttl= "3600"
dns.cache.NAPTR.1.order="1"
dns.cache.NAPTR.1.preference="1"
dns.cache.NAPTR.1.flag="s"
dns.cache.NAPTR.1.service=" SIP+D2U"
dns.cache.NAPTR.1.regexp=""
dns.cache.NAPTR.1.replacement="_sip._udp.sipserver.example.com"

dns.cache.SRV.1.name="_sip._udp.sipserver.example.com "
dns.cache.SRV.1.ttl= "3600"
dns.cache.SRV.1.priority="1"
dns.cache.SRV.1.weight="1"
dns.cache.SRV.1.port="5075"
dns.cache.SRV.1.target="primary.sipserver.example.com"

dns.cache.SRV.2.name="_sip._udp.sipserver.example.com "
dns.cache.SRV.2.ttl= "3600"
dns.cache.SRV.2.priority="2"
dns.cache.SRV.2.weight="1"
dns.cache.SRV.2.port="5075"
dns.cache.SRV.2.target="secondary.sipserver.example.com

dns.cache.A.1.name="primary.sipserver.example.com"
dns.cache.A.1.ttl="3600"
dns.cache.A.1.address="172.23.0.140"

dns.cache.A.2.name="secondary.sipserver.example.com"
dns.cache.A.2.ttl="3600"
dns.cache.A.2.address="172.23.0.150"
```

Note The `reg.1.server.1.port`, `reg.1.server.2.port`, `reg.1.server.1.transport`, and `reg.1.server.2.transport` values in this example are set to null to force NAPTR lookups.

Display of Warnings from SIP Headers

The Warning Field from a SIP header may be used to cause the phone to display a three second "pop-up" to the user. For example, this feature can be used to inform the user of information such as the reason that a call transfer action failed (bad extension entered, for example). (For more information, refer to Header Support on page B-4.)

These messages are displayed in any language supported by the phone for three seconds unless overridden by another message or action.

For example, if a user parks a call, the following message appears on their phone:

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Turn this feature on or off and specify which warnings are displayable. • For more information, refer to SIP <SIP/> on page A-11.

Quick Setup of SoundPoint IP / SoundStation IP / VVX Phones

In the SIP 3.1.2 release, a Quick Setup feature was added to simplify the process of entering the provisioning (boot) server parameters from the phone's user interface. This feature is designed to make it easier for on-site, "out of the box" provisioning of SoundPoint IP, SoundStation IP, and VVX phones.

When enabled, this feature will present a **QSetup** soft key to the user. When the user presses the **QSetup** soft key, a new menu will immediately appear that allows them to configure the necessary parameters for the phone to access the provisioning server for configuration. The **QSetup** soft key may be disabled using a configuration file setting such that it does not appear after it has been successfully configured.

The Quick Setup feature is supported on all SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670 desktop phones, SoundStation IP 6000 and 7000 conference phones, and Polycom VVX 1500 phones.

System administrators can enable the Quick Setup feature through the use of a new parameter in **sip.cfg** configuration file (or through the phone's menu).

For details on how to configure SoundPoint IP, SoundStation IP, and VVX phones for quick setup, refer to "Technical Bulletin 45460: Using Quick Setup with SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Turn this feature on or off . • For more information, refer to Provisioning <prov/> on page A-114.

Setting Up Audio Features

Proprietary state-of-the-art digital signal processing (DSP) technology is used to provide an excellent audio experience.

This section provides information for making configuration changes for the following audio-related features:

- Low-Delay Audio Packet Transmission
- Jitter Buffer and Packet Error Concealment
- Voice Activity Detection
- DTMF Tone Generation
- DTMF Event RTP Payload
- Acoustic Echo Cancellation
- Audio Codecs
- Background Noise Suppression
- Comfort Noise Fill
- Automatic Gain Control
- IP Type-of-Service
- IEEE 802.1p/Q
- Voice Quality Monitoring
- Dynamic Noise Reduction
- Treble/Bass Controls

Low-Delay Audio Packet Transmission

The phone is designed to minimize latency for audio packet transmission.

There are no related configuration changes.

Jitter Buffer and Packet Error Concealment

The phone employs a high-performance jitter buffer and packet error concealment system designed to mitigate packet inter-arrival jitter and out-of-order or lost (lost or excessively delayed by the network) packets. The jitter buffer is adaptive and configurable for different network environments. When packets are lost, a concealment algorithm minimizes the resulting negative audio consequences.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Set the jitter buffer tuning parameters including minimum and maximum size and shrink aggression. • For more information, refer to Codec Profiles <audioProfile/> on page A-49.
Local	Web Server (if enabled)	Set the jitter buffer tuning parameters including minimum and maximum size and shrink aggression. Navigate to http://<phoneIPAddress>/coreConf.htm#au Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Voice Activity Detection

The purpose of voice activity detection (VAD) is to conserve network bandwidth by detecting periods of relative "silence" in the transmit data path and replacing that silence efficiently with special packets that indicate silence is occurring. For those compression algorithms without an inherent VAD function, such as G.711, the phone is compatible with the comprehensive codec-independent comfort noise transmission algorithm specified in RFC 3389. This algorithm is derived from G.711 Appendix II, which defines a comfort noise (CN) payload format (or bit-stream) for G.711 use in packet-based, multimedia communication systems. The phone generates CN packets (also known as Silence Insertion Descriptor (SID) frames) and also decodes CN packets, efficiently regenerating a facsimile of the background noise at the remote end.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Enable or disable VAD and set the detection threshold. • For more information, refer to Voice Activity Detection <vad/> on page A-60.

DTMF Tone Generation

The phone generates dual tone multi-frequency (DTMF) tones in response to user dialing on the dial pad. These tones are transmitted in the real-time transport protocol (RTP) streams of connected calls. The phone can encode the DTMF tones using the active voice codec or using RFC 2833 compatible encoding. The coding format decision is based on the capabilities of the remote end point.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Set the DTMF tone levels, autodialing on and off times, and other parameters. • For more information, refer to Dual Tone Multi-Frequency <DTMF/> on page A-35.

DTMF Event RTP Payload

The phone is compatible with RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*. RFC 2833 describes a standard RTP-compatible technique for conveying DTMF dialing and other telephony events over an RTP media stream. The phone generates RFC 2833 (DTMF only) events but does not regenerate, nor otherwise use, DTMF events received from the remote end of the call.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Enable or disable RFC 2833 support in SDP offers and specify the payload value to use in SDP offers. • For more information, refer to Dual Tone Multi-Frequency <DTMF/> on page A-35.

Acoustic Echo Cancellation

The phone employs advanced acoustic echo cancellation (AEC) for hands-free operation. Both linear and non-linear techniques are employed to aggressively reduce echo yet provide for natural full-duplex communication patterns.

When using the handset on any SoundPoint IP phones, AEC is not normally required. In certain situations, where echo is experienced by the far-end party, when the user is on the handset, AEC may be enabled to reduce/avoid this echo. To achieve this, make the following changes in the **sip.cfg** configuration file (default settings for these parameters are disabled):

```
voice.aec.hs.enable = 1
voice.aes.hs.enable = 1
voice.ns.hs.enable = 1
voice.ns.hs.signalAttn = -6
voice.ns.hs.silenceAttn = -9
```

For more information, refer to Acoustic Echo Cancellation <aec/> on page A-44, Acoustic Echo Suppression <aes/> on page A-55, and Background Noise Suppression <ns/> on page A-56.

Audio Codecs

The following table shows which audio codecs are support by each of the SoundPoint IP, SoundStation IP, and Polycom VVX phones:

Phone	Supported Audio Codecs
SoundPoint IP 430	G.711μ-law, G.711a-law, G.729AB
SoundPoint IP 320/321/330/331	G.711μ-law, G.711a-law, G.729AB, iLBC
SoundPoint IP 335, 450, 550, 560, 650, and 670	G.711μ-law, G.711a-law, G.722, G.729AB, iLBC
SoundStation IP 6000	G.711μ-law, G.711a-law, G.722, G.722.1, G.722.1C, G.729AB, Siren14, iLBC
SoundStation IP 7000	G.711μ-law, G.711a-law, G.722, G.722.1, G.722.1C, G.729AB, Lin16, Siren14, Siren22, iLBC
Polycom VVX 1500	G.711μ-law, G.711a-law, G.719, G.722, G.722.1, G.722.1C, G.729AB, Lin16, Siren14, iLBC

The following table summarizes the supported audio codecs:

Algorithm	MIME Type	Ref.	Bit Rate	Sample Rate	Frame Size	Effective audio bandwidth
G.711μ-law	PMCU	RFC 1890	64 Kbps	8 Ksps	10ms - 80ms	3.5KHz
G.711a-law	PCMA	RFC 1890	64 Kbps	8 Ksps	10ms - 80ms	3.5KHz
G.719	G719/48000	RFC 5404	32 Kbps, 48 Kbps, 64 Kbps	48 kHz	20ms	20kHz
G.722	G722/8000	RFC 1890	64 Kbps	16 Ksps	10ms - 80ms	7 KHz
G.722.1	G7221/16000	RFC 3047	16 Kbps, 24 Kbps, 32 Kbps	16 Ksps	20ms - 80ms	7 KHz
G.722.1C	G7221/ 32000	G7221C	24 Kbps 32 Kbps 48 Kbps	32 Ksps	20ms - 80ms	14 KHz
G.729AB	G729	RFC 1890	8 Kbps	8 Ksps	10ms - 80ms	3.5KHz
SID	CN	RFC 3389	N/A	N/A	N/A	N/A

Algorithm	MIME Type	Ref.	Bit Rate	Sample Rate	Frame Size	Effective audio bandwidth
Lin16	L16/8000 L16/16000 L16/32000 L16/44100 L16/48000	RFC 1890	128 Kbps 256 Kbps 512 Kbps 705.6 Kbps 768 Kbps	8 Ksps 16 Ksps 32 Ksps 44.1 Ksps 48 Ksps	10ms	3.5 KHz 7 KHz 14 KHz 20 KHz 22 KHz
Siren14	SIREN14/ 16000	SIREN14	24 Kbps 32 Kbps 48 Kbps	32 Ksps	20ms - 80ms	14 KHz
Siren22	SIREN22/ 48000	SIREN22	32 Kbps 48 Kbps 64 Kbps	48 Ksps	20ms - 80ms	22 KHz
RFC 2833	phone-event	RFC 2833	N/A	N/A	N/A	N/A
iLBC	iLBC	RFC 3951	13.33Kbps 15.2Kbps	8 Ksps	30ms - 60ms 20ms - 80ms	3.5KHz

Note The network bandwidth necessary to send the encoded voice is typically 5-10% higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48kbps consumes about 100kbps of network bandwidth (two-way audio).

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. • For more information, refer to Codec Preferences <codecPref/> on page A-45 and Codec Profiles <audioProfile/> on page A-49.
Local	Web Server (if enabled)	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. Navigate to http://<phoneIPAddress>/coreConf.htm#au Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Background Noise Suppression

Background noise suppression (BNS) is designed primarily for hands-free operation and reduces background noise to enhance communication in noisy environments.

There are no related configuration changes.

Comfort Noise Fill

Comfort noise fill is designed to help provide a consistent noise level to the remote user of a hands-free call. Fluctuations in perceived background noise levels are an undesirable side effect of the non-linear component of most AEC systems. This feature uses noise synthesis techniques to smooth out the noise level in the direction toward the remote user, providing a more natural call experience.

There are no related configuration changes.

Automatic Gain Control

Automatic Gain Control (AGC) is applicable to hands-free operation and is used to boost the transmit gain of the local talker in certain circumstances. This increases the effective user-phone radius and helps with the intelligibility of soft-talkers.

There are no related configuration changes.

IP Type-of-Service

The "type of service" field in an IP packet header consists of four type-of-service (TOS) bits and a 3-bit precedence field. Each TOS bit can be set to either 0 or 1. The precedence field can be set to a value from 0 through 7. The type of service can be configured specifically for RTP packets and call control packets, such as SIP signaling packets.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify protocol-specific IP TOS settings. • For more information, refer to IP TOS <IP/> on page A-72.
Local	Web Server (if enabled)	Specify IP TOS settings. Navigate to: http://<phoneIPAddress>/netConf.htm#qo

IEEE 802.1p/Q

The phone will tag all Ethernet packets it transmits with an 802.1Q VLAN header for one of the following reasons:

- When it has a valid VLAN ID set in its network configuration

- When it is instructed to tag packets through Cisco Discovery Protocol (CDP) running on a connected Ethernet switch
- When a VLAN ID is obtained from DHCP (refer to DHCP Menu on page 3-8)

The 802.1p/Q user_priority field can be set to a value from 0 to 7. The user_priority can be configured specifically for RTP packets and call control packets, such as SIP signaling packets, with default settings configurable for all other packets.

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify default and protocol-specific 802.1p/Q settings. • For more information, refer to Ethernet IEEE 802.1p/Q <ethernet/> on page A-71.
Local	Web Server (if enabled)	Specify 802.1p/Q settings. Navigate to http://<phoneIPAddress>/netConf.htm#qo
	Local Phone User Interface	Specify whether CDP is to be used or manually set the VLAN ID or configure DHCP VLAN Discovery. Phase 1: BootRom - Navigate to: SETUP menu during auto-boot countdown. Phase 2: SIP Application - Navigate to: ***Menu>Settings>Advanced>Admin Settings>Network Configuration*** • For more information, refer to Setting Up the Network on page 3-2.

Voice Quality Monitoring

Note This feature requires a license key for activation except for the Polycom VVX 1500. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

The SoundPoint IP phones can be configured to generate various quality metrics for listening and conversational quality. These metrics can be sent between the phones in RTCP XR packets. The metrics can also be sent as SIP PUBLISH messages to a central voice quality report collector. The collection of these metrics is supported on the SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670 phones and the Polycom VVX 1500 phone.

Note Voice Quality Monitoring is not supported on the SoundStation IP 6000 and 7000 conference phones at this time. Only Voice Quality Monitoring of the audio portion is supported on the Polycom VVX 1500 at this time.

The RTCP XR packets are compliant with *RFC 3611 - RTP Control Extended Reports (RTCP XR).* The packets are sent to a report collector as specified in draft RFC *draft-ietf_sipping_rtcp-summary-02.*

Three types of quality reports can be enabled:

- Alert—Generated when the call quality degrades below a configurable threshold.
- Periodic—Generated during a call at a configurable period.
- Session—Generated at the end of a call.

A wide range of performance metrics are generated. Some are based on current values, such as jitter buffer nominal delay and round trip delay, while others cover the time period from the beginning of the call until the report is sent, such as network packet loss. Some metrics are computed using other metrics as input, such as listening Mean Opinion Score (MOS), conversational MOS, listening R-factor, and conversational R-factor.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the location of the central report collector, how often the reports are generated, and the warning and critical threshold values that will cause generation of alert reports. • For more information, refer to Quality Monitoring <quality monitoring/> on page A-61.

Dynamic Noise Reduction

Dynamic noise reduction (DNR) provides maximum microphone sensitivity, while automatically reducing background noise— from fans, projectors, heating and air conditioning—for clearer sound and more efficient conferencing.

There are no related configuration changes.

Treble/Bass Controls

The treble and bass controls equalize the tone of the high and low frequency sound from the speakers.

The SoundStation IP 7000 phone's treble and bass controls can be modified by the user (through **Menu > Settings > Basic > Audio > Treble EQ** or **Bass EQ**).

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the user's preferences for treble and bass. • For more information, refer to User Preferences <up/> on page A-31.

Setting Up Video Features

The Polycom VVX 1500 phone supports transmission and reception of high quality video images. The video is compatible with RFC 3984 - RTP Payload Format for H.264 Video, RFC 4629 - RTP Payload Format for ITU-T Rec. H.263 Video, and RFC 5168 - XML Schema for Media Control.

This section provides information for making configuration changes for the following video-related features:

- Video Transmission
- Video Codecs
- H.323 Protocol

Video Transmission

By default, at the start of a video call, the Polycom VVX 1500 phone transmits an RTP encapsulated video stream with images captured from the local camera. Users can stop and start video transmission by pressing the **Video** key, and then selecting the appropriate soft key.

You can control of the following features of the Polycom VVX 1500 phone's camera:

- Flicker avoidance
- Frame rate
- Brightness level
- Saturation level
- Contrast level
- Sharpness level

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Turn video transmission off at the near end when calls start and transmit still image if video not available. • For more information, refer to Video Settings <video/> on page A-64.
		Specify camera parameters. • For more information, refer to Camera Controls <camera/> on page A-70. Determine how the local camera is displayed. • For more information, refer to Local Camera View <localCameraView/> on page A-71.
Local	Local Phone User Interface	The user can set the individual video settings from the menu through **Settings > Basic > Video > Video Call Settings**, **Video Screen Mode**, and **Local Camera View**. The user can set the individual camera settings from the menu through **Settings > Basic > Video > Camera Settings**.

Video Codecs

The following table summarizes the Polycom VVX 1500 phone's video codec support:

Algorithm	MIME Type	Bit Rate	Frame Rate	Frame Size	Effective video bandwidth
H.261	H261/90000	64 kbps to 768 kbps	5 fps to 30 fps	Tx Frame size: CIF, QCIF RX Frame size: CIF, QCIF	Refer to Bit Rate column.
H.263	H263/90000, H263-1998/90000, H263-2000/90000			Tx Frame size:CIF, QCIF Rx Frame size:CIF, QCIF, SQCIF, QVGA, SVGA, SIF	
H.264	H264/90000				

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify codec priority, payload type and jitter buffer tuning parameters. • For more information, refer to Codec Preferences <codecPref/> on page A-66 and Codec Profiles <profile/> on page A-66.

H.323 Protocol

Note This feature requires a license key for activation on the Polycom VVX 1500. Using this feature may require purchase of a license key or activation by Polycom channels. For more information, contact your Certified Polycom Reseller.

As of SIP 3.2.2, telephony signaling support via the H.323 family of protocols enabling direct communication with H.323 endpoints, gatekeepers, call and media servers, and signaling gateways is supported on the Polycom VVX 1500 phone.

SIP and H.323 signaling can be supported at the same time, including bridging both types of calls during multi-party conference calls. Automatic detection of the correct or optimal signaling protocol is available when dialing from the contact or corporate directories. While SIP supports server redundancy and several transport options, only a single configured H.323 gatekeeper address per phone is supported. H.323 gatekeepers are optional, but if available, they must be used. If a gatekeeper is not configured or unavailable, calls can still be made if so configured.

If the H.323 protocol is disabled, there will be no visible evidence in the user interface of the Polycom VVX 1500 phone.

Support of the SIP protocol for telephony signaling can be disabled on the Polycom VVX 1500 such that all calls would be routed via the H.323 protocol.

This section provides detailed information on:

- Supported Standards
- Supported Polycom Interoperability
- Configuration File Changes
- Useful Tips
- Examples

Supported Standards

The following standards are supported by the implementation of this feature:

Standard	Description
ITU-T Recommendation H.323 (2003)	Packet-based multimedia communications systems
ITU-T Recommendation Q.931 (1998)	ISDN user-network interface layer 3 specification for basic call control
ITU-T Recommendation H.225.0 (2003)	Call signaling protocols and media stream packetization for packet based multimedia communications systems

Standard	Description
ITU-T Recommendation H.245 (5/2003)	Control protocol for multimedia communication
ITU-T Recommendation H.235.0 - H.235.9 (2005)	Security and encryption for H Series (H.323 and other H.245 based) multimedia terminals
ITU-T Recommendation H.350.1 (8/2003)	Directory services architecture for H.323

Supported Polycom Interoperability

Video calls are supported to the following Polycom endpoints/bridges/call servers (or gatekeepers)/media servers:

Make/Model	Protocol	Software Version
Polycom RMX 1000™	H.323	SW 1.1.1.8787
Polycom RMX 2000™	H.323	SW 4.0.2.7
Polycom HDX 9000™ series	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 8000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 7000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 6000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 4000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom VSX™ 8000	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 7000s and VSX™ 7000e	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 6000 and 6000a	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 5000	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 3000	SIP/ISDN/H.323	SW 9.0.5.1
Polycom V700™	SIP/ISDN/H.323	SW 9.0.5.1
Polycom V500™	SIP/ISDN/H.323	SW 9.0.5.1

Note Refer to the *Release Notes* for the latest list of supported Polycom endpoints/bridges/call servers (or gatekeepers)/media servers and any supported third party products. Any issues (and possible workarounds) with any of the above-mentioned products are also documented in the *Release Notes*.

Configuration File Changes

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify H.323 protocol, server, gatekeeper, and DTMF signaling parameters. • For more information, refer to Protocol <voIpProt/> on page A-7. Specify call routing parameters. • For more information, refer to User Preferences <up/> on page A-31 and Call Handling Configuration <call/> on page A-80. Specify the video call rate. • For more information, refer to Video Settings <video/> on page A-64. Specify auto-answer parameters. • For more information, refer to Call Handling Configuration <call/> on page A-80. Specify H.323 media encryption parameters. • For more information, refer to H.235 <H235/> on page A-112.
	Configuration file: **phone1.cfg**	Specify the registration protocol. • For more information, refer to Registration <reg/> on page A-134. Specify the auto off-hook protocol for dual-protocol lines. • For more information, refer to Calls <call/> on page A-139.

Useful Tips

The following information should be noted:

- If the phone has only the H.323 protocol enabled, it cannot be used to answer SIP calls.
- If the phone has only the SIP protocol enabled, it cannot be used to answer H.323 calls.
- If both SIP and H.323 protocols are disabled by mistake, the phone will continue to work as a SIP-only phone; however, the phone is not registered (you are able to send and receive SIP URL calls).
- The protocol that is used to place a call is stored in the placed call list of the user's phone.
- The protocol to be used when placing a call from the user's local contact directory is unspecified by default. The user can select SIP or H.323.
- The protocol that is used when placing a call from the user's corporate directory depends on the order of the attributes in the corporate directory. If only SIP_address is defined, then the SIP protocol is used. If only H323_address is defined, then the H.323 protocol is used. If both are defined, then the one that is defined first is used. For example, if `dir.corp.attribute.4.type` is **SIP_address** and

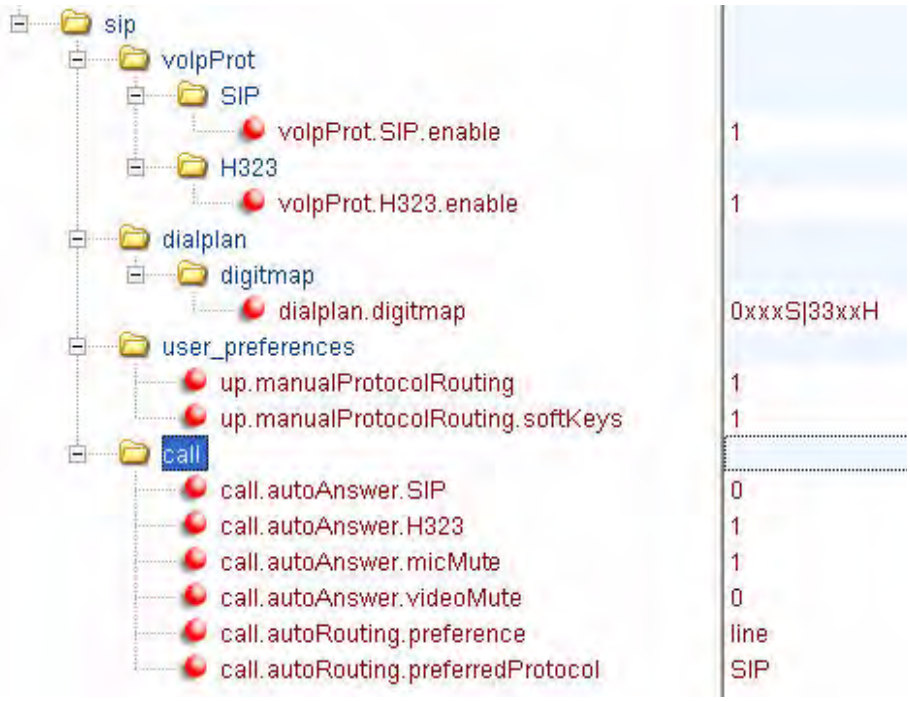
`dir.corp.attribute.5.type` is **H323_address**, then the SIP protocol is used.

- By default, the user is presented with protocol routing choices when a call could be placed with more than one protocol from its current context. The user must choose between SIP and H.323 to place a call by pressing the appropriate soft key.
- Calls made using H.323 cannot be forwarded or transferred.
  - The **Transfer** and **Forward** soft keys are not displayed during an H.323 call on a Polycom VVX 1500 phone. The **Forward** soft key is not displayed on the idle display of a Polycom VVX 1500 phone if the primary line is an H.323 line.
  - If a Polycom VVX 1500 user presses the **Transfer** soft key during an H.323 call on a Polycom VVX 1500 phone, no action is taken.
  - The auto-divert field in the local contact directory entry is ignored when a call is placed to that contact using H.323.
  - If a conference host ends a three-way conference the call and one of the party is connected by H.323, that party is not transferred to the other party.

Examples

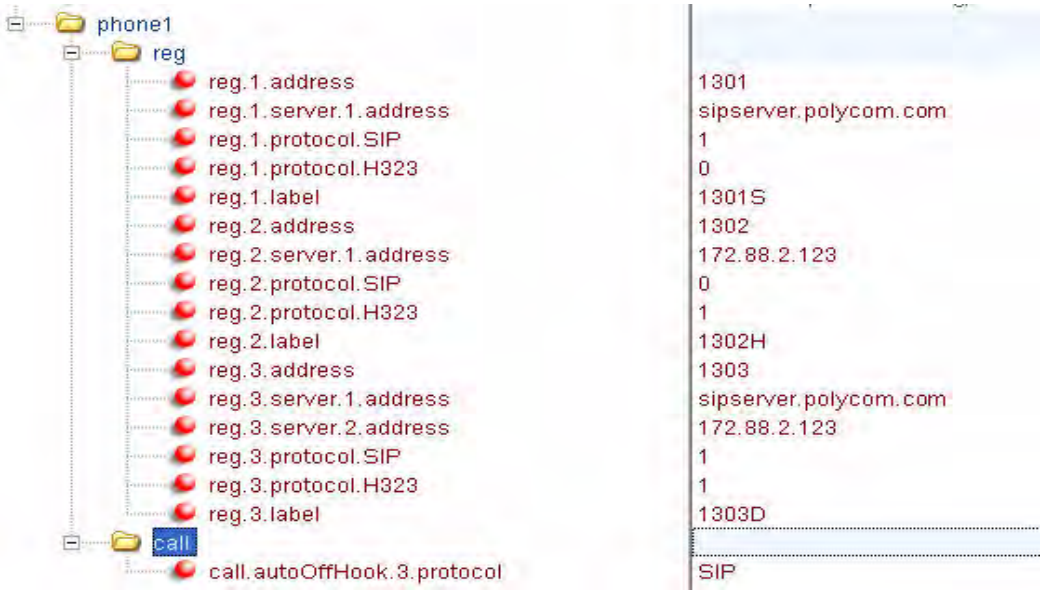
The following example **sip.cfg** configuration file shows the relevant parameters:

- To configure both SIP and H.323 protocols
- To set up a SIP and H.323 dial plan—Numbers with the format "0xxx" are placed on a SIP line and numbers with the format "33xx" are placed on an H.323 line
- To set up manual protocol routing using soft keys—If the protocol to use to place a call cannot be determined, the **Use SIP** and **Use H.323** soft keys appear, and the user must select one for the call to be placed.
- To configure auto-answering on H.323 calls only
- To set the preferred protocol to SIP



The following example **phone1.cfg** configuration file shows the relevant parameters:

- To set to configure one SIP line, one H.323 line, and a dual protocol line—both SIP and H.323 can be used.
- To set the preferred protocol for off-hook calls on the third (dual protocol) line to SIP



Setting Up Security Features

This section provides information for making configuration changes for the following security-related features:

- Local User and Administrator Privilege Levels
- Custom Certificates
- Incoming Signaling Validation
- Secure Real-Time Transport Protocol
- Configuration File Encryption
- Digital Certificates
- Mutual TLS Authentication

Local User and Administrator Privilege Levels

Several local settings menus are protected with two privilege levels, user and administrator, each with its own password. The phone will prompt for either the user or administrator password before granting access to the various menu options. When the user password is requested, the administrator password will also work. The web server is protected by the administrator password (refer to Configuring SoundPoint IP / SoundStation IP / VVX Phones Locally on page 4-98).

Configuration changes can be performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify the minimum lengths for the user and administrator passwords. • For more information, refer to Password Lengths <pwd/><length/> on page A-109.
Local	Web Server (if enabled)	None.
	Local Phone User Interface	The user and administrator passwords can be changed under the Settings menu or through configuration parameters (refer to Flash Parameter Configuration on page A-157). Passwords can consist of ASCII characters 32-127 (0x20-0x7F) only. Changes are saved to local flash but are not backed up to ***<Ethernet address>*-phone.cfg** on the provisioning server for security reasons.

Custom Certificates

The phone trusts certificates issued by widely recognized certificate authorities when trying to establish a connection to a provisioning server for application provisioning. Refer to Trusted Certificate Authority List on page C-1.

In addition, custom certificates can be added to the phone. This is done by using the SSL Security menu on the phone to provide the URL of the custom certificate then select an option to use this custom certificate.

Note For more information on using custom certificates, refer to “Technical Bulletin 17877: Using Custom Certificates With SoundPoint IP Phones” at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed locally:

Local	Local Phone User Interface	The custom certificate can be specified and the type of certificate to trust can be set under the Settings menu.

Incoming Signaling Validation

The three optional levels of security for validating incoming network signaling are:

- Source IP address validation
- Digest authentication
- Source IP address validation and digest authentication

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration File: **sip.cfg**	Specify the type of validation to perform on a request-by-request basis, appropriate to specific event types in some cases. • For more information, refer to Request Validation <requestValidation/> on page A-19.

Secure Real-Time Transport Protocol

Secure Real-Time Transport Protocol (SRTP) provides means of encrypting the audio stream(s) of VoIP phone calls to avoid interception and eavesdropping on phone calls. Both RTP and RTCP signaling may be encrypted using an AES algorithm as described in RFC3711. When this feature is enabled, phones will negotiate with the other end-point whether and what type of encryption or

authentication to utilize for the session. This negotiation process is compliant with RFC4568 (Session Description Protocol (SDP) Security Descriptions for Media Streams).

Authentication proves to the phone receiving the RTP/RTCP stream that the packets are from the expected source and have not been tampered with. Encryption modifies the data in the RTP/RTCP streams so that, if the data is captured or intercepted, it cannot be understood—it sounds like noise. Only the receiver knows the key to restore the data.

A number of configuration parameters have been added to allow you to turn off authentication and encryption for RTP and RTCP streams. This is done mainly to allow the system administrator to reduce the CPU usage on "legacy" Polycom phones in certain deployment scenarios (for example, if three-way local conferencing is required).

Note When using SRTP with Polycom VVX 1500 phone, limit the video bandwidth on the Polycom VVX 1500 to 384kbps; otherwise you will experience performance issues.

If the call is completely secure (RTP authentication and encryption and RTCP authentication and RTCP encryption are enabled), then the user sees a padlock symbol appearing in the last frame of the connected context animation (two arrow moving towards each other).

In SIP 2.2, the SRTP feature has been implemented in a very configurable manner. The reason for this is to allow deployment in a mixed environment where some elements of the solution are SRTP capable and some are not.

In SIP 3.2, `sec.srtp.requireMatchingTag` was added to **sip.cfg** as a flag to force a check of the tag value in the crypto attribute in an SDP answer.

For detailed configuration instructions, refer to "Technical Bulletin 25751: Secure Real-Time Transport Protocol on SoundPoint IP Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuration changes can be performed centrally at the boot server:

Central (boot server)	Configuration File: **sip.cfg**	Specify the parameters to enable and disable SRTP. • For more information, refer to SRTP <srtp/>on page A-110.

Configuration File Encryption

Configuration files (excluding the master configuration file), contact directories, and configuration override files can all be encrypted.

Note

Encrypted configuration files can be decrypted on the SoundPoint IP 32x, 33x, 430, 450, 550, 560, 650, and 670, the SoundStation IP 6000, and 7000, and the Polycom VVX 1500 phones.

The master configuration file cannot be encrypted on the provisioning server. This file is downloaded by the BootROM that does not recognize encrypted files. For more information, refer to Master Configuration Files on page A-2.

For more information on encrypting configuration files including determining whether an encrypted file is the same as an unencrypted file and using the SDK to facilitate key generation, refer to Encrypting Configuration Files on page C-4.

Configuration changes can be performed centrally at the provisioning server:

Central (provisioning server)	Configuration File: **sip.cfg**	Specify the phone-specific contact directory and the phone-specific configuration override file. • For more information, refer to Encryption <encryption/> on page A-109.
	Configuration file: **<device>.cfg**	Change the encryption key. • For more information, refer to refer to Flash Parameter Configuration on page A-157.

Digital Certificates

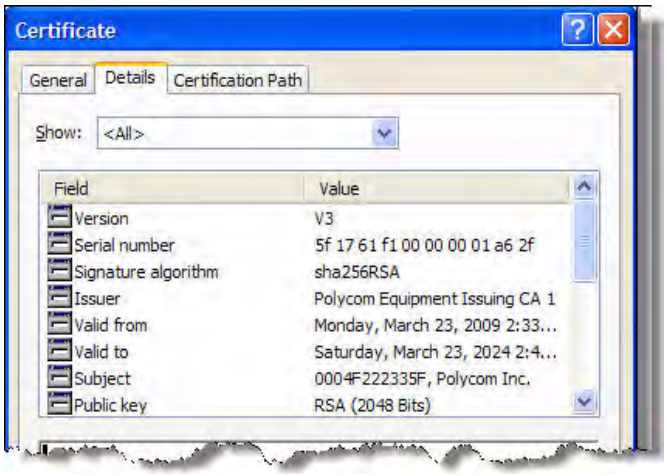
Starting in May 2009, Polycom is installing a digital certificate on certain SoundPoint IP phone models at the manufacturing facility. Over time, other SoundPoint IP phone models as well as all SoundStation IP and Polycom VVX phone models will have a digital certificate. Refer to "Technical Bulletin 37148: Device Certificates on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones"at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

This X.509 digital certificate is 'signed' by the Polycom Root CA and may be used for a server to authenticate the phone when initiating Transport Layer Security (TLS) based communications such as those used for HTTPS provisioning and TLS SIP signaling encryption. The Polycom Root CA can be downloaded from http://pki.polycom.com/pki . The X.509 digital certificates are set to expire on March 9, 2044.

An X.509 digital certificate is a digitally signed statement. The X.509 standard defines what information can go into a certificate. All X.509 certificates have the following fields, in addition to the signature:

- **Version**. This identifies which version of the X.509 standard applies to this certificate, which affects what information can be specified in it.
- **Serial Number**. The entity that created the certificate is responsible for assigning it a serial number to distinguish it from other certificates it issues.
- **Signature Algorithm Identifier**. This identifies the algorithm used by the Certificate Authority (CA) to sign the certificate.
- **Issuer Name**. The X.500 name of the entity that signed the certificate. This is normally a CA. Using this certificate implies trusting the entity that signed this certificate.
- **Validity Period**. Each certificate is valid only for a limited amount of time. This period is described by a start date and time and an end date and time, and can be as short as a few seconds or almost as long as a century.
- **Subject Name**. The name of the entity whose public key the certificate identifies. This name uses the X.500 standard, so it is intended to be unique across the Internet.
- **Subject Public Key Information**. This is the public key of the entity being named, together with an algorithm identifier which specifies which public key cryptographic system this key belongs to and any associated key parameters.

The following is an example of a Polycom device certificate (if opened with the Microsoft Internet Explorer 7 or Firefox 3.5 browser on a computer running Microsoft XP Service Pack 3):



The device certificate and associated private key are stored on the phone in its non-volatile memory as part of the manufacturing process.

For more information on digital certificates, refer to http://www.ietf.org/html.charters/pkix-charter.html and http://www.ietf.org/rfc/rfc2459.txt .

To determine if there is a digital certificate on a SoundPoint IP, SoundStation IP, or Polycom VVX phone:

1. Press the **Menu** key, and then select **Status > Platform > Phone**.
2. Scroll down to the bottom of screen.

   One of three messages will be displayed:

   - "Device Certificate: Installed" is displayed if the certificate is available in flash memory, all the certificate fields are valid (listed above) and certificate has not expired.
   - "Device Certificate: Not Installed" is displayed if the certificate is not available in flash memory (or the flash memory location where the device certificate is to be stored is blank).
   - "Device Certificate: Invalid" is displayed if the certificate is not valid (if any of the fields listed above are not correct).

Mutual TLS Authentication

Mutual Transport Layer Security (TLS) authentication is a process in which both entities in a communications link authenticate each other. In a network environment, the phone authenticates the server and vice-versa. In this way, phone users can be assured that they are doing business exclusively with legitimate entities and servers can be certain that all would-be users are attempting to gain access for legitimate purposes.

This feature requires that the phone being used has a Polycom factory-installed device certificate. Refer to the previous section, Digital Certificates.

Prior to SIP 3.2, and in cases where the phones do not have factory-installed device certificates, the phone will authenticate to the server as part of the TLS authentication, but the server cannot cryptographically-authenticate the phone. This is sometime referred to as Server Authentication or single-sided Authentication.

Mutual TLS authentication is optional and is initiated by the server. When the phone acts as a TLS client and the server is configured to require mutual TLS, the server will request, and then validate the client certificate during the handshake. If the server is configured to require mutual TLS, a device certificate and an associated private key must be loaded on the phone.

The digital certificate, stored on the phone, is used by:

- HTTPS device configuration, if the server is configured for Mutual Authentication

- SIP signaling—when the selected transport protocol is TLS and the server is configured for Mutual Authentication
- Syslog—when the selected transport protocol is TLS and the server is configured for Mutual Authentication
- Corporate Directory—when the selected transport protocol is TLS and the server is configured for Mutual Authentication

Note At this time, the user will not be able to modify or update the digital certificate or the associated private key stored on the phone during manufacturing.

The Polycom Root CA can be downloaded from http://pki.polycom.com . The location of the Certificate Revocation List (CRL)—a list of all expired certificates signed by the Polycom Root CA—is part of the Polycom Root CA digital certificate. If Mutual TLS is enabled, the Polycom Root CA must be downloaded onto the HTTPS server.

At this time, the following operating systems/web servers combinations are supported:

- Microsoft Internet Information Services 6.0 on Microsoft Windows Server 2003
- Apache v1.3 on Microsoft Windows XP

For more information on using Mutual TLS with Microsoft® Internet Information Services (IIS) 6.0, refer to "Technical Bulletin 52609: Mutual Transport Layer Security Provisioning Using Microsoft Internet Information Services 6.0" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Configuring SoundPoint IP / SoundStation IP / VVX Phones Locally

A local phone-based configuration web server is available, unless it is disabled through **sip.cfg**. It can be used as the only method of modifying phone configuration or as a distributed method of augmenting a centralized provisioning model. For more information, refer to Web Server <httpd/> on page A-79.

The phone's local user interface also permits many application settings to be modified, such as SIP server address, ring type, or regional settings such as time/date format and language.

Local Web Server Access	Point your web browser to http://<phoneIPAddress>/. Configuration pages are accessible from the menu along the top banner. The web server will issue an authentication challenge to all pages except for the home page. Credentials are (case sensitive): User Name: Polycom Password: The administrator password is used for this.
Local Settings Menu Access	Some items in the Settings menu are locked to prevent accidental changes. To unlock these menus, enter the user or administrator passwords. The administrator password can be used anywhere that the user password is used. (Polycom recommends that you change the administrative password from the default value.) Factory default passwords are: User password: 123 Administrator password: 456
Passwords:	
Administrator password required.	Network Configuration Line Configuration Call Server Configuration SSL Security settings Reset to Default - local configuration, device settings, and file system format
User password required.	Reboot Phone

Changes made through the web server or local user interface are stored internally as overrides. These overrides take precedence over settings contained in the configuration obtained from the provisioning server.

If the provisioning server permits uploads, these override setting will be saved in a file called **<*Ethernet address*>-phone.cfg** on the provisioning server as well as in flash memory.

Warning Local configuration changes will continue to override the provisioning server-derived configuration until deleted through the Reset Local Config menu selection or configured using the ‘device set ‘ procedure.

For more information, refer to Modifying Phone’s Configuration Using the Web Interface on page C-27.

5

Troubleshooting Your SoundPoint IP / SoundStation IP / VVX Phones

This chapter provides you with some tools and techniques for troubleshooting SoundPoint IP / SoundStation IP / VVX phones and installations. The phone can provide feedback in the form of on-screen error messages, status indicators, and log files for troubleshooting issues.

This chapter includes information on:

- BootROM Error Messages
- SIP Application Error Messages
- Status Menu
- Log Files
- Testing Phone Hardware

This chapter also presents phone issues, likely causes, and corrective actions. Issues are grouped as follows:

- Power and Startup
- Controls
- Access to Screens and Systems
- Calling
- Displays
- Audio
- Licensable Features
- Upgrading

Review the latest *Release Notes* for the SIP application for known problems and possible workarounds. For the latest *Release Notes* and the latest version of this Administrator's Guide, go to Polycom Technical Support at http://www.polycom.com/support/voice/.

If a problem is not listed in this chapter nor described in the latest *Release Notes*, contact your Certified Polycom Reseller for support.

Error Messages

There are several different error messages that can be displayed on the phone when it is booting. Some of these errors are fatal, meaning that the phone will not able to boot until this issue has been resolved, and some are recoverable, meaning the phone will continue booting after the error, but the configuration of the phone may not be what you were expecting.

BootROM Error Messages

Most of these errors are also logged on the phone's boot log, however, if you are having trouble connecting to the provisioning server, the phone will likely not be able to upload the boot log for you to examine.

Failed to get boot parameters via DHCP

The phone does not have an IP address and therefore cannot boot. Check that all cables are connected, the DHCP server is running and that the phone has not been put into a VLAN which is different from the DHCP server. Check the DHCP configuration.

Application <file name> is not compatible with this phone!

When the BootROM displays an error like "The application is not compatible", it means an application file was downloaded from the provisioning server, but it cannot be installed on this phone. This issue can usually be resolved by finding a software image that is compatible with the hardware or the BootROM being used and installing this on the provisioning server. There are various different hardware and software dependencies. Refer to the latest *Release Notes* for details on the version you are using.

Could not contact boot server, using existing configuration

The phone could not contact the provisioning server, but the causes may be numerous. It may be cabling issue, it may be related to DHCP configuration, or it could be a problem with the provisioning server itself. The phone can recover from this error so long as it previously downloaded a valid application BootROM image and all of the necessary configuration files.

Error, application is not present!

There is no application stored in flash memory and the phone cannot boot. A compatible SIP application must be downloaded into the phone using one of the supported provisioning protocols. You need to resolve the issue of connecting to the provisioning server. This error is typically a result of one of the above errors. This error is fatal, but recoverable. Contact your system administrator.

Not all configuration files were present on the server

Similarly, a message about configuration files not being present, means that the phone was able to reach the provisioning server, but that it was not able to find all the necessary files. So long as the files exist in flash memory, the phone can boot following this error. The probable cause of this error is a misconfiguration of the **<MACaddress>.cfg** file.

Note This error does not occur with BootROM 3.2.2 B or later.

Error loading <file name>

When the required file does not exist in flash memory and cannot be found on the provisioning server, the "Error loading" message will tell you which file could not be found. This error only remains on the screen for a few seconds so you need to watch closely. The phone reboots.

Note This error does not occur with BootROM 3.2.2 B or later.

SIP Application Error Messages

Config file error. Error is <Hex #>

If there is an error in the configuration file, you will not be able to reboot the phones. You must review the provisioning server configuration, make the correction, and reapply the configuration file by restarting the phones. This error also happens when phone does a restart (not a reboot) and finds a newer version of BootROM or application; this triggers a reboot. Usually this error is self-recoverable.

Network link is down

Since the SoundPoint IP / SoundStation IP / VVX phones do not have an LED indicating network LINK status like many networking devices, if a link failure is detected while the phone is running a message saying "Network link is down" will be displayed. This message will be shown on the screen whenever

the phone is not in the menu system and will remain on screen until the link problem is resolved. Call related functions (for example, soft keys and feature keys) disabled when the network is down; however the menu works.

Status

When the phone is unable to register with the call control server, the icon is shown (outline). Once the phone is registered, the icon is shown (solid). On the SoundStation IP 7000, the icons are and . On the Polycom VVX 1500, the icons are and .

Flashing Time

If the phone has not been able to contact the SNTP server or if one has not been configured, the date/time display will flash until this is fixed. If an SNTP is not available, the data/time display can be turned off so that the flashing display is not a distraction.

Status Menu

Debugging of single phone may be possible through an examination of the phone's status menu. Press **Menu**, select ***Status***, and then press the **Select** soft key.

Under the ***Platform*** selection, you can get details on the phone's serial number or MAC address, the current IP address, the BootROM version, the application version, the name of the configuration files in use, and the address of the provisioning server.

In the ***Network*** menu, the phone will provide information about TCP/IP setting, Ethernet port speed, connectivity status of the PC port, and statistics on packets sent and received since last boot. This would also be a good place to look and see how long it has been since the phone rebooted. The Call Statistics screen shows packets sent and received on the last call.

The ***Lines*** menu will give you details about the status of each line that has been configured on the phone.

Finally, the ***Diagnostics*** menu offers a series of hardware tests to verify correct operation of the microphone, speaker, handset, and third party headset, if present. It will also let you test that each of the keys on the phone is working, and it will display the function that has been assigned to each of the keys in the configuration. This is also where you can test the LCD for faulty pixels.

In addition to the hardware tests, the Diagnostics menu has a series of real-time graphs for CPU, network and memory utilization that can be helpful in diagnosing performance issues.

Log Files

SoundPoint IP and SoundStation IP phones will log various events to files stored in the flash file system and will periodically upload these log files to the provisioning server. The files are stored in the phone's home directory or a user-configurable directory. You can also configure a phone to send log messages to a syslog server.

There is one log file for the BootROM and one for the application. When a phone uploads its log files, they are saved on the provisioning server with the MAC address of the phone prepended to the file name. For example, **0004f200360b-boot.log** and **0004f200360b-app.log** are the files associated with MAC address 00f4f200360b. The BootROM log file is uploaded to the provisioning server after every reboot. The application log file is uploaded periodically or when the local copy reaches a predetermined size. Refer to Basic Logging <level/><change/> and <render/> on page A-106.

Both log files can be uploaded on demand using a multiple key combination described in Multiple Key Combinations on page C-11. The phone uploads four files, namely, **mac-boot.log**, **app-boot.log**, **mac-now-boot.log**, and **mac-now-app.log**. The "now_" logs are uploaded manually unless they are empty.

The amount of logging that the phone performs can be tuned for the application to provide more or less detail on specific components of the phone's software. For example, if you are troubleshooting a SIP signaling issue, you are not likely interested in DSP events. Logging levels are adjusted in the configuration files or via the web interface. You should not modify the default logging levels unless directed to by Polycom Technical Support. Inappropriate logging levels can cause performance issues on the phone.

In addition to logging events, the phone can be configured to automatically execute command-line instructions at specified intervals that output run-time information such as memory utilization, task status, or network buffer contents to the log file. These techniques should only be used in consultation with Polycom Technical Support.

Application Logging Options

Each of the components of the application software is capable of logging events of different severity. This allows you to capture lower severity events in one part of the application, while still only getting high severity event for other components.

The parameters for log level settings are found in the **sip.cfg** configuration file. They are `log.level.change.module_name`. Log levels range from 1 to 6 (1 for the most detailed logging, 6 for critical errors only). There are currently 27 different log types that can be adjusted to assist with the investigation of different problems.

When testing is complete, remember to return all logging levels to the default value of 4.

There are other logging parameters that you may wish to modify. Changing these parameters does not have the same impact as changing the logging levels, but you should still understand how your changes will affect the phone and the network.

- `log.render.level`—Sets the lowest level that can be logged (default=1)
- `log.render.file.size`—Maximum size before log file is uploaded (default=16 kb)
- `log.render.file.upload.period`—Frequency of log uploads (default is 172800 seconds = 48 hours)
- `log.render.file.upload.append`—Controls if log files on the provisioning server are overwritten or appended, not supported by all servers
- `log.render.file.upload.append.sizeLimit`—Controls the maximum size of log files on the provisioning server (default=512 kb)
- `log.render.file.upload.append.limitMode`—Controls action to take when server log reaches max size, actions are stop and delete

Scheduled Logging

Scheduled logging is a powerful tool for anyone who is trying to troubleshoot an issue with the phone that only occurs after some time in operation.

The output of these instructions is written to the application log, and can be examined later (for trend data).

The parameters for scheduled logging are found in the **sip.cfg** configuration file. They are `log.sched.module_name`.

The following figure shows an example of a configuration file and the resulting log file.

Manual Log Upload

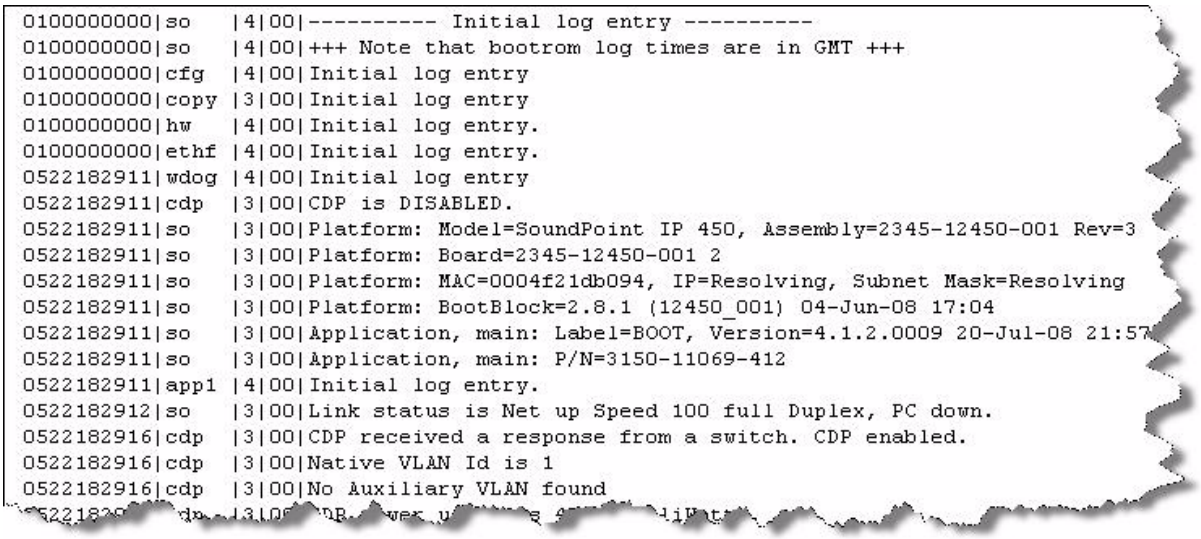
If you want to look at the log files without having to wait for the phone to upload them (which could take as long as 24 hours or more), initiate an upload by pressing correct combination of keys on the phone.

For more information, refer to Multiple Key Combinations on page C-11.

When the log files are manually uploaded, the word “now” is inserted into the name of the file, for example, 0004f200360b-now-boot.log .

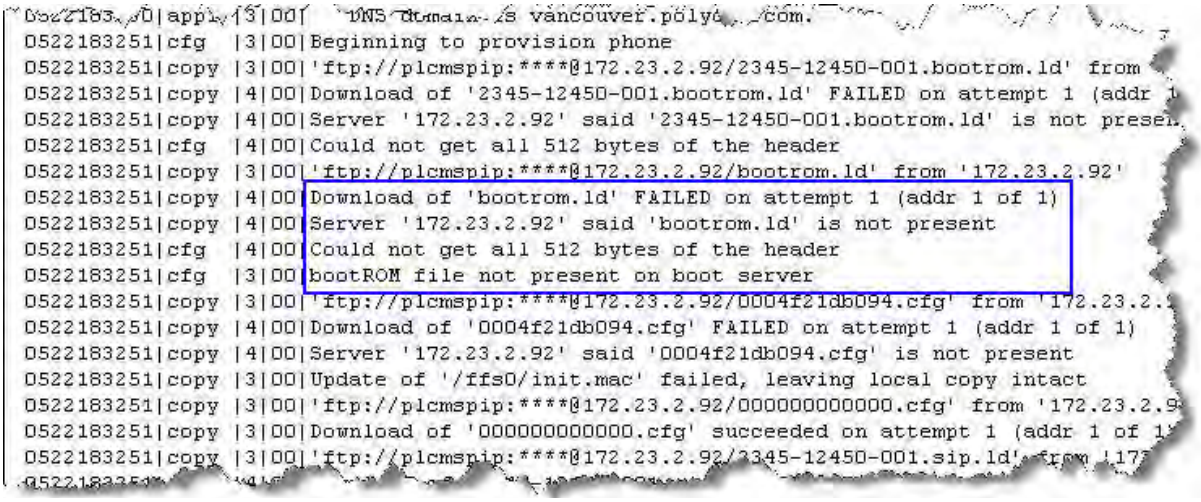
Reading a Boot Log

The following figure shows a portion of a boot log file:



Boot Failure Messages

The following figure shows a number of boot failure messages:



Reading an Application Log

The following figure shows portions of an application log file:

Reading a Syslog

The following shows a portion of a syslog log file—the messages look identical to the normal log with the addition of a timestamp and IP address:

```
Jan  0 00:00:00 172.23.7.249 0100000000|so   |4|00|---------- Initial log entry ----------
Jan  0 00:00:00 172.23.7.249 0100000000|so   |4|00|+++ Note that bootrom log times are in GMT +++
Jan  0 00:00:00 172.23.7.249 0100000000|cfg  |4|00|Initial log entry
Jan  0 00:00:00 172.23.7.249 0100000000|copy |3|00|Initial log entry
Jan  0 00:00:00 172.23.7.249 0100000000|hw   |4|00|Initial log entry.
Jan  0 00:00:00 172.23.7.249 0100000000|ethf |4|00|Initial log entry.
Feb 13 01:12:39 172.23.7.249 0213011239|wdog |4|00|Initial log entry
Feb 13 01:12:39 172.23.7.249 0213011239|cdp  |3|00|CDP is DISABLED.
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Platform: Model=SoundPoint IP 650, Assembly=2345-12600-
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Platform: Board=2345-12600-001 1
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Platform: MAC=0004f2111511, IP=Resolving, Subnet Mask=
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Platform: BootBlock=2.7.0 (12600_001) 30-May-06 15:58
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Application, main: Label=BOOT, Version=4.1.0.0219 10-D
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Application, main: P/N=3150-11069-410
Feb 13 01:12:39 172.23.7.249 0213011239|appl |4|00|Initial log entry.
Feb 13 01:12:40 172.23.7.249 0213011240|so   |3|00|Link status is Net down, PC down.
Feb 13 01:12:41 172.23.7.249 0213011241|so   |3|00|Link status is Net up Speed 100 half Duplex, PC down.
Feb 13 01:12:41 172.23.7.249 0213011241|cdp  |3|00|CDP is disabled.
Feb 13 01:12:45 172.23.7.249 0213011245|appl |3|00|DNS resolver servers are '172.23.0.200' '172.23.0.239'
Feb 13 01:12:45 172.23.7.249 0213011245|appl |3|00|DNS resolver search domain is 'vancouver.polycom.com'
Feb 13 01:12:45 172.23.7.249 0213011245|appl |3|00|Bootline: esw(3,0)bootHost:flash e=172.23.7.249:ffff00
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|Time has been set from 172.23.0.200 (172.23.0.200).
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|DHCP returned result 0x3E7 from server 172.23.0.232.
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|   Phone IP address is 172.23.7.249.
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|   Subnet mask is 255.255.0.0.
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|   Gateway address is 172.23.2.240.
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|   Time server is 172.23.0.200.
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|   GMT offset is -28800 seconds.
Apr 15 22:32:22 172.23.7.249 0415223222|appl |3|00|   DHCP time offset is 0 seconds.
```

Testing Phone Hardware

To obtain more detailed troubleshooting information, you can access certain menus on the SoundPoint IP and SoundStation IP phone that test the phone hardware.

From the diagnostics menu, you can test:

- The phone's microphones, speaker, handset, and any third-party handset (if present)
- Keypad mapping—You can verify the function assign to each key.
- Graphic display—You can test the LCD for faulty pixels.

To test the phone hardware:

>> Press **Menu**, and then select **Status** > **Diagnostics** > **Test Hardware** > **Audio Diagnostics**, **Keypad Diagnostics**, or **Display Diagnostics**.

Power and Startup

Symptom	Problem	Corrective Action
There are power issues.	The SoundPoint IP / SoundStation IP / VVX family SIP phone has no power.	Do one of the following: • Verify that no lights appear on the unit when it is powered up. • Check if the phone is properly plugged into a functional AC outlet. • Make sure that the phone isn't plugged into a plug controlled by a light switch that is off. • If plugged into a power strip, try plugging directly into a wall outlet instead. • Try the phone in another room where the electricity is known to be working on a particular outlet. • If using PoE, the power supply voltage may be too high or too low.

Controls

Symptom	Problem	Corrective Action
The dial pad does not work.	The dial pad on the SoundPoint IP / SoundStation IP / VVX family SIP phone does not respond.	Do one of the following: • Check for a response from other feature keys or from the dial pad. • Place a call to the phone from a known working telephone. Check for display updates. • Press the **Menu** key followed by **System Status** and **Server Status** to check if the telephone is correctly registered to the server. • Press the **Menu** key followed by **System Status** and **Network Statistics**. Scroll down to see if LAN port shows active or Inactive. • Check the termination at the switch or hub end of the network LAN cable. Ensure that the switch/hub port connected to the telephone is operational (if not accessible, contact your system administrator). • Before restarting your phone, contact your system administrator, since this may allow more detailed troubleshooting to occur before losing any current status information.

Access to Screens and Systems

Symptom	Problem	Corrective Action
There is no response from feature key presses.	The SoundPoint IP / SoundStation IP / VVX family SIP phone is not in active state.	Do one of the following: • Press the keys more slowly. • Check to see whether or not the key has been mapped to a different function or disabled. • Make a call to the phone to check for inbound call display and ringing as normal. If successful, try to press feature keys within the call to access Directory or Buddy Status, for example. • Press **Menu** followed by **Status > Lines** to confirm line is actively registered to the call server. • Reboot the phone to attempt re-registration to the call server (refer to Rebooting the Phone on page C-11).
The display shows “Network Link is Down”.	The LAN cable is not properly connected.	Do one of the following: • Check termination at the switch or hub (furthest end of the cable from the phone). • Check that the switch or hub is operational (flashing link/status lights) or contact your system administrator. • Press **Menu** followed by **Status > Network**. Scroll down to verify that the LAN is active. • Ping phone from another machine. • Reboot the phone to attempt re-registration to the call server (refer to Rebooting the Phone on page C-11).

Calling

Symptom	Problem	Corrective Action
There is no dial tone.	Power is not correctly applied to the SoundPoint IP family SIP phone.	Do one of the following: • Check that the display is illuminated. • Make sure the LAN cable is inserted properly at the rear of the phone (try unplugging and re-inserting the cable). • If using in-line powering, have your system administrator check that the switch is supplying power to the phone.
	Dial tone is not present on one of audio paths.	Do one of the following: • Switch between Handset, Headset (if present) or Hands-Free Speakerphone to see if dial tone is present on another paths. • If dial tone exists on another path, connect a different handset or headset to isolate the problem. • Check configuration for gain levels.
	The phone is not registered.	Contact your system administrator.
The phone does not ring.	Ring setting or volume is low.	Do one of the following: • Adjust the ringing level from the front panel using the volume up/down keys. • Check same status of handset, headset (if connected) and through the Hands-Free Speakerphone.
	Outbound or inbound calling is unsuccessful.	Do one of the following: • Place a call to the phone under investigation. Check that the display indicates incoming call information. • Lift the handset. Ensure dial tone is present and place a call to another extension or number. Check that the display changes in response.
The line icon shows an unregistered line icon.	The phone line is unregistered.	Contact your system administrator.

Displays

Symptom	Problem	Corrective Action
There is no display. The display is incorrect. The display has bad contrast.	Power is not correctly applied to the SoundPoint IP family SIP phone.	Do one of the following: • Check that the display is illuminated. • Make sure the LAN cable is inserted properly at the rear of the phone (try unplugging and re-inserting the cable). • If using in-line powering, have your system administrator check that the switch is supplying power to the phone. • Use the screen capture feature. Refer to Capturing Phone's Current Screen on page C-30.
	The contrast needs adjustment.	Do one of the following: • Refer to the appropriate SoundPoint IP / SoundStation IP / VVX SIP phone User Guide. • Reboot the phone to obtain a default level of contrast (refer to Rebooting the Phone on page C-11). • Use the screen capture feature. Refer to Capturing Phone's Current Screen on page C-30.
	Outbound or inbound calling is unsuccessful.	Do one of the following: • Place a call to the phone under investigation. Check that the display indicates incoming call information. • Lift the handset. Ensure dial tone is present and place a call to another extension or number. Check that the display changes in response. • Use the screen capture feature. Refer to Capturing Phone's Current Screen on page C-30.
The display is flickering.	Certain type of older fluorescent lighting causes the display to appear to flicker.	Do one of the following: • Move the SoundPoint IP / SoundStation IP / VVX SIP phone away from the lights. • Replace the lights. • Use the screen capture feature. Refer to Capturing Phone's Current Screen on page C-30.

Audio

Symptom	Problem	Corrective Action
There is no audio on the headset.	The connections are not correct.	Do one of the following: • Ensure the headset is plugged into the jack marked Headset at the rear of the phone. • Ensure the headset amplifier (if present) is turned on and/or the volume is correctly adjusted).
There are audio and echo issues on the headset.	Possible issues include: • Echo on external calls through a gateway. • Internal calls (no gateway), handsfree echo. • Internal calls (no gateway), handset to handset echo.	Refer to "Technical Bulletin 16249: Troubleshooting Audio and Echo Issues on SoundPoint® IP Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Licensable Features

Symptom	Problem	Corrective Action
A user is trying to access one of the following features, but it is not available on their phone: • Corporate Directory • Recording and Playback of Audio Calls • Managing Conferences • Voice Quality Monitoring • H.323	The license is not installed on the phone or it has expired.	Do the following: • Press the **Menu** key, then select **Status > Licenses**. • Using the arrow keys, verify that the feature in question has a valid license. If no licenses are installed, the "No license installed." message appears.

Upgrading

Symptom	Problem	Corrective Action
SoundPoint IP 300, 301, 500, 501, 600, and/or 601 and/or SoundStation IP 4000 behave incorrectly or do not display new features.	New features are not supported on the SoundPoint IP 300, 301, 500, 501, 600, and 601 and SoundStation IP 4000 and the configuration files have not been correctly modified. These phones will not 'understand' the new configuration parameters, and will attempt to load the new application.	The attempt to load the new application will fail since there is no 300/301/500/501/600/601/4000 image contained within the **sip.ld** file, so the phone will continue on and run the current version of application that it has in memory. It will however use the new configuration files. Refer to Supporting SoundPoint IP 300, 301, 500, 501, 600 and 601 and SoundStation IP 4000 Phones on page 3-23.

A

Configuration Files

This appendix provides detailed descriptions of certain configuration files used by the Session Initiation Protocol (SIP) application. It is a reference for all parameters that are configurable when using the centralized provisioning installation model.

This appendix contains information on:

- Master Configuration Files (**MAC-address.cfg** or **000000000000.cfg**)
- Application Configuration (**sip.cfg**)
- Per-Phone Configuration (**phone1.cfg**)
- Flash Parameter Configuration

The application configuration files dictate the behavior of the phone once it is running the executable specified in the master configuration file.

Warning Configuration files should only be modified by a knowledgeable system administrator. Applying incorrect parameters may render the phone unusable. The configuration files which accompany a specific release of the SIP software must be used together with that software. Failure to do this may render the phone unusable.

Note In the tables in the subsequent sections, “Null” should be interpreted as the empty string, that is, attributeName=“” when the file is viewed in an XML editor.

To enter special characters in a configuration file, enter the appropriate sequence using an XML editor:

- & as &
- ” as "
- ’ as '
- < as <
- > as >

Note The various .hd. parameters in **sip.cfg** (such as `voice.aec.hd.enable`, `voice.ns.hd.enable`, and `voice.agc.hd.enable`) are headset parameters. They are not connected to high definition or HD voice.

Note

You can make changes to the configuration files through the web interface to the phone. Using your chosen browser, enter the phone's IP address as the browser address. For more information, refer to Modifying Phone's Configuration Using the Web Interface on page C-27.

Changes made through the web interface are written to the override file (highest priority). These changes remain active until **Reset Local Config** is performed.

Master Configuration Files

The master configuration files can be one of:

- Specified master configuration file—The master configuration file can be explicitly specified in the provisioning server address, for example, http://usr:pwd@server/dir/example1.cfg. The filename must end with **.cfg** and be at least five characters long. If this file cannot be downloaded, the phone will search for the per-phone master configuration file (described next).
- Per-phone master configuration file—If per-phone customization is required, the file should be named **<Ethernet address>.cfg**, where Ethernet address is the MAC address of the phone in question. For A-F hexadecimal digits, use upper or lower case, for example, **0004f200106c.cfg**. The Ethernet address can be viewed using the **About** soft key during the auto-restart countdown of the BootROM or through the **Menu > Status > Platform > Phone** menu in the application. It is also printed on a label on the back of the phone. If this file cannot be downloaded, the phone will search for the default master configuration file (described next).
- Default master configuration file—For systems in which the configuration is identical for all phones (no per-phone **<Ethernet address>.cfg** files), the default master configuration file may be used to set the configuration for all phones. The file named **000000000000.cfg** (<12 zeros>.cfg) is the default master configuration file and it is recommended that one be present on the provisioning server. If a phone does not find its own **<Ethernet address>.cfg** file, it will use this one, and establish a baseline configuration. This file is part of the standard Polycom distribution of configuration files. It should be used as the template for the **<Ethernet address>.cfg** files.

  The default master configuration file, **000000000000.cfg**, for SIP 3.2.2 is shown below:

```
    <?xml version="1.0" standalone="yes"?>
    <!-- Default Master SIP Configuration File -->
    <!-- For information on configuring Polycom VoIP phones please
refer to the -->
    <!-- Configuration File Management white paper available from: -->
```

```
    <!--
http://www.polycom.com/common/documents/whitepapers/configuration_file
_management_on_soundpoint_ip_phones.pdf -->
    <!-- $RCSfile: 000000000000.cfg,v $  $Revision: 1.21 $ -->
    <APPLICATION APP_FILE_PATH="sip.ld" CONFIG_FILES="phone1.cfg,
sip.cfg" MISC_FILES="" LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY="" LICENSE_DIRECTORY="">
    <APPLICATION_SPIP300 APP_FILE_PATH_SPIP300="sip_212.ld"
CONFIG_FILES_SPIP300="phone1_212.cfg, sip_212.cfg"/>
    <APPLICATION_SPIP500 APP_FILE_PATH_SPIP500="sip_212.ld"
CONFIG_FILES_SPIP500="phone1_212.cfg, sip_212.cfg"/>
    <APPLICATION_SPIP301 APP_FILE_PATH_SPIP301="sip_313.ld"
CONFIG_FILES_SPIP301="phone1_313.cfg, sip_313.cfg"/>
    <APPLICATION_SPIP501 APP_FILE_PATH_SPIP501="sip_313.ld"
CONFIG_FILES_SPIP501="phone1_313.cfg, sip_313.cfg"/>
    <APPLICATION_SPIP600 APP_FILE_PATH_SPIP600="sip_313.ld"
CONFIG_FILES_SPIP600="phone1_313.cfg, sip_313.cfg"/>
    <APPLICATION_SPIP601 APP_FILE_PATH_SPIP601="sip_313.ld"
CONFIG_FILES_SPIP601="phone1_313.cfg, sip_313.cfg"/>
    <APPLICATION_SSIP4000 APP_FILE_PATH_SSIP4000="sip_313.ld"
CONFIG_FILES_SSIP4000="phone1_313.cfg, sip_313.cfg"/>
    </APPLICATION>
```

Master configuration files contain the following XML attributes:

- APP_FILE_PATH—The path name of the application executable. It can have a maximum length of 255 characters. This can be a URL with its own protocol, user name and password, for example http://usr:pwd@server/dir/sip.ld.
- CONFIG_FILES—A comma-separated list of configuration files. Each file name has a maximum length of 255 characters and the list of file names has a maximum length of 2047 characters, including commas and white space. Each configuration file can be specified as a URL with its own protocol, user name and password, for example ftp://usr:pwd@server/dir/phone2034.cfg.
- MISC_FILES—A comma-separated list of other required files. Dictionary resource files listed here will be stored in the phone's flash file system. So if the phone reboots at a time when the provisioning server is unavailable, it will still be able to load the preferred language.
- LOG_FILE_DIRECTORY—An alternative directory to use for log files if required. A URL can also be specified. This is blank by default.
- CONTACTS_DIRECTORY—An alternative directory to use for user directory files if required. A URL can also be specified. This is blank by default.
- OVERRIDES_DIRECTORY—An alternative directory to use for configuration overrides files if required. A URL can also be specified. This is blank by default.

- LICENSE_DIRECTORY — An alternative directory to use for license files if required. A URL can also be specified. This is blank by default.

Warning

The order of the configuration files listed in CONFIG_FILES is significant:

- The files are processed in the order listed (left to right).
- The same parameters may be included in more than one file.
- The parameter found first in the list of files will be the one that is effective.

This provides a convenient means of overriding the behavior of one or more phones without changing the baseline configuration files for an entire system.

For more information, refer to the "Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

If you have a requirement for different application loads on different phones on the same provisioning server, you can create a variable in the master configuration file that is replaced by the MAC address of each phone when it reboots. An example is shown below:

```
    <?xml version="1.0" standalone="yes"?>
    <!-- Default Master SIP Configuration File -->
    <!-- For information on configuring Polycom VoIP phones please
refer to the -->
    <!-- Configuration File Management white paper available from: -->
    <!--
http://www.polycom.com/common/documents/whitepapers/configuration_file
_management_on_soundpoint_ip_phones.pdf -->
    <!-- $RCSfile: 000000000000.cfg,v $  $Revision: 1.21 $ -->
    < APPLICATION APP_FILE_PATH="sip[MACADDRESS].ld"
CONFIG_FILES="phone1[MACADDRESS].cfg, sip.cfg" MISC_FILES=""
LOG FILE DIRECTORY="" OVERRIDES_DIRECTORY="" CONTACTS_DIRECTORY=""
LICENSE_DIRECTORY=""/>
```

If you have a requirement for separate application loads on different phones on the same provisioning server, you can modify the application that is loaded when each phone reboots. An example is below:

```
    <?xml version="1.0" standalone="yes"?>
    <!-- Default Master SIP Configuration File -->
    <!-- For information on configuring Polycom VoIP phones please
refer to the -->
    <!-- Configuration File Management white paper available from: -->
    <!--
http://www.polycom.com/common/documents/whitepapers/configuration_file
_management_on_soundpoint_ip_phones.pdf -->
    <!-- $RCSfile: 000000000000.cfg,v $  $Revision: 1.21 $ -->
    < APPLICATION APP_FILE_PATH="[PHONE_PART_NUMBER].sip.ld"
CONFIG_FILES="phone1.cfg, sip.cfg" MISC_FILES="" LOG FILE DIRECTORY=""
OVERRIDES_DIRECTORY="" CONTACTS_DIRECTORY="" LICENSE_DIRECTORY=""/>
```

You can also use the substitution strings PHONE_MODEL, PHONE_PART_NUMBER, MACADDRESS, and PHONE_MAC_ADDRESS in the master configuration file. For more information, refer to Product, Model, and Part Number Mapping on page C-26.

You can also direct phone upgrades to a software image and configuration files based on the phone model number and part number. All XML attributes can be modified in this manner. An example is below:

```
    <?xml version="1.0" standalone="yes"?>
    <!-- Default Master SIP Configuration File -->
    <!-- For information on configuring Polycom VoIP phones please
refer to the -->
    <!-- Configuration File Management white paper available from: -->
    <!--
http://www.polycom.com/common/documents/whitepapers/configuration_file
_management_on_soundpoint_ip_phones.pdf -->
    <!-- $RCSfile: 000000000000.cfg,v $  $Revision: 1.21 $ -->
    <APPLICATION APP_FILE_PATH="sip.ld" CONFIG_FILES="phone1.cfg,
sip.cfg" MISC_FILES="" LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY=""
CONTACTS_DIRECTORY="" LICENSE_DIRECTORY="" />
    <APPLICATION APP_FILE_PATH_SPIP300="SPIP300.sip.ld"
CONFIG_FILES_SPIP300="phone1_SPIP300.cfg, sip_SPIP300.cfg" />
    <APPLICATION APP_FILE_PATH_SPIP500="SPIP500.sip.ld"
CONFIG_FILES_SPIP500="phone1_SPIP500.cfg, sip_SPIP500.cfg" />
```

For more information:

- Refer to "Technical Bulletin 35311: Supporting SoundPoint IP 300, 301, 500, 501, 600, and 601 and SoundStation IP 4000 Phones with SIP 2.2.0 or SIP 3.2.0 and Later Releases" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .
- Refer to "Technical Bulletin 35361: Overriding Parameters in Master Configuration File on SoundPoint IP Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html.

Application Configuration

The configuration file **sip.cfg** contains SIP protocol and core configuration settings that would typically apply to an entire installation and must be set before the phones will be operational, unless changed through the local web server interface or local menu settings on the phone. These settings include the local port used for SIP signaling, the address and ports of a cluster of SIP application servers, voice codecs, gains, and tones, and other parameters.

Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first.

For more information, refer to the"Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

These parameters include:

- Protocol <voIpProt/>
- Dial Plan <dialplan/>
- Localization <lcl/>
- User Preferences <up/>
- Tones <tones/>
- Sampled Audio for Sound Effects <saf/>
- Sound Effects <se/>
- Voice Settings <voice/>
- Video Settings <video/>
- Basic TCP/IP <TCP_IP/>
- Web Server <httpd/>
- Call Handling Configuration <call/>
- Directory <dir/>
- Presence <pres/>
- Fonts <font/>
- Keys <key/>
- Backgrounds <bg/>
- Bitmaps <bitmap/>
- Indicators <ind/>
- Event Logging <log/>
- Security <sec/>
- License <license/>
- Provisioning <prov/>
- RAM Disk <ramdisk/>

- Request <request/>
- Feature <feature/>
- Resource <res/>
- Microbrowser <mb/>
- Applications <apps/>
- Peer Networking <pnet/>
- DNS Cache <dns/>
- Soft Keys <softkey/>
- LCD Power Saving <powerSaving/>

Protocol <volpProt/>

This attribute includes:

- Server <server/>
- SIP <SIP/>
- H.323 <H323/>

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.server.dhcp.available	0 or 1	0	If set to 1, check with the DHCP server for SIP server IP address. If set to 0, do not check with DHCP server.
voIpProt.server.dhcp.option	128 to 255	Null	Option to request from the DHCP server if `voIpProt.server.dhcp.available` = 1. There is no default value for this parameter, it must be filled in with a valid value. ***Note:*** *If the reg.x.server.y.address parameter in Registration <reg/> on page A-134 is non-Null, it takes precedence even if the DHCP server is available.*
voIpProt.server.dhcp.type	0 or 1	Null	If set to 0, IP request address. If set to 1, request string. Type to request from the DHCP server if voIpProt.server.dhcp.available = 1. There is no default value for this parameter, it must be filled in with a valid value.

Attribute	Permitted Values	Default	Interpretation
voIpProt.server.x.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server that accepts registrations. Multiple servers can be listed starting with x=1, 2, ... for fault tolerance. If port is 0 or Null: `If voIpProt.server.x.address` is a hostname and `voIpProt.server.x.transport` is set to DNSnaptr, do NAPTR then SRV lookups. If `voIpProt.server.x.transport` is set to TCPpreferred or UDPOnly, then use 5060 and don't advertise the port number in signaling. If `voIpProt.server.x.address` is an IP address, there is no DNS lookup and 5060 is used for the port but it is not advertised in signaling. If port is 1 to 65535: This value is used and it is advertised in signaling. ***Note:*** *If the reg.x.server.y.address parameter in Registration <reg/> on page A-134 is non-Null, all of the reg.x.server.y.xxx parameters will override the voIpProt.server parameters.* ***Note****: The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP.*
voIpProt.server.x.port	0, Null, 1 to 65535	Null	
voIpProt.server.x.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	If set to Null or DNSnaptr: If `voIpProt.server.x.address` is a hostname and voIpProt.server.x.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If `voIpProt.server.x.address` is an IP address, or a port is given, then UDP is used. If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails. If set to UDPOnly: Only UDP will be used. If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. If set to TCPOnly: Only TCP will be used.

Attribute	Permitted Values	Default	Interpretation
voIpProt.server.x.expires	positive integer, minimum 10	3600	The phone's requested registration period in seconds. ***Note:*** *The period negotiated with the server may be different. The phone will attempt to re-register at the beginning of the overlap period. For example, if "expires"=300 and "overlap"=5, the phone will re-register after 295 seconds (300-5).*
voIpProt.server.x.expires.overlap	positive integer, minimum 5, maximum 65535	60	The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone will try to re-register at half the expiration time returned by the server if that value is less than the configured overlap value.
voIpProt.server.x.register	0 or 1	1	If set to 0, calls can be routed to an outbound proxy without registration. Refer to `reg.x.server.y.register` in Registration <reg/>on page A-134. For more information, refer to "Technical Bulletin 5844: SIP Server Fallback Enhancements on SoundPoint IP Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html
voIpProt.server.x.retryTimeOut	Null or non-negative integer	0	If set to 0 or Null, use standard RFC 3261 signaling retry behavior. Otherwise retryTimeOut determines how often retries will be sent. Units = milliSeconds. (Finest resolution = 100ms).
voIpProt.server.x.retryMaxCount	Null or non-negative integer	3	If set to 0 or Null, 3 is used. retryMaxCount retries will be attempted before moving on to the next available server.
voIpProt.server.x.expires.lineSeize	positive integer, minimum 10	30	Requested line-seize subscription period.
voIpProt.server.x.lcs	0 or 1	0	This attribute overrides the `voIpProt.SIP.lcs` . If set to 1, the proprietary "epid" parameter is added to the From field of all requests to support Microsoft Live Communications Server.

Attribute	Permitted Values	Default	Interpretation
voIpProt.server.H323.x.address	dotted-decimal IP address or host name	Null	Address of the H.323 gatekeeper. ***Note**: Only one H.323 gatekeeper per phone is supported; if more than one is configured, only the first is used.*
voIpProt.server.H323.x.port	0, Null, 1 to 65535	Null	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. ***Note**: The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP.*
voIpProt.server.H323.x.expires	positive integer	Null	Desired registration period.

SIP <SIP/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.enable	0 or 1	1	Flag to determine whether or not the SIP protocol is used for call routing, dial plan, DTMF, and URL dialing. If set to 1, the SIP protocol is used.
voIpProt.SIP.local.port	0 to 65535	5060	Local port to be used for SIP signaling. Local port for sending and receiving SIP signaling packets. If set to 0 or Null, 5060 is used for the local port but it is not advertised in the SIP signaling. If set to some other value, that value is used for the local port and it is advertised in the SIP signaling.
voIpProt.SIP.useContactInReferTo	0 or 1	0	If set to 0, the “To URI” is used in the REFER. If set to 1, the “Contact URI” is used in the REFER.
voIpProt.SIP.useRFC2543hold	0 or 1	0	If set to 1, use SDP media direction attributes (such as a=sendonly) per RFC 3264 when initiating a call, otherwise use the obsolete c=0.0.0.0 RFC2543 technique. In either case, the phone processes incoming hold signaling in either format. ***Note:** `voIpProt.SIP.useRFC2543hold` is effective only when the call is initiated.*

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.useSendonlyHold	0 or 1	1	If set to 1, the phone will send a reinvite with a stream mode attribute of "sendonly" when a call is put on hold. This is the same as the previous behavior. If set to 0, the phone will send a reinvite with a stream mode attribute of "inactive" when a call is put on hold. ***NOTE:*** *The phone will ignore the value of this parameter if set to 1 when the parameter* `voIpProt.SIP.useRFC2543hold` *is also set to 1 (default is 0).*
voIpProt.SIP.lcs	0 or 1	0	If set to 1, the proprietary "epid" parameter is added to the From field of all requests to support Microsoft Live Communications Server.
voIpProt.SIP.ms-forking	0 or 1	0	If set to 0, support for MS-forking is disabled. If set to 1, support for MS-forking is enabled and the phone will reject all Instant Message INVITEs. This parameter is relevant for Microsoft Live Communications Server server installations. Note that if any end point registered to the same account has MS-forking disabled, all other end points default back to non-forking mode. Windows Messenger does not use MS-forking so be aware of this behavior if one of the end points is Windows Messenger.
voIpProt.SIP.sendCompactHdrs	0 or 1	0	If set to 0, SIP header names generated by the phone use the long form, for example 'From'. If set to 1, SIP header names generated by the phone use the short form, for example 'f'.
voIpProt.SIP.keepalive. sessionTimers	0 or 1	0	If set to 1, the session timer will be enabled. If set to 0, the session timer will be disabled, and the phone will not declare "timer" in "Support" header in INVITE. The phone will still respond to a re-INVITE or UPDATE. The phone will not try to re-INVITE or do UPDATE even if remote end point asks for it.
voIpProt.SIP.requestURI.E164. addGlobalPrefix	0 or 1	0	If set to 1, '+' global prefix is added to E.164 user parts in sip: URIs:.

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP. allowTransferOnProceeding	0 or 1	1	If set to 1, a transfer can be completed during the proceeding state of a consultation call. If set to 0, a transfer is not allowed during the proceeding state of a consultation call. If set to Null, the default value is used.
voIpProt.SIP.pingInterval	0 to 3600	0	The number in seconds to send "PING" message. This feature is disabled by default.
voIpProt.SIP.useContactInReferTo	0 or 1	0	If set to 1, the Contact URI is used. If set to 0, the TO URI is used (previous behavior).
voIpProt.SIP.serverFeatureControl.cf	0 or 1	Null	If set to 1, server-based call forwarding is enabled. The call server has control of call forwarding. If set to 0 or Null, server-based call forwarding is not enabled. This is the old behavior.
voIpProt.SIP.serverFeatureControl.localProcessing.cf	0 or 1	Null	If set to 0 and `voIpProt.SIP.serverFeatureControl.cf="1"`, the phone will not perform local Call Forward behavior. If set to 1 or Null, the phone will perform local Call Forward behavior on all calls received.
voIpProt.SIP.serverFeatureControl. dnd	0 or 1	Null	If set to 1, server-based DND is enabled. The call server has control of DND. If set to 0 or Null, server-based DND is not enabled. This is the old behavior.
voIpProt.SIP.serverFeatureControl. localProcessing.dnd			If set to 0 and `voIpProt.SIP.serverFeatureControl.dnd="1"`, the phone will not perform local DND call behavior. If set to 1 or Null, the phone will perform local DND call behavior on all calls received.

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.authOptimizedInFailover	0,1	0	If set to 1, when failover occurs, the first new SIP request is sent to the server that sent the proxy authentication request. If set to 0, when failover occurs, the first new SIP request is sent to the server with the highest priority in the server list. If `reg.x.auth.optimizedInFailover` set to Null, this attribute is checked. If `voIpProt.SIP.authOptimizedInFailover` is Null, then this feature is disabled. If both attributes are set, the value of `reg.x.auth.optimizedInFailover` takes precedence.
voIpProt.SIP.csta	0 or 1	0	If set to 1, uaCSTA is enabled.
voIpProt.SIP.strictLineSeize	0 or 1	Null	If set to 1, forces the phone to wait for 200 OK response when receiving a TRYING notify. If set to 0 or Null, this is old behavior.
voIpProt.SIP.strictUserValidation	0 or 1	Null	If set to 1, forces the phone to match user portion of signaling exactly. If set to 0 or Null, phone will use first registration if the user part does not match any registration.
voIpProt.SIP.lineSeize.retries	3 to 10	10	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA).
voIpProt.SIP.header.diversion.enable	0 or 1	0	If set to 1, the diversion header is displayed if received. If set to 0 or Null, the diversion header is not displayed.
voIpProt.SIP.header.diversion.list.useFirst	0 or 1	1	If set to 1 or Null, the first diversion header is displayed. If set to 0, the last diversion header is displayed.

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.header.warning.codes.accept	comma separated list	Null	A list of accepted warning codes. If set to Null, all codes are accepted. Only codes between 300 and 399 are supported. For example, if you want to accept only codes 325 to 330: `voIpProt.SIP.header.warning.codes.accept = 325,326,327,328,329,330` Text will be shown in the appropriate language. For more information, refer to `lcl.ml.lang.tags.x` in Multilingual <ml/> on page A-28.
voIpProt.SIP.header.warning.enable	0 or 1	0	If set to 1, the warning header is displayed if received. If set to 0 or Null, the warning header is not displayed.
voIpProt.SIP.acd.signalingMethod	0 or 1	0	If set to 0 or Null, the 'SIP-B' signaling is supported. (This is the older ACD functionality.) If set to 1, the feature synchronization signaling is supported. (This is the new ACD functionality.)
voIpProt.SIP.tcpFastFailover	0 or 1	Null	If set to 1, failover occurs based on the values of `reg.x.server.y.retryMaxCount` `voIpProt.server.x.retryTimeOut`. If set to 0, this is old behavior. If `reg.x.tcpFastFailover` is Null, this attribute is checked. If `voIpProt.SIP.tcpFastFailover` is Null, then this feature is disabled. If both attributes are set, the value of `reg.x.tcpFastFailover` takes precedence.

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.strictReplacesHeader	0 or 1	Null	This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources. If set to 1 or Null, the phone requires call-id,to-tag, and from-tag to perform a directed call-pickup when call.directedCallPickupMethod is configured as "native". If set to 0, all that is required to perform the directed call pick-up is a call-id.
voIpProt.SIP.use486forReject	0 or 1	0	If set to1 and the phone is indicating a ringing inbound call appearance, phone will transmit a 486 response to the received INVITE when the **Reject** soft key is pressed. If set to 0, no 486 response is transmitted.
voIpProt.SIP.dtmfViaSignaling. rtc2976	0 or 1	0	If set to 1, DTMF digit information is sent in RFC2976 SIP INFO packets during a call. If set to 0 or Null, no DTMF digit information is sent.

This attribute also includes:

- SDP <SDP/>
- Outbound Proxy <outboundProxy/>
- Alert Information <alertInfo/>
- Request Validation <requestValidation/>
- Special Events <specialEvent/>
- Conference Setup <conference/>
- Dialog <dialog/>
- Connection Reuse<dialog/>
- Music on Hold <musicOnHold/>
- Compliance <compliance/>

SDP <SDP/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SDP. useLegacyPayloadTypeNegotiation	0 or 1	Null	If set to 1, the phone transmits and receives RTP using the payload type identified by the first codec listed in the SDP of the codec negotiation answer. If set to 0 or Null, RFC 3264 is followed for transmit and receive RTP payload type values.
voIpProt.SDP.answer. useLocalPreferences	0 or 1	0	If set to 1, the phones uses its own preference list when deciding which codec to use rather than the preference list in the offer. If set to 0, it is disabled. ***Note:*** *If a H.323 call from a Polycom VVX 1500 selects a lower-quality codec (H.261) but the called device also support H.264, this parameter should be enabled to resolve the situation.*
voIpProt.SDP.iLBC.13_33kbps. includeMode	0 or 1	Null	If set to 1 or Null, the phone should include the mode=30 FMTP attribute in SDP offers: • If `voice.codecPref.iLBC.13_33kbps` is set and `voice.codecPref.iLBC.15_2kbps` is Null. • If `voice.codecPref.iLBC.13_33kbps` and `voice.codecPref.iLBC.15_2kbps` are both set, but iLBC 13.33 kbps codec is set to a higher preference. If set to 0, the phone should not include the mode=30 FTMP attribute in SDP offers even if iLBC 13.33 kbps codec is being advertised. Refer to Codec Preferences <codecPref/> on page A-45.
voIpProt.SDP.early.answerOrOffer	0 or 1	Null	If set to 1, an SDP offer or answer is generated in a provisional reliable response and PRACK request and response. If set to 0, an SDP offer or answer is not generated. ***Note****: An SDP offer or answer is not generated if the user (reg.x) is configured for the Music On Hold. Refer to Music on Hold <musicOnHold/> on page A-21.*

Outbound Proxy <outboundProxy/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.outboundProxy.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server to which the phone shall send all requests.
voIpProt.SIP.outboundProxy.port	0 to 65535	5060	
voIpProt.SIP.outboundProxy.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	If set to Null or DNSnaptr: If voIpProt.SIP.outboundProxy.address is a hostname and `voIpProt.SIP.outboundProxy.port` is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If `voIpProt.SIP.outboundProxy.address` is an IP address, or a port is given, then UDP is used. If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails. If set to UDPOnly: Only UDP will be used. If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. If set to TCPOnly: Only TCP will be used. ***NOTE:*** *TLS is not supported on SoundPoint IP 300 and 500 phones.*

Alert Information <alertInfo/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.alertInfo.x.value	string to compare against the value of Alert-Info headers in INVITE requests	Null	Alert-Info fields from INVITE requests will be compared against as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class (refer to Ring type <rt/> on page A-43) will be applied.
voIpProt.SIP.alertInfo.x.class	positive integer	Null	

Request Validation <requestValidation/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.requestValidation.x.request	One of: “INVITE”, “ACK”, “BYE” “REGISTER”, “CANCEL”, “OPTIONS”, “INFO”, “MESSAGE”, “SUBSCRIBE” “NOTIFY”, “REFER”, “PRACK”, or “UPDATE”	Null	Sets the name of the method for which validation will be applied. ***WARNING**: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases, therefore, use it wisely.*
voIpProt.SIP.requestValidation.x.method	Null or one of: “source”, “digest” or “both”/”all”	Null	If Null, no validation is done. Otherwise this sets the type of validation performed for the request: *source*: ensure request is received from an IP address of a server belonging to the set of target registration servers; *digest:* challenge requests with digest authentication using the local credentials for the associated registration (line); *both* or *all*: apply both of the above methods

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.requestValidation.x.request.y.event	A valid string	Null	Determines which events specified with the Event header should be validated; only applicable when `voIpProt.SIP.requestValidation.x.request` is set to "SUBSCRIBE" or "NOTIFY". If set to Null, all events will be validated.
voIpProt.SIP.requestValidation.digest.realm	A valid string	Polycom SPIP	Determines string used for Realm.

Special Events <specialEvent/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.specialEvent.lineSeize.nonStandard	0 or 1	1	If set to 1, process a 200 OK response for a line-seize event SUBSCRIBE as though a line-seize NOTIFY with Subscription State: active header had been received, this speeds up processing.
voIpProt.SIP.specialEvent.checkSync.alwaysReboot	0 or 1	0	If set to 1, always reboot when a NOTIFY message is received from the server with event equal to check-sync. If set to 0, only reboot if any of the files listed in **<MAC-address>.cfg** have changed on the FTP server when a NOTIFY message is received from the server with event equal to check-sync.

Conference Setup <conference/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.conference.address	ASCII string up to 128 characters long	Null	If Null, conferences are set up on the phone locally. If set to some value, conferences are set up by the server using the conferencing agent specified by this address. The acceptable values depend on the conferencing server implementation policy.

Dialog <dialog/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.dialog.useSDP	0 or 1	0	If set to 0, new dialog event package draft is used (no SDP in dialog body). If set to 1, for backwards compatibility, use this setting to send SDP in dialog body.
voIpProt.SIP.dialog.usePvalue	0 or 1	0	If set to 0, phone uses "pval" field name in Dialog. This obeys the draft-ietf-sipping-dialog-package-06.txt draft. If set to 1, phone uses a field name of "pvalue".

Connection Reuse<dialog/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.connectionReuse. useAlias	0 or 1	0	If set to 0, this is the old behavior. If set to 1, phone uses the connection reuse draft which introduces "alias".

Music on Hold <musicOnHold/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.musicOnHold.uri	string	Null	A URI that provides the media stream to play for the remote party on hold. If `reg.x.musicOnHold` is set to Null, this attribute is checked. ***Note**: The SIP URI parameter `transport` is supported when configured with the values of UDP, TCP, or TLS.*

Compliance <compliance/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.compliance.RFC3261.validate.contentLanguage	0 or 1	Null	If set to 1, validation of the SIP header content language is enabled. If set to 0 or Null, validation is disabled.
voIpProt.SIP.compliance.RFC3261.validate.uriScheme	0 or 1	Null	If set to 1 or Null, validation of the SIP header URI scheme is enabled. If set to 0, validation is disabled.

H.323 <H323/>

Note At this time, this attribute is used with the Polycom VVX 1500 phone only.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voIpProt.H323.enable	0 or 1	0	Flag to determine whether or not the H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing. If set to 1, the H.323 protocol is used.
voIpProt.H323.local.port	0 to 65535	1720	Local port to be used for H.323 signaling. Local port for sending and receiving H.323 signaling packets. If set to 0 or Null, 1720 is used for the local port but it is not advertised in the H.323 signaling. If set to some other value, that value is used for the local port and it is advertised in the H.323 signaling.
voIpProt.H323.autoGatekeeperDiscovery	0 or 1	0	If set to 1, the phone will attempt to discover an H.323 gatekeeper address via the standard multicast technique, provided that a statically configured gatekeeper address is not available. If set to 0, the phone will no send out any gatekeeper discovery messages.
voIpProt.H323.dtmfViaSignaling.enabled	0 or 1	1	If set to 1, the phone will use the H.323 signaling channel for DTMF key press transmission.

Attribute	Permitted Values	Default	Interpretation
voIpProt.H323.dtmfViaSignaling. H245alphanumericMode	0 or 1	1	If set to 1, the phone will support H.245 signaling channel alphanumeric mode DTMF transmission. ***Note**: If both alphanumeric and signal mode are enabled, the phone will give preference to sending DTMF in alphanumeric mode where there is the possibility of sending in both modes.*
voIpProt.H323.dtmfViaSignaling. H245signalMode	0 or 1	1	If set to 1, the phone will support H.245 signaling channel signal mode DTMF transmission.

Dial Plan <dialplan/>

Note The dial plan is not applied against Placed Call List, VoiceMail, last call return, remote control dialed numbers, and on-hook dialing.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.applyToCallListDial	0 or 1	0	This attribute covers dialing from Received Call List and Missed Call List including dialing from Edit or Info sub- menus. If set to 0, the digit map replacement operations are not applied against the dialed number. if set to 1, the digit map replacement operations are applied against the dialed number.
dialplan.applyToDirectoryDial	0 or 1	0	This attribute covers dialing from Directory as well as Speed Dial List. Value interpretation is the same as for `dialplan.applyToCallListDial`. ***Note:** An Auto Call Contact number is considered a dial from directory.*
dialplan.applyToUserDial	0 or 1	1	This attribute covers the case when the user presses the **Dial** soft key to send dialed number when in idle state display. Value interpretation is the same as for `dialplan.applyToCallListDial`.

Attribute	Permitted Values	Default	Interpretation
dialplan.applyToUserSend	0 or 1	1	This attribute covers the case when the user presses the **Send** soft key to send the dialed number. Value interpretation is the same as for `dialplan.applyToCallListDial`.
dialplan.impossibleMatchHandling	0, 1 or 2	0	Affects digits entered while in dial mode. For example, the digits are affected after a user has picked up the handset, headset, or pressed the dial key, and not when hot dialing, contact dialing, or call list dialing. If set to 0, the digits entered up to and including the point where an impossible match occurred are sent to the server immediately. If set to 1, give reorder tone. If set to 2, allow user to accumulate digits and dispatch call manually with the Send soft key.
dialplan.removeEndOfDial	0 or 1	1	If set to 1, strip trailing # digit from digits sent out.
dialplan.applyToTelUriDial	0 or 1	1	A flag to determine if the dial plan applies to uses of the tel:// URI. If set to 1 or Null, the dial plan applies. If set to 0, the dial plan does not apply.
dialplan.applyToRemoteDialing	0 or 1	0	A flag to determine if the dial plan applies to for calls made through the Polycom HDX or SoundStructure systems. If set to 1, the dial plan applies. If set to 0 or Null, the dial plan does not apply.

This attributes also includes:

- Digit Map <digitmap/>
- Routing <routing/>

Digit Map <digitmap/>

A digit map is defined either by a "string" or by a list of strings. Each string in the list is an alternative numbering scheme, specified either as a set of digits or timers, or as an expression over which the gateway will attempt to find a shortest possible match.

Digit map extension letter “R” indicates that certain matched strings are replaced. Digit map timer letter “T” indicates a timer expiry. Digit map protocol letters “S” and “H” indicate the protocol to use when placing a call. The following examples shows the semantics of the syntax:

- `R9RRxxxxxxx`—Remove 9 at the beginning of the dialed number
  - For example, if a customer dials 914539400, the first 9 is removed when the call is placed.
- `RR604Rxxxxxxx`—Prepend 604 to all seven digit numbers
  - For example, if a customer dials 4539400, 604 is added to the front of the number, so a call to 6044539400 is placed.
- `R9R604Rxxxxxxx`—Replaces 9 with 604
- `R999R911R`—Convert 999 to 911
- `xxR601R600Rxx`—When applied on 1160122 gives 1160022
- `xR60xR600Rxxxxxxx`—Any 60x will be replaced with 600 in the middle of the dialed number that matches
  - For example, if a customer dials 16092345678, a call is placed to 16002345678.
- 911xxx.T— A period (".") which matches an arbitrary number, including zero, of occurrences of the preceding construct
  - For example:
    911123 with waiting time to comply with T is a match
    9111234 with waiting time to comply with T is a match
    91112345 with waiting time to comply with T is a match
    and the number can grow indefinitely given that pressing the next digit takes less than T.
- 0xxxS|33xxH—All four digit numbers starting with a 0 are placed using the SIP protocol, whereas all four digit numbers starting with 33 are placed using the H.323 protocol.

Note

Only Polycom VVX 1500 phones will match the “H”. It is ignored by all other phones and the user will need to press the **Send** soft key to complete dialing. For example, if the digit map is “33xxH”, the result is as follows:

- If a Polycom VVX 1500 user dials “3302” on an H.323 or dual protocol line, the call will be placed after the user dials the last digit.
- If a SoundPoint IP 650 user dials “3307”, the user must press the **Send** soft key to complete dialing.

The following guidelines should be noted:

- The letters (x, T, R, S, H) are case sensitive.
- You must use only *, #, +, or 0-9 between second and third R

- If a digit map does not comply, it is not included in the digit plan as a valid one. That is, no matching is done against it.
- There is no limitation on the number of R triplet sets in a digit map. However, a digit map that contains less than full number of triplet sets (for example, a total of 2Rs or 5Rs) is considered an invalid digit map.
- Using T in the left part of RRR syntax is not recommended. For example, R0TR322R should be avoided.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.digitmap	string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. String is limited to 768 bytes and 30 segments; a comma is also allowed; when reached in the digit map, a comma will turn dial tone back on;'+' is allowed as a valid digit; extension letter 'R' is used as defined above.	[2-9]11|0T| +011xxx.T| 0[2-9]xxxxxxxxx| +1[2-9]xxxxxxxx| [2-9]xxxxxxxxx| [2-9]xxxT	When this attribute is present, number-only dialing during the setup phase of new calls will be compared against the patterns therein and if a match is found, the call will be initiated automatically eliminating the need to press Send. Attributes `dialplan.applyToCallListDial`, `dialplan.applyToDirectoryDial`, `dialplan.applyToUserDial`, and `dialplan.applyToUserSend` control the use of match and replace in the dialed number in the different scenarios.
dialplan.digitmap.timeOut	string of positive integers separated by '|'	3 | 3 | 3 | 3 | 3 | 3	Timeout in seconds for each segment of digit map. ***Note:*** *If there are more digit maps than timeout values, the default value of 3 will be used. If there are more timeout values than digit maps, the extra timeout values are ignored.*

Routing <routing/>

This attribute allows the user to create a specific routing path for outgoing SIP calls independent of other "default" configurations.

This attribute also includes:

- Server <server/>

- Emergency <emergency/>

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.routing.server.x. address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server that will be used for routing calls. Multiple servers can be listed starting with x=1, 2, ... for fault tolerance.
dialplan.routing.server.x.port	1 to 65535	5060	

Emergency <emergency/>

In the following attributes, *x* is the index of the emergency entry description and *y* is the index of the server associated with emergency entry *x*. For each emergency entry (index *x*), one or more server entries (indexes (*x*,*y*)) can be configured. *x* and *y* must both use sequential numbering starting at 1.

Attribute	Permitted Values	Default	Interpretation
dialplan.routing.emergency.x. value	Single entry representing a SIP URL	for x =1, value = “911”, Null for all others	This determines the URLs that should be watched for. When one of these defined URLs is detected as having been dialed by the user, the call will automatically be directed to the defined emergency server.
dialplan.routing.emergency.x. server.y	positive integer	for x=1, y =1, Null for all others	Index representing the server defined in Server <server/> on page A-27 that will be used for emergency routing.

Localization <lcl/>

The phone has a multilingual user interface. It supports both North American and international time and date formats. The call progress tones can also be customized. For more information, refer to Chord-Sets <chord/> on page A-36, and Call Progress Patterns on page A-40.

This attribute includes:

- Multilingual <ml/>
- Date and Time <datetime/>

Multilingual <ml/>

The multilingual feature is based on string dictionary files downloaded from the provisioning server. These files are encoded in standalone XML format. Several eastern European and Asian languages are included with the distribution.

Attribute	Permitted Values	Interpretation
lcl.ml.lang	Null OR An exact match for one of the label names stored in `lcl.ml.lang.menu.x.label`.	If Null, the default internal language (US English) will be used, otherwise, the language to be used may be specified in the format of `lcl.ml.lang.menu.x.label`. For example, to get the phone to boot up in German: `lcl.ml.lang = "Deutsch (de-de)"`.

lcl.ml.lang.menu.x
lcl.ml.lang.menu.x.label
	String in the format *language_region*	Multiple `lcl.ml.lang.menu.x attributes` are supported - as many languages as are desired. However, the `lcl.ml.lang.menu.x` attributes must be sequential (`lcl.ml.lang.menu.1`, `lcl.ml.lang.menu.2`, `lcl.ml.lang.menu.3`, ..., `lcl.ml.lang.menu.N`) with no gaps and the strings must exactly match a folder name under the SoundPointIPLocalization folder on the provisioning server for the phone to be able to locate the dictionary file.
		For example:
		`lcl.ml.lang.menu.8="German_Germany"`
		`lcl.ml.lang.menu.8.label="Deutsch (de-de)"`

lcl.ml.lang.clock.x.24HourClock	0,1	If attribute present, overrides `lcl.datetime.time.24HourClock.` If 1, display time in 24-hour clock mode rather than am/pm.
lcl.ml.lang.clock.x.format	string which includes 'D', 'd' and 'M' and two optional commas	If attribute present, overrides lcl.datetime.date.format; D = day of week d = day M = month Up to two commas may be included. For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time. For example: "D,,dM" is illegal.

Attribute	Permitted Values	Interpretation
lcl.ml.lang.clock.x.longFormat	0 or 1	If attribute present, overrides `lcl.datetime.date.longFormat`. If 1, display the day and month in long format (Friday/November), otherwise use abbreviations (Fri/Nov).
lcl.ml.lang.clock.x.dateTop	0 or 1	If attribute present, overrides `lcl.datetime.date.dateTop`. If 1, display date above time, otherwise display time above date.
lcl.ml.lang.y.list	"All" or a comma-separated list	A list of the languages supported on hardware platform 'y' where 'y' can be IP_500, IP_600, or IP4000. The IP_500 platform does not support any Asian languages. The IP_4000 platform does not support Slovenian.
lcl.ml.lang.tags.x	string in the format *language_region, language; preference level*	The format is: • The first two letters are the ISO-639 language abbreviation. • The next two letters are the ISO-3166 country code. • The next two letters are the ISO-639 language abbreviation. • The remainder of the string is the preference level for the display of the language, or English if the language is not available For example: `lcl.ml.lang.tags.1 = "zh-cn,zh;q=0.9,en;q=0.8"` For more information, refer to the Accept-Language header definition in the HTTP RFC 2616 at http://www.w3.org/Protocols/rfc2616/rfc2616-sec14.html#sec14.4

To add new languages to those included with the distribution:

1. Create a new dictionary file based on an existing one.
2. Change the strings making sure to encode the XML file in UTF-8 but also ensuring the UTF-8 characters chosen are within the Unicode character ranges indicated in the tables below.
3. Place the file in an appropriately named folder according to the format *language_region* parallel to the other dictionary files under the SoundPointIPLocalization folder on the provisioning server.

4. Add a `lcl.ml.lang.clock.menu.x` attribute to the configuration file.
5. Add `lcl.ml.lang.clock.x.24HourClock`, `lcl.ml.lang.clock.x.format`, `lcl.ml.lang.clock.x.longFormat` and `lcl.ml.lang.clock.x.dateTop` attributes and set them according to the regional preferences.
6. (Optional) Set `lcl.ml.lang` to be the new *language_region* string.

Basic character support includes the following Unicode character ranges	
Name	**Range**
C0 Controls and Basic Latin	U+0000 - U+007F
C1 Controls and Latin-1 Supplement	U+0080 - U+00FF
Cyrillic (partial)	U+0400 - U+045F

Extended character support available on SoundPoint IP 600 and SoundStation IP 4000 and 7000 platforms includes the following Unicode character ranges	
Name	**Range**
CJK Symbols and Punctuation	U+3000 - U+303F
Hiragana	U+3040 - U+309F
Katakana	U+30A0 - U+30FF
Bopomofo	U+3100 - U+312F
Hangul Compatibility Jamo	U+3130 - U+318F
Bopomofo Extended	U+31A0 - U+31BF
Enclosed CJK Letters and Months	U+3200 - U+327F
CJK Compatibility	U+3300 - U+33FF
CJK Unified Ideographs	U+4E00 - U+9FFF
Hangul Syllables	U+AC00 - U+D7A3
CJK Compatibility Ideographs	U+F900 - U+FAFF
CJK Half-width forms	U+FF00 - U+FFFF

Note Within a Unicode range, some characters may not be supported due to their infrequent usage

Date and Time <datetime/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Interpretation
lcl.datetime.time.24HourClock	0,1	If set to 1, display time in 24-hour clock mode rather than a.m./p.m.
lcl.datetime.date.format	string which includes ‘D’, ‘d’ and ‘M’ and two optional commas	Controls format of date string. D = day of week d = day M = month Up to two commas may be included. For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time. For example: “D,,dM” is illegal.
lcl.datetime.date.longFormat	0,1	If set to 1, display the day and month in long format (Friday/November), otherwise, use abbreviations (Fri/Nov).
lcl.datetime.date.dateTop	0 or 1	If set to 1, display date above time else display time above date.

User Preferences <up/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
up.headsetMode	0 or 1	0	If set to 1, the headset will be selected as the preferred transducer after its first use until the headset key is pressed again; otherwise, hands-free will be selected preferentially over the headset.
up.useDirectoryNames	0 or 1	0	If set to 1, the name fields of the local contact directory entries which match incoming calls will be used for caller identification display and in the call lists instead of the name provided through network signaling. ***Note**: There is no matching of outgoing calls. There is no matching to corporate directory entries.*
up.oneTouchVoiceMail	0 or 1	0	If set to 1, the voice mail summary display is bypassed and voice mail is dialed directly (if configured).

Attribute	Permitted Values	Default	Interpretation
up.welcomeSoundEnabled	0 or 1	1	If set to 1, play welcome sound effect after a reboot.
up.welcomeSoundOnWarmBootEnabled	0 or 1	0	If set to 1, play welcome sound effect on warm and cold boots. If set to 0, only a cold reboot will trigger the welcome sound effect.
up.localClockEnabled	0 or 1	1	If set to 1, display the date and time on the idle display.
up.backlight.onIntensity	0 (off), 1 (low), 2 (medium), 3 (high)	Null	This parameter controls the intensity of the LCD backlight when it turns on during normal use of the phone. The default value is medium.
up.backlight.idleIntensity	0 (off), 1 (low), 2 (medium), 3 (high)	Null	This parameter controls the intensity of the LCD backlight when the phone is idle. The default value is low. ***Note:*** *If idleIntensity is set higher than onIntensity, it will be replaced with the onIntensity value*.
up.toneControl.bass	-4 to 4, Null	0	Bass equalization control. Each step is an increment of 1 dB at 225 kHz and 2 dB < 225 Hz.
up.toneControl.treble	-4 to 4, Null	0	Treble equalization control. Each step is an increment of 1 dB at 3.7 kHz and 2 dB > 10 kHz.
up.audioSetup.auxInput	0 - Other Input, 1 - Polycom Wireless Mic, 2 - off	Null	Auxiliary audio input. If set to Null, default value is 2.
up.audioSetup.auxOutput	0 - Other Input, 1 - Polycom Wireless Mic, 2 - off	Null	Auxiliary audio output. If set to Null, default value is 2.
up.idleTimeout	positive integer, seconds	Null	Timeout for the idle display or default call handling display. If set to 0, there is no timeout. If set to Null, the default timeout of 40 seconds is used. If set to value greater than 0, the timeout is for that number of seconds (maximum 65535).

Attribute	Permitted Values	Default	Interpretation
up.mwiVisible	0 or 1	0	If set is 0 or Null, the incoming MWI notifications for lines where the MWI callback mode is disabled (`msg.mwi.x.callBackMode` is set to 0) are ignored, and do not appear in the message retrieval menus. If set to 1, the MWI for lines whose MWI is disabled is displayed (pre-SIP 2.1 behavior), even though MWI notifications have been received for those lines.
up.handsfreeMode	0 or 1	1	If set to 1 or Null, hands-free speakerphone is enabled. If set to 0, hands-free speakerphone is disabled.
up.numberFirst CID	0 or 1	0	If set to 0 or Null, caller ID display will show caller's name first. If set to 1, caller ID display will show caller's number first.
up.idleBrowser.enabled	0 or 1	Null	A flag to determine whether or not the background takes priority over the idle browser. Used in conjunction with `up.prioritizeBackground.enable` .
up.prioritizeBackgroundMenuItem .enable	0 or 1	1	If set to 1, the "Prioritize Background" menu is available to the user. The user can then decide whether or not the background takes priority over the idle browser. Used in conjunction with `up.idleBrowser.enabled` .
up.screenCapture.enabled	0 or 1	0	A flag to determine whether or not the user can get a screen capture of the current screen shown on a phone. The flag is cleared when the phone reboots. If set to 1, the "Screen Capture" menu is available to the user. Refer to Capturing Phone's Current Screen on page C-30.
up.manualProtocolRouting	0 or 1	1	If set to 1, the user is presented with protocol routing choices when a call could be placed with more than one protocol from its current context. The user must choose between SIP and H.323 to place a call. ***Note**: This parameter is supported on the Polycom VVX 1500 only.*

Attribute	Permitted Values	Default	Interpretation
up.manualProtocolRouting.softKeys	0 or 1	1	If set to 1 and `up.manualProtocolRouting` is set to 1, soft keys are used to provide the user with a routing choice. If set to 0, a routing confirmation dialog is presented with a choice for each possible routing. ***Note**: This parameter is supported on the Polycom VVX 1500 only.*
up.callTypePromptPref	0 or 1	1	A flag to determine which interface is used to place calls. If set to 1 or Null, the voice interface is used. If set to 0, the video interface is used to place calls during hot-dialing or if the use presses the off-hook key on the SoundStation IP 7000 phone. ***Note**: This parameter is supported on the SoundStation IP 7000 only.*
up.enableCallTypePrompt	0 or 1	1	Enable the call type prompt. If set to 1 or Null, the call type prompt is enabled. If set to 0, the call type prompt is disabled. ***Note**: This parameter is supported on the SoundStation IP 7000 only.*

Tones <tones/>

This attribute describes configuration items for the tone resources available in the phone.

This attribute includes:

- Dual Tone Multi-Frequency <DTMF/>
- Chord-Sets <chord/>

Dual Tone Multi-Frequency <DTMF/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
tone.dtmf.level	-33 to -3	-15	Level of the high frequency component of the DTMF digit measured in dBm0; the low frequency tone will be two dB lower.
tone.dtmf.onTime	positive integer	50	When a sequence of DTMF tones is played out automatically, this is the length of time in milliseconds the tones will be generated for; this is also the minimum time the tone will be played for when dialing manually (even if key press is shorter).
tone.dtmf.offTime	positive integer	50	When a sequence of DTMF tones is played out automatically, this is the length of time in milliseconds the phone will pause between digits; this is also the minimum inter-digit time when dialing manually.
tone.dtmf.chassis.masking	0 or 1	0	If set to 1, DTMF tones will be substituted with a non-DTMF pacifier tone when dialing in hands-free mode. This prevents DTMF digits being broadcast to other surrounding telephony devices or being inadvertently transmitted in-band due to local acoustic echo. ***Note:*** *tone.dtmf.chassis.masking should only be enabled when tone.dtmf.viaRtp is disabled.*
tone.dtmf.stim.pac.offHookOnly	0 or 1	0	Not currently used.

Attribute	Permitted Values	Default	Interpretation
tone.dtmf.viaRtp	0 or 1	1	If set to 1, encode DTMF in the active RTP stream, otherwise, DTMF may be encoded within the signaling protocol only when the protocol offers the option. ***Note:*** *tone.dtmf.chassis.masking should be enabled when tone.dtmf.viaRtp is disabled.*
tone.dtmf.rfc2833Control	0 or 1	1	If set to 1, the phone will indicate a preference for encoding DTMF through RFC 2833 format in its Session Description Protocol (SDP) offers by showing support for the phone-event payload type; this does not affect SDP answers, these will always honor the DTMF format present in the offer since the phone has native support for RFC 2833.
tone.dtmf.rfc2833Payload	96-127	127	The phone-event payload encoding in the dynamic range to be used in SDP offers.

Chord-Sets <chord/>

Chord-sets are the building blocks of sound effects that use synthesized rather than sampled audio (most call progress and ringer sound effects). A chord-set is a multi-frequency note with an optional on/off cadence. A chord-set can contain up to four frequency components generated simultaneously, each with its own level.

There are three blocks of chord sets:

- callProg (used for call progress sound effect patterns)
- ringer
- misc (miscellaneous)

All three blocks use the same chord set specification format.

In the following table, *x* is the chord-set number and *cat* is one of callProg, ringer, or misc.

Attribute	Permitted Values	Interpretation
tone.chord.cat.x.freq.y	0-1600	Frequency for this component in Hertz; up to four chord-set components can be specified (y=1, 2, 3, 4).
tone.chord.cat.x.level.y	-57 to 3	Level of this component in dBm0.
tone.chord.cat.x.onDur	positive integer	On duration in milliseconds, 0=infinite.
tone.chord.cat.x.offDur	positive integer	Off duration in milliseconds, 0=infinite.
tone.chord.cat.x.repeat	positive integer	Specifies how many times the ON/OFF cadence is repeated, 0=infinite.

Sampled Audio for Sound Effects <saf/>

The following sampled audio WAVE file (.wav) formats are supported:

- mono 8 kHz G.711 μ-Law
- G.711 A-Law
- L16/16000 (16-bit, 16 kHz sampling rate, mono)
- L16/32000 (16-bit, 32 kHz sampling rate, mono)
- L16/48000 (16-bit, 48 kHz sampling rate, mono)

Note L16/32000 and L16/48000 are supported on SoundStation IP 6000 and 7000 phones.

The phone uses built-in wave files for some sound effects. The built-in wave files can be replaced with files downloaded from the provisioning server or from the Internet, however, these are stored in volatile memory so the files will need to remain accessible should the phone need to be rebooted. Files will be truncated to a maximum size of 300 kilobytes.

In the following table, *x* is the sampled audio file number.

Attribute	Permitted Values	Interpretation
saf.x	Null OR valid path name OR an RFC 1738-compliant URL to a HTTP, FTP, or TFTP wave file resource. ***Note:*** *Refer to the above wave file format restrictions.*	If Null, the phone will use a built-in file. If set to a path name, the phone will attempt to download this file at boot time from the provisioning server. If set to a URL, the phone will attempt to download this file at boot time from the Internet. ***Note:*** *A TFTP URL is expected to be in the format: tftp://<host>/[pathname]<filename>, for example: tftp://somehost.example.com/sounds/example.wav .*

The following table defines the default usage of the sampled audio files with the phone:

Sampled Audio File	Default use within phone (pattern reference)
1	Ringer 12 (se.pat.misc.7)
2	Ringer 13 (se.pat.ringer.13)
3	Ringer 14 (se.pat.ringer.14)
4	Ringer 15 (se.pat.ringer.15)
5	Ringer 16 (se.pat.ringer.16)
6	Ringer 17 (se.pat.ringer.17)
7	Ringer 18 (se.pat.ringer.18)
8	Ringer 19 (se.pat.ringer.19)
9	Ringer 20 (se.pat.ringer.20)
10	Ringer 21 (se.pat.ringer.21)
11	Ringer 22 (se.pat.ringer.22)
12-24	Not used.

Note In SIP 3.1, the SoundPoint IP welcome sound was removed from `saf.1` . If you want the welcome sound to be played when a phone reboots or restarts, set `saf.1` to **SoundPointIPWelcome.wav** .

Sound Effects <se/>

The phone uses both synthesized (based on the chord-sets, refer to Chord-Sets <chord/> on page A-36) and sampled audio sound effects. Sound effects are defined by patterns: rudimentary sequences of chord-sets, silence periods, and wave files.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
se.stutterOnVoiceMail	0 or 1	1	If set to 1, stuttered dial tone is used in place of normal dial tone to indicate that one or more messages (voice mail) are waiting at the message center.
se.appLocalEnabled	0 or 1	1	If set to 1, local user interface sound effects such as confirmation/error tones, will be enabled.

This attribute also includes:

- Patterns <pat/>
- Ring type <rt/>

Patterns <pat/>

Patterns use a simple script language that allows different chord sets or wave files to be strung together with periods of silence. The script language uses the following instructions:

Instruction	Meaning	Example
sampled (n)	Play sampled audio file n	`se.pat.callProg.x.inst.y.type` ="sampled" (sampled audio file instruction type) `se.pat.callProg.x.inst.y.value` ="3" (specifies sampled audio file 3)
chord (n, d)	Play chord set n (d is optional and allows the chord set ON duration to be overridden to d milliseconds)	`se.pat.callProg.x.inst.y.type` = "chord" (chord set instruction type) `se.pat.callProg.x.inst.y.value` = "3" (specifies call progress chord set 3) `se.pat.callProg.x.inst.y.param` = "2000" (override ON duration of chord set to 2000 milliseconds)
silence (d)	Play silence for d milliseconds (Rx audio is not muted)	`se.pat.callProg.x.inst.y.type` = "silence" (silence instruction type) `se.pat.callProg.x.inst.y.value` = "300" (specifies silence is to last 300 milliseconds)
branch (n)	Advance n instructions and execute that instruction (n must be negative and must not branch beyond the first instruction)	`se.pat.callProg.x.inst.y.type` = "branch" (branch instruction type) `se.pat.callProg.x.inst.y.value` = "-5" (step back 5 instructions and execute that instruction)

Note Currently, patterns that use the *sampled* instruction are limited to the following format: *sampled* followed by optional *silence* and optional *branch* back to the beginning.

In the following table, *x* is the pattern number, *y* is the instruction number. Both *x* and *y* need to be sequential. There are three categories of sound effect patterns: *callProg* (Call Progress Patterns), *ringer* (Ringer Patterns) and *misc* (Miscellaneous Patterns).

Attribute	Permitted Values	Interpretation	
se.pat.callProg.x.name	UTF-8 encoded string	Used for identification purposes in the user interface (currently used for ringer patterns only); for patterns that use a sampled audio file which has been overridden by a downloaded replacement, the se.pat.ringer.x.name parameter will be overridden in the user interface by the file names of the wave file.	
se.pat.callProg.x.inst.y.type	sampled OR chord OR silence OR branch	As above.	
se.pat.callProg.x.inst.y. value	integer	Instruction type: sampled chord silence branch	Interpretation: sampled audio file number chord set number silence duration in ms number of instructions to advance
se.pat.callProg.x.inst.y. param	positive integer	If instruction type is chord, this optional parameter specifies the on duration to be used, overriding the on duration specified in the chord-set definition.	

Call Progress Patterns

The following table maps call progress patterns to their usage within the phone.

Call progress pattern number	Use within phone
1	dial tone
2	busy tone
3	ring back tone
4	reorder tone
5	stuttered dial tone
6	call waiting tone

Call progress pattern number	Use within phone
7	alternate call waiting tone (distinctive)
8	confirmation tone
9	howler tone (off-hook warning)
10	record warning
11	message waiting tone
12	alerting
13	intercom announcement tone
14	barge-in tone
15	secondary dial tone

Ringer Patterns

The following table maps ringer pattern numbers to their default descriptions.

Ringer pattern number	Default description
1	Silent Ring
2	Low Trill
3	Low Double Trill
4	Medium Trill
5	Medium Double Trill
6	High Trill
7	High Double Trill
8	Highest Trill
9	Highest Double Trill
10	Beeble
11	Triplet
12	Ringback-style
13	Sampled audio file 1
14	Sampled audio file 2
15	Sampled audio file 3
16	Sampled audio file 4
17	Sampled audio file 5

Ringer pattern number	Default description
18	Sampled audio file 6
19	Sampled audio file 7
20	Sampled audio file 8
21	Sampled audio file 9
22	Sampled audio file 10

Note

Silent Ring will only provide a visual indication of an incoming call, but no audio indication.

Sampled audio files 1-21 all use the same built-in file unless that file has been replaced with a downloaded file. For more information, refer to Sampled Audio for Sound Effects <saf/> on page A-37.

Miscellaneous Patterns

The following table maps miscellaneous patterns to their usage within the phone.

Miscellaneous pattern number	Use within phone
1	new message waiting indication
2	new instant message
3	Not used
4	local hold notification
5	positive confirmation
6	negative confirmation
7	welcome (boot up)

Ring type <rt/>

Ring type is used to define a simple class of ring to be applied based on some credentials that are usually carried within the network protocol. The ring class includes attributes such as call-waiting and ringer index, if appropriate. The ring class can use one of four types of ring that are defined as follows:

ring	Play a specified ring pattern or call waiting indication.
visual	Provide only a visual indication (no audio indication) of incoming call (no ringer needs to be specified).
answer	Provide auto-answer on incoming call.
ring-answer	Provide auto answer on incoming call after a ring period.

Note The auto-answer on incoming call is currently only applied if there is no other call in progress on the phone at the time.

In the following table, *x* is the ring class number. The *x* index needs to be sequential.

Attribute	Permitted Values	Interpretation
se.rt.enabled	0,1	Set to 1 to enable the ring type feature within the phone, 0 otherwise.
se.rt.modification.enabled	0,1	Set to 1 to allow user modification through local user interface of the pre-defined ring type enabled for modification.
se.rt.x.name	UTF-8 encoded string	Used for identification purposes in the user interface.
se.rt.x.type	ring OR visual OR answer OR ring-answer	As defined in table above.
se.rt.x.ringer	integer - only relevant if the type is set to ‘ring’ or ‘ring-answer’	The ringer index to be used for this class of ring. The ringer index should match one of Ringer Patterns on page A-41.
se.rt.x.callWait	integer - only relevant if the type is set to ‘ring’ or ‘ring-answer’	The call waiting index to be used for this class of ring. The call waiting index should match one defined in Call Progress Patterns on page A-40.
se.rt.x.timeout	positive integer - only relevant if the type is set to ‘ring-answer’. Default value is 2000.	The duration of the ring in milliseconds before the call is auto answered. If this field is omitted or is left blank, a value of 2000 is used.
se.rt.x.mod	0,1	Set to 1 if the user interface should allow for modification by the user of the ringer index used for this ring class.

Note Modification of `se.rt.modification.enabled` and `se.rt.x.name` parameters through the user interface will be implemented in a future release.

Voice Settings <voice/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.txPacketFilter	0 or 1	Null	Flag to determine whether or not narrowband Tx high-pass filtering should be enabled. If set to 1, narrowband Tx high-pass filter is enabled. If set 0 or Null, no Tx filtering is performed.

This attribute includes:

- Voice Coding Algorithms <codecs/>
- Volume Persistence <volume/>
- Gains <gain/>
- Acoustic Echo Cancellation <aec/>
- Acoustic Echo Suppression <aes/>
- Background Noise Suppression <ns/>
- Automatic Gain Control <agc/>
- Receive Equalization <rxEq/>
- Transmit Equalization <txEq/>
- Voice Activity Detection <vad/>
- Quality Monitoring <quality monitoring/>

Voice Coding Algorithms <codecs/>

These codecs include:

- Codec Preferences <codecPref/>
- Codec Profiles <audioProfile/>

Codec Preferences <codecPref/>

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.G711Mu	Null, 1-3	1	Specifies the codec preferences for SoundPoint IP 32x/33x, and 430 phones. 1 = highest 3 = lowest Null = do not use Give each codec a unique priority, this will dictate the order used in SDP negotiations. ***Note**: iLBC is not supported on the SoundPoint IP 430.*
voice.codecPref.G711A		2	
voice.codecPref.G729AB		3	
voice.codecPref.iLBC.13_33kbps		Null	
voice.codecPref.iLBC.15_2kbps		Null	
voice.codecPref.IP_650.G711Mu	Null, 1-4	2	Specifies the codec preferences for the SoundPoint IP 450, 550, 560, 650, and 670 phones. Interpretation as above.
voice.codecPref.IP_650.G711A		3	
voice.codecPref.IP_650.G729AB		4	
voice.codecPref.IP_650.G722		1	
voice.codecPref.iLBC.IP_650. 13_33kbps		Null	
voice.codecPref.iLBC.IP_650.15_2kbps		Null	

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_6000.G711Mu	Null, 1-13	5	Specifies the codec preferences for the SoundStation IP 6000 phone. Interpretation as above.
voice.codecPref.IP_6000.G711A		6	
voice.codecPref.IP_6000.G722		3	
voice.codecPref.IP_6000.G7221.16kbps		Null	
voice.codecPref.IP_6000.G7221.24kbps		Null	
voice.codecPref.IP_6000.G7221.32kbps		4	
voice.codecPref.IP_6000.G729AB		7	
voice.codecPref.IP_6000.G7221C. 24kbps		Null	
voice.codecPref.IP_6000.G7221C. 32kbps		Null	
voice.codecPref.IP_6000.G7221C. 48kbps		1	
voice.codecPref.IP_6000.Siren14. 24kbps		Null	
voice.codecPref.IP_6000.Siren14. 32kbps		Null	
voice.codecPref.IP_6000.Siren14. 48kbps		2	
voice.codecPref.iLBC.IP_6000. 13_33kbps		Null	
voice.codecPref.iLBC.IP_6000. 15_2kbps		Null	

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_7000.G711Mu	Null, 1-16	6	Specifies the codec preferences for the SoundStation IP 7000 phone. Interpretation as above.
voice.codecPref.IP_7000.G711A		7	
voice.codecPref.IP_7000.G722		4	
voice.codecPref.IP_7000.G7221.16kbps		Null	
voice.codecPref.IP_7000.G7221.24kbps		Null	
voice.codecPref.IP_7000.G7221.32kbps		5	
voice.codecPref.IP_7000.G7221C. 24kbps		Null	
voice.codecPref.IP_7000.G7221C. 32kbps		Null	
voice.codecPref.IP_7000.G7221C. 48kbps		2	
voice.codecPref.IP_7000.G729AB		8	
voice.codecPref.IP_7000.Lin16.16ksps		Null	
voice.codecPref.IP_7000.Lin16.32ksps		Null	
voice.codecPref.IP_7000.Lin16.48ksps		Null	
voice.codecPref.IP_7000.Siren22. 32kbps		Null	
voice.codecPref.IP_7000.Siren22. 48kbps		Null	
voice.codecPref.IP_7000.Siren22. 64kbps		1	
voice.codecPref.IP_7000.Siren14. 24kbps		Null	
voice.codecPref.IP_7000.Siren14. 32kbps		Null	
voice.codecPref.IP_7000.Siren14. 48kbps		3	
voice.codecPref.iLBC.IP_7000. 13_33kbps		Null	
voice.codecPref.iLBC.IP_7000. 15_2kbps		Null	

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.VVX_1500.G711Mu	Null, 1-16	4	Specifies the audio codec preferences for the Polycom VVX 1500 phone. Interpretation as above.
voice.codecPref.VVX_1500.G711A		5	
voice.codecPref.VVX_1500. G719.32kbps		Null	
voice.codecPref.VVX_1500. G719.48kbps		Null	
voice.codecPref.VVX_1500. G719.64kbps		Null	
voice.codecPref.VVX_1500.G722		3	
voice.codecPref.VVX_1500. G7221.16kbps		Null	
voice.codecPref.VVX_1500. G7221.24kbps		Null	
voice.codecPref.VVX_1500. G7221.32kbps		2	
voice.codecPref.VVX_1500. G7221C.24kbps		Null	
voice.codecPref.VVX_1500. G7221C.32kbps		Null	
voice.codecPref.VVX_1500. G7221C.48kbps		1	
voice.codecPref.VVX_1500.G729AB		6	
voice.codecPref.VVX_1500. Lin16.16ksps		Null	
voice.codecPref.VVX_1500. Lin16.32ksps		Null	
voice.codecPref.VVX_1500. Lin16.44_1ksps		Null	
voice.codecPref.VVX_1500. Lin16.48ksps		Null	
voice.codecPref.VVX_1500.Lin16.8ksps		Null	
voice.codecPref.VVX_1500. Siren14.24kbps		Null	
voice.codecPref.VVX_1500. Siren14.32kbps		Null	

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.VVX_1500. Siren14.48kbps		Null	
voice.codecPref.iLBC.VVX_1500. 13_33kbps		Null	
voice.codecPref.iLBC.VVX_1500. 15_2kbps		Null	

Note Codecs with a default of Null are available for test purposes only and are not expected to be used in your deployment.

Codec Profiles <audioProfile/>

The following profile attributes can be adjusted for each of the five supported codecs. In the table, *x*=G711Mu, G711A, G719, G722, G7221, G7221C, G729AB, Lin16, Siren14, Siren22, and iLBC.

Attribute	Permitted Values	Interpretation
voice.audioProfile.x.payloadSize	10, 20, 30, ...80	Preferred Tx payload size in milliseconds to be provided in SDP offers and used in the absence of ptime negotiations. This is also the range of supported Rx payload sizes. The payload size for G719, G7221, G7221C, Siren14, Siren22, and iLBC are further subdivided.
voice.audioProfile.x.jitterBufferMin	20, 40, 50, 60, ... (multiple of 10)	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter. The IP4000 values are the same as the IP30x values.
voice.audioProfile.x.jitterBufferShrink	10, 20, 30, ... (multiple of 10)	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (1000 ms) to minimize the delay on known good networks. Use larger values to minimize packet loss on networks with large jitter (3000 ms).

Attribute	Permitted Values	Interpretation
voice.audioProfile.x.jitterBufferMax	> jitterBufferMin, multiple of 10, <=300 for IP 32x, 33x, 430, 550, 600, and 650	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.
voice.audioProfile.x.payloadType	96 - 127 (default)	The codec payload encoding in the dynamic range to be used in SDP offers.

Volume Persistence <volume/>

The user's selection of the receive volume during a call can be remembered between calls. This can be configured per termination (handset, headset and hands-free/chassis). In some countries regulations exist which dictate that receive volume should be reset to nominal at the start of each call on handset and headset.

Attribute	Permitted Values	Default	Interpretation
voice.volume.persist.handset	0 or 1	0	If set to 1, the receive volume will be remembered between calls. If set to 0, the receive volume will be reset to nominal at the start of each call.
voice.volume.persist.headset	0 or 1	0	
voice.volume.persist.handsfree	0 or 1	1	

Gains <gain/>

The default gain settings have been carefully adjusted to comply with the TIA-810-A digital telephony standard.

Polycom recommends that you do not change these values.

Attribute	Default
voice.gain.rx.analog.handset	0
voice.gain.rx.analog.handset.VVX_1500	-2
voice.gain.rx.analog.headset	0
voice.gain.rx.analog.headset.VVX_1500	-2
voice.gain.rx.analog.chassis	0

Attribute	Default
voice.gain.rx.analog.chassis.IP_330	0
voice.gain.rx.analog.chassis.IP_430	0
voice.gain.rx.analog.chassis.IP_450	0
voice.gain.rx.analog.chassis.IP_650	0
voice.gain.rx.analog.chassis.IP_6000	0
voice.gain.rx.analog.chassis.IP_7000	0
voice.gain.rx.analog.chassis.VVX_1500	-3
voice.gain.rx.analog.ringer	0
voice.gain.rx.analog.ringer.IP_330	0
voice.gain.rx.analog.ringer.IP_430	0
voice.gain.rx.analog.ringer.IP_450	0
voice.gain.rx.analog.ringer.IP_650	0
voice.gain.rx.analog.ringer.IP_6000	0
voice.gain.rx.analog.ringer.IP_7000	0
voice.gain.rx.analog.ringer.VVX_1500	0
voice.gain.rx.digital.handset	-15
voice.gain.rx.digital.headset	-21
voice.gain.rx.digital.chassis	0
voice.gain.rx.digital.chassis.IP_450	5
voice.gain.rx.digital.chassis.IP_6000	5
voice.gain.rx.digital.chassis.IP_7000	5
voice.gain.rx.digital.chassis.VVX_1500	0
voice.gain.rx.digital.ringer	-21
voice.gain.rx.digital.ringer.IP_330	-12
voice.gain.rx.digital.ringer.IP_430	-12
voice.gain.rx.digital.ringer.IP_450	-12
voice.gain.rx.digital.ringer.IP_650	-12
voice.gain.rx.digital.ringer.IP_6000	-21
voice.gain.rx.digital.ringer.IP_7000	-21
voice.gain.rx.digital.ringer.VVX_1500	-21

Attribute	Default
voice.gain.rx.analog.handset.sidetone	-20
voice.gain.rx.analog.handset.sidetone.VVX_1500	-15
voice.gain.rx.analog.headset.sidetone	-24
voice.gain.rx.analog.headset.sidetone.VVX_1500	-31
voice.gain.tx.analog.handset	6
voice.gain.tx.analog.handset.VVX_1500	-48
voice.gain.tx.analog.headset	3
voice.gain.tx.analog.headset.VVX_1500	-47
voice.gain.tx.analog.chassis	3
voice.gain.tx.analog.chassis.IP_330	36
voice.gain.tx.analog.chassis.IP_430	36
voice.gain.tx.analog.chassis.IP_450	36
voice.gain.tx.analog.chassis.IP_650	36
voice.gain.tx.analog.chassis.IP_6000	0
voice.gain.tx.analog.chassis.IP_7000	0
voice.gain.tx.analog.chassis.VVX_1500	-25
voice.gain.tx.digital.handset	0
voice.gain.tx.digital.handset.IP_330	10
voice.gain.tx.digital.handset.IP_430	6
voice.gain.tx.digital.handset.IP_450	6
voice.gain.tx.digital.handset.IP_650	6
voice.gain.tx.digital.handset.VVX_1500	12
voice.gain.tx.digital.headset	0
voice.gain.tx.digital.headset.IP_330	10
voice.gain.tx.digital.headset.IP_430	10
voice.gain.tx.digital.headset.IP_450	6
voice.gain.tx.digital.headset.IP_650	6
voice.gain.tx.digital.headset.VVX_1500	12
voice.gain.tx.digital.chassis	3
voice.gain.tx.digital.chassis.IP_330	12

Attribute	Default
voice.gain.tx.digital.chassis.IP_430	12
voice.gain.tx.digital.chassis.IP_450	12
voice.gain.tx.digital.chassis.IP_650	12
voice.gain.tx.digital.chassis.IP_6000	6
voice.gain.tx.digital.chassis.IP_7000	6
voice.gain.tx.digital.chassis.VVX_1500	3
voice.gain.tx.analog.preamp.handset	23
voice.gain.tx.analog.preamp.headset	23
voice.gain.tx.analog.preamp.chassis	32
voice.gain.tx.analog.preamp.chassis.IP_601	32
voice.handset.rxag.adjust.IP_330	1
voice.handset.rxag.adjust.IP_430	1
voice.handset.rxag.adjust.IP_450	1
voice.handset.rxag.adjust.IP_650	1
voice.handset.txag.adjust.IP_330	18
voice.handset.txag.adjust.IP_430	18
voice.handset.txag.adjust.IP_450	18
voice.handset.txag.adjust.IP_650	18
voice.handset.sidetone.adjust.IP_330	3
voice.handset.sidetone.adjust.IP_430	3
voice.handset.sidetone.adjust.IP_450	0
voice.handset.sidetone.adjust.IP_650	0
voice.headset.rxag.adjust.IP_330	4
voice.headset.rxag.adjust.IP_430	1
voice.headset.rxag.adjust.IP_450	1
voice.headset.rxag.adjust.IP_650	1
voice.headset.txag.adjust.IP_330	21
voice.headset.txag.adjust.IP_430	21
voice.headset.txag.adjust.IP_450	21
voice.headset.txag.adjust.IP_650	21

Attribute	Default
voice.headset.sidetone.adjust.IP_330	-3
voice.headset.sidetone.adjust.IP_430	-3
voice.headset.sidetone.adjust.IP_450	-3
voice.headset.sidetone.adjust.IP_650	-3

Acoustic Echo Cancellation <aec/>

These settings control the performance of the speakerphone acoustic echo canceller.

Polycom recommends that you do not change these values.

Attribute	Default
voice.aec.hs.enable	1
voice.aec.hs.lowFreqCutOff	100
voice.aec.hs.highFreqCutOff	7000
voice.aec.hs.erlTab_0_300	-24
voice.aec.hs.erlTab_300_600	-24
voice.aec.hs.erlTab_600_1500	-24
voice.aec.hs.erlTab_1500_3500	-24
voice.aec.hs.erlTab_3500_7000	-24
voice.aec.hd.enable	0
voice.aec.hd.lowFreqCutOff	100
voice.aec.hd.highFreqCutOff	7000
voice.aec.hd.erlTab_0_300	-24
voice.aec.hd.erlTab_300_600	-24
voice.aec.hd.erlTab_600_1500	-24
voice.aec.hd.erlTab_1500_3500	-24
voice.aec.hd.erlTab_3500_7000	-24
voice.aec.hf.enable	1
voice.aec.hf.lowFreqCutOff	100

Attribute	Default
voice.aec.hf.highFreqCutOff	7000
voice.aec.hf.erlTab_0_300	-6
voice.aec.hf.erlTab_300_600	-6
voice.aec.hf.erlTab_600_1500	-6
voice.aec.hf.erlTab_1500_3500	-6
voice.aec.hf.erlTab_3500_7000	-6

Acoustic Echo Suppression <aes/>

Acoustic Echo Suppression (AES) provides non-linear processing of the microphone signal to remove any residual echo remaining after linear AEC processing. Because AES depends on AEC, AES should only be enabled when AEC is also enabled. Normally, AES should be used whenever AEC is used for handsfree or handset and both are enabled by default for those terminations

These settings control the performance of the speakerphone acoustic echo suppressor.

Polycom recommends that you do not change these values.

Attribute	Default
voice.aes.hs.enable	1
voice.aes.hs.duplexBalance	7
voice.aes.hd.enable	0
voice.aes.hd.duplexBalance	0
voice.aes.hf.enable	1
voice.aes.hf.duplexBalance.0	7
voice.aes.hf.duplexBalance.1	7
voice.aes.hf.duplexBalance.2	6
voice.aes.hf.duplexBalance.3	6
voice.aes.hf.duplexBalance.4	5
voice.aes.hf.duplexBalance.5	4
voice.aes.hf.duplexBalance.6	4

Attribute	Default
voice.aes.hf.duplexBalance.7	3
voice.aes.hf.duplexBalance.8	2
voice.aes.hf.duplexBalance.IP_4000.0	10
voice.aes.hf.duplexBalance.IP_4000.1	9
voice.aes.hf.duplexBalance.IP_4000.2	8
voice.aes.hf.duplexBalance.IP_4000.3	7
voice.aes.hf.duplexBalance.IP_4000.4	6
voice.aes.hf.duplexBalance.IP_4000.5	5
voice.aes.hf.duplexBalance.IP_4000.6	4
voice.aes.hf.duplexBalance.IP_4000.7	3
voice.aes.hf.duplexBalance.IP_4000.8	2

Background Noise Suppression <ns/>

These settings control the performance of the transmit background noise suppression feature.

Polycom recommends that you do not change these values.

Attribute	Default
voice.ns.hs.enable	1
voice.ns.hs.signalAttn	-6
voice.ns.hs.silenceAttn	-9
voice.ns.hd.enable	0
voice.ns.hd.signalAttn	0
voice.ns.hd.silenceAttn	0
voice.ns.hf.enable	1
voice.ns.hf.signalAttn	-6
voice.ns.hf.silenceAttn	-9

Attribute	Default
voice.ns.hf.IP_4000.enable	1
voice.ns.hf.IP_4000.signalAttn	-6
voice.ns.hf.IP_4000.silenceAttn	-9

Automatic Gain Control <agc/>

These settings control the performance of the transmit automatic gain control feature.

Note Automatic Gain Control will be implemented in a future release.

Polycom recommends that you do not change these values.

Attribute	Default
voice.agc.hs.enable	0
voice.agc.hd.enable	0
voice.agc.hf.enable	0

Receive Equalization <rxEq/>

These settings control the performance of the receive equalization feature.

Polycom recommends that you do not change these values.

Attribute	Default
voice.rxEq.hs.IP_330.preFilter.enable	1
voice.rxEq.hs.IP_430.preFilter.enable	1
voice.rxEq.hs.IP_450.preFilter.enable	1
voice.rxEq.hs.IP_650.preFilter.enable	1
voice.rxEq.hs.VVX_1500.preFilter.enable	1

Attribute	Default
voice.rxEq.hs.IP_330.postFilter.enable	0
voice.rxEq.hs.IP_430.postFilter.enable	0
voice.rxEq.hs.IP_450.postFilter.enable	0
voice.rxEq.hs.IP_650.postFilter.enable	0
voice.rxEq.hs.VVX_1500.postFilter.enable	0
voice.rxEq.hd.IP_330.preFilter.enable	0
voice.rxEq.hd.IP_430.preFilter.enable	0
voice.rxEq.hd.IP_450.preFilter.enable	0
voice.rxEq.hd.IP_650.preFilter.enable	1
voice.rxEq.hd.VVX_1500.preFilter.enable	0
voice.rxEq.hd.IP_330.postFilter.enable	0
voice.rxEq.hd.IP_430.postFilter.enable	0
voice.rxEq.hd.IP_450.postFilter.enable	0
voice.rxEq.hd.IP_650.postFilter.enable	0
voice.rxEq.hd.VVX_1500.postFilter.enable	0
voice.rxEq.hf.IP_330.preFilter.enable	1
voice.rxEq.hf.IP_430.preFilter.enable	1
voice.rxEq.hf.IP_450.preFilter.enable	1
voice.rxEq.hf.IP_650.preFilter.enable	1
voice.rxEq.hf.IP_6000.preFilter.enable	0
voice.rxEq.hf.IP_7000.preFilter.enable	0
voice.rxEq.hf.VVX_1500.preFilter.enable	1
voice.rxEq.hf.IP_330.postFilter.enable	0
voice.rxEq.hf.IP_430.postFilter.enable	0
voice.rxEq.hf.IP_450.postFilter.enable	0
voice.rxEq.hf.IP_650.postFilter.enable	0
voice.rxEq.hf.IP_6000.postFilter.enable	0
voice.rxEq.hf.IP_7000.postFilter.enable	0
voice.rxEq.hf.VVX_1500.postFilter.enable	0

Transmit Equalization <txEq/>

These settings control the performance of the hands-free transmit equalization feature.

Polycom recommends that you do not change these values.

Attribute	Default
voice.txEq.hs.IP_330.preFilter.enable	0
voice.txEq.hs.IP_430.preFilter.enable	0
voice.txEq.hs.IP_450.preFilter.enable	0
voice.txEq.hs.IP_650.preFilter.enable	1
voice.txEq.hs.VVX_1500.preFilter.enable	0
voice.txEq.hs.IP_330.postFilter.enable	1
voice.txEq.hs.IP_430.postFilter.enable	1
voice.txEq.hs.IP_450.postFilter.enable	1
voice.txEq.hs.IP_650.postFilter.enable	1
voice.txEq.hs.VVX_1500.postFilter.enable	1
voice.txEq.hd.IP_330.preFilter.enable	0
voice.txEq.hd.IP_430.preFilter.enable	0
voice.txEq.hd.IP_450.preFilter.enable	0
voice.txEq.hd.IP_650.preFilter.enable	1
voice.txEq.hd.VVX_1500.preFilter.enable	0
voice.txEq.hd.IP_330.postFilter.enable	0
voice.txEq.hd.IP_430.postFilter.enable	0
voice.txEq.hd.IP_450.postFilter.enable	0
voice.txEq.hd.IP_650.postFilter.enable	0
voice.txEq.hd.VVX_1500.postFilter.enable	0
voice.txEq.hf.IP_330.preFilter.enable	0
voice.txEq.hf.IP_430.preFilter.enable	0
voice.txEq.hf.IP_450.preFilter.enable	0
voice.txEq.hf.IP_650.preFilter.enable	1

Attribute	Default
voice.txEq.hf.IP_6000.preFilter.enable	0
voice.txEq.hf.IP_7000.preFilter.enable	0
voice.txEq.hf.VVX_1500.preFilter.enable	0
voice.txEq.hf.IP_330.postFilter.enable	1
voice.txEq.hf.IP_430.postFilter.enable	1
voice.txEq.hf.IP_450.postFilter.enable	1
voice.txEq.hf.IP_650.postFilter.enable	1
voice.txEq.hf.IP_6000.postFilter.enable	0
voice.txEq.hf.IP_7000.postFilter.enable	0
voice.txEq.hf.VVX_1500.postFilter.enable	1

Voice Activity Detection <vad/>

These settings control the performance of the voice activity detection (silence suppression) feature.

Attribute	Permitted Values	Default	Interpretation
voice.vadEnable	0 or 1	0	If set to 1, enable VAD.
voice.vadThresh	integer from 0 to 30	15	The threshold for determining what is active voice and what is background noise in dB. This does not apply to G.729AB codec operation which has its own built-in VAD function.
voice.vad. signalAnnexB	0 or 1	Null	If set to 1 or Null and `voice.vadEnable` is set to 1, Annex B is used. A new line can be added to SDP depending on the setting of this parameter and the `voice.vadEnable` parameter. • If `voice.vadEnable` is set to 1, add attribute line a=fmtp:18 annexb="yes" below a=rtpmap… attribute line (where '18' could be replaced by another payload). • If `voice.vadEnable is set to 0`, add attribute line a=fmtp:18 annexb="no" below a=rtpmap… attribute line (where '18' could be replaced by another payload). If set to 0, there is no change to SDP.

Quality Monitoring <quality monitoring/>

This attribute includes:

- Central Report Collector <collector/>
- Alert Reports <alert/>
- Server <server/>
- RTCP-XR <rtcpxr/>

Central Report Collector <collector/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.enable.periodic	0, 1	0	Enables generation of periodic quality reports throughout a call.
voice.qualityMonitoring.collector.enable.session	0, 1	0	Enables generation of a quality report at the end of each call.
voice.qualityMonitoring.collector.enable.triggeredPeriodic	0, 1, 2	0	Controls the generation of periodic quality reports triggered by alert states. If set to 0, alert states do not cause periodic reports to be generated. If set to 1, periodic reports will be generated when an alert state is critical. If set to 2, periodic reports will be generated when an alert state is either warning or critical. ***Note:*** *This parameter is ignored when* `qualityMonitoring.collector.enable.periodic` *is set 1, since periodic reports are sent throughout the duration of a call.*
voice.qualityMonitoring.collector.period	5 to 20	20	The time interval between successive periodic quality reports.

Alert Reports <alert/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.alert.moslq.threshold.warning	Null, 15 to 40	Null	Threshold value of listening MOS score (MOS-LQ) that causes phone to send a warning alert quality report. Configure the desired MOS value multiplied by 10. If set to Null, warning alerts are not generated due to MOS-LQ. For example, a configured value of 35 corresponds to the MOS score 3.5.
voice.qualityMonitoring.collector.alert.moslq.threshold.critical	Null, 15 to 40	Null	Threshold value of listening MOS score (MOS-LQ) that causes phone to send a critical alert quality report. Configure the desired MOS value multiplied by 10. If set to Null, critical alerts are not generated due to MOS-LQ. For example, a configured value of 28 corresponds to the MOS score 2.8.
voice.qualityMonitoring.collector.alert.delay.threshold.warning	Null, 10 to 2000	Null	Threshold value of one way delay (in ms) that causes phone to send a critical alert quality report. If set to Null, warning alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.
voice.qualityMonitoring.collector.alert.delay.threshold.critical	Null, 10 to 2000	Null	Threshold value of one way delay (in ms) that causes phone to send a critical alert quality report. If set to Null, critical alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.collector.server.x.address	Dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages. Set x to 1as only one report collector is supported at this time.
voice.qualityMonitoring.collector.server.x.port	0, Null, 1 to 65535	5060	If port is 0 or Null, port 5060 will be used. Set x to 1as only one report collector is supported at this time.

RTCP-XR <rtcpxr/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
voice.qualityMonitoring.rtcpxr.enable	0, 1	0	Enables generation of RTCP-XR packets.

Video Settings <video/>

Note This attribute is only supported for use on the Polycom VVX 1500.

These configuration attributes are defined as follows:

Attribute	Permitted Values	Default	Interpretation
video.enable	0=Disable, 1=Enable	1	Flag to determine whether or not video calls are established. This applies to all calls, between two Polycom VVX 1500s and between Polycom VVX 1500 and any other video device. If set to 1 (enabled) or Null, video is sent in outgoing calls and received in incoming calls. If set to 0, video is not sent in outgoing calls and not received in incoming calls. All calls are audio only.
video.autoStartVideoTx	0 or 1	1	Flag to determine whether or not video transmission occurs when a call starts. If set to 0, video transmission does not start. If set to 1or Null, video transmission from the near end starts when a call starts.
video.screenMode	"normal", "full", "crop"	normal	Applies to the video window shown in the normal mode. If set to "normal" or Null, all pixels are displayed, black bars appear on the top, bottom, or sides of the window, if necessary, to maintain the correct aspect ratio. If set to "full", all pixels are displayed and the image is stretched linearly and independently to fill the video frame. If set to "crop", the black bars do not appear, the image size is re-sized to maintain the correct aspect ratio, and any parts of the image that do not fit in the display are cropped.
video.screenModeFS	"normal", "full", "crop"	normal	Applies to the video window in Full Screen mode. The image is re-sized to maintain the correct aspect ratio and any parts of the image that do not fit in the display are cropped.
video.quality	"motion", "sharpness"	Null	Determine the quality of video shown in a call or conference. Use "motion" for people or other video with motion. Use "sharpness" or Null for video with little or no movement. Moderate to heavy motion can cause some frames to be dropped.

Attribute	Permitted Values	Default	Interpretation
video.callRate	128 - 1024 kbps	448	The maximum call rate in kbps to use when initially negotiating the bandwidth for a video call. This value cannot exceed `video.maxCallRate`.
video.maxCallRate	128 - 1024 kbps	512	Limits the maximum network bandwidth used in a call. It is used in the SDP bandwidth signaling. If honored by the far end, both Rx and Tx network bandwidth used in a call will not exceed this value (in kbps). If set to Null, the value 1024 is used.
video.autoFullScreen	0 or 1	Null	Flag to determine whether or not video calls use the full screen layout. If set to 1, video calls will use the full screen layout by default. When a video call is first created (upon discovery that far-end is video capable) or when an audio call transitions to a video call (through far-end transfer), the full screen layout will be used. If set to 0 or Null, video calls only use the full screen layout if it is selected by the user.
video.forceRtcpVideoCodecControl	0 or 1	0	If set to 1, force the Polycom VVX 1500 to send RTCP feedback messages to request fast update I-frames for all video calls.

These attributes also include:

- Video Coding Algorithms <codecs/>
- Camera Controls <camera/>
- Local Camera View <localCameraView/>

Video Coding Algorithms <codecs/>

These codecs include:

- Codec Preferences <codecPref/>
- Codec Profiles <profile/>

Codec Preferences <codecPref/>

Attribute	Permitted Values	Default	Interpretation
video.codecPref.H261	1 to 4	4	Specifies the video codec preferences for the Polycom VVX 1500 phone.
video.codecPref.H264	1 to 4	1	
video.codecPref.H2631998	1 to 4	2	
video.codecPref.H263	1 to 4	3	

Note Codecs with a default of Null are available for test purposes only and are not expected to be used in your deployment.

Codec Profiles <profile/>

The profile attributes can be adjusted for each of the new supported video codecs.

Attribute	Permitted Values	Interpretation
video.profile.H261.jitterBufferMax	(`video.profile.H261.jitterBufferMin` + 500ms) to 2500ms, default 2000ms	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.
video.profile.H261.jitterBufferMin	33ms to 1000ms, default 150ms	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter.
video.profile.H261.jitterBufferShrink	33ms to 1000ms, default 70ms	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (33 ms) to minimize the delay on known good networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).

Attribute	Permitted Values	Interpretation
video.profile.H261.CifMpi	1 (default) to 32	This value is H261 format parameter CIF used to signal Polycom VVX 1500 receiving capability in SDP. This value also controls the TX frame size. If set to 1, CIF is used (provided the far end supports CIF=1); otherwise QCIF is used.
video.profile.H261.QcifMpi	1 (default) to 32	This value is H261 format parameter QCIF used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H261.annexD	0 or 1 default Null	This value is H261 format parameter ANNEXD used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H264.jitterBufferMax	(`video.profile.H264.jitterBufferMin` + 500ms) to 2500ms, default 2000ms	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.
video.profile.H264.jitterBufferMin	33ms to 1000ms, default 150ms	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter.
video.profile.H264.jitterBufferShrink	33ms to 1000ms, default 70ms	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (33 ms) to minimize the delay on known good networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).
video.profile.H264.payloadType	96 to 127, default 109	RTP payload format type for H264/90000 MIME type.
video.profile.H264.profileLevel	1, 1b, 1.1, 1.2, 1.3 (default)	This value is H.264's level used in the phone. The Level is a constraint set to selected key algorithm parameters, codec in different level has different ability, at this time Polycom VVX 1500 support these level (1,1b,1.1,1.2,1.3), as to detailed level definition. For more information, refer to ITU-T H.264.

Attribute	Permitted Values	Interpretation
video.profile.H263.jitterBufferMax	(`video.profile.H263.jitterBufferMin` + 500ms) to 2500ms, default 2000ms	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.
video.profile.H263.jitterBufferMin	33ms to 1000ms, default 150ms	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter.
video.profile.H263.jitterBufferShrink	33ms to 1000ms, default 70ms	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (33 ms) to minimize the delay on known good networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).
video.profile.H263.CifMpi	1 (default) to 32	This value is H263/90000 format parameter CIF used to signal Polycom VVX 1500 receiving capability in SDP. This value also controls the TX frame size. If set to 1, CIF is used (provided the far end supports CIF=1); otherwise QCIF is used.
video.profile.H263.QcifMpi	1 (default) to 32	This value is H263/90000 format parameter QCIF used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H263.SqcifMpi	1 (default) to 32	This value is H263/90000 format parameter SQCIF used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.jitterBufferMax	(`video.profile.H2631998.jitterBufferMin` + 500ms) to 2500ms, default 2000ms	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.

Attribute	Permitted Values	Interpretation
video.profile.H2631998.jitterBufferMin	33ms to 1000ms, default 150ms	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter.
video.profile.H2631998.jitterBufferShrink	33ms to 1000ms, default 70ms	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (33 ms) to minimize the delay on known good networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).
video.profile.H2631998.payloadType	96 (default) to 127	RTP payload format type for H263-1998/90000 MIME type.
video.profile.H2631998.CifMpi	1 (default) to 32	This value is H263-1998/90000 format parameter CIF used to signal Polycom VVX 1500 receiving capability in SDP. This value also controls the TX frame size. If set to 1, CIF is used (provided the far end supports CIF=1); otherwise QCIF is used.
video.profile.H2631998.QcifMpi	1 (default) to 32	This value is H263-1998/90000 format parameter QCIF used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.SqcifMpi	1 (default) to 32	This value is H263-1998/90000 format parameter SQCIF used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.annexF	0 or 1 default Null	This value is H263-1998/90000 format parameter ANNEXF used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.annexI	0 or 1 default Null	This value is H263-1998/90000 format parameter ANNEXI used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.annexJ	0 or 1 default Null	This value is H263-1998/90000 format parameter ANNEXJ used to signal Polycom VVX 1500 receiving capability in the SDP.

Attribute	Permitted Values	Interpretation
video.profile.H2631998.annexT	0 or 1 default Null	This value is H263-1998/90000 format parameter ANNEXT used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.annexK	0 or 1 (default), 2, 3, 4	This value is H263-1998/90000 format parameter ANNEXK used to signal Polycom VVX 1500 receiving capability in the SDP.
video.profile.H2631998.annexN	0 or 1 (default), 2, 3, 4	This value is H263-1998/90000 format parameter ANNEXN used to signal Polycom VVX 1500 receiving capability in the SDP.

Camera Controls <camera/>

These settings control the performance of the camera.

These configuration attributes are defined as follows:

Attribute	Permitted Values	Default	Interpretation
video.camera. flickerAvoidance	0 to 2	Null	Set flicker avoidance. If set to 0 or Null, flicker avoidance is automatic. If set to 1, 50hz AC power frequency flicker avoidance (Europe/Asia). If set to 2, 60hz AC power frequency flicker avoidance (North America).
video.camera.frameRate	5 to 30 frames per second	Null	Set target frame rate. Values indicate a fixed frame rate, from 5 (least smooth) to 30 (most smooth). If set to Null, the value 25 is used.
video.camera.brightness	0 to 6	Null	Set brightness level. The value range is from 0 (Dimmest) to 6 (Brightest). If set to Null, the value 3 is used.
video.camera.saturation	0 to 6	Null	Set saturation level. The value range is from 0 (Lowest) to 6 (Highest). If set to Null, the value 3 is used.

Attribute	Permitted Values	Default	Interpretation
video.camera.contrast	0 to 4	Null	Set contrast level. The value range is from 0 (No contrast increase) to 3 (Most contrast increase, and 4 (Noise reduction contrast). If set to Null, the value 0 is used.
video.camera.sharpness	0 to 6	Null	Set sharpness level. The value range is from 0 (Lowest) to 6 (Highest). If set to Null, the value 3 is used.

Local Camera View <localCameraView/>

These settings control how the local camera is viewed on the screen.

These configuration attributes are defined as follows:

Attribute	Permitted Values	Default	Interpretation
video.localCameraView. fullscreen.enabled	0=Disable, 1=Enable	Null	Determines whether the local camera view is shown in the full screen layout . If set to 0, the local camera view is not shown. If set to 1 or Null, the local camera view is shown.
video.localCameraView. fullscreen.mode	“pip” or Null	Null	How the local camera view is shown. If set to “pip”, the local camera view appears as a picture-in-picture with the far end window. If set to Null, the local camera view appears side-by-side with the far end window.

Quality of Service <QOS/>

These settings control the Quality of Service (QOS) options.

This attribute includes:

- Ethernet IEEE 802.1p/Q <ethernet/>
- IP TOS <IP/>

Ethernet IEEE 802.1p/Q <ethernet/>

The following settings control the 802.1p/Q user_priority field:

- RTP <RTP/>
- Call Control <callControl/>

- Other <other/>

RTP <RTP/>

These parameters apply to RTP packets.

Attribute	Permitted Values	Default	Interpretation
qos.ethernet.rtp.user_priority	0-7	5	User-priority used for Voice RTP packets.
qos.ethernet.rtp.video.user_priority	0-7	5	User-priority used for Video RTP packets.

Call Control <callControl/>

These parameters apply to call control packets, such as the network protocol signaling.

Attribute	Permitted Values	Default	Interpretation
qos.ethernet.callControl.user_priority	0-7	5	User-priority used for call control packets.

Other <other/>

These default parameter values are used for all packets which are not set explicitly.

Attribute	Permitted Values	Default	Interpretation
qos.ethernet.other.user_priority	0-7	2	User-priority used for packets that do not have a per-protocol setting.

IP TOS <IP/>

The following settings control the "type of service" field in outgoing packets:

- RTP <rtp/>
- Call Control <callControl/>

RTP <rtp/>

These parameters apply to RTP packets.

Attribute	Permitted Values	Default	Interpretation
qos.ip.rtp.dscp	0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Null	This parameter allows the DSCP of packets to be specified. If set to a value, this will override the other `qos.ip.rtp`... parameters. Default of Null which means the other `qos.ip.rtp`... parameters will be used.
qos.ip.rtp.min_delay	0 or 1	1	If set to 1, set min-delay bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.max_throughput	0 or 1	1	If set to 1, set max-throughput bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.max_reliability	0 or 1	0	If set to 1, set max-reliability bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.min_cost	0 or 1	0	If set to 1, set min-cost bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.precedence	0-7	5	If set to 1, set precedence bits in the IP TOS field of the IP header, or else don't set them.
qos.ip.rtp.video.dscp	0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Null	This parameter allows the DSCP of packets to be specified. If set to a value, this will override the other `qos.ip.rtp.video`... parameters. Default of Null which means the other `qos.ip.rtp.video`... parameters will be used.
qos.ip.rtp.video.min_delay	0 or 1	1	If set to 1, set min-delay bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.video.max_throughput	0 or 1	1	If set to 1, set max-throughput bit in the IP TOS field of the IP header, or else don't set it.

Attribute	Permitted Values	Default	Interpretation
qos.ip.rtp.video.max_reliability	0 or 1	0	If set to 1, set max-reliability bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.video.min_cost	0 or 1	0	If set to 1, set min-cost bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.rtp.video.precedence	0-7	5	If set to 1, set precedence bits in the IP TOS field of the IP header, or else don't set them.

Call Control <callControl/>

These parameters apply to call control packets, such as the network protocol signaling.

Attribute	Permitted Values	Default	Interpretation
qos.ip.callControl.dscp	0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Null	This parameter allows the DSCP of packets to be specified. If set to a value this will override the other `qos.ip.callControl...` parameters. Default of Null which means the other `qos.ip.callControl...` parameters will be used.
qos.ip.callControl.min_delay	0 or 1	1	If set to 1, set min-delay bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.max_throughput	0 or 1	0	If set to 1, set max-throughput bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.max_reliability	0 or 1	0	If set to 1, set max-reliability bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.min_cost	0 or 1	0	If set to 1, set min-cost bit in the IP TOS field of the IP header, or else don't set it.
qos.ip.callControl.precedence	0-7	5	If set to 1, set precedence bits in the IP TOS field of the IP header, or else don't set them.

Basic TCP/IP <TCP_IP/>

This attribute includes:

- Network Monitoring <netMon/>
- Time Synchronization <sntp/>
- Port <port/>
- Keep-Alive <keepalive/>

Network Monitoring <netMon/>

Polycom recommends that you do not change these values.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default
tcpIpApp.netMon.enabled	0 or 1	1
tcpIpApp.netMon.period	1 to 86400	30

Time Synchronization <sntp/>

The following table describes the parameters used to set up time synchronization and daylight savings time. The defaults shown will enable daylight savings time (DST) for North America.

Daylight savings defaults:

- Do not use fixed day, use first or last day of week in the month.
- Start DST on the second Sunday in March at 2 am.
- Stop DST on the first Sunday in November at 2 am.

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.sntp.resyncPeriod	positive integer	86400 (24 hours)	Time in seconds between Simple Network Time Protocol (SNTP) re-syncs.
tcpIpApp.sntp.address	valid host name or IP address	clock	Address of the SNTP server.

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.sntp.address.overrideDHCP	0 or 1	0	These parameters determine whether configuration file parameters override DHCP parameters for the SNTP server address and Greenwich Mean Time (GMT) offset. If set to 0, DHCP values will override configuration file parameters. If set to 1, the configuration file parameters will override DHCP values.
tcpIpApp.sntp.gmtOffset	positive or negative integer	-28800 (Pacific time)	Offset in seconds of the local time zone from GMT. 3600 seconds = 1 hour
tcpIpApp.sntp.gmtOffset.overrideDHCP	0 or 1	0	These parameters determine whether configuration file parameters override DHCP parameters for the SNTP server address and GMT offset. If set to 0, DHCP values will override configuration file parameters. If set to 1, the configuration file parameters will override DHCP values.
tcpIpApp.sntp.daylightSavings.enable	0 or 1	1	If set to 1, apply daylight savings rules to displayed time.
tcpIpApp.sntp.daylightSavings.fixedDayEnable	0 or 1	0	If set to 0, month, date, and dayOfWeek are used in DST date calculation. If set to 1, then only month and date are used.
tcpIpApp.sntp.daylightSavings.start.month	1-12	3 (March)	Month to start DST. Mapping: 1=Jan, 2=Feb, ..., 12=Dec

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.sntp.daylightSavings.start.date	1-31	8	If `fixedDayEnable` is set to 1, use as day of the month to start DST. If `fixedDayEnable` is set to 0, us the mapping: 1 = the first occurrence of a given day-of-the-week in a month, 8 = the second occurrence of a given day-of-the-week in a month, 15 = the third occurrence of a given day-of-the-week in a month, 22 = the fourth occurrence of a given day-of-the-week in a month
tcpIpApp.sntp.daylightSavings.start.time	0-23	2	Time of day to start DST in 24 hour clock. Mapping: 2=2 am, 14=2 pm
tcpIpApp.sntp.daylightSavings.start.dayOfWeek	1-7	1	Day of week to apply DST. Mapping: 1=Sun, 2=Mon, ..., 7=Sat
tcpIpApp.sntp.daylightSavings.start.dayOfWeek.lastInMonth	0 or 1	0	If set to 1 and fixedDayEnable is set to 0, DST starts on the last day (specified by `start.dayOfWeek`) of the week in the month. The `start.date` is ignored.
tcpIpApp.sntp.daylightSavings.stop.month	1-12	11	Month to stop DST.
tcpIpApp.sntp.daylightSavings.stop.date	1-31	1	Day of the month to stop DST.
tcpIpApp.sntp.daylightSavings.stop.time	0-23	2	Time of day to stop DST in 24 hour clock.
tcpIpApp.sntp.daylightSavings.stop.dayOfWeek	1-7	1	Day of week to stop DST.
tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth	0 or 1	0	If set to 1 and fixedDayEnable set to 0, DST stops on the last day (specified by `stop.dayOfWeek`) of the week in the month. The `stop.date` is ignored.

Port <port/>

This attribute includes:

- RTP <rtp/>

RTP <rtp/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.port.rtp.filterByIp	0 or 1	1	If set to 1, reject RTP packets arriving from (sent from) a non-negotiated (through SDP) IP address.
tcpIpApp.port.rtp.filterByPort	0 or 1	0	If set to 1, reject RTP packets arriving from (sent from) a non-negotiated (through SDP) port.
tcpIpApp.port.rtp.forceSend	Null, 1024-65534	Null	When non-Null, send all RTP packets to, and expect all RTP packets to arrive on, the specified port. ***Note:*** *both tcpIpApp.port.rtp.filterByIp and tcpIpApp.port.rtp.filterByPort must be enabled for this to work.*
tcpIpApp.port.rtp.mediaPortRangeStart	Null, even integer from 1024-65534	Null	If set to Null, the value 2222 will be used for the first allocated RTP port, otherwise, the specified port will be used. Ports will be allocated from a pool starting with the specified port up to a value of (start-port + 47) for a voice-only phone or (start-port + 95) for a video phone.

Keep-Alive <keepalive/>

Allowing for the configuration of TCP keep-alive on SIP TLS connections, the phone can detect a failures quickly (in minutes) and attempt to re-register with the SIP call server (or its redundant pair).

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.keepalive.tcp.idleTransmitInterval	10 to 7200	Null	After idle x seconds, the keep-alive message is sent to the call server. If set to Null, the default value is 30 seconds. ***Note**: If this parameter is set to a value that is out of range, the default value is used.*
tcpIpApp.keepalive.tcp. noResponseTrasmitInterval	5 to 120	Null	If no response is received to keep-alive message, another keep-alive message is sent to the call server after x seconds. If set to Null, the default value to 20 seconds. ***Note**: If this parameter is set to a value that is out of range, the default value is used.*
tcpIpApp.keepalive.tcp.sip.tls.enable	0 or 1	0	If set to 1, enable TCP keep-alive for SIP signaling connections that use TLS transport. If set to 0, disable TCP keep-alive for SIP signaling connections that use TLS transport.

Web Server <httpd/>

The phone contains a local web server for user and administrator features. This can be disabled for applications where it is not needed or where it poses a security threat. The web server supports both basic and digest authentication. The authentication user name and password are not configurable for this release.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
httpd.enabled	0 or 1	1	If set to 1, the HTTP server will be enabled.
httpd.cfg.enabled	0 or 1	1	If set to 1, the HTTP server configuration interface will be enabled.
httpd.cfg.port	1-65535	80	Port is 80 for HTTP servers. Care should be taken when choosing an alternate port.

Call Handling Configuration <call/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.rejectBusyOnDnd	0 or 1	1	If set to 1, reject all incoming calls with the reason "busy" if do-not-disturb is enabled. ***Note:*** *This attribute is ignored when the line is configured as shared. The reason being that even though one party has turned on DND, the other person/people sharing that line do not necessarily want all calls to that number diverted away.* ***Note:*** *If server-based DND is enabled, this parameter is disabled*.
call.enableOnNotRegistered	0 or 1	1	If set to 1, calls will be allowed when the phone is not successfully registered, otherwise, calls will not be permitted without a valid registration. ***Note****: Setting this parameter to 1 can allow Polycom VVX 1500 phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured*.
call.offeringTimeOut	positive integer	60	Time in seconds to allow an incoming call to ring before dropping the call, 0=infinite. ***Note:*** *The call diversion, no answer feature will take precedence over this feature if enabled. For more information, refer to* *No Answer <noanswer/> on page A-144.*
call.ringBackTimeOut	positive integer	60	Time in seconds to allow an outgoing call to remain in the ringback state before dropping the call, 0=infinite.

Attribute	Permitted Values	Default	Interpretation
call.dialtoneTimeOut	Null, positive integer	60	Time in seconds to allow the dial tone to be played before dropping the call. If set to 0, the call is not dropped. If set to Null, call dropped after 60 seconds.
call.lastCallReturnString	string of maximum length 32	*69	The string sent to the server when the user selects the “last call return” action.
call.callsPerLineKey	1 to 24 OR 1 to 8	34, 24, 8 OR 4	For the SoundPoint IP 650 and 670, the permitted range is 1 to 34 and the default is 34. For the SoundPoint IP 550 and 560, the permitted range is 1 to 24 and the default is 24. For the SoundPoint IP 32x/33x and 430, the permitted range is 1 to 8 and the default is 4. For all other phones, the permitted range is 1 to 8 and the default is 8. This is the number of calls that may be active or on hold per line key on the phone. Note that this may be overridden by the per-registration attribute of `reg.x.callsPerLineKey`. Refer to Registration <reg/> on page A-134.
call.stickyAutoLineSeize	Null, 0, or 1	0	If set to 1, makes the phone use "sticky" line seize behavior. This will help with features that need a second call object to work with. The phone will attempt to initiate a new outgoing call on the same SIP line that is currently in focus on the LCD (this was the behavior in SIP 1.6.5). Dialing through the call list when there is no active call will use the line index for the previous call. Dialing through the call list when there is an active call will use the current active call line index. Dialing through the contact directory will use the current active call line index. If set to 0 or Null, the feature is disabled (this was the behavior in SIP 1.6.6). Dialing through the call list will use the line index for the previous call. Dialing through the contact directory will use a random line index. ***Note:*** *This may fail due to glare issues in which case the phone may select a different available line for the call.*

Attribute	Permitted Values	Default	Interpretation
call.stickyAutoLineSeize. onHookDialing	Null, 0, or 1	Null	If `call.stickyAutoLineSeize` is set to 1, this parameter has no effect. The regular stickyAutoLineSeize behavior is followed. If `call.stickyAutoLineSeize` is set to 0 or Null and this parameter is set to 1, this overrides the stickyAutoLineSeize behavior for hot dial only. (Any new call scenario seizes the next available line.) If `call.stickyAutoLineSeize` is set to 0 or Null and this parameter is set to 0 or Null, there is no difference between hot dial and new call scenarios. ***Note**: A hot dial occurs on the line which is currently in the call appearance. Any new call scenario seizes the next available line.*
call.singleKeyPressConference	0,1	0	If set to 1, the conference will be setup after a user presses the **Conference** soft key or **Conference** key the first time. Also, all sound effects (dial tone, DTMF tone while dialing and ringing back) are heard by all existing participants in the conference. If set to 0 or Null, sound effects are only heard by conference initiator (old behavior). Only supported for SoundPoint IP 550, 560,650 and 670 and SoundStation IP 7000. For all others, set to 0.
call.localConferenceCallHold	0 or 1	0	If set to 0, a hold will happen for all legs when conference is put on hold. (old behavior). If set to 1, only the host is out of the conference, all other parties in conference continue to talk. (new behavior). If set to Null, the default value is 0. Only supported for the SoundPoint IP 550, 560,650 and 670 and the SoundStation IP 7000 (refer to Manage Conferences on page 4-22). For all others, set to 0.
call.transfer.blindPreferred	0,1	Null	If set to 1, the blind transfer is the default mode. The **Normal** soft key is available to switch to a consultative transfer. If set to 0 or Null, the consultative transfer is the default mode. The **Blind** soft key is available to switch to a blind transfer. ***Note:** This parameter is supported on the SoundPoint IP 32x/33x only.*

Attribute	Permitted Values	Default	Interpretation
call.directedCallPickupString	star code	*97	The star code to initiate a directed call pickup. ***Note:*** *The default value supports the BroadWorks calls server only. You must change the value if your organization uses a different call server.*
call.directedCallPickupMethod	“native” or “legacy”	Null	The method the phone will use to perform a directed call pick-up of a BLF resource's inbound ringing call. “native” indicates the phone will use a native protocol method (in this case SIP INVITE with the Replaces header [4]). “legacy” indicates the phone will use the method specified in `call.directedCallPickupString`.
call.parkedCallRetrieveMethod	“native” or “legacy”	Null	The method the phone will use to retrieve a BLF resource's call which has dialog state confirmed. “native” indicates the phone will use a native protocol method (in this case SIP INVITE with the Replaces header [4]). “legacy” indicates the phone will use the method specified in `call.parkedCallRetrieveString` .
call.parkedCallRetrieveString	star code	Null	The star code used to initiate retrieve of a parked call.
call.autoAnswer.micMute	0 or 1	1	If set to 1, the microphone is initially muted after a call is auto-answered.
call.autoAnswer.videoMute	0 or 1	0	If set to 1, video Tx is initially disabled after a call is auto-answered. ***Note****: This parameter is supported on the Polycom VVX 1500 only.*
call.autoAnswer.SIP	0 or 1	0	If set to 1, auto-answer is enabled for all SIP calls. ***Note****: This parameter is supported on the Polycom VVX 1500 only.*
call.autoAnswer.H323	0 or 1	0	If set to 1, auto-answer is enabled for all H.323 calls. ***Note****: This parameter is supported on the Polycom VVX 1500 only.*
call.autoAnswer.ringClass	positive integer	4	The ring class (`se.rt.x`) to use when a call is to be automatically answered using the auto-answer feature. If set to a ring class with a type other than “answer” or “ring-answer”, the setting will be overridden such that a ring type of “visual” (no ringer) applies.

Attribute	Permitted Values	Default	Interpretation
call.autoRouting.preference	"line" or "protocol"	"line	If set to **line**, calls are placed via the first available line, regardless of its protocol capabilities. If the first available line has both SIP and H.323 capabilities, the preferred protocol will be used preferentially (`call.autoRouting.preferredProtocol`). If set to **protocol**, the first available line with the preferred protocol activated is used, if available, and if not available, the first available line will be used. ***Note**: Auto-routing is used when manual routing selection features are disabled. Refer to User Preferences <up/> on page A-31.* ***Note**: This parameter is supported on the Polycom VVX 1500 only.*
call.autoRouting. preferredProtocol	"SIP or "H323"	SIP	If set to **SIP**, calls are placed via SIP if available, or via H.323 if SIP is not available. If set to **H323**, calls are placed via H.323 if available, or via SIP if H.323 is not available. ***Note**: This parameter is supported on the Polycom VVX 1500 only.*

This attribute also includes:

- Shared Calls <shared/>
- Hold, Local Reminder <hold/><localReminder/>

Shared Calls <shared/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.shared.disableDivert	0 or 1	1	If set to 1, disable diversion feature for shared lines. ***Note:** This feature is disabled on most call servers*.
call.shared.seizeFailReorder	0 or 1	1	If set to 1, play re-order tone locally on shared line seize failure.

Attribute	Permitted Values	Default	Interpretation
call.shared.oneTouchResume	0 or 1	0	If set to 1, when a shared line has a call on hold the remote user can press that line and resume the call. If more than one call is on hold on the line then the first one will be selected and resumed automatically. If set to 0, pressing the shared line will bring up a list of the calls on that line and the user can select which call the next action should be applied to. ***Note:*** *This parameter affects the SoundStation IP 6000 and 7000 phones. For other phones, a quick press and release of the line key will resume a call whereas pressing and holding down the line key will show a list of calls on that line.*
call.shared.exposeAutoHolds	0 or 1	0	If set to 1, on a shared line, when setting up a conference, a re-INVITE will be sent to the server. If set to 0, no re-INVITE will be sent to the server.

Hold, Local Reminder <hold/><localReminder/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.hold.localReminder.enabled	0 or 1	0	If set to 1, periodically notify the local user that calls have been on hold for an extended period of time.
call.hold.localReminder.period	non-negative integer	60	Time in seconds between subsequent reminders.
call.hold.localReminder.startDelay	non-negative integer	90	Time in seconds to wait before the initial reminder.

Directory <dir/>

This attribute includes:

- Local Directory <local/>
- Corporate Directory <corp/>

Local Directory <local/>

The local directory is stored in either flash memory or RAM on the phone. The local directory size is limited based on the amount of flash memory in the phone. (Different phone models have variable flash memory.)

When the volatile storage option is enabled, ensure that a properly configured provisioning server that allows uploads is available to store a back-up copy of the directory or its contents will be lost when the phone reboots or loses power.

Attribute	Permitted Values	Default	Interpretation
dir.local.volatile.2meg	0 or 1	0	Applies to platforms with 2 Mbytes of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
dir.local.nonVolatile.maxSize.2meg	1 to 20	20	Applies to platforms with 2 Mbytes of flash memory. Maximum size in Kbytes of non-volatile storage that the directory will be permitted to consume.
dir.local.volatile.4meg	0 or 1	0	Applies to platforms with 4 Mbytes of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
dir.local.nonVolatile.maxSize.4meg	1 to 50	50	Applies to platforms with 4 Mbytes of flash memory. Maximum size in Kbytes of non-volatile storage that the directory will be permitted to consume.
dir.local.volatile.maxSize	1 to 200	200	When the volatile storage option is set (refer to see `dir.local.volatile.4meg` and `dir.local.volatile.8meg`), this attribute is the maximum size of contact directory file that the phone supports. Note that phones with 16 MB RAM support up to 50 Kbytes of directory file, and phones with more than 16 MB RAM support up to 200 Kbytes of directory file. When the value specified for this attribute exceeds the limit, the limit will be used as the max. directory size.

Attribute	Permitted Values	Default	Interpretation
dir.local.volatile.8meg	0 or 1	0	Attribute applies only to platforms with 8 Mbytes or more of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
dir.local.nonVolatile.maxSize.8meg	1 to 100	100	Attribute applies only to platforms with 8 Mbytes or more of flash memory. This is the maximum size of non-volatile storage that the directory will be permitted to consume.
dir.local.readonly	0 or 1	1	Specifies whether or not local contact directory is read only. If set to 0 or Null, the local contact directory is editable. If set to 1, the local contact directory is read only. ***Note:*** *If the local contact directory is read only, speed dial entry on the SoundPoint IP 32x/33x is disabled (enter the speed dial index followed by "#").*
dir.search.field	0 or 1	Null	Specifies how to search the contact directory. If set to 1, search by contact's first name. If set to 0, search by contact's last name.

Corporate Directory <corp/>

A portion of the corporate directory is stored in flash memory on the phone. The size is based on the amount of flash memory in the phone. (Different phone models have variable flash memory.)

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dir.corp.address	dotted-decimal IP address or host name or FQDN	Null	The IP address or host name of the LDAP server interface to the corporate directory. For example, host.domain.com.
dir.corp.port	0, Null, 1 to 65535	389 (TCP) 636 (TLS)	This parameter is used to specify the port to connect to on the server, if a full URL is not provided.
dir.corp.transport	TCP, TLS, Null	TCP	This parameter is used to specify whether a TCP or TLS connection is made with the server, if a full URL is not provided.
dir.corp.baseDN	UTF-8 encoded string	Null	The base domain name is the starting point for making queries on the LDAP server.
dir.corp.user	UTF-8 encoded string	Null	The username used to authenticate to the LDAP server.
dir.corp.password	UTF-8 encoded string	Null	The password used to authenticate to the LDAP server.
dir.corp.filterPrefix	UTF-8 encoded string	(objectclass=person)	Predefined filter string. If set to Null or invalid, "(objectclass=person)" is used.
dir.corp.scope	"one", "sub", "base"	"sub"	Type of search. If set to "one", a search of the level one below the baseDN is performed. If set to "sub" or Null, a recursive search (of all levels below the baseDN) is performed. If set to "base", a search at the baseDN level is performed.
dir.corp.attribute.x.name	UTF-8 encoded string	Null	The name of the attribute to match on the server. Each name must be unique, however, an LDAP entry can have multiple attributes with the same name. Up to eight attributes can be configured (x = 1 to 8).
dir.corp.attribute.x.label	UTF-8 encoded string	Null	A UTF-8 encoded string that is used as the label when data is displayed.

Attribute	Permitted Values	Default	Interpretation
dir.corp.attribute.x.type	first_name, last_name, phone_number, SIP_address, H323_address URL, other	last_name	This parameter defines how the attribute is interpreted by the phone. Entries can have multiple attributes of the same type. Type 'other' is used for display purposes only. If the user saves the entry to the local contact directory on the phone, first_name, last_name, and phone_number are copied. The user can place a call to the phone_number and SIP_address from the corporate directory.
dir.corp.attribute.x.sticky	0 or 1	Null	If set to 0 or Null, the filter criteria for this attribute is reset after a reboot. If set to 1, the filter criteria for this attribute is retained through a reboot. Such attributes are denoted with a "*" before the label when displayed on the phone.
dir.corp.attribute.x.filter	UTF-8 encoded string	Null	The filter string for this attribute, which is edited when searching.
dir.corp.attribute.x.searchable	0 or 1	0	A flag to determine if the attribute is searchable through quick search. This flag applies for x = 2 or greater. If set to 0 or Null, quick search on this attribute is disabled. If set to 1, quick search on this attribute is enabled.
dir.corp.backGroundSync	0 or 1	0	If set to 0 or Null, there will be no background downloading from the LDAP server. If set to 1, there will be background downloading of data from the LDAP server.
dir.corp.backGroundSync.period	3600 to 604800 seconds	86400	The corporate directory cache is refreshed after the corporate directory feature has not been used for this period of time. The default period is 24 hours. The minimum is 1 hour and the maximum is 7 days.

Attribute	Permitted Values	Default	Interpretation
dir.corp.viewPersistence	0 or 1	1	If set to 0, the browse position in the data on the LDAP server and the attribute filters are reset for subsequent usage of the corporate directory. If set to 1or Null, the browse position in the data and the attribute filters are retained for subsequent usage of the corporate directory.
dir.corp.cacheSize	8 to 256	128	The maximum number of entries that can be cached locally on the phone.
dir.corp.pageSize	8 to 64	32	The maximum number of entries requested from the corporate directory server with each query.
dir.corp.vlv.allow	0 or 1	0	A flag to determine whether or not VLV queries can be made if the LDAP server supports VLV. If set to 0, VLV queries are disabled. If set to 1 or Null, VLV queries are enabled. ***Note**: If VLV is enabled, `dir.corp.attribute.x.searchable` is ignored.*
dir.corp.vlv.sortOrder	list of attributes	Null	The list of attributes (in the exact order) to be used by the LDAP server when indexing. For example, sn, givenName, telephoneNumber.

Attribute	Permitted Values	Default	Interpretation
dir.corp.autoQuerySubmitTimeout	0 to 60 seconds	0	To control if there is a timeout after the user stops entering characters in the quick search and, if there is, how long the timeout is. If set to 0, there is not (disabled).
dir.corp.sortCtrl	0 or 1	Null	Controls how client makes queries and does it sort entries locally. It should not be used by users. If set to 0 or Null, leave sorting as negotiated between client and server. If set to 1, force sorting of queries. ***Note**: Polycom does not recommend setting `dir.corp.sortCtrl` to 1 as it causes excessive LDAP queries. It should be used to diagnose LDAP servers with sorting problems only.*

Presence <pres/>

The parameter `pres.reg` is the line number used to send SUBSCRIBE. If this parameter is missing, the phone will use the primary line to send SUBSCRIBE.

Attribute	Permitted Values	Default	Interpretation
pres.reg	positive integer	1	Specifies the line/registration number used to send SUBSCRIBE for presence. Must be a valid line/registration number. If the number is not a valid line/registration number, it is ignored.
pres.idleSoftkeys	0 or 1	Null	If set to Null or 0, the presence idle soft keys (**MyStat** and **Buddies**) do not appear. If set to Null or 1, the presence idle soft keys appear.

Fonts <font/>

These settings control the phone's ability to dynamically load an external font file during boot up. Loaded fonts can either overwrite pre-existing fonts embedded within the software (not recommended) or can extend the phone's

font support for Unicode ranges not already embedded. The font file must be a Microsoft **.fnt** file format. The font file name must follow a specific pattern as described:

- Font filename: `<fontName>_<fontHeightInPixels>_<fontRange>.<fontExtension>`
- <fontName> is a free string of characters that typically carries the meaning of the font. Examples are "fontFixedSize" for a fixed-size font, or "fontProportionalSize" for a proportional size font.
- <fontHeightInPixels> describes the font height in number of screen pixels.
- <fontRange> describes the Unicode range covered by this font. Since **.fnt** are 256 characters based blocks, the <fontRange> is Uxx00_UxxFF (**.fnt** file). For more information, refer to Multilingual User Interface on page 4-29.
- <fontExtension> describes the file type. Either **.fnt** for single 256 characters font .

If it is necessary to overwrite an existing font, use these <fontName>_<fontHeightInPixels>:

SoundPoint IP 32x, 33x, 430, 450	
"fontProp_10"	This is the font used widely in the current implementation.
"fontPropSoftkey_10"	This is the soft key specific font.
SoundPoint IP 550, 560, 650, and 670	
"fontProp_19"	This is the font used widely in the current implementation including for soft keys.
"fontProp_26"	This is the font used to display time (but not date).
"fontProp_x"	This is a small font used for the CPU/Load/Net utilization graphs, this is the same as the "fontProp_10" for the SoundPoint IP 500.

If the <fontName>_<fontHeightInPixels> does not match any of the names above, then the downloaded font will be applied against all fonts defined in the phone, which means that you may lose the benefit of fonts being calibrated differently depending on their usage. For example, the font used to display the time on the SoundPoint IP 650 is a large font, larger than the one used to display the date, and if you overwrite this default font with a unique font, you lose this size aspect. For example:

- to overwrite the font used for SoundPoint IP 500 soft keys for ASCII, the name should be **fontPropSoftkey_10_U0000_U00FF.fnt** .
- to add support for a new font that will be used everywhere and that is not currently supported. For example, for the Eastern/Central European Czech language, this is Unicode range 100-17F, the name could be **fontCzechIP500_10_U0100_U01FF.fnt** and **fontCzechIP600_19_U0100_U01FF.fnt** .

The font delimiter is important to retrieve the different scrambled **.fnt** blocks. This font delimiter must be placed in the "copyright" attribute of the **.fnt** header. If you are simply adding or changing a few fonts currently in use, multiple **.fnt** files are recommended since they are easier to work with individually.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.delimiter	string up to 256 ASCII characters	Null	Delimiter required to retrieve different grouped **.fnt** blocks.

This attribute also includes:

- IP_330 font <IP_330/>
- IP_400 font <IP_400/>
- IP_500 font <IP_500/>
- IP_600 font <IP_600/>

IP_330 font <IP_330/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_330.x.name	**fontName_height_Uxx00_UxxFF.fnt**	Null	Defines the font file that will be loaded from provisioning server during boot up. ***Note:*** *When several font.IP_330.x.name are defined, the index x must follow consecutive increasing order.*

IP_400 font <IP_400/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_400.x.name	**fontName_height_Uxx00_UxxFF.fnt**	Null	Defines the font file that will be loaded from provisioning server during boot up. ***Note:*** *When several font.IP_400.x.name are defined, the index x must follow consecutive increasing order.*

IP_500 font <IP_500/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_500.x.name	**fontName_height_Uxx00_UxxFF.fnt**	Null	Defines the font file that will be loaded from provisioning server during boot up. ***Note:*** *When several font.IP_500.x.name are defined, the index x must follow consecutive increasing order.*

IP_600 font <IP_600/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
font.IP_600.x.name	**fontName_height_Uxx00_UxxFF.fnt**	Null	Defines the font file that will be loaded from provisioning server during boot up. ***Note:*** *When several font.IP_600.x.name are defined, the index x must follow consecutive increasing order.*

Keys <key/>

These settings control the scrolling behavior of keys and can be used to change key functions.

Attribute	Permitted Values	Default	Interpretation
key.scrolling.timeout	positive integer	1	The time-out after which a key that is enabled for scrolling will go into scrolling mode until the key is released. Keys enabled for scrolling are menu navigation keys (left, right, up, down arrows), volume keys, and some context-specific soft keys. The value is an integer multiple of 500 milliseconds (1=500ms).

SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670, SoundStation IP 6000 and 7000, and Polycom VVX 1500 key functions can be changed from the factory defaults, although this is typically not necessary. For each key whose function you wish to change, add an XML attribute in the format described in the following table to the <keys .../> element of the configuration file. These will override the built-in assignments.

Polycom does not recommend the remapping for keys.

In the following table, *x*=IP_ IP_330, IP 430, IP_450, IP_550, IP_650, and IP_6000, and IP_7000 and *y* is the key number. Note that IP_330 parameters affect SoundPoint IP 32x/33x phones, IP_430 parameters affect SoundPoint IP 430 phones, IP_550 parameters affect SoundPoint IP 550 and 560 phones, IP_650 parameters affect SoundPoint IP 650 and 670 phones, IP_6000 parameters affect the SoundStation IP 6000 phones, IP_7000 parameters affect the SoundStation IP 7000 phones, and the VVX_1500 parameters affect the Polycom VVX 1500 phones. IP 330: *y*=1-34; IP 430: *y*=1-35; IP_450: *y*=1-35; IP_550: *y*=1-40; IP_650:*y*=1-42; IP_6000:*y*=1-29; IP_7000:*y*=1-30; VVX_1500:*y*=1-42.

Attribute	Permitted Values	Interpretation
key.x.y.function.prim	Functions listed below.	Sets the function for key y on platform x.
key.x.y.subPoint.prim	positive integer	Sets the sub-identifier for key functions with a secondary array identifier such as SpeedDial.

The following table lists the functions that are available:

Functions			
ArrowDown	Dialpad5	Line2	Select
ArrowLeft	Dialpad6	Line3	Setup
ArrowRight	Dialpad7	Line4	SoftKey1
ArrowUp	Dialpad8	Line5	SoftKey2
BuddyStatus	Dialpad9	Line6	SoftKey3
CallList	DialpadStar	Messages	SoftKey4
Conference	DialpadPound	Menu	SpeedDial
Delete	Directories	MicMute	SpeedDialMenu
Dialpad0	DoNotDisturb	MyStatus	Transfer
Dialpad1	Handsfree	Null	Video
Dialpad2	Headset	Offline	VolDown
Dialpad3	Hold	Redial	VolUp
Dialpad4	Line1	Release	

Backgrounds <bg/>

The backgrounds used by the SoundPoint IP 450, 550, 560, 650, and 670 and the Polycom VVX 1500 phones are defined in this section. In the following table, *w*=1 to 3, *x*=1 to 6. `hiRes` parameters are used by SoundPoint IP 550, 560, 650, and 670 phones, `medRes` parameters are used by SoundPoint IP 450 phones, and VVX_1500 parameters are used by Polycom VVX 1500 phones.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
bg.VVX_1500.color.selection	w,x	1,1	Specify which type of background (w) and index for that type (x) is selected on reboot where w=1 to 3, x=1 to 6. The type of backgrounds are built-in (w=1), solids (w=2), and bitmaps (w=3). w=2 is used when selecting any image as a background. w=3 is used when selecting any image from the Digital Picture Frame as a background. This image is stored under "Local File". Only one local file at a time is supported.
bg.VVX_1500.color.bm.x.name	any string	Null	Graphic files for display on the phone. For example, if you set `bg.VVX_1500.color.bm.1.name` to **Polycom.bmp**, the user will be able to select "Polycom.bmp" as a background on the phone.
bg.hiRes.color.selection	w,x	1,1	Specify which type of background (w) and index for that type (x) is selected on reboot where w=1 to 3, x=1 to 6.
bg.hiRes.color.pat.solid.x. name	any string		Solid pattern name. For x=1: Light Blue, x=2: Teal, x=3: Tan, x=4:Null The screen background layouts. For x=1, red (151), green, (207), blue (249) For x=2, red (73), green (148), blue (148) For x=3, red (245), green (157), blue (69) For x=4, red (Null), green (Null), blue (Null)
bg.hiRes.color.pat.solid.x.red	0 to 255		
bg.hiRes.color.pat.solid.x. green	0 to 255		
bg.hiRes.color.pat.solid.x.blue	0 to 255		

Attribute	Permitted Values	Default	Interpretation
bg.hiRes.color.bm.x.name	any string	built-in value of “Thistle”	Graphic files for display on the phone and Expansion Module. For x=1: • name is “Leaf.jpg” name is “LeafEM.jpg” For x=2: • name is “Sailboat.jpg” name is “SailboatEM.jpg” For x=3: • name is “Beach.jpg” name is “BeachEM.jpg” For x=4: • name is “Palm.jpg” name is “PalmEM.jpg” For x=5: • name is “Jellyfish.jpg” name is “JellyfishEM.jpg” For x=6: • name is “Mountain.jpg” name is “MountainEM.jpg” ***Note:*** *If the file is missing or unavailable, the built-in default solid pattern is displayed.*
bg.hiRes.color.bm.x.em.name	any string		
bg.hiRes.gray.selection	w,x	2,1	Specify which type of background (w) and index (x) for that type is selected on reboot.
bg.hiRes.gray.pr.x.adj		-3	Specify the brightness adjustment to the graphic.
bg.hiRes.gray.pat.solid.x.name	any string	White	Solid pattern name. For x=1: White, x=2: Light Gray, x=3, 4: Null
bg.hiRes.gray.pat.solid.x.red	0 to 255		The screen background layouts. For x=1, red (255), green, (255), blue (255) For x=2, red (160), green (160), blue (160) For x=3 and 4, all values are Null. ***Note****: The values for red, green, and blue must be the same to display correctly on grayscale.*
bg.hiRes.gray.pat.solid.x.green	0 to 255		
bg.hiRes.gray.pat.solid.x.blue	0 to 255		

Attribute	Permitted Values	Default	Interpretation
bg.hiRes.gray.bm.x.name	any string		Graphic files for display on the phone and Expansion Module and also the brightness adjustment to the graphic. For x=1: • name is "Leaf.jpg" name is "LeafEM.jpg" adjustment is "0" For x=2: • name is "Sailboat.jpg" name is "SailboatEM.jpg" adjustment is "-3" For x=3: • name is "Beach.jpg" name is "BeachEM.jpg" adjustment is "0" For x=4: • name is "Palm.jpg" name is "PalmEM.jpg" adjustment is "-3" For x=5: • name is "Jellyfish.jpg" name is "JellyfishEM.jpg" adjustment is "-2" For x=6: • name is "Mountain.jpg" name is "MountainEM.jpg" adjustment is "0" ***Note:*** *If the file is missing or unavailable, the built-in default solid pattern is displayed.* ***Note:*** *The adjustment value is changed on each individual phone when the user lightens or darkens the graphic during preview.*
bg.hiRes.gray.bm.x.em.name	any string		
bg.hiRes.gray.bm.x.adj	integer		
bg.medRes.gray.selection	w,x	2,1	Specify which type of background (w) and index (x) for that type is selected on reboot.
bg.medRes.gray.pr.x.adj		-3	Specify the brightness adjustment to the graphic.
bg.medRes.gray.pat.solid.x.name	any string	White	Solid pattern name. For x=1: White, x=2: Light Gray, x=3, 4: Null
bg.medRes.gray.pat.solid.x.red	0 to 255		The screen background layouts. For x=1, red (255), green, (255), blue (255) For x=2, red (160), green (160), blue (160) For x=3 and 4, all values are Null. ***Note****: The values for red, green, and blue must be the same to display correctly on grayscale.*
bg.medRes.gray.pat.solid.x.green	0 to 255		
bg.medRes.gray.pat.solid.x.blue	0 to 255		

Attribute	Permitted Values	Default	Interpretation
bg.medRes.gray.bm.x.name	any string		Graphic files for display on the phone and Expansion Module and also the brightness adjustment to the graphic. For x=1: • name is “Leaf256x116.jpg” adjustment is “0” For x=2: • name is “Sailboat256x116.jpg” adjustment is “-3” For x=3: • name is “Beach256x116.jpg” adjustment is “0” For x=4: • name is “Palm256x116.jpg” adjustment is “-3” For x=5: • name is “Jellyfish256x116.jpg” adjustment is “-2” For x=6: • name is “Mountain256x116.jpg” adjustment is “0” ***Note:*** *If the file is missing or unavailable, the built-in default solid pattern is displayed.* ***Note:*** *The adjustment value is changed on each individual phone when the user lightens or darkens the graphic during preview.*
bg.medRes.gray.bm.x.em. name	any string		
bg.medRes.gray.bm.x.adj	integer		
button.color.selection.x.y. modify	any string		The label color for soft keys and line key labels associated with the defined colored backgrounds. These values can be modified locally by the user. The format is: “rgbHILo, <parameter list>”. For example: “rbgHiLo, 51, 255, 68, 255, 0, 119” is the default button color associated with the built-in background.
button.gray.selection.x.y. modify	any string		The label color for soft keys and line key labels associated with the defined gray backgrounds. These values can be modified locally by the user. The format is: “rgbHILo, <parameter list>”. By default, all defaults are set to “none”.

Bitmaps <bitmap/>

The bitmaps used by each phone model are defined in this section.

Platform <IP 330/>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/>

In the following table, *x*=IP_330, IP_400, IP_450, IP_600, IP_4000, or IP_7000 and *y* is the bitmap number. Note that IP_330 parameters affect SoundPoint IP 32x/33x phones, IP_400 parameters affect SoundPoint IP 430 phones, IP_450 parameters affect SoundPoint IP 450 phones, IP_600 parameters affect SoundPoint IP 550, 560, 600, 601, and 650, and 670 phones, IP_4000 parameters affect SoundStation IP 6000 phones, and IP_7000 parameters affect SoundStation IP 7000 phones.

Attribute	Permitted Values	Interpretation
bitmap.x.y.name	The name of a bitmap to be used.	This is the name of a bitmap to be used for creating an animation. If the bitmap is to be downloaded from the provisioning server, its name must: • Be different from any name already in use in **sip.cfg**. • Match the name of the corresponding <fileName>.bmp to be retrieved from the provisioning server.

Indicators <ind/>

The following indicators are used by the phone:

- Animations <anim/> <IP_330/>, <IP_335/>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/>
- Patterns <pattern/>
- Classes <class/>
- Assignments

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
ind.idleDisplay.mode	1 (default), 2, 3	Null	The idle display animation screen layouts. For example, for the SoundPoint IP 330/320: • If set to 1 or Null, the idle display animation size is 87 x 11 pixels. • If set to 2, the idle display animation size is 87 x 22 pixels. • If set to 3, the idle display animation size is 102 x 22 pixels.
ind.idleDisplay.enabled	0 or 1	0	If set to 1, the idle display may support presentation of a custom animation if configured in the animation section of **sip.cfg**.

Animations <anim/> <IP_330/>, <IP_335/>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/>

This section defines bitmap animations composed of bitmap/duration couples. In the following table, *x*=IP_330, IP_335, IP_400, IP_450, IP_600, IP_4000 or IP_7000, *y* is the animation number, *z* is the step in the animation. Note that IP_330 parameters affect SoundPoint IP 32x/33x phones, IP_335 parameters affect SoundPoint IP 335 phones, IP_400 parameters affect SoundPoint IP 430 phones, IP_450 parameters affect SoundPoint IP 450 phones, IP_600 parameters affect SoundPoint IP 550, 560, 600, 601, 650, and 670 phones, IP_4000 parameters affect SoundStation IP 6000 phones, and IP_7000 parameters affect SoundStation IP 7000 phones.

Note As of SIP 2.2.0, a maximum of 24 frames per animation is supported.

Attribute	Permitted Values	Interpretation
ind.anim.x.y.frame.z.bitmap	A bitmap name defined previously.	Bitmap to use. Note that it must be defined already, refer to Platform <IP 330/>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/> on page A-100.
ind.anim.x.y.frame.z.duration	positive integer	Duration in milliseconds for this step. 0=infinite.

Patterns <pattern/>

This section defines patterns for the LED indicators. In the following table, *x* is the pattern number, *y* is the step in the pattern.

Attribute	Permitted Values	Interpretation
ind.pattern.x.step.y.state	On or Off	Turn LED on or off for this step.
ind.pattern.x.step.y.duration	positive integer	Duration in milliseconds for this step. 0=infinite
ind.pattern.x.step.y.colour	Red or Green (default is Red if not specified)	For bi-color LEDs, specify color.

Classes <class/>

This section defines the available classes for the LED and graphical icon indicator types. In the following table, *x* is the class number, *y* is the identifier of the state number for that class.

Attribute	Permitted Values	Interpretation
ind.class.x.state.y.index	positive integer	For LED type indicators, index refers to the pattern index, such as index x in the Patterns <pattern/> tag above. For Graphic Icon type indicators, index refers to the animation index, such as index y in the Animations <anim/> <IP_330/>, <IP_335/>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/> tag above.

Assignments

This attribute assigns a type and a class to an indicator. In the case of the Graphic Icon type, it also assigns a physical location and size in pixels on the LCD display (refer to the next section). In the case of the LED type, it assigns a physical LED number (refer to Graphic Icons <gi/> <IP_330>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/> on page A-103).

LEDs <led/>

In the following table, *x* is the LED number.

Attribute	Permitted Values	Interpretation
ind.led.x.index		This is for internal usage only and should not be changed (this is the logical index).
ind.led.x.class	positive integer	Assigns the class (defined in Classes <class/> on page A-102) for this indicator.
ind.led.x.physNum		This maps the logical index to a specific physical LED.

Graphic Icons <gi/> <IP_330>, <IP_400/>, <IP_450/>, <IP_600/>, <IP_4000/>, and <IP_7000/>

In the following table, *x*=IP_330, IP_400, IP_500, IP_600, IP_4000, or IP_7000, *y* is the graphic icon number. Note that IP_330 parameters affect SoundPoint IP 32x/33x phones, IP_400 parameters affect SoundPoint IP 430 phones, IP_450 parameters affect SoundPoint IP 450 phones, and IP_600 parameters affect SoundPoint IP 550, 560, 600, 601, 650, and 670 phones, IP_4000 parameters affect SoundStation IP 6000 phones, and IP_7000 parameters affect SoundStation IP 7000 phones.

Attribute	Permitted Values	Interpretation
ind.gi.x.y.index		This is for internal usage only and should not be changed (this is the logical index).
ind.gi.x.y.class	positive integer	Assigns the class (defined in Classes <class/> on page A-102) for this indicator.
ind.gi.x.y.physX	IP 330: 0-101 IP 400: 0-122 IP 450: 0-238 IP 600: 0-319 IP 4000: 0-247 IP 7000: 0-255	For Graphic Icon type indicators, this is the x-axis location of the upper left corner of the indictor measured in pixels from left to right.

Attribute	Permitted Values	Interpretation
ind.gi.x.y.physY	IP 330: 0-19 IP 400: 0-45 IP 450: 0-89 IP 600: 0-159 IP 4000: 0-67 IP 7000: 0-127	For Graphic Icon type indicators, this is the y-axis location of the upper left corner of the indicator measured in pixels from top to bottom.
ind.gi.x.y.physW	IP 330: 1-87 IP 400: 1-102 IP 450: 1-170 IP 600: 1-320 IP 4000: 1-248 IP 7000: 1-256	For Graphic Icon type indicators, this is the width of the indicator measured in pixels.
ind.gi.x.y.physH	IP 330: 1-20 IP 400: 1-23 IP 450: 1-73 IP 600: 1-160 IP 4000: 1-68 IP 7000: 1-128	For Graphic Icon type indicators, this is the height of the indicator measured in pixels.

Event Logging <log/>

Warning Logging parameter changes can impair system operation. Do not change any logging parameters without prior consultation with Polycom Technical Support.

The event logging system supports the following classes of events:

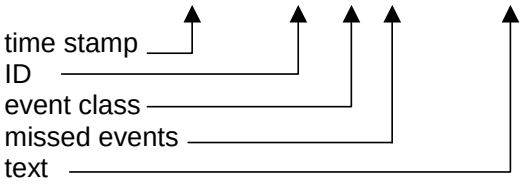
Level	Interpretation
0	Debug only
1	High detail event class
2	Moderate detail event class
3	Low detail event class
4	Minor error - graceful recovery
5	Major error - will eventually incapacitate the system
6	Fatal error

Each event in the log contains the following fields separated by the | character:

- time or time/date stamp
- 1-5 character component identifier (such as “so”)
- event class
- cumulative log events missed due to excessive CPU load
- free form text - the event description

Example:

```
011511.006|so   |2|00|soCoreAudioTermChg: chassis -> idle
```



Three formats are available for the event timestamp:

Type	Example
0 - seconds.milliseconds	`011511.006` -- 1 hour, 15 minutes, 11.006 seconds since booting.
1 - absolute time with minute resolution	`0210281716` -- 2002 October 28, 17:16
2 - absolute time with seconds resolution	`1028171642` -- October 28, 17:16:42

Two types of logging are supported:

- Basic Logging <level/><change/> and <render/>
- Scheduled Logging Parameters <sched/>

Basic Logging <level/><change/> and <render/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
log.level.change.xxx	0-5	4	Control the logging detail level for individual components. These are the input filters into the internal memory-based log system. Possible values for xxx are so, app1, sip, sspsc, ssps, pps, net, cfg, cdp, pmt, ftp, ares, dns, cxss, httpd, rdisk, copy, slog, res, key, log, curl, rtos, mb, ib, sotet, ttrs, srtp, usb, efk, clink, ldap, and peer, pnetm, cmp, cmr, usbio, pres, pwrsv, and lldp, wmgr, push, poll, and h323.
log.render.level	0-6	1	Specifies the lowest class of event that will be rendered to the log files. This is the output filter from the internal memory-based log system. The `log.render.level` maps to syslog severity as follows: 0 -> SeverityDebug (7) 1 -> SeverityDebug (7) 2 -> SeverityInformational (6) 3 -> SeverityInformational (6) 4 -> SeverityError (3) 5 -> SeverityCritical (2) 6 -> SeverityEmergency (0) 7 -> SeverityNotice (5) For more information, refer to Syslog Menu on page 3-13.
log.render.type	0-2	2	Refer to above table for timestamp type.
log.render.realtime	0 or 1	1	Set to 1. ***Note:*** *Polycom recommends that you do not change this value.*
log.render.stdout	0 or 1	1	Set to 1. ***Note:*** *Polycom recommends that you do not change this value.*
log.render.file	0 or 1	1	Set to 1. ***Note:*** *Polycom recommends that you do not change this value.*

Attribute	Permitted Values	Default	Interpretation
log.render.file.size	positive integer, 1 to 179.5	16	Maximum local application log file size in Kbytes. When this size is exceeded, the file is uploaded to the provisioning server and the local copy is erased.
log.render.file.upload.period	positive integer	172800	Time in seconds between log file uploads to the provisioning server. ***Note:*** *The log file will not be uploaded if no new events have been logged since the last upload*.
log.render.file.upload.append	0 or 1	1	If set to 1, use append mode when uploading log files to server. ***Note:*** *HTTP and TFTP don't support append mode unless the server is set up for this*.
log.render.file.upload.append.sizeLimit	positive integer	512	Maximum log file size on provisioning server in Kbytes.
log.render.file.upload.append.limitMode	delete, stop	delete	Behavior when server log file has reached its limit. delete=delete file and start over stop=stop appending to file

Scheduled Logging Parameters <sched/>

The phone can be configured to schedule certain advanced logging tasks on a periodic basis. These attributes should be set in consultation with Polycom Technical Support. Each scheduled log task is controlled by a unique attribute set starting with log.sched.x where *x* identifies the task.

Attribute	Permitted Values	Interpretation
log.sched.x.name	alphanumeric string	Name of an internal system command to be periodically executed. To be supplied by Polycom.
log.sched.x.level	0-5	Event class to assign to the log events generated by this command. This needs to be the same or higher than log.level.change.slog for these events to appear in the log.
log.sched.x.period	positive integer	Seconds between each command execution. 0=run once

Attribute	Permitted Values	Interpretation
log.sched.x.startMode	abs, rel	Start at *abs*olute time or *rel*ative to boot.
log.sched.x.startTime	positive integer OR hh:mm	Seconds since boot when startMode is *rel* or the start time in 24-hour clock format when startMode is *abs*.
log.sched.x.startDay	1-7	When startMode is *abs*, specifies the day of the week to start command execution. 1=Sun, 2=Mon, ..., 7=Sat

Security <sec/>

This attribute's settings affect security aspects of the phone.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.tagSerialNo	0 or 1	Null	If set to 1, the phone may advertise its serial number (Ethernet address) through protocol signaling. If set to 0 or Null, the phone does advertise its serial number.

This attribute also includes:

- Encryption <encryption/>
- Password Lengths <pwd/><length/>
- SRTP <srtp/>
- H.235 <H235/>

Encryption <encryption/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.encryption.upload.dir	0 or 1	0	If set to 0, the phone-specific contact directory is uploaded to the server unencrypted regardless of how it was downloaded. This will replace whatever phone-specific contact directory is on the server even if it is encrypted. If set to 1, the phone-specific contact directory is uploaded encrypted regardless of how it was downloaded. This will replace whatever phone-specific contact directory is on the server even if it is unencrypted.
sec.encryption.upload.overrides	0 or 1	0	If set to 0, the phone-specific configuration override file (<Ethernet Address>-phone.cfg) is uploaded unencrypted regardless of how it was downloaded. This will replace the override file on the server even if it is encrypted. If set to 1, the phone-specific configuration override file is uploaded encrypted regardless of how it was downloaded. This will replace the override file on the server even if it is unencrypted.

Password Lengths <pwd/><length/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.pwd.length.admin	0-32	1	Password changes will need to be at least this long. Use 0 to allow null passwords.
sec.pwd.length.user	0-32	2	

SRTP <srtp/>

Note As per RFC 3711, you cannot turn off authentication of RTCP.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.srtp.enable	0 or 1	Null	If set to 1 or Null, the phone accepts SRTP offers. If set to 0, the phone always declines SRTP offers. ***Note**: The default behavior changed in SIP 3.2.0 . In previous SIP releases, the default value was 0 when null or not defined.*
sec.srtp.offer	0 or 1	Null	If set to 1 or Null, the phone includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE. This is for the phone initiating (offering) a phone call. If set to 0, no secure media stream is included in SDP of a SIP invite.
sec.srtp.require	0 or 1	Null	If set to 1, the phone is only allowed to use secure media streams. Any offered SIP INVITEs must include a secure media description in the SDP or the call will be rejected. For outgoing calls, only a secure media stream description is include in the SDP of the SIP INVITE, meaning that the non-secure media description is not included. If `sec.srtp.require` is set to 1, `sec.srtp.offer` is logically set to 1 no matter what the value in the configuration file. If set to 0 or Null, secure media streams are not required.
sec.srtp.offer. HMAC_SHA1_80	0 or 1	Null	If set to 1 or Null, a crypto line with the AES_CM_128_HMAC_SHA1_80 crypto-suite will be included in offered SDP. If set to 0, the crypto line is not included. ***Note:** This parameter was added in SIP 2.2.1 .*
sec.srtp.offer. HMAC_SHA1_32	0 or 1	Null	If set to 1, a crypto line with the AES_CM_128_HMAC_SHA1_32 crypto-suite will be included in offered SDP. If set to 0 or Null, the crypto line is not included. ***Note:** This parameter was added in SIP 2.2.1 .*

Attribute	Permitted Values	Default	Interpretation
sec.srtp.key.lifetime	0, positive integer minimum 1024	Null	The master key lifetime used for the cryptographic attribute in the SDP. The value specified is the number of SRTP packets. If set to 0 or Null, the master key lifetime is not set. If set to 1 or greater, master key lifetime is set. The default setting should be suitable for most installations. When the lifetime is set greater than 0, a re-invite with a new key will be sent when the number of SRTP packets sent for an outgoing call exceeds half the value of the master key lifetime. ***Note**: Setting this parameter to a non-zero value may affect performance of the phone.*
sec.srtp.mki.enabled	0 or 1	Null	The master key identifier (MKI) is an optional parameter for the cryptographic attribute in the SDP that uniquely identifies the SRTP stream within an SRTP session. MKI is expressed as a pair of decimal numbers in the form: |mki:mki_length| where mki is the MKI value and mki_length its length in bytes. If set to 1, a four-byte MKI parameter is sent within the SDP message of the SIP INVITE / 200 OK. If set to 0 or Null, the MKI parameter is not sent.
sec.srtp.sessionParams.noAuth.offer	0 or 1	Null	If set to 1, no authentication of RTP is offered. A session description that includes the UNAUTHENTICATED_SRTP session parameter is sent when initiating a call. If set to 0 or Null, authentication is offered.
sec.srtp.sessionParams.noAuth.require	0 or 1	Null	If set to 1, no authentication of RTP is required. A call placed to a phone configured with `noAuth.require` must offer the UNAUTHENTICATED_SRTP session parameter in its SDP. If `sec.srtp.sessionParams.noAuth.require` is set to 1, `sec.srtp.sessionParams.noAuth.offer` is logically set to 1 no matter what the value in the configuration file. If set to 0 or Null, authentication is required.
sec.srtp.sessionParams.noEncrypRTCP.offer	0 or 1	Null	If set to 1, no encryption of RTCP is offered. A session description that includes the UNENCRYPTED_SRTCP session parameter is sent when initiating a call. If set to 0, or Null, encryption of RTCP is offered.

Attribute	Permitted Values	Default	Interpretation
sec.srtp.sessionParams.noEncrypRTCP.require	0 or 1	Null	If set to 1, no encryption of RTCP is required. A call placed to a phone configured with noAuth.require must offer the UNENCRYPTED_SRTCP session parameter in its SDP. If `sec.srtp.sessionParams.noEncryptRTCP.require` is set to 1, `sec.srtp.sessionParams.noEncryptRTCP.offer` is logically set to 1 no matter what the value in the configuration file. If set to 0 or Null, encryption of RTCP is required.
sec.srtp.sessionParams.noEncrypRTP.offer	0 or 1	Null	If set to 1, no encryption of RTP is offered. A session description that includes the UNENCRYPTED_SRTP session parameter is sent when initiating a call. If set to 0, or Null, encryption of RTP is offered.
sec.srtp.sessionParams.noEncrypRTP.require	0 or 1	Null	If set to 1, no encryption of RTP is required. A call placed to a phone configured with `noAuth.require` must offer the UNENCRYPTED_SRTP session parameter in its SDP. If `sec.srtp.sessionParams.noEncryptRTP.require` is set to 1, `sec.srtp.sessionParams.noEncryptRTP.offer` is logically set to 1 no matter what the value in the configuration file. If set to 0 or Null, encryption of RTP is required.
sec.srtp.requireMatchingTag	0 or 1	Null	A flag to determine whether or not to check the tag value in the crypto attribute in an SDP answer. If set to 1 or Null, the tag values must match. If set to 0, the tag value is ignored.

H.235 <H235/>

Note

At this time, this attribute is used with the Polycom VVX 1500 phone only.

The H.235 Voice Profile implementation is Polycom HDX-compatible. OpenSSL-based Diffie-Hellman key exchange and AES-128 CBC encryption algorithms is used to encrypt the RTP media

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
sec.H235. mediaEncryption. enabled	0 or 1	1	If set to 1, H.235 Voice Profile RTP media encryption will be enabled. When enabled, media encryption will be negotiated when such encryption is requested by the far end.
sec.H235. mediaEncryption.offer	0 or 1	0	If set to 1 and `sec.H235.mediaEncryption.enabled` is also set to 1, media encryption negotiations will be initiated with the far end;however, successful negotiations is not a requirement for the call to complete.
sec.H235. mediaEncryption.require	0 or 1	0	If set to 1 and `sec.H235.mediaEncryption.enabled` is also set to 1, media encryption negotiations will be initiated or completed with the far end, and if negotiations fail, the call will be dropped.

License <license/>

This attribute's settings control aspects of the feature licensing system.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
license.polling.time	00:00 – 23:59	2:00am	The time to check whether or not the license has expired.

Provisioning <prov/>

This attribute's settings control aspects of the phone's provisioning server provisioning system.

Attribute	Permitted Values	Default	Interpretation
prov.fileSystem.rfs0.minFreeSpace	5-512	5	Minimum free space in Kbytes to reserve in the file system when downloading files from the provisioning server. ***Note:*** *Polycom recommends that you do not change these parameters*. ***Note****: For the SoundPoint IP 650 phone,* `prov.fileSystem.ffs0.8meg.minFreeSpace` *is internally replaced by 2X the value.* ***Note****: For the SoundPoint IP 7000 phone,* `prov.fileSystem.rfs0.minFreeSpace` *is internally replaced by 4X the value.*
prov.fileSystem.ffs0.4meg.minFreeSpace		420	
prov.fileSystem.ffs0.2meg.minFreeSpace		48	
prov.fileSystem.ffs0.8meg.minFreeSpace		512	
prov.polling.enabled	0 or 1	0	If set to 1, automatic periodic provisioning server polling for upgrades is enabled.
prov.polling.mode	abs, rel	abs	Polling mode is *abs*olute or *rel*ative.
prov.polling.period	integer greater than 3600	86400	Polling period in seconds. Rounded up to the nearest number of days in *abs* mode. Measured relative to boot time in *rel* mode.
prov.polling.time	Format is hh:mm	03:00	Only used in *abs* mode. Polling time.
prov.quickSetup.enabled	0 or 1	Null	If set to 1, the quick setup feature is enabled. If set to 0 or Null, the quick setup feature is disabled.

RAM Disk <ramdisk/>

This attribute's settings control the phone's internal RAM disk feature.

Polycom recommends that you do not change these values.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
ramdisk.enable	0 or 1	1	If set to 1, RAM disk will be available. The RAM disk is used to cache downloaded wave files, and other resources for the user interface.
ramdisk.bytesPerBlock	0, 32, 33, ..., 1024	0	These four parameters use internal defaults when value is set to 0. ***Note**: For the SoundPoint IP 650 phone, ramdisk.bytesPerBlock is internally replaced by 2X the value.* ***Note**: For the SoundPoint IP 7000 phone, ramdisk.bytesPerBlock is internally replaced by 4X the value.*
ramdisk.blocksPerTrack	0, 1, 2, ..., 65536	0	
ramdisk.nBlocks	0, 1, 2, ..., 65536	4096	
ramdisk.nBlocks.IP_650	0, 1, 2, ..., 65536	2048	
ramdisk.minsize	50 to 16384	50	Smallest size in Kbytes of RAM disk to create before returning an error. RAM disk size is variable depending on the amount of device memory.
ramdisk.minfree	512 to 16384	3150	Minimum amount of free space that must be left after the RAM disk has been created. The RAM disk's size will be reduced as necessary in order to leave this amount of free RAM.

Request <request/>

This attribute includes:

- Delay <delay/>

Delay <delay/>

These settings control the phone's behavior when a request for restart or reconfiguration is received.

Attribute	Permitted Values	Default	Interpretation
request.delay.type	Null, "audio", or "call"	call	Defines the strategy to adopt before a request gets executed. If set to "audio", a request can be executed as soon as there is no active audio on the phone, independently of any call state. If set to "call", a request can be executed as soon as there are no calls in any state on the phone.

Feature <feature/>

These settings control the activation or deactivation of a feature at run time. In the table below, *x* is the feature number.

Attribute	Permitted Values	Interpretation
feature.x.name	"presence"	"presence" is the presence feature including management of buddies and own status
	"messaging"	"messaging" is the instant messaging feature
	"directory"	"directory" is the local directory feature
	"calllist"	"calllist" is the locally controlled call lists ***Note:*** *The "call list" feature can be disabled on all SoundPoint IP, SoundStation IP, and VVX phones except the SoundPoint IP 32x/33x.*
	"ring-download"	"ring-download" is run-time downloading of ringers
	"calllist-received"	"calllist-received" is the received-calls list feature (the "calllist" feature must be enabled for this feature to be available)
	"calllist-placed"	"calllist-placed" is the placed-calls list feature (the "calllist" feature must be enabled for this feature to be available)
	"calllist-missed"	"calllist-missed" is the missed-calls list feature (the "calllist" feature must be enabled for this feature to be available)
	"url-dialing"	"url-dialing" controls whether URL/name dialing is available from a private line (it is never available from a shared line) ***Note:*** *The "url-dialing" feature must be disabled by setting* `feature.9.enabled` *to 0 in order to prevent unknown callers from being identified on the display by an IP address*.
	"call-park"	"call-park" is the call park and park-retrieve features

Attribute	Permitted Values	Interpretation
feature.x.name (continued)	“group-call-pickup”	“group-call-pickup” is the group call pickup feature
	“directed-call-pickup”	“directed-call-pickup” is the directed call pickup feature
	“last-call-return”	“last-call-return” is the last call return feature
	“acd-login-logout”	“acd-login-logout” is the ACD login/logout feature
	acd-agent-available”	“acd-agent-available” is the ACD agent available/unavailable feature
	“nway-conference”	“nway-conference” is the conference managing feature ***Note:*** *For* `feature.16.name =“nway-conference”`*:* • *If set to 0, the n-way conferencing feature is disabled, meaning that three-way conferencing can exist, but there is no manage conference page.* • *If set to 1, the n-way conferencing feature is enabled, the maximum number of conference parties for the platform can exist, and there is a manage conference page.* ***Note****: The manage conference feature is always disabled on the SoundPoint IP 32x/33x and 430 phone. The manage conference feature is always enabled on the SoundStation IP 7000 and the Polycom VVX 1500 phone.*
	“call-recording”	“call-recording” is the call recording and playback feature
	“enhanced-feature-keys”	“enhanced-feature-keys” is the enhanced feature keys feature ***Note****: This feature must be enabled to use the configurable soft keys feature*.
	“corporate-directory”	“corporate-directory” is the corporate directory feature
	“picture-frame”	“picture-frame“ is the digital picture frame feature ***Note****:* `feature.20.name = “picture-frame”` *is only supported on the Polycom VVX 1500.*
feature.x.enabled	0 or 1 (default) except for x=9	If set to 0, the feature will be disabled. If set to 1, the feature will be enabled and usable by the local user.

Note `feature.16.name =“nway-conference”`, `feature.17.name = “call-recording”`, and `feature.19.name =“corporate-directory”` are charged for separately. To activate these features, you must go to the Polycom Resource Center (http://extranet.polycom.com/csnprod/signon.html) to retrieve the activation code. However, these feature are included on the Polycom VVX 1500.

Resource <res/>

This attribute's settings control the maximum size or an external resource retrieved at run time.

For more information, refer to "Technical Bulletin 35704: Allocating Adequate Memory for resources on SoundPoint IP and SoundStation IP Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

This attribute also includes:

- Finder <finder/>
- Quotas <quotas/>

Finder <finder/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
res.finder.sizeLimit	positive integer	300	If a resource that is being downloaded to the phone is larger than this value * 1024 bytes (= the maximum size), the resource will be automatically truncated to the maximum size defined. ***Note**: For the SoundPoint IP 550, 560, 650, and 670 phones, this value is internally replaced by 2X the value. For the SoundStation IP 6000 and 7000 phones, this value is internally replaced by 4X the value.*
res.finder.minfree	1 to 2048	600	A resource will not be downloaded to the phone if the amount of free memory is less than this value * 1024 bytes (= the minimum size). This parameter is used for 16MB SDRAM platforms and scaled up for platforms with more SDRAM. If set to 0 or Null, the default value of 600 is used. ***Note**: For the SoundPoint IP 550, 560, 650, and 670 phones, this value is internally replaced by 2X the value. For the SoundStation IP 6000 and 7000 phones, this value is internally replaced by 4X the value.*

Quotas <quotas/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Interpretation
res.quotas.x.name	1="tone", 2= "bitmap", 3="font", 5="background"	The name of the sub-application for which the particular quota will apply: "tone" relates to all downloaded tones and sound effects "bitmap" relates to all downloaded bitmaps "font" relates to all downloaded fonts "background" relates to all downloaded backgrounds
res.quotas.x.value	positive integer	When a particular resource (one of category "font", "bitmap", or "font") is downloaded to the phone, a quota equal to this value * 1024 bytes of compound data size is applied for that category. If downloading a resource would exceed the quota for that category, the resource will not be downloaded and a predefined default will be used instead. For `res.quotas.x.value`, the default is 300 KB for tones, 10 KB for bitmaps and fonts, and 600KB for backgrounds. ***Note**: For the SoundPoint IP 550, 560, 650, and 670 phones, this value is internally replaced by 2X the value. For the SoundStation IP 6000 and 7000 phones, this value is internally replaced by 4X the value.*

Microbrowser <mb/>

This attribute's settings control the home page, proxy and size limits to be used by the Microbrowser and Browser when it is selected to provide services. The Microbrowser is supported on the SoundPoint IP 430, 450, 550, 560, 601, 650, and 670, and the SoundStation IP 6000 and 7000 phones, and the Browser is supported on the Polycom VVX 1500 phones.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
mb.proxy	Null or domain name or IP address in the format <address>:<port>	Null. Default port = 8080	Address of the desired HTTP proxy to be used by the Microbrowser. If blank, normal unproxied HTTP is used by the Microbrowser.

Attribute	Permitted Values	Default	Interpretation
mb.ssawc.enabled	0 or 1	Null	If set to 0 or Null, spontaneous display of web content is disabled. If set to 1, spontaneous display of web content is enabled.
mb.ssawc.call.mode	Active, Passive	Null	Control the spontaneous display of web content. If set to passive or Null, the web content is displayed only when requested by the user. If set to active, the web content is displayed immediately.

This attribute also includes:

- Idle Display <idleDisplay/>
- Main Browser <main/>
- Browser Limits <limits/>

Idle Display <idleDisplay/>

The Microbrowser can be used to create a display that will be part of the phone's idle display. These settings control the home page and the refresh rate.

Attribute	Permitted Values	Default	Interpretation
mb.idleDisplay.home	Null or any fully formed valid HTTP URL. Length up to 255 characters.	Null	URL used for Microbrowser idle display home page. For example: http://www.example.com/xhtml/frontpage.cgi?page=home. If empty, there will be no Microbrowser idle display feature. Note that the Microbrowser idle display will displace the idle display indicator (refer to `ind.idleDisplay.enabled` in Indicators <ind/> on page A-100). ***Note**: If `ind.idleDisplay.enabled` is enabled, miscellaneous XML errors can occur on SoundPoint IP 430, 550, 560, 650, and 670 and SoundStation IP 6000 and 7000 phones.*
mb.idleDisplay.refresh	0 or an integer > 5	0	The period in seconds between refreshes of the idle display Microbrowser's content. If set to 0, the idle display Microbrowser is not refreshed. The minimum refresh period is 5 seconds (values from 1 to 4 are ignored, and 5 is used). ***Note:** If an HTTP Refresh header is detected, it will be respected, even if this parameter is set to 0. The refresh parameter will be respected only in the event that a refresh fails. Once a refresh is successful, the value in the HTTP refresh header, if available, will be used.*

Main Browser <main/>

This setting controls the home page used by the Microbrowser when that function is selected.

Attribute	Permitted Values	Default	Interpretation
mb.main.home	Any fully formed valid HTTP URL. Length up to 255 characters.	Null	URL used for Microbrowser home page. If blank, the browser will notify the user that a blank home-page was used. For example: http://www.example.com/xhtml/frontpage.cgi?page=home.
mb.main.statusbar	0 or 1	Null	Flag to determine whether or not to turn off display of status messages. If set to 1, the display of the status bar is enabled. If set to 0, or Null, the display of the status bar is disabled.
mb.main.idleTimeout	0 - 600, seconds	Null	Timeout for the interactive browser. If the interactive browser remains idle for a defined period of time, the phone should return to the idle browser. If set to 0, there is no timeout. If set to Null, the value from `up.idleTimeout` is used. Refer to User Preferences <up/> on page A-31. If `mb.main.idleTimeout` and `up.idleTimeout` are Null, the timeout is 40 seconds. If set to value greater than 0 and less than 600, the timeout is for that number of seconds.
mb.main.autoBackKey	0 or 1	1	If set to 1, the phone will automatically supply a **Back** soft key in all main browser screens, which if pressed will take the user back through the browser history. This is the null default behavior (for backward compatibility). If set to 0, the phone will not provide a Back soft key. All soft keys will be created and controlled by the application.

Browser Limits <limits/>

These settings limit the size of object which the Microbrowser will display by limiting the amount of memory available for the Microbrowser.

Attribute	Permitted Values	Default	Interpretation
mb.limits.nodes	positive integer	256	Limits the number of tags that the XML parser will handle. This limits the amount of memory used by complicated pages. A maximum total of 500 (256 each) is recommended. This value is used as referent values for 16MB of SDRAM. ***Note:*** *Increasing this value may have a detrimental effect on performance of the phone.*
mb.limits.cache	positive integer	200	Limits the total size of objects downloaded for each page (both XHTML and images). Once this limit is reached, no more images are downloaded until the next page is requested. Units = kBytes. This value is used as referent values for 16MB of SDRAM. ***Note:*** *Increasing this value may have a detrimental effect on performance of the phone.*

Applications <apps/>

This attribute's settings control the telephone notification events, state polling events, and the push server controls. For more information, refer to the *Web Application Developer's Guide,* which can be found at http://www.polycom.com/voicedocumentation/.

This attribute also includes:

- Telephone Notification <telNotification/>
- State Polling <statePolling/>
- Push <push/>

Telephone Notification <telNotification/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
apps.telNotification.URL	URL	Null	The URL to which the phone sends notifications of specified events. The protocol used can be either HTTP or HTTPS.
apps.telNotification. incomingEvent	0 or 1	0	If set to 0, incoming call notification is disabled. If set to 1, incoming call notification is enabled.
apps.telNotification. outgoingEvent	0 or 1	0	If set to 0, outgoing call notification is disabled. If set to 1, outgoing call notification is enabled.
apps.telNotification. offhookEvent	0 or 1	0	If set to 0, offhook notification is disabled. If set to 1, offhook notification is enabled.
apps.telNotification. onhookEvent	0 or 1	0	If set to 0, onhook notification is disabled. If set to 1, onhook notification is enabled.

State Polling <statePolling/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
apps.statePolling.URL	URL	Null	The URL to which the phone sends call processing state/device/network information. The protocol used can be either HTTP or HTTPS. ***Note**: To enable state polling, the attributes `apps.statePolling.URL`, `apps.statePolling.username`, and `apps.statePolling.password` must be set to non-Null values*.
apps.statePolling. username	string	Null	The user name to access the state polling URL..
apps.statePolling. password	string	Null	The password to access the state polling URL.

Push <push/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
apps.push.messageType	0 to 3	0	Select the allowable push priority messages on phone. The values are: • 0: (None) Discard push messages • 1: (Critical) Allows only critical push messages • 2: (Normal) Allows only normal push messages • 3: (Both) Allows both critical and normal push messages
apps.push. serverRootURL	URL	Null	The relative URL (received from HTTP URL Push message) is appended to the application server root URL and the resultant URL is sent to the Microbrowser. For example, if the application server root URL is http://172.24.128.85:8080/sampleapps and the relative URL is /examples/sample.html, the URL that is sent to the Microbrowser is http://172.24.128.85:8080/sampleapps/examples/sample.html. The protocol used can be either HTTP or HTTPS.
apps.push.username	string	Null	The user name to access the push server URL. ***Note**: To enable the push functionality, the attributes `apps.push.username` and `apps.push.password` must be set to non-Null values.*
apps.push.password	string	Null	The password to access the push server URL.

Peer Networking <pnet/>

Peer networking manages communications between Polycom devices. For the SoundStation IP 7000 conference phone, it manages daisy-chaining and video integation with the Polycom HDX video systems.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
pnet.role	“auto” “standAlone” “masterOnly” “masterPreferred” “slaveOnly” “slavePreferred”	Null	The role of the SoundStation IP 7000 when communicating with other Polycom devices. If the attribute is not defined or is null, the default value is “auto” meaning that the configuration of the peer role is automatic. The other value definitions are: • “standAlone” - IP 7000 is always only standalone. • “masterOnly” - IP 7000 is always the master. • “masterPreferred” - The configuration is automatic, but if the call capability of the daisy-chained IP 7000 is the same as this one, this one is the master. • “slaveOnly” - IP 7000 is always the slave. • “slavePreferred” - The configuration is automatic, but if the call capability of the daisy-chained IP 7000 is the same as this one, this one is the slave.
pnet.hdx.ext	string	Null	The HDX Extension Number to be displayed on the IP 7000 when it is connected to an HDX system.
pnet.remoteCall. callProgAtten	-60 to 0	Null	The attenuation applied to tones played by the IP 7000 for POTS calls when it is connected to an HDX system when the HDX is the active speaker. If set to Null, the default is -15.
pnet.remoteCall. localDialTone	0 or 1	Null	A flag to determine whether or not a dialtone is played when the IP 7000 makes an outgoing POTS call when it is connected to an HDX. If set to 1, a dial tone is played. If set to 0 or Null, a dial tone is not played.

DNS Cache <dns/>

In the tables below, a maximum of 12 entries of NAPTR, SRV, and A record can be added.

This attribute includes:

- NAPTR <NAPTR/> attribute
- SRV <SRV/>
- A <A/>

NAPTR <NAPTR/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dns.cache.NAPTR.x. name	domain name string	Null	The domain name to which this resource record refers.
dns.cache.NAPTR.x.ttl	300 to 65535	300	Specifies the time interval (in seconds) that the resource record may be cached before the source of the information should again be consulted.
dns.cache.NAPTR.x. order	0 to 65535	0	A 16-bit unsigned integer specifying the order in which the NAPTR records must be processed to ensure the correct ordering of rules.
dns.cache.NAPTR.x. preference	0 to 65535	0	A 16-bit unsigned integer that specifies the order in which NAPTR records with equal "order" values should be processed, low numbers being processed before high numbers.
dns.cache.NAPTR.x. flags	string		Flags to control aspects of the rewriting and interpretation of the fields in the record. Flags are single characters from the set [A-Z, 0-9]. The alphabetic characters are case insensitive. At this time only four flag, "S", "A", "U", and "P" are defined. For more information, go to http://tools.ietf.org/html/rfc2915 .
dns.cache.NAPTR.x. service	string		Specifies the service(s) available down this rewrite path. For more information, go to http://tools.ietf.org/html/rfc2915 .

Attribute	Permitted Values	Default	Interpretation
dns.cache.NAPTR.x. regexp	string	Null	A string containing a substitution expression that is applied to the original string held by the client in order to construct the next domain name to lookup. The grammar of the substitution expression is given in RFC 2915. ***Note**: This parameter is currently unused.*
dns.cache.NAPTR.x. replacement	domain name string with SRV prefix	Null	The next name to query for NAPTR, SRV, or address records depending on the value of the flags field. It must be a fully qualified domain-name.

SRV <SRV/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dns.cache.SRV.x.name	domain name string	Null	The domain name string with SRV prefix.
dns.cache.SRV.x.ttl	0 to 65535, seconds	300	Specifies the time interval that the resource record may be cached before the source of the information should again be consulted.
dns.cache.SRV.x.priority	0 to 65535	0	The priority of this target host. For more information, go to http://tools.ietf.org/html/rfc2782 .
dns.cache.SRV.x.weight	0 to 65535	0	A server selection mechanism. For more information, go to http://tools.ietf.org/html/rfc2782 .
dns.cache.SRV.x.port	0 to 65535	0	The port on this target host of this service. For more information, go to http://tools.ietf.org/html/rfc2782 .
dns.cache.SRV.x.target	domain name string	Null	The domain name of the target host. For more information, go to http://tools.ietf.org/html/rfc2782 .

A <A/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dns.cache.A.x.name	valid hostname	Null	Hostname
dns.cache.A.x.ttl	0 to 65535	300	Specifies the time interval that the resource record may be cached before the source of the information should again be consulted.
dns.cache.A.x.address	dotted-decimal IP version 4 address	Null	IP address that hostname `dns.cache.A.x.name` maps to.

Soft Keys <softkey/>

Note `feature.20.name = "enhanced-feature-keys"` must be enabled to use the Configurable Soft Key feature. Refer to Feature <feature/> on page A-116.

This configuration attribute is defined as follows (where x =1 to maximum number of defined soft keys):

Attribute	Permitted Values	Default	Interpretation
softkey.x.label	string	Null	This is the text displayed with the soft key. If set to Null, the label to display is determined as follows: • If the soft key is mapped to a enhanced feature key macro, the label of the enhanced feature key macro will be used. • If the soft key is mapped to a speed dial, the label of the corresponding directory entry will be used. If this label does not exist as well and the directory entry is a enhanced feature key macro, then the label of the enhanced feature key macro will be used. • If the soft key is mapped to chained actions, only the first one is considered for label, using the rules above. • If no labels are found after the above steps, the soft key label will be blank.
softkey.x.action	string	Null	The same syntax as the enhanced feature key action. For more information, refer to Macro Definition on page 4-42.

Attribute	Permitted Values	Default	Interpretation
softkey.x.enable	0 (default) 1	Null	If set to 0 or Null, the soft key is disabled. If set to 1, the soft key is enabled.
softkey.x.precede	0 (default) 1	Null	If set to 0 or Null, the soft key replaces any empty space from the leftmost position. If set to 1, the soft key is displayed before the first standard soft key.
softkey.x.use.idle	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the idle state. If set to 1, the soft key is displayed in the idle state.
softkey.x.use.active	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the active call state. If set to 1, the soft key is displayed in the active call state.
softkey.x.use.alerting	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the alerting state. If set to 1, the soft key is displayed in the alerting state.
softkey.x.use.dialtone	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the dialtone state. If set to 1, the soft key is displayed in the dialtone state.
softkey.x.use.proceeding	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the proceeding state. If set to 1, the soft key is displayed in the proceeding state.
softkey.x.use.setup	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the setup state. If set to 1, the soft key is displayed in the setup state.
softkey.x.use.hold	0 (default) 1	Null	If set to 0 or Null, the soft key is not displayed in the hold state. If set to 1, the soft key is displayed in the hold state.
softkey.feature.newcall	0 1 (default)	Null	If set to 0, the **New Call** soft key is not displayed when there is another way to place a call. If set to 1 or Null, the **New Call** soft key is displayed.

Attribute	Permitted Values	Default	Interpretation
softkey.feature.endcall	0 1 (default)	Null	If set to 0, the **End Call** soft key is not displayed. If set to 1 or Null, the **EndCall** soft key is displayed.
softkey.feature.split	0 1 (default)	Null	If set to 0, the **Split** soft key is not displayed. If set to 1 or Null, the **Split** soft key is displayed.
softkey.feature.join	0 1 (default)	Null	If set to 0, the **Join** soft key is not displayed. If set to 1 or Null, the **Join** soft key is displayed.
softkey.feature.forward	0 1 (default)	Null	If set to 0, the **Forward** soft key is not displayed. If set to 1 or Null, the **Forward** soft key is displayed.
softkey.feature.directories	0 1 Null (default)	Null	If set to Null, the **Dir** soft key is displayed on the SoundPoint IP 32x/33x phone, but not on any other phone. If set to 0, the **Dir** soft key is not displayed on any phone. If set to 1, the **Dir** soft key is displayed on all phones as follows: • In the idle state, it is displayed after the **New Call** and **Callers** soft keys. • In the dialtone state, it is displayed after the **End Call** and **Callers** soft keys. • During a conference or transfer, it is displayed after the **Callers** and **Cancel** soft keys.

Attribute	Permitted Values	Default	Interpretation
softkey.feature.callers	0 1 Null (default)	Null	If set to Null, the **Callers** soft key is displayed on the SoundPoint IP 32x/33x phone, but not on any other phone. If set to 0, the **Callers** soft key is not displayed on any phone. If set to 1, the **Callers** soft key is displayed on all phones as follows: • In the idle state, it is displayed after the **New Call** soft key and before the **Dir** soft key. • In the dialtone state, it is displayed after the **End Call soft key** and before the **Dir** soft key. • During a conference or transfer, it is displayed before the **Cancel** soft key.
softkey.feature.mystatus	0 or 1	1	If set to 0, the **MyStatus** soft key is not displayed. If set to 1 or Null, the **MyStatus** soft key is displayed. ***Note**:* `pres.idleSoftKeys` *must be set to 1 for this soft key to be displayed*.
softkey.feature.buddies	0 or 1	1	If set to 0, the **Buddies** soft key is not displayed. If set to 1 or Null, the **Buddies** soft key is displayed. ***Note**:* `pres.idleSoftKeys` *must be set to 1 for this soft key to be displayed*.
softkey.feature. basicCallManagement.redundant	0 or 1	1	If set to 0 and the phone has hard keys mapped for **Hold**, **Transfer**, and **Conference** functions (all must be mapped), all of these soft keys are not displayed. If set to 1 or Null, all of these soft keys are displayed.

LCD Power Saving <powerSaving/>

Note This attribute is supported on the Polycom VVX 1500 only.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
powerSaving.enabled	0 or 1	1	If set to 1 or Null, the LCD power saving feature is enabled. If set to 0, the LCD power saving feature is disabled.
powerSaving.officeHours. startHour.xxx	0 to 23	8	The starting hour for the day's office hours, where xxx is one of "monday", "tuesday", "wednesday", "thursday", "friday", "saturday", and "sunday". If set to Null, the default value is 8.
powerSaving.officeHours. duration.xxx	0 to 12	10 or 0	The duration of the day's office hours, where xxx is one of "monday", "tuesday", "wednesday", "thursday", "friday", "saturday", and "sunday" . If set to Null, the default value for the week days is 10 (hours) and the default value for Saturday and Sunday is 0 (hours).
powerSaving.idleTimeout. officeHours	1 to 600	10	The office hours mode idle timeout (in minutes). If set to Null, the default value is 10.
powerSaving.idleTimeout. offHours	1 to 10	1	The off hours mode idle timeout (in minutes). If set to Null, the default value is 1.
powerSaving.idleTimeout. userInputExtension	1 to 20	10	The minimum idle timeout after user input events (in minutes). If set to Null, the default value is 10.

Attribute	Permitted Values	Default	Interpretation
powerSaving. userDetectionSensitivity. officeHours	0 to 10	7	The sensitivity of the algorithm used to detect the presence of the phone's user during office hours. If set to 0, this feature is disabled. If set to Null, the default value is 7. This value was chosen for good performance in a typically office environment and is biased for easy detection during office hours.
powerSaving. userDetectionSensitivity. offHours	0 to 10	2	The sensitivity of the algorithm used to detect the presence of the phone's user during off hours. If set to 0, this feature is disabled. If set to Null, the default value is 2. This value was chosen for good performance in a typically office environment and is biased for difficult detection during off hours.

Per-Phone Configuration

This section covers the parameters in the per-phone example configuration file **phone1.cfg**. This file would normally be used as a template for the per-phone configuration files. For more information, refer to Deploying Phones From the Provisioning Server on page 3-17.

Polycom recommends that you create another file with your organization's modifications. If you must change any Polycom templates, back them up first.

For more information, refer to the "Configuration File Management on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones" white paper at http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf .

The parameters include:

- Registration <reg/>
- Calls <call/>
- Diversion <divert/>
- Dial Plan <dialplan/>
- Messaging <msg/>
- Network Address Translation <nat/>
- Attendant <attendant/>

- Roaming Buddies <roaming_buddies/>
- Roaming Privacy <roaming_privacy/>
- User Preferences <up/>
- Automatic Call Distribution <acd/>

Registration <reg/>

SoundPoint IP 32x/33x and 430 support a maximum of two unique registrations, SoundPoint IP 450 supports three, the SoundPoint IP 550 and 560 supports four, and SoundPoint IP 650 and 670 and the Polycom VVX 1500 support six. Up to three SoundPoint IP Expansion Modules can be added to a single host SoundPoint IP 650 and 670 phone increasing the total number of buttons to 34 registrations on the IP 650 and 670. Each registration can optionally be associated with a private array of servers for completely segregated signaling. The SoundStation IP 6000, and 7000 supports a single registration.

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6; IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
reg.x.csta	0 or 1	Null	If set to 1, uaCSTA is enabled. If `reg.x.csta` is not Null, this attribute overrides the global CSTA flag in the **sip.cfg** configuration file.
reg.x.displayName	UTF-8 encoded string	Null	Display name used in SIP signaling as the default caller ID. Display name used in SIP signaling and/or H.323 alias as the default caller ID.
reg.x.address	string in the format userPart from userPart@domain	Null	The user part or the user and the host part of the phone's SIP URI or the H.323 ID/extension. For example (SIP): reg.x.address="1002" from 1002@polycom.com or reg.x.address="1002@polycom.com". For example (H.323): reg.x.address="23456"
reg.x.label	UTF-8 encoded string	Null	Text label to appear on the display adjacent to the associated line key. If omitted, the label will be derived from the user part of reg.x.address.
reg.x.lcs	0 or 1	0	If set to 1, the Microsoft Live Communications Server is supported for registration x.

Attribute	Permitted Values	Default	Interpretation
reg.x.type	private OR shared	private	If set to private, use standard call signaling. If set to shared, augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.
reg.x.thirdPartyName	string in the same format as reg.x.address	Null	This field must match the `reg.x.address` value of the other registration which makes up the bridged line appearance (BLA). It must be Null in all other cases.
reg.x.auth.userId	string	Null	User ID to be used for authentication challenges for this registration. If non-Null, will override the “Reg User x” parameter entered into the Authentication submenu off of the Settings menu on the phone.
reg.x.auth.password	string	Null	Password to be used for authentication challenges for this registration. If non-Null, will override the “Reg Password x” parameter entered into the Authentication submenu off of the Settings menu on the phone.
reg.x.acd-login-logout	0 or 1	0	If both parameters are set to 1 for a registration, the ACD feature will be enabled for that registration.
reg.x.acd-agent-available	0 or 1	0	
reg.x.ringType	1 to 22	2	The ringer to be used for calls received by this registration. Default is the first non-silent ringer.
reg.x.lineKeys	1 to *max*	1	*max* = the number of line keys on the phone. *max* = 1 on IP 6000, 7000, *max* = 2 on IP 32x/33x, 430, *max* = 3 on IP 450, *max* = 4 on IP 550, 560, *max* = 6 on VVX 1500, *max* = 34 on IP 650, 670 (without any Expansion Modules attached, only 6 line keys are available) The number of line keys on the phone to be associated with registration ‘x’.

Attribute	Permitted Values	Default	Interpretation
reg.x.callsPerLineKey	1 to 34 OR 1 to 24 OR 1 to 8 OR 1 to 4	34 OR 24 OR 8 OR 4	For the SoundPoint IP 650 and 670, the permitted range is 1 to 34 and the default is 34. For the SoundPoint IP 550 and 560 and the VVX 1500, the permitted range is 1 to 24 and the default is 24. For the SoundPoint IP 430 the permitted range is 1 to 4 and the default is 4. For all other phones the permitted range is 1 to 8 and the default is 8. This is the number of calls or conferences which may be active or on hold per line key associated with this registration. Note that this overrides `call.callsPerLineKey` for this registration. Refer to Call Handling Configuration <call/> on page A-80. If `reg.1.callsPerLineKey` is set to 1, call waiting can be disabled. ***Note:*** *A call active on another phone on a shared line counts as a call for every phone sharing that registration.*
reg.x.bargeInEnabled	0 or 1	Null	Allow remote user of SCA to interrupt call. (Works in a similar way to resume.) If set to 1, barge-in is enabled for line x. If set to 0 or Null, barge-in is disabled for line x.
reg.x.outboundProxy.address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server to which the phone shall send all requests.
reg.x.outboundProxy.port	1 to 65535	5060	

Attribute	Permitted Values	Default	Interpretation
reg.x.outboundProxy.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	If set to Null or DNSnaptr: If reg.x.outboundProxy.address is a hostname and reg.x.outboundProxy.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If reg.x.outboundProxy.address is an IP address, or a port is given, then UDP is used. If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails. If set to UDPOnly: Only UDP will be used. If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. If set to TCPOnly: Only TCP will be used.
reg.x.proxyRequire	string	Null	The string that needs to appear in the “Proxy-Require” header. If Null, no "Proxy-Require" will be sent.
reg.x.serverFeatureControl.cf	0 or 1	0	If set to 1, server-based call forwarding is enabled. The call server has control of call forwarding. If set to 0, server-based call forwarding is not enabled. This is the old behavior. If `reg.x.serverFeatureControl.cf` is not Null, this attribute overrides the global server-based call forwarding flag in the **sip.cfg** configuration file.
reg.x.serverFeatureControl.dnd	0 or 1	0	If set to 1, server-based DND is enabled. The call server has control of DND. If set to 0, server-based DND is not enabled. This is the old behavior. If `reg.x.serverFeatureControl.dnd` is not Null, this attribute overrides the global server-based call forwarding flag in the **sip.cfg** configuration file.

Attribute	Permitted Values	Default	Interpretation
reg.x.auth.optimizedInFailover	0 or 1	0	If set to 1, when failover occurs, the first new SIP request is sent to the server that sent the proxy authentication request. If set to 0, when failover occurs, the first new SIP request is sent to the server with the highest priority in the server list. If this parameter is Null, `voIpProt.SIP.authOptimizedInFailover` is checked. If both parameters are set, this parameter takes precedence.
reg.x.strictLineSeize	0 or 1	Null	If set to 1, forces phone to wait for 200 OK on registration x when receiving a TRYING notify. If set to 0 or Null, this is old behavior. If this parameter is Null, `voIpProt.SIP.strictLineSeize` is checked. If both parameters are set, this parameter takes precedence.
reg.x.musicOnHold.uri	string	Null	A URI that provides the media stream to play for the remote party on hold. When present, and if `reg.x.musicOnHold` is not Null, this attribute overrides the global Music on Hold defined in the **sip.cfg** configuration file.
reg.x.tcpFastFailover	0 or 1	Null	If set to 1, failover occurs based on the values of `reg.x.server.y.retryMaxCount` `voIpProt.server.x.retryTimeOut`. If set to 0 or Null, this is old behavior. If this parameter is Null, `voIpProt.SIP.tcpFastFailover` is checked. If both parameters are set, this parameter takes precedence.
reg.x.protocol.SIP	0 or 1	1	If set to 1, SIP signaling is enabled for this line registration. If set to 0, SIP signaling is not enabled for this line registration.
reg.x.protocol.H323	0 or 1	0	If set to 1, H.323 signaling is enabled for this line registration. If set to 0, H.323 signaling is not enabled for this line registration.

Attribute	Permitted Values	Default	Interpretation
reg.x.server.y.address	dotted-decimal IP address or host name	Null	Optional IP address or host name, port, transport, registration period, fail-over parameters and line seize subscription period of a SIP server that accepts registrations. Multiple servers can be listed starting with y=1, 2, ... for fault tolerance. If specified, these servers may override the servers specified in **sip.cfg** in Server <server/> on page A-8. ***Note:*** *If the reg.x.server.y.address parameter is non-Null, all of the reg.x.server.y.xxx parameters will override the parameters specified in **sip.cfg** in Server <server/> on page A-8.* ***Note:*** *If the reg.x.server.y.address parameter is non-Null, it takes precedence even if the DHCP server is available.*
reg.x.server.y.port	0, Null, 1 to 65535	Null	
reg.x.server.y.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	
reg.x.server.y.expires	positive integer	Null	
reg.x.server.y.register	0 or 1	Null	
reg.x.server.y.expires.overlap	positive integer, minimum 5, maximum 65535	60	
reg.x.server.y.retryTimeOut	Null or non-negative integer	Null	
reg.x.server.y.retryMaxCount	Null or non-negative integer	Null	
reg.x.server.y.expires.lineSeize	positive integer	Null	
reg.x.server.y.lcs	0 or 1	0	This attribute overrides the `reg.x.lcs`. If set to 1, the Microsoft Live Communications Server is supported for registration x.
reg.x.server.H323.y.address	dotted-decimal IP address or host name	Null	Address of the H.323 gatekeeper.
reg.x.server.H323.y.port	0, Null, 1 to 65535	Null	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used.
reg.x.server.H323.y.expires	postive integer	Null	Desired registration period.

Calls <call/>

This attribute affects the call-oriented per-phone configuration.

This attribute includes:

- Do Not Disturb <donotdisturb/>
- Automatic Off-Hook Call Placement <autoOffHook/>

- Missed Call Configuration <serverMissedCall/>
- Missed Call Tracking <missedCallTracking/>
- Call Waiting <callWaiting/>

Do Not Disturb <donotdisturb/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.donotdisturb.perReg	0 or 1	0	If set to 1, the DND feature will allow selection of DND on a per-registration basis. ***NOTE:*** *If* `voIpProt.SIP.serverFeatureControl.dnd` *is set to 1 (enabled), this parameter is ignored. For more information, refer to* *SIP <SIP/> on page A-11*.

Automatic Off-Hook Call Placement <autoOffHook/>

An optional per-registration feature is supported which allows automatic call placement when the phone goes off-hook.

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6;
IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
call.autoOffHook.x.enabled	0 or 1	0	If set to 1, a call will be automatically placed to the contact specified upon going off-hook on this registration.
call.autoOffHook.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	
call.autoOffHook.x.protocol	"SIP" or "H323"	Null	On a dual-protocol line only, specifies the routing protocol to use for the auto off-hook dialing. The strings are case sensitive. If set to Null, the value of `call.autoRouting.preferredProtocol` is used. ***Note****: If a line is single-protocol configured, the configured protocol will be used in the auto off-hook dialing and any value in its* `call.autoOffHook.x.protocol` *field will be ignored.*

Missed Call Configuration <serverMissedCall/>

The phone supports a per-registration configuration of which events will cause the locally displayed "missed calls" counter to be incremented.

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6; IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
call.serverMissedCall.x.enabled	0 or 1	0	If set to 0, all missed-call events will increment the counter. If set to 1, only missed-call events sent by the server will increment the counter. ***NOTE:*** *This feature is supported with the Sylantro call server only.*

Missed Call Tracking <missedCallTracking/>

You can enable/disable missed call tracking on a per-line basis.

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6; IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
call.missedCallTracking.x.enabled	0 or 1	1	If set to 1 or Null, missed call tracking is enabled. If `call.missedCallTracking.x.enabled` is set to 0, then missedCall counter is not updated regardless of what `call.serverMissedCalls.x.enabled` is set to (and regardless of how the server is configured). There is no Missed Call List provided under Menu > Features of the phone. If `call.missedCallTracking.x.enabled` is set to 1 and call.serverMissedCalls.x.enabled is set to 0, then the number of missedCall counter is incremented regardless of how the server is configured. If `call.missedCallTracking.x.enabled` is set to 1 and `call.serverMissedCalls.x.enabled` is set to 1, then the handling of missedCalls depends on how the server is configured.

Call Waiting <callWaiting/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
call.callWaiting.ring	beep, ring, silent	beep	Specifies the ring tone heard on an incoming call when another call is active. If set to Null, the default value is beep.

Diversion <divert/>

The phone has a flexible call forward/diversion feature for each registration. In all cases, a call will only be diverted if a non-Null contact has been configured.

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6; IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
divert.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com	Null	The forward-to contact used for all automatic call diversion features unless overridden by a specific contact of a per-call diversion feature (refer to below).
divert.x.autoOnSpecificCaller	0 or 1	1	If set to 1, calls may be diverted using the Auto Divert feature of the directory. This is a global flag. ***Note:*** *If server-based call forwarding is enabled, this parameter is disabled*.
divert.x.sharedDisabled	0 or 1	1	If set to 1, all diversion features on that line will be disabled if the line is configured as shared.

This attribute also includes:

- Forward All <fwd/>
- Busy <busy/>
- No Answer <noanswer/>
- Do Not Disturb <dnd/>

Forward All <fwd/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
divert.fwd.x.enabled	0 or 1	1	If set to 1, the user will be able to enable universal call forwarding through the soft key menu. ***Note:*** *If server-based call forwarding is enabled, this parameter is enabled*.

Busy <busy/>

Calls can be automatically diverted when the phone is busy.

Attribute	Permitted Values	Default	Interpretation
divert.busy.x.enabled	0 or 1	1	If set to 1, calls will be forwarded on busy to the contact specified below. ***Note:*** *If server-based call forwarding is enabled, this parameter is disabled*.
divert.busy.x.timeout	positive integer	60	Time in seconds to allow altering before initiating the diversion.
divert.busy.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com	Null	Forward-to contact for calls forwarded due to busy status, if Null, `divert.x.contact` will be used.

No Answer <noanswer/>

The phone can automatically divert calls after a period of ringing.

Attribute	Permitted Values	Default	Interpretation
divert.noanswer.x.enabled	0 or 1	1	If set to 1, calls will be forwarded on no answer to the contact specified. ***Note:*** *If server-based call forwarding is enabled, this parameter is disabled*.
divert.noanswer.x.timeout	positive integer	55	Time in seconds to allow altering before initiating the diversion.
divert.noanswer.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Forward-to contact used for calls forwarded due to no answer, if Null, `divert.x.contact` will be used.

Do Not Disturb <dnd/>

The phone can automatically divert calls when Do Not Disturb (DND) is enabled.

Attribute	Permitted Values	Default	Interpretation
divert.dnd.x.enabled	0 or 1	0	If set to 1, calls will be forwarded on DND to the contact specified below. ***Note:*** *If server-based DND or server-base call forwarding is enabled, this parameter is disabled*.
divert.dnd.x.contact	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Forward-to contact used for calls forwarded due to DND status, if Null `divert.x.contact` will be used.

Dial Plan <dialplan/>

Per-registration dial plan configuration is supported.

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6; IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

Attribute	Permitted Values	Default	Interpretation
dialplan.x.applyToCallListDial	0 or 1	0	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.
dialplan.x.applyToDirectoryDial	0 or 1	0	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.
dialplan.x.applyToUserDial	0 or 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.
dialplan.x.applyToUserSend	0 or 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.
dialplan.x.impossibleMatchHandling	0, 1 or 2	0	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.

Attribute	Permitted Values	Default	Interpretation
dialplan.x.removeEndOfDial	0 or 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.
dialplan.x.applyToTelUriDial	0 or 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan <dialplan/> on page A-23.

This attribute also includes:

- Digit Map <digitmap/>
- Routing <routing/>

Digit Map <digitmap/>

For more information on digit map syntax, refer to Digit Map <digitmap/> on page A-24.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.digitmap	A string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435; string is limited to 768 bytes and 30 segments; a comma is also allowed; a comma is also allowed; when reached in the digit map, a comma will turn dial tone back on;'+' is allowed as a valid digit; extension letter 'R' is used as defined above.	Null	When present, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For more information, refer to Digit Map <digitmap/> on page A-24.
dialplan.x.digitmap.timeOut	string of positive integers separated by '|'	Null	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For more information, refer to Digit Map <digitmap/> on page A-24.

Routing <routing/>

This attribute allows specific routing paths for outgoing SIP calls to be configured independent of other 'default' configuration.

This attribute includes:

- Server <server/>
- Emergency <emergency/>

Server <server/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.routing.server.y. address	dotted-decimal IP address or host name	Null	IP address or host name and port of a SIP server that will be used for routing calls. Multiple servers can be listed starting with y=1, 2, ... for fault tolerance.
dialplan.x.routing.server.y.port	1 to 65535	5060	

Emergency <emergency/>

In the following attributes, *y* is the index of the emergency entry description and *z* is the index of the server associated with the emergency entry *y*. For each emergency entry (index *y*), one or more server entry (indexes (*y*,*z*)) can be configured. *y* and *z* must both follow single step increasing numbering starting at 1.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.routing.emergency. y.value	Comma separated list of entries or single entry representing a or a combination of SIP URL.	Null Example: "15,17,18", "911", "sos".	This represents the URLs that should be watched for emergency routing. When one of these defined URL is detected as being dialed by the user, the call will be automatically directed to the defined emergency server.
dialplan.x.routing.emergency. y.server.z	positive integer	Null	Index representing the server defined in Server <server/> on page A-148 that will be used for emergency routing.

Messaging <msg/>

Message-waiting indication is supported on a per-registration basis.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
msg.bypassInstantMessage	0 or 1	0	If set to 1, the display offering a choice of “Message Center” and “Instant Messages” will be bypassed when pressing the Messages key. The phone will act as if “Message Center” was chosen. Refer to Voice Mail Integration on page 4-56. Instant Messages will still be accessible from the Main Menu.

This attribute also includes:

- Message Waiting Indicator <mwi/>

Message Waiting Indicator <mwi/>

In the following table, x is the registration number. IP 32x/33x, 430: *x*=1-2; IP 450: *x*=1-3; IP 550, 560: *x*=1-4; VVX 1500: *x*=1-6; IP 650, 670: *x*=1-34; IP 6000: *x*=1; IP 7000: *x*=1.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
msg.mwi.x.subscribe	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	If non-Null, the phone will send a SUBSCRIBE request to this contact after boot-up.
msg.mwi.x. callBackMode	contact or registration or disabled	"registration"	Configures message retrieval and notification for the line. If set to "contact", a call will be placed to the contact specified in the callback attribute when the user invokes message retrieval. If set to "registration", a call will be placed using this registration to the contact registered (the phone will call itself). If set to "disabled", message retrieval and message notification are disabled.
msg.mwi.x.callBack	ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)	Null	Contact to call when retrieving messages for this registration.

Network Address Translation <nat/>

These parameters define port and IP address changes used in NAT traversal. The port changes will change the port used by the phone, while the IP entry simply changes the IP advertised in the SIP signaling. This allows the use of simple NAT devices that can redirect traffic, but do not allow for port mapping. For example, port 5432 on the NAT device can be sent to port 5432 on an internal device, but not port 1234.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
nat.ip	dotted-decimal IP address	Null	IP address to advertise within SIP signaling - should match the external IP address used by the NAT device.
nat.signalPort	1024 to 65535	Null	If non-Null, this port will be used by the phone for SIP signaling, overriding the value set for `voIpProt.local.Port` in **sip.cfg**.
nat.mediaPortStart	1024 to 65535	Null	If non-Null, this attribute will be used to set the initially allocated RTP port, overriding the value set for `tcpIpApp.port.rtp.mediaPortRangeStart` in **sip.cfg**. Refer to RTP <rtp/> on page A-78.
nat.keepalive.interval	0 to 3600	Null	If non-Null (or 0), the keepalive interval in seconds. This parameter is used to set the interval at which phones will send a keep-alive packet to the gateway/NAT device to keep the communication port open so that NAT can continue to function as setup initially. The Microsoft Live Communications Server 2005 keepalive feature will override this interval. If you want to deploy phones behind a NAT and connect them to Live Communications Server, the keepalive interval received from the Live Communications Server must be short enough to keep the NAT port open. Once the TCP connection is closed, the phones stop sending keep-alive packets.

Attendant <attendant/>

Note These attributes are available on SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670 phones only.

The Busy Lamp Field (BLF) / attendant console feature enhances support for a phone-based attendant console.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
attendant.uri	string	Null	For attendant console / busy lamp field (BLF) feature. This specifies the list SIP URI on the server. If this is just a user part, the URI is constructed with the server host name/IP. ***Note**: If `attendant.uri` is set, then the individually addressed users configured by `attendant.resourceList` and `attendant.behaviors` attributes are ignored.*
attendant.reg	positive integer	1	For attendant console / BLF feature. This is the index of the registration which will be used to send a SUBSCRIBE to the list SIP URI specified in attendant.uri. For example, `attendant.reg = 2` means the second registration will be used.
attendant.ringType	1 to 22	1	The ring tone to play when a BLF dialog is in the offering state.

This attribute also includes:

- Resource List <resourceList/>
- Behaviors <behaviors/>

Resource List <resourceList/>

In the following table, x is the monitored user number. For IP 450: x=1-2; IP 550, IP 560: X=1-3; IP 650, IP 670: x=1-47.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
attendant.resourceList.x.address	string that constitutes a valid SIP URI (sip:6416@polycom.com) or contains the user part of a SIP URI (6416)	Null	The user referenced by `attendant.reg=""` will subscribe to this URI for dialog. If a user part is present, the phone will subscribe to a sip URI constructed from user part and the domain of the user referenced by `attendant.reg`.
attendant.resourceList.x.label	UTF-8 encoded string	Null	Text label to appear on the display adjacent to the associated line key. If set to Null, the label will be derived from the user part of `attendant.resourceList.x.address` .

Attribute	Permitted Values	Default	Interpretation
attendant.resourceList.x.type	"normal" or "automata"	“normal”	Type of resource being monitored. If set to **normal**, the default action when pressing the line key adjacent to this monitored user is to initiate a call if the user is idle or busy and to perform a directed call pickup if the user is ringing. Any active calls are first placed on hold. If set to **automata**, the default action when pressing the line key adjacent to this monitored user is to perform a park/blind transfer of any currently active call. If there is no active call and the monitored user is ringing/busy, an attempt to perform a directed call pickup/park retrieval is made.

Behaviors <behaviors/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
attendant.behaviors.display. spontaneousCallAppearances.normal	0 or 1	1	A flag to determine whether or not a call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource. The information displayed after a press-and-hold of a resource's line key is unchanged by this parameter. If set to 1, the display is enabled.
attendant.behaviors.display. spontaneousCallAppearances.automata	0 or 1	0	
attendant.behaviors.display. remoteCallerID.normal	0 or 1	1	A flag to determine whether or not remote party caller ID information is presented to the attendant. If set to 0 (disabled), the string "unknown" would be substituted for both name and number information.
attendant.behaviors.display. remoteCallerID.automata	0 or 1	1	

Roaming Buddies <roaming_buddies/>

Note This attribute is used in conjunction with Microsoft Live Communications Server 2005 only.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
roaming_buddies.reg	positive integer	Null	Specifies the line/registration number which has roaming buddies support enabled. If Null, roaming buddies is disabled. If value < 1, then value is replaced with 1. ***Warning:*** *This parameter must be enabled (value > 0) if the call server is Microsoft Live Communications Server 2005.*

Roaming Privacy <roaming_privacy/>

Note This attribute is used in conjunction with Microsoft Live Communications Server 2005 only.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
roaming_privacy.reg	positive integer	Null	Specifies the line/registration number which has roaming privacy support enabled. If Null, roaming privacy is disabled. If value < 1, then value is replaced with 1.

User Preferences <up/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
up.analogHeadsetOption	0, 1, or 2	0	Selects optional external hardware for use with a headset attached to the phone's analog headset jack. If set to 0, no compatible headset is attached. If set to 1, a DHSG-compatible headset is attached and can be used as an electronic hookswitch. If set to 2, a Plantronics compatible headset is attached and can be used an electronic hookswitch.
up.offHookAction.none	0 or 1	Null	If set to 0 or Null, the behavior introduced in SIP 2.1.2 occurs. When users go off-hook, the phone tries to seize a line. Which line is seized depends on `voIpProt.SIP.strictLineSeize`, `voIPProt.SIP.lineSeize.retries`, and `reg.x.strictLineSeize`. If set to 1, the behavior from SIP 1.6.7 occurs. When users go off-hook, the phone does not seize a line or answer a ringing call. The user must use the line keys to either make a new call or answer a ringing call. This will apply under all ringer settings, not just SilentRing.
up.pictureFrame.folder	string	Null	The path name for images. The maximum length is 40 characters. If set to Null, images stored in the root folder on the USB flash drive are displayed. For example, if the images are stored in the “/images/phone” folder on the USB flash drive, set `up.pictureFrame.folder` to **images/phone** . ***Note**: This parameter is supported on the Polycom VVX 1500 only.*
up.pictureFrame. timePerImage	3 to 300 seconds	Null	The time to display the image. If set to Null, the default time is 5 seconds. ***Note**: This parameter is supported on the Polycom VVX 1500 only.*

Attribute	Permitted Values	Default	Interpretation
up.screenSaver.enabled	0 or 1	0	If set to 1, a USB flash drive is attached to the phone, and the idle browser is not configured, a slide show will cycle through the images from the USB flash drive. The images must stored in the appropriate directory of the USB flash drive (`up.pictureFrame.folder` in **phone1.cfg**). The slide show does not appear when the phone is in the active state. If set to 1, but there is no USB flash drive attached to the phone, there is not change on the screen. However, the screen saver will start working once a USB flash drive is attached. If set to 0, the feature is disabled. ***Note**: This parameter is supported on the Polycom VVX 1500 only.* ***Note**: If the idle browser is also enabled, the idle browser is displayed until the screen saver times out; then the screen saver appears. When the screen saver exits, the idle browser is displayed again and is up to date (it is refreshed in the background).*
up.screenSaver.waitTime	1 to 9999	Null	The time to wait (In minutes) in the idle state (until the screen saver starts). If set to Null, the default time is 15 minutes. ***Note**: This parameter is supported on the Polycom VVX 1500 only.*

Automatic Call Distribution <acd/>

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
acd.reg	1 to 34	1	The registration index used to support BroadSoft server-based ACD. If set to Null, line 1 is used.
acd.stateAtSignIn	0 or 1	1	The state of the user when signing in. If set to 1 or Null, the user is available. If set to 0, the user is unavailable.

Flash Parameter Configuration

Any field in the BootROM setup menu and the application Line Configuration and Call Server Configuration menus can be set through a configuration file.

A DHCP server can be configured to point the phones to a provisioning server that has the required configuration files. The new settings will be downloaded by the phones and used to configure them. This removes the need for manual interaction with phones to configure basic settings. This is especially useful for initial installation of multiple phones.

These device settings are detected when the application starts. If the new settings would normally cause a reboot if they were changed in the application Network Configuration menu, then they will cause a reboot when the application starts.

Warning

The parameters for this feature should be put in separate configuration files to simplify maintenance. Do not add them to existing configuration files (such as **sip.cfg**). One new configuration file will be required for parameters that should apply to all phones, and individual configuration files will be required for phone-specific parameters such as SIP registration information.

The global `device.set` parameter must be enabled when the initial installation is done, and then it should be disabled. This prevents subsequent reboots by individual phones triggering a reset of parameters on the phone that may have been tweaked since the initial installation.

Warning

This feature is very powerful and should be used with caution. For example, an incorrect setting could set the IP Address of multiple phones to the same value.

Note that some parameters may be ignored, for example if DHCP is enabled it will still override the value set with `device.net.ipAddress`.

Individual parameters are checked to see whether they are in range, however, the interaction between parameters is not checked. If a parameter is out of range, an error message will appear in the log file and parameter will not be used.

Incorrect configuration could cause phones to get into a reboot loop. For example, server A has a configuration file that specifies that server B should be used, which has a configuration file that specifies that server A should be used.

Polycom recommends that you test the new configuration files on two phones before initializing all phones. This should detect any errors including IP address conflicts.

The flash attributes are defined as follows:

Name	Possible Values	Description
device.set	0 or 1 default = 0	If set to 0, do not use any `device.xxx.yyy` fields to set any parameters. Set this to 0 after the initial installation. If set to 1, use the `device.xxx.yyy` fields that have `device.xxx.yyy.set` = 1. Set this to 1 for the initial installation only.
device.xxx.yyy.set	0 or 1 default = 0	If set to 0, do not use the `device.xxx.yyy` value. If set to 1, use the `device.xxx.yyy` value. For example, if `device.net.ipAddress.set` = 1, then use the contents of the `device.net.ipAddress` field.
device.net.ipAddress	dotted-decimal IP address	Phone's IP address. ***Note:*** *This field is not used when DHCP client is enabled.*
device.net.subnetMask	dotted-decimal IP address	Phone's subnet mask. ***Note:*** *This field is not used when DHCP client is enabled.*
device.net.IPgateway	dotted-decimal IP address	Phone's default router / IP gateway. ***Note:*** *This field is not used when DHCP client is enabled.*
device.net.vlanId	Null, 0 to 4094	Phone's 802.1Q VLAN identifier. ***Note:*** *Null = no VLAN tagging*
device.net.cdpEnabled	0 or 1	If set to 1, the phone will attempt to determine its VLAN ID and negotiate power through CDP.
device.dhcp.enabled	0 or 1	For description, refer to DHCP or Manual TCP/IP Setup on page 3-2.
device.dhcp. bootSrvUseOpt	0 to 3	For descriptions, refer to DHCP Menu on page 3-8.
device.dhcp.bootSrvOpt	128 to 254 (Cannot be the same as VLAN ID Option)	
device.dhcp. bootSrvOptType	0 or 1	
device.dhcp. dhcpVlanDiscUseOpt	0 to 2	
device.dhcp. dhcpVlanDiscOpt	128 to 254 (Cannot be the same as provisioning server Option)	

Name	Possible Values	Description
device.prov.serverName	any string	For descriptions, refer to Server Menu on page 3-10.
device.prov.serverType	0 to 4	
device.prov.user	any string	
device.prov.password	any string	
device.prov.appProvType	0 or 1	
device.prov.appProvString	any string	
device.prov. redunAttemptLimit	10, Null	
device.prov. redunInterAttemptDelay	300, Null	
device.prov. maxRedunServers	1 to 8	
device.sntp.serverName	any string	Can be dotted-decimal IP address or domain name string. SNTP server from which the phone will obtain the current time
device.sntp.gmtOffset	-43200 to 46800	GMT offset in seconds, corresponding to -12 to +13 hours.
device.dns.serverAddress	dotted-decimal IP address	Primary server to which the phone directs Domain Name System queries.
device.dns.altSrvAddress	dotted-decimal IP address	Secondary server to which the phone directs Domain Name System queries.
device.dns.domain	any string	The phone's DNS domain.
device.auth. localAdminPassword	any string	The phone's local administrator password.
device.auth. localUserPassword	any string	The phone user's local password.
device.auth.regUserx	any string	The SIP registration user name for registration x where x = 1 to 48.
device.auth.regPasswordx	any string	The SIP registration password for registration x where x = 1 to 48.
device.sec. configEncryption.key	any string	Configuration encryption key that is used for encryption of configuration files.
device.syslog.serverName	dotted-decimal IP address OR domain name string	The syslog server IP address or host name. The default value is NULL.

Name	Possible Values	Description
device.syslog.transport	None=0, UDP=1, TCP=2, TLS=3	The protocol that the phone will use to write to the syslog server. If set to "None", transmission is turned off, but the server address is preserved.
device.syslog.facility	0 to 23	A description of what generated the log message. For more information, refer to section 4.1.1 of RFC 3165. The default value is 16, which maps to "local 0".
device.syslog.renderLevel	0 to 6	Specifies the lowest class of event that will be rendered to syslog. It is based on `log.render.level` and can be a lower value. Refer to Basic Logging <level/><change/> and <render/> on page A-106.
device.syslog.prependMac	Enabled, Disabled	If enabled, the phone's MAC address is prepended to the log message sent to the syslog server.
device.em.power	Enabled, Disabled, Null	Refer to the EM Power parameter in Main Menu on page 3-7.
device.net.etherVlanFilter	Enabled, Disabled	Refer to the VLAN Filtering parameter in Ethernet Menu on page 3-12.
device.net.etherStormFilter	Enabled, Disabled	Refer to the Storm Filtering parameter in Ethernet Menu on page 3-12.
device.net.etherModeLAN	-1 to 5	Refer to the LAN Port Mode parameter in Ethernet Menu on page 3-12.
device.net.etherModePC	-1 to 5	Refer to the PC Port Mode parameter in Ethernet Menu on page 3-12.
device.serial.enable	0,1	Enables the debug serial port. The default value is 1.
device.sec.SSL.certList	all, custom, default	The type of certificate list.
device.sec.SSL.customCert	X.509 certificate	The certificate value.
device.net.lldpEnabled	0 or 1	If set to 1, the phone will attempt to determine its VLAN ID and negotiate power through LLDP. If set to 0, the phone will not attempt to determine its VLAN ID or power management through LLDP.
device.prov.clinkEnabled	0 or 1 (default)	If set to 1, enable the CLink subsystem when the Ethernet cable is not attached. If set to 0, disable the CLink subsystem when the Ethernet cable is not attached.

B

Session Initiation Protocol (SIP)

This chapter provides a description of the basic Session Initiation Protocol (SIP) and the protocol extensions that are supported by the current SIP application. To find the applicable Request For Comments (RFC) document, go to http://www.ietf.org/rfc.html and enter the RFC number.

This chapter contains information on:

- Basic Protocols—All the basic calling functionality described in the SIP specification is supported. Transfer is included in the basic SIP support.
- Protocol Extensions—Extensions add features to SIP that are applicable to a range of applications, including reliable 1xx responses and session timers.

For information on supported RFC's and Internet drafts, refer to the following section, RFC and Internet Draft Support.

This chapter also describes:

- Request Support
- Header Support
- Response Support
- Hold Implementation
- Reliability of Provisional Responses
- Transfer
- Third Party Call Control
- SIP for Instant Messaging and Presence Leveraging Extensions
- Shared Call Appearance Signaling
- Bridged Line Appearance Signaling

RFC and Internet Draft Support

The following RFC's and Internet drafts are supported:

- RFC 1321—The MD5 Message-Digest Algorithm
- RFC 2327—SDP: Session Description Protocol
- RFC 2387—The MIME Multipart / Related Content-type
- RFC 2976—The SIP INFO Method
- RFC 3261—SIP: Session Initiation Protocol (replacement for RFC 2543)
- RFC 3262—Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3263—Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264—An Offer / Answer Model with the Session Description Protocol (SDP)
- RFC 3265—Session Initiation Protocol (SIP) - Specific Event Notification
- RFC 3311—The Session Initiation Protocol (SIP) UPDATE Method
- RFC 3325—SIP Asserted Identity
- RFC 3515—The Session Initiation Protocol (SIP) Refer Method
- RFC 3555 — MIME Type of RTP Payload Formats
- RFC 3611 — RTP Control Protocol Extended reports (RTCP XR)
- RFC 3665—Session Initiation Protocol (SIP) Basic Call Flow Examples
- draft-ietf-sip-cc-transfer-05.txt—SIP Call Control - Transfer
- RFC 3725—Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3842—A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC 3856—A Presence Event Package for Session Initiation Protocol (SIP)
- RFC 3891—The Session Initiation Protocol (SIP) "Replaces" Header
- RFC 3892—The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 3959—The Early Session Disposition Type for the Session Initiation Protocol (SIP)
- RFC 3960—Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
- RFC 3968—The Internet Assigned Number Authority (IANA) Header Field Parameter Registry for the Session Initiation Protocol (SIP)

- RFC 3969—The Internet Assigned Number Authority (IANA) Uniform Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP)
- RFC 4028—Session Timers in the Session Initiation Protocol (SIP)
- RFC 4235—An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- draft-levy-sip-diversion-08.txt—Diversion Indication in SIP
- draft-anil-sipping-bla-02.txt—Implementing Bridged Line Appearances (BLA) Using Session Initiation Protocol (SIP)
- draft-ietf-sip-privacy-04.txt—SIP Extensions for Network-Asserted Caller Identity and Privacy within Trusted Networks
- draft-ietf-sipping-cc-conferencing-03.txt—SIP Call Control - Conferencing for User Agents
- draft-ietf-sipping-rtcp-summary-02.txt —Session Initiation Protocol Package for Voice Quality Reporting Event
- draft-ietf-sip-connect-reuse-04.txt—Connection Reuse in the Session Initiation Protocol (SIP)

Request Support

The following SIP request messages are supported:

Method	Supported	Notes
REGISTER	Yes	
INVITE	Yes	
ACK	Yes	
CANCEL	Yes	
BYE	Yes	
OPTIONS	Yes	
SUBSCRIBE	Yes	
NOTIFY	Yes	
REFER	Yes	
PRACK	Yes	

Method	Supported	Notes
INFO	Yes	RFC 2976, the phone does not generate INFO requests, but will issue a final response upon receipt. No INFO message bodies are parsed.
MESSAGE	Yes	Final response is sent upon receipt. Message bodies of type text/plain are sent and received.
UPDATE	Yes	

Header Support

The following SIP request headers are supported:

Note In the following table, a "Yes" in the Supported column means the header is sent and properly parsed.

Header	Supported	Notes
Accept	Yes	
Accept-Encoding	No	
Accept-Language	Yes	
Access-Network-Info	No	
Alert-Info	Yes	
Allow	Yes	
Allow-Events	Yes	
Authentication-Info	No	
Authorization	Yes	
Call-ID	Yes	
Call-Info	Yes	
Contact	Yes	
Content-Disposition	No	
Content-Encoding	No	
Content-Language	No	
Content-Length	Yes	
Content-Type	Yes	
CSeq	Yes	

Header	Supported	Notes
Date	No	
Diversion	Yes	
Error-Info	No	
Event	Yes	
Expires	Yes	
From	Yes	
In-Reply-To	No	
Max-Forwards	Yes	
Min-Expires	No	
Min-SE	Yes	
MIME-Version	No	
Organization	No	
P-Asserted-Identity	Yes	
P-Preferred-Identity	Yes	
Priority	No	
Privacy	No	
Proxy-Authenticate	Yes	
Proxy-Authorization	Yes	
Proxy-Require	Yes	
RAck	Yes	
Record-Route	Yes	
Refer-To	Yes	
Referred-By	Yes	
Referred-To	Yes	
Remote-Party-ID	Yes	
Replaces	Yes	
Reply-To	No	
Requested-By	No	
Require	Yes	
Response-Key	No	

Header	Supported	Notes
Retry-After	Yes	
Route	Yes	
RSeq	Yes	
Server	Yes	
Session-Expires	Yes	
Subject	Yes	
Subscription-State	Yes	
Supported	Yes	
Timestamp	Yes	
To	Yes	
Unsupported	Yes	
User-Agent	Yes	
Via	Yes	
Warning	Yes	Only warning codes 300 to 399
WWW-Authenticate	Yes	

Response Support

The following SIP responses are supported:

Note In the following table, a "Yes" in the Supported column means the header is sent and properly parsed. The phone may not actually generate the response.

1xx Responses - Provisional

Response	Supported	Notes
100 Trying	Yes	
180 Ringing	Yes	
181 Call Is Being Forwarded	No	
182 Queued	No	
183 Session Progress	Yes	

2xx Responses - Success

Response	Supported	Notes
200 OK	Yes	
202 Accepted	Yes	In REFER transfer.

3xx Responses - Redirection

Response	Supported	Notes
300 Multiple Choices	Yes	
301 Moved Permanently	Yes	
302 Moved Temporarily	Yes	
305 Use Proxy	No	
380 Alternative Service	No	

4xx Responses - Request Failure

Note All 4xx responses for which the phone does not provide specific support will be treated the same as 400 Bad Request.

Response	Supported	Notes
400 Bad Request	Yes	
401 Unauthorized	Yes	
402 Payment Required	No	
403 Forbidden	No	
404 Not Found	Yes	
405 Method Not Allowed	Yes	
406 Not Acceptable	No	
407 Proxy Authentication Required	Yes	
408 Request Timeout	No	
410 Gone	No	
413 Request Entity Too Large	No	
414 Request-URI Too Long	No	

Response	Supported	Notes
415 Unsupported Media Type	Yes	
416 Unsupported URI Scheme	No	
420 Bad Extension	No	
421 Extension Required	No	
423 Interval Too Brief	No	
480 Temporarily Unavailable	Yes	
481 Call/Transaction Does Not Exist	Yes	
482 Loop Detected	Yes	
483 Too Many Hops	No	
484 Address Incomplete	Yes	
485 Ambiguous	No	
486 Busy Here	Yes	
487 Request Terminated	Yes	
488 Not Acceptable Here	Yes	
491 Request Pending	No	
493 Undecipherable	No	

5xx Responses - Server Failure

Response	Supported	Notes
500 Server Internal Error	Yes	
501 Not Implemented	Yes	
502 Bad Gateway	No	
503 Service Unavailable	No	
504 Server Time-out	No	
505 Version Not Supported	No	
513 Message Too Large	No	

6xx Responses - Global Failure

Response	Supported	Notes
600 Busy Everywhere	No	
603 Decline	Yes	
604 Does Not Exist Anywhere	No	
606 Not Acceptable	No	

Hold Implementation

The phone supports both currently accepted means of signaling hold.

The first method, no longer recommended due in part to the RTCP problems associated with it, is to set the "c" destination addresses for the media streams in the SDP to zero, for example, c=0.0.0.0.

The second, and preferred, method is to signal the media directions with the "a" SDP media attributes sendonly, recvonly, inactive, or sendrecv. The hold signaling method used by the phone is configurable (refer to SIP <SIP/>on page A-11), but both methods are supported when signaled by the remote end point.

Note Even if the phone is set to use c=0.0.0.0, it will not do so if it gets any sendrecv, sendonly, or inactive from the server. These flags will cause it to revert to the other hold method.

Reliability of Provisional Responses

The phone fully supports RFC 3262 - *Reliability of Provisional Responses*.

Transfer

The phone supports transfer using the REFER method specified in draft-ietf-sip-cc-transfer-05 and RFC 3515.

Third Party Call Control

The phone supports the delayed media negotiations (INVITE without SDP) associated with third party call control applications.

When used with an appropriate server, the User Agent Computer Supported Telecommunications Applications (uaCSTA) feature on the phone may be utilized for remote control of the phone from computer applications such as Microsoft Office Communicator.

The phone is compliant with "Using CSTA for SIP Phone User Agents (uaCSTA), ECMA TR/087" for the Answer Call, Hold Call, and Retrieve Call functions and "Services for Computer Supported Telecommunications Applications Phase III", ECMA – 269 for the Conference Call function.

This feature is enabled by configuration parameters described in SIP <SIP/> on page A-11 and Registration <reg/> on page A-134 and needs to be activated by a feature application key.

SIP for Instant Messaging and Presence Leveraging Extensions

The phone is compatible with the Presence and Instant Messaging features of Microsoft Windows Messenger 5.1. In a future release, support for the Presence and Instant Message recommendations in the SIP Instant Messaging and Presence Leveraging Extensions (SIMPLE) proposals will be provided by the following Internet drafts or their successors:

- draft-ietf-simple-cpim-mapping-01
- draft-ietf-simple-presence-07
- draft-ietf-simple-presencelist-package-00
- draft-ietf-simple-winfo-format-02
- draft-ietf-simple-winfo-package-02

Shared Call Appearance Signaling

A shared line is an address of record managed by a call server. The server allows multiple end points to register locations against the address of record.

The phone supports shared call appearances (SCA) using the SUBSCRIBE-NOTIFY method in the "SIP Specific Event Notification" framework (RFC 3265). The events used are:

- "call-info" for call appearance state notification
- "line-seize for the phone to ask to seize the line

Bridged Line Appearance Signaling

A bridged line is an address of record managed by a server. The server allows multiple end points to register locations against the address of record.

The phone supports bridged line appearances (BLA) using the SUBSCRIBE-NOTIFY method in the "SIP Specific Event Notification" framework (RFC 3265). The events used are:

- "dialog" for bridged line appearance subscribe and notify

C

Miscellaneous Administrative Tasks

This appendix provides information required by varied aspects of the Session Initiation Protocol (SIP) application. This includes:

- Trusted Certificate Authority List
- Encrypting Configuration Files
- Adding a Background Logo
- BootROM/SIP Application Dependencies
- Supported SoundStation IP 7000 / Polycom HDX Software Interoperability
- Multiple Key Combinations
- Default Feature Key Layouts
- Internal Key Functions
- Assigning a VLAN ID Using DHCP
- Parsing Vendor ID Information
- Product, Model, and Part Number Mapping
- Disabling PC Ethernet Port
- Modifying Phone's Configuration Using the Web Interface
- Capturing Phone's Current Screen
- LLDP and Supported TLVs

Trusted Certificate Authority List

The following certificate authorities are trusted by the phone by default:

- ABAecom (sub., Am. Bankers Assn.) Root CA
- ANX Network CA by DST

- American Express CA
- American Express Global CA
- BelSign Object Publishing CA
- BelSign Secure Server CA
- Deutsche Telekom AG Root CA
- Digital Signature Trust Co. Global CA 1
- Digital Signature Trust Co. Global CA 2
- Digital Signature Trust Co. Global CA 3
- Digital Signature Trust Co. Global CA 4
- Entrust Worldwide by DST
- Entrust.net Premium 2048 Secure Server CA
- Entrust.net Secure Personal CA
- Entrust.net Secure Server CA
- Equifax Premium CA
- Equifax Secure CA
- Equifax Secure eBusiness CA 1
- Equifax Secure eBusiness CA 2
- Equifax Secure Global eBusiness CA 1
- GeoTrust Primary Certification Authority
- GeoTrust Global CA
- GeoTrust Global CA 2
- GeoTrust Universal CA
- GeoTrust Universal CA 2
- GTE CyberTrust Global Root
- GTE CyberTrust Japan Root CA
- GTE CyberTrust Japan Secure Server CA
- GTE CyberTrust Root 2
- GTE CyberTrust Root 3
- GTE CyberTrust Root 4
- GTE CyberTrust Root 5

- GTE CyberTrust Root CA
- GlobalSign Partners CA
- GlobalSign Primary Class 1 CA
- GlobalSign Primary Class 2 CA
- GlobalSign Primary Class 3 CA
- GlobalSign Root CA
- National Retail Federation by DST
- TC TrustCenter, Germany, Class 1 CA
- TC TrustCenter, Germany, Class 2 CA
- TC TrustCenter, Germany, Class 3 CA
- TC TrustCenter, Germany, Class 4 CA
- Thawte Personal Basic CA
- Thawte Personal Freemail CA
- Thawte Personal Premium CA
- Thawte Premium Server CA
- Thawte Server CA
- Thawte Universal CA Root
- UPS Document Exchange by DST
- ValiCert Class 1 VA
- ValiCert Class 2 VA
- ValiCert Class 3 VA
- VeriSign Class 4 Primary CA
- Verisign Class 1 Public Primary Certification Authority
- Verisign Class 1 Public Primary Certification Authority - G2
- Verisign Class 1 Public Primary Certification Authority - G3
- Verisign Class 2 Public Primary Certification Authority
- Verisign Class 2 Public Primary Certification Authority - G2
- Verisign Class 2 Public Primary Certification Authority - G3
- Verisign Class 3 Public Primary Certification Authority
- Verisign Class 3 Public Primary Certification Authority - G2

- Verisign Class 3 Public Primary Certification Authority - G3
- Verisign Class 4 Public Primary Certification Authority - G2
- Verisign Class 4 Public Primary Certification Authority - G3
- Verisign/RSA Commercial CA
- Verisign/RSA Secure Server CA

Polycom endeavors to maintain a built-in list of the most commonly used CA Certificates. Due to memory constraints, we cannot keep as thorough a list as some other applications (for example, browsers). If you are using a certificate from a commercial Certificate Authority not in the list above, you may submit a Feature Request for Polycom to add your CA to the trusted list by visiting https://jira.polycom.com:8443//secure/CreateIssue!default.jspa?os_username=jiraguest&os_password=polycom. At this point, you can use the Custom Certificate method to load your particular CA certificate into the phone (refer to "Technical Bulletin 17877: using Custom Certificates on SoundPoint IP Phones" at http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html).

Encrypting Configuration Files

The phone can recognize encrypted files, which it downloads from the provisioning server and it can encrypt files before uploading them to the provisioning server. There must be an encryption key on the phone to perform these operations. Configuration files (excluding the master configuration file), contact directories, and configuration override files can be encrypted.

A separate SDK, with a readme file, is provided to facilitate key generation and configuration file encryption and decrypt on a UNIX or Linux server. The utility is distributed as source code that runs under the UNIX operating system. For more information, contact Polycom Technical Support.

A key is generated by the utility and must be downloaded to the phone so that it can decrypt the files that were encrypted on the server. The `device.sec.configEncryption.key` configuration file parameter is used to set the key on the phone. The utility generates a random key and the encryption is Advanced Encryption Standard (AES) 128 in Cipher Block Chaining (CBC) mode. An example key would look like this:

```
Crypt=1;KeyDesc=companyNameKey1;Key=06a9214036b8a15b512e03d534120006;
```

If the phone doesn't have a key, it must be downloaded to the phone in plain text (a potential security hole if not using HTTPS). If the phone already has a key, a new key can be downloaded to the phone encrypted using the old key (refer to Changing the Key on the Phone on page C-6). At a later date, new phones from the factory will have a key pre-loaded in them. This key will be changed at regular intervals to enhance security

It is recommended that all keys have unique descriptive strings in order to allow simple identification of which key was used to encrypt a file. This makes provisioning server management easier.

After encrypting a configuration file, it is useful to rename the file to avoid confusing it with the original version, for example rename **sip.cfg** to **sip.enc**. However, the directory and override filenames cannot be changed in this manner.

You can check whether an encrypted file is the same as an unencrypted file by:

1. Run the configFileEncrypt utility on the unencrypted file with the "-d" option. This shows the "digest" field.
2. Look at the encrypted file using WordPad and check the first line that shows a "Digest=…." field. If the two fields are the same, then the encrypted and unencrypted file are the same.

Note If a phone downloads an encrypted file that it cannot decrypt, the action is logged, an error message displays, and the phone reboots. The phone will continue to do this until the provisioning server provides an encrypted file that can be read, an unencrypted file, or the file is removed from the master configuration file list.

Note Encrypted configuration files can only be decrypted on the SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670, the SoundStation IP 6000 and 7000 phones, and the Polycom VVX 1500 phones.

The master configuration file cannot be encrypted on the provisioning server. This file is downloaded by the BootROM that does not recognize encrypted files. For more information, refer to Master Configuration Files on page 2-5.

The following configuration file changes are required to modify this feature:

Central (provisioning server)	Configuration File: **sip.cfg**	Specify the phone-specific contact directory and the phone-specific configuration override file. For more information, refer to Encryption <encryption/> on page A-109.
	Configuration file: **<device>.cfg**	Change the encryption key. For more information, refer to Flash Parameter Configuration on page A-157.

Changing the Key on the Phone

For security purposes, it may be desirable to change the key on the phones and the server from time to time.

To change a key:

1. Put the new key into a configuration file that is in the list of files downloaded by the phone (specified in **000000000000.cfg** or **<Ethernet address>.cfg**).

   Use the `device.sec.configEncryption.key` parameter to specify the new key.

2. Manually reboot the phone so that it will download the new key. The phone will automatically reboot a second time to use the new key.

   At this point, the phone expects all encrypted configuration files on the provisioning server to use the new key and it will continue to reboot until this is the case. The files on the server must be updated to the new key or they must be made available in unencrypted format. Updating to the new key requires decrypting the file with the old key, then encrypting it with the new key.

   Note that configuration files, contact directory files and configuration override files may all need to be updated if they were already encrypted. In the case of configuration override files, they can be deleted from the provisioning server so that the phone will replace them when it successfully boots.

Adding a Background Logo

Note Background logos are not supported on the Polycom VVX 1500 phone.

This section provides instructions on how to add a background logo to all SoundPoint IP phones in your organization. You must be running at least BootROM 2.x.x and SIP 1.x.x. One bitmap file is required for each model.

Model	Width	Height	Color Depth
IP 32x/33x	102	23	monochrome
IP 430	94	23	monochrome
IP 450	256	116	4-bit grayscale or monochrome
IP 550/560/650	209	109	4-bit grayscale or monochrome

Model	Width	Height	Color Depth
IP 670	209	109	12-bit color
IP 6000	150	33	32-bit grayscale or monochrome
IP 7000	255	128	32-bit grayscale or monochrome

Logos smaller than described in the table above are acceptable, but larger logos may be truncated or interfere with other areas of the user interface.

Color	RGB Values (Decimal)	RGB Values (Hexadecimal)
Black	0,0,0	00,00,00
Dark Gray	96,96,96	60,60,60
Light Gray	160,160,160	A0,A0,A0
White	255,255,255	FF,FF,FF

The SoundPoint IP 450/550/560/650 phone support a 4-bit grayscale, which is a smooth gradient from black (0, 0, 0) to white (FF, FF, FF).

The SoundPoint IP 670 phone support a 12-bit color scale from black (0, 0, 0) to white (FFFF, FFFF, FFFF).

The SoundStation IP 6000 phone is the same as the IP 7000.

The SoundStation IP 7000 phone supports a 32-bit grayscale, which is a smooth gradient from black (0, 0, 0) to white (FF, FF, FF).

Configuration File Changes

In the `<bitmaps>` section of **sip.cfg**, find the end of each model's bitmap list and add your bitmap to the end; do not include the **.bmp** extension.

Model	Associate Parameter
IP 32x/33x	`bitmap.IP_330.68.name`
IP 430	`bitmap.IP_400.61.name`
IP 450	`bitmap.IP_450.82.name`
IP 550, 560, 650, 670	`bitmap.IP_600.83.name`
IP 6000	`bitmap.IP_4000.83.name`
IP 7000	`bitmap.IP_7000.84.name`

For example:

```
<bitmaps>
  <IP_330 … bitmap.IP_330.68.name="logo-330" />
  <IP_400 … bitmap.IP_400.61.name="logo-430" />
  <IP_450 … bitmap.IP_450.82.name="logo-450" />
  <IP_600 … bitmap.IP_600.83.name="logo-650" />
  <IP_4000 … bitmap.IP_4000.83.name="logo-6000" />
  <IP_7000 … bitmap.IP_7000.84.name="logo-7000" />
</bitmaps>
```

Next, enable the idle display feature and modify the idle display "animation" for each model to point to your bitmap (again without the **.bmp** extension):

```
<indicators ind.idleDisplay.enabled="1">
  <Animations>
    <IP_330>
    …
    <IDLE_DISPLAY ind.anim.IP_330.30.frame.1.bitmap="logo-330"
ind.anim.IP_330.30.frame.1.duration="0"/>
    …
    </IP_330>
    <IP_400>
      …
      <IDLE_DISPLAY ind.anim.IP_400.30.frame.1.bitmap="logo-400"
ind.anim.IP_400.30.frame.1.duration="0"/>
      …
    </IP_400>
    <IP_450>
      …
      <IDLE_DISPLAY ind.anim.IP_450.45.frame.1.bitmap="logo-450"
ind.anim.IP_450.45.frame.1.duration="0"/>
      …
    </IP_450>
    <IP_600>
      …
      <IDLE_DISPLAY ind.anim.IP_600.46.frame.1.bitmap="logo-650"
ind.anim.IP_600.46.frame.1.duration="0"/>
      …
    </IP_600>
    <IP_4000>
      …
      <IDLE_DISPLAY ind.anim.IP_4000.45.frame.1.bitmap="logo-6000"
ind.anim.IP_4000.45.frame.1.duration="0"/>
      …
    </IP_4000>
    <IP_7000>
      …
      <IDLE_DISPLAY ind.anim.IP_7000.46.frame.1.bitmap="logo-7000"
ind.anim.IP_7000.46.frame.1.duration="0"/>
      …
```

```
    </IP_7000>
  </Animations>
  …
</indicators>
```

BootROM/SIP Application Dependencies

Not withstanding the hardware backward compatibility mandate, there have been times throughout the life of the SoundPoint IP / SoundStation IP / VVX phones where certain dependencies on specific BootROM and application versions have been necessitated.

This table summarizes some the major dependencies that you are likely to encounter:

Model	BootROM	SIP Application
IP 320/330	3.2.3B or later	2.1.1 or later
IP 321/331	4.1.3 or later	3.1.3C or later
IP 335	4.2.0B or later	4.1.2B or later
IP 430	3.1.3C or later	1.6.6 or later
IP 450	4.1.2 or later	3.1.0C or later
IP 550[1]	3.2.2B or later	2.1 or later
IP 560[1]	4.0.1 or later	2.2.2 or later
IP 650/EM[1]	3.2.2B or later	2.0.3B or later
IP 650/BEM	4.0.1 or later	2.2.2 or later
IP 670/CEM	4.1.1 or later	3.0.3 or later
IP 6000	4.1.1 or later	3.0.2 or later
IP 7000[2]	4.1.1 or later	3.0.2 or later
VVX 1500	4.1.2 or later	3.1.2B or later

Note

1. SoundPoint IP 550, 560 and 650 phones manufactured as of February 2009 have additional BootROM/SIP application dependencies. For more information, refer to “Technical Bulletin TB 46440: Notice of Product Shipping Configuration Change” at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .
2. If the SoundStation IP 7000 is connected to a Polycom HDX system, the BootROM must be 4.1.2 or later.

Migration Dependencies

In addition to the BootROM and application dependencies, there are certain restrictions with regard to upgrading or downgrading from one BootROM release to another BootROM release. These restrictions are typically caused by the addition of features that change the way BootROM provisioning is done, so the older version become incompatible.

There is always a way to move forward with BootROM releases, although it may be a two or three step procedure sometimes, but there are cases where it is impossible to move backward. Make special note of these cases before upgrading.

For the latest information, refer to the latest *Release Notes*.

Supported SoundStation IP 7000 / Polycom HDX Software Interoperability

To operate your SoundStation IP 7000 phone in this environment, Polycom recommends that you run the following combination of software on the phone and the HDX system:

- SoundStation IP7000 running SIP 3.2.1 and BootROM 4.2.0
- Polycom HDX systems running HDX Version 2.5.0.2_29-3382 (hot fix 29)

The following table shows all supported software for the SoundStation IP 7000 and Polycom HDX systems.

For Polycom HDX 4000 HDX 8000 HDX 9000	• **Recommended:** SIP 3.2.1/BootROM 4.2.0 in combination with HDX 2.5.0.2_29-3382 (hot fix 29) • **Compatible:** SIP 3.1.1RevB / BootROM 4.1.2 through SIP 3.1.3RevC / BootROM 4.1.3 in combination with HDX 2.5 through 2.5.0.5
For Polycom HDX 7000	• **Recommended:** SIP 3.2.1/BootROM 4.2.0 in combination with HDX 2.5.0.2_29-3382 (hot fix 29) • **Compatible:** SIP 3.1.2RevC / BootROM 4.1.2 through SIP 3.1.3RevC / BootROM 4.1.3 in combination with HDX 2.5 through 2.5.0.5
For Polycom HDX 6000	• **Recommended:** SIP 3.2.1/BootROM 4.2.0 in combination with HDX 2.5.0.7 planned for Q4 2009

Multiple Key Combinations

On SoundPoint IP and SoundStation IP phones, certain multiple key combinations can be used to reboot the phone and restore factory defaults.

For other methods for resetting and rebooting your SoundPoint IP, SoundStation IP, or Polycom VVX phones, refer to "Quick Tip 18298: Resetting and Rebooting SoundPoint IP / SoundStation IP / VVX Phones" at http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html .

Rebooting the Phone

For the key combination, press and hold certain key combinations (depending on the phone model) simultaneously until a confirmation tone is heard or for about three seconds:

- IP 32x/33x: Volume-, Volume+, Hold, and Hands-free
- IP 430: Volume-, Volume+, Hold, and Messages
- IP 450, 550, 560, 600, 601, and 650, and 670: Volume-, Volume+, Mute, and Messages
- IP 6000: *, #, Volume+, and Select
- IP 7000: *, #, Volume-, and Volume+
- VVX 1500: Delete, Volume-, Volume+, and Select

Note As of SIP 3.2.0, users can restart their phones by pressing the **Menu** key, and then selecting **Settings > Basic > Restart Phone**. Any new BootROM and SIP applications will be downloaded to the phone as a result of this restart.

Restoring Factory Defaults

For the key combination, press and hold certain key combinations (depending on the phone model) simultaneously during the countdown process in the BootROM until the password prompt appears:

- IP 450, 550, 600, 601, and 650, and 670 and VVX 1500: 4, 6, 8 and * dial pad keys
- IP 32x/33x, 430, 560, 7000: 1, 3, 5, and 7 dial pad keys
- IP 6000: 6, 8 and * dial pad keys

Enter the administrator password to initiate the reset. Resetting to factory defaults will also reset the administrator password (factory default password is 456). Polycom recommends that you change the administrative password from the default value.

Uploading Log Files

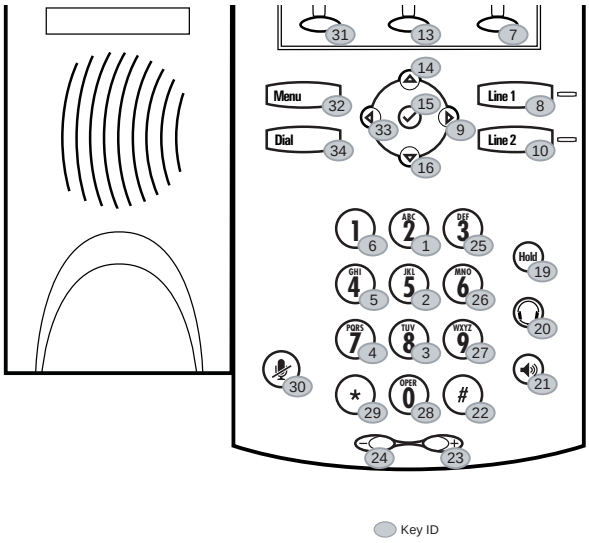
For the key combination, press and hold certain key combinations (depending on the phone model) simultaneously until a confirmation tone is heard or for about three seconds:

- IP 32x/33x: Menu, Dial, and the two Line keys
- IP 430, 450, 550, 560, 600, 601, 650, 670, and 7000 and VVX 1500: Up, Down, Left, and Right arrow keys
- IP 6000: Menu, Exit, Off-hook/Hands-free, Redial

Default Feature Key Layouts

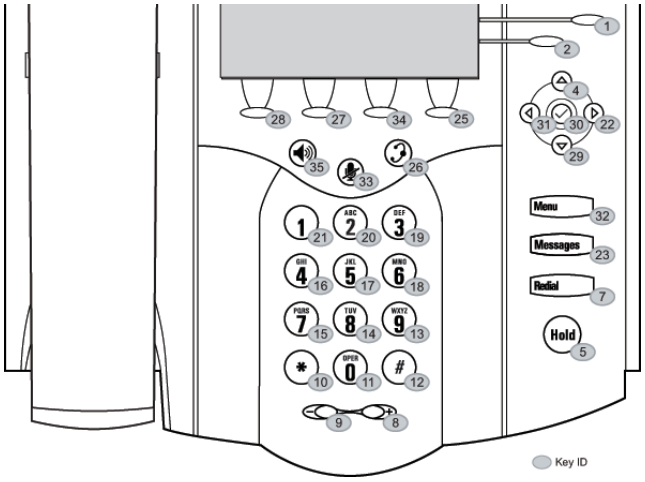
The following figures and tables show the default SIP key layouts for the SoundPoint IP 32x/33x, 430, 450, 550, 560, 650, and 670, SoundStation IP 6000 and 7000, and Polycom VVX 1500 models.

SoundPoint IP 320/321/330/331/335



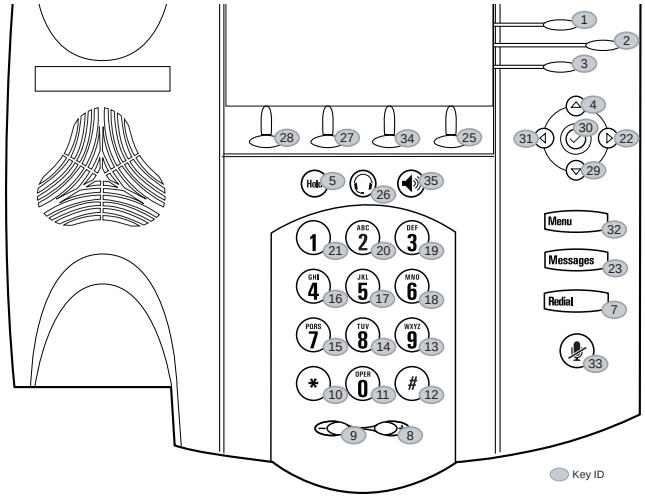
Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Dialpad2	12	n/a	23	VolUp	34	Menu
2	Dialpad5	13	SoftKey2	24	VolDown	35	n/a
3	Dialpad8	14	ArrowUp	25	Dialpad3	36	n/a
4	Dialpad7	15	Select	26	Dialpad6	37	n/a
5	Dialpad4	16	ArrowDown	27	Dialpad9	38	n/a
6	Dialpad1	17	n/a	28	Dialpad0	39	n/a
7	SoftKey3	18	n/a	29	DialpadStar	40	n/a
8	Line1	19	Hold	30	MicMute	41	n/a
9	ArrowRight	20	Headset	31	SoftKey1	42	n/a
10	Line2	21	Handsfree	32	Dial		
11	n/a	22	DialpadPound	33	ArrowLeft		

SoundPoint IP 430



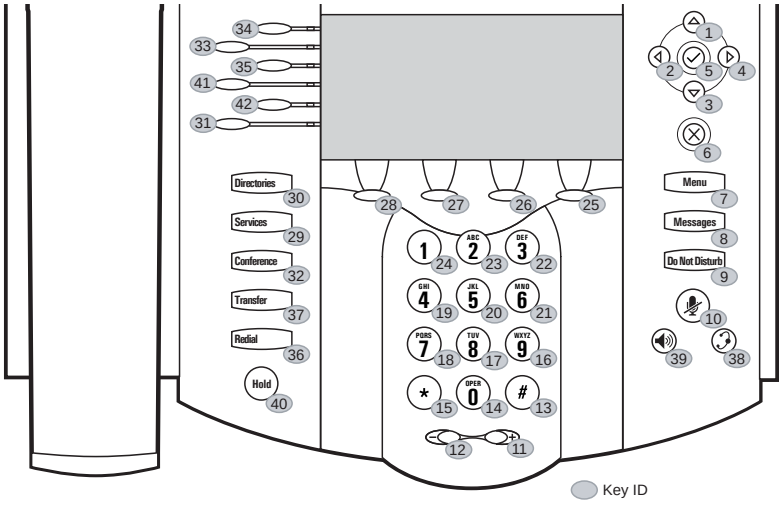
Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Line1	12	DialpadPound	23	Messages	34	Softkey3
2	Line2	13	Dialpad9	24	n/a	35	Handsfree
3	n/a	14	Dialpad8	25	SoftKey4	36	n/a
4	ArrowUp	15	Dialpad7	26	Headset	37	n/a
5	Hold	16	Dialpad4	27	SoftKey2	38	n/a
6	n/a	17	Dialpad5	28	SoftKey1	39	n/a
7	Redial	18	Dialpad6	29	ArrowDown	40	n/a
8	VolUp	19	Dialpad3	30	Select	41	n/a
9	VolDown	20	Dialpad2	31	ArrowLeft	42	n/a
10	DialpadStar	21	Dialpad1	32	Menu		
11	Dialpad0	22	ArrowRight	33	MicMute		

SoundPoint IP 450



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Line1	12	DialpadPound	23	Messages	34	SoftKey3
2	Line2	13	Dialpad9	24	n/a	35	Handsfree
3	Line3	14	Dialpad8	25	Softkey4	36	n/a
4	ArrowUp	15	Dialpad7	26	Headset	37	n/a
5	Hold	16	Dialpad4	27	SoftKey2	38	n/a
6	n/a	17	Dialpad5	28	SoftKey1	39	n/a
7	Redial	18	Dialpad6	29	ArrowDown	40	n/a
8	VolUp	19	Dialpad3	30	Select	41	n/a
9	VolDown	20	Dialpad2	31	ArrowLeft	42	n/a
10	DialpadStar	21	Dialpad1	32	Menu		
11	Dialpad0	22	ArrowRight	33	MicMute		

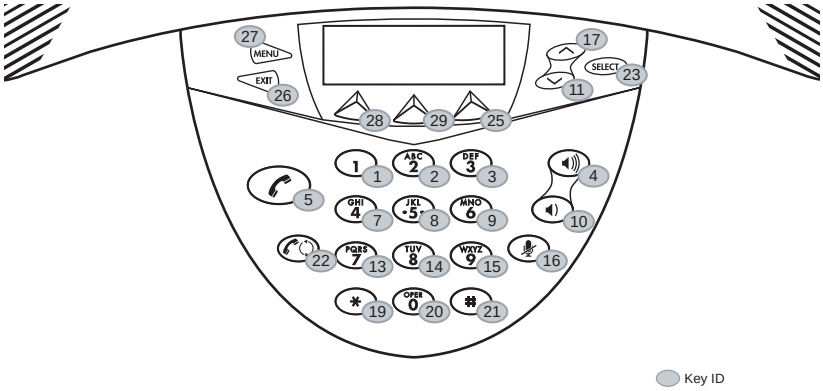
SoundPoint IP 550/560/650/670



Note The SoundPoint IP 550 and 560 has have only the top four lines keys. Key IDs 31 and 42 are not used on SoundPoint IP 550 and 560 phones.

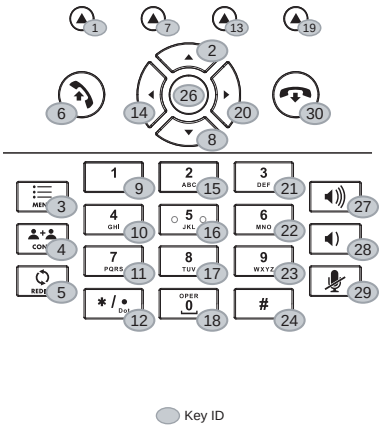
Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	ArrowUp	12	VolDown	23	Dialpad2	34	Line1
2	ArrowLeft	13	DialpadPound	24	Dialpad1	35	Line3
3	ArrowDown	14	Dialpad0	25	SoftKey4	36	Redial
4	ArrowRight	15	DialpadStar	26	SoftKey3	37	Transfer
5	Select	16	Dialpad9	27	SoftKey2	38	Headset
6	Delete	17	Dialpad8	28	SoftKey1	39	Handsfree
7	Menu	18	Dialpad7	29	Applications	40	Hold
8	Messages	19	Dialpad4	30	Directories	41	Line4
9	DoNotDisturb	20	Dialpad5	31	Line6	42	Line5
10	MicMute	21	Dialpad6	32	Conference		
11	VolUp	22	Dialpad3	33	Line2		

SoundStation IP 6000



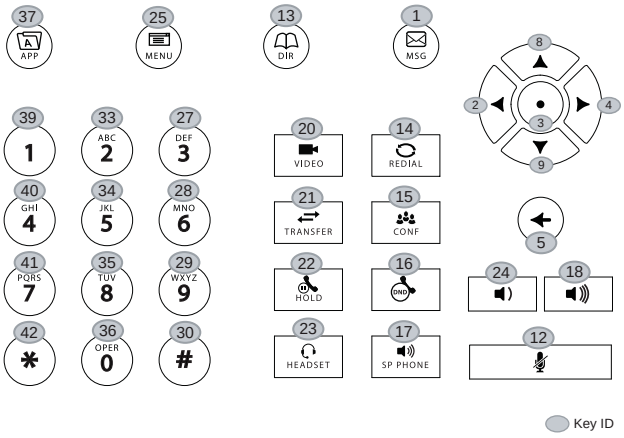
Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Dialpad1	12	n/a	23	Select	34	n/a
2	Dialpad2	13	Dialpad7	24	n/a	35	n/a
3	Dialpad3	14	Dialpad8	25	SoftKey3	36	n/a
4	VolUp	15	Dialpad9	26	Exit	37	n/a
5	Handsfree	16	MicMute	27	Menu	38	n/a
6	n/a	17	ArrowUp	28	SoftKey1	39	n/a
7	Dialpad4	18	n/a	29	SoftKey2	40	n/a
8	Dialpad5	19	DialpadStar	30	n/a	41	n/a
9	Dialpad6	20	Dialpad0	31	n/a	42	n/a
10	VolDown	21	DialpadPound	32	n/a		
11	ArrowDown	22	Redial	33	n/a		

SoundStation IP 7000



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	SoftKey1	12	DialpadStar	23	Dialpad9	34	n/a
2	ArrowUp	13	SoftKey3	24	DialpadPound	35	n/a
3	Menu	14	ArrowLeft	25	n/a	36	n/a
4	Conference	15	Dialpad2	26	Select	37	n/a
5	Redial	16	Dialpad5	27	VolUp	38	n/a
6	Handsfree	17	Dialpad8	28	VolDown	39	n/a
7	SoftKey2	18	Dialpad0	29	MicMute	40	n/a
8	ArrowDown	19	SoftKey4	30	Release	41	n/a
9	Dialpad1	20	ArrowRight	31	n/a	42	n/a
10	Dialpad4	21	Dialpad3	32	n/a		
11	Dialpad7	22	Dialpad6	33	n/a		

Polycom VVX 1500



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	Messages	12	MicMute	23	Headset	34	Dialpad5
2	ArrowLeft	13	Directories	24	n/a	35	Dialpad8
3	Select	14	Redial	25	Menu	36	Dialpad0
4	ArrowRight	15	Conference	26	n/a	37	Applications
5	Delete	16	DoNotDisturb	27	Dialpad3	38	n/a
6	n/a	17	Handsfree	28	Dialpad6	39	Dialpad1
7	n/a	18	VolUp	29	Dialpad9	40	Dialpad4
8	ArrowUp	19	n/a	30	DialpadPound	41	Dialpad7
9	ArrowDown	20	Video	31	n/a	42	DialpadStar
10	n/a	21	Transfer	32	n/a		
11	n/a	22	Hold	33	Dialpad2		

Internal Key Functions

A complete list of internal key functions for enhanced feature keys and hard key mappings is shown in the following table.

The following guidelines should be noted:

- The **Label** value is case sensitive.
- Some functions are dependent on call state. Generally, if the soft key appears on a call screen, the soft key function is executable. There are some exceptions on the SoundPoint IP 32x/33x phone (because it does not display as many soft keys).

- On the SoundPoint IP 32x/33x phone, CallPickup and ParkedPickup refer to the same function. On other phones, CallPickup refers to the soft key function that provides the menu with separate soft keys for parked pickup, directed pickup, and group pickup.
- Some functions depend on the feature being enabled. For example, BuddyStatus and MyStatus require the presence feature to be enabled.
- Hard key remappings do not require the Enhanced Feature key feature to be enabled. This include the SpeedDial function on older platforms. On newer platforms, use line key functions.
- The table below shows only Line1 to Line6 functions. For the SoundPoint IP 650 and 670 phones with attached Expansion Modules, Line7 to Line48 functions are also supported.

Label	Function	Notes
ACDAvailable	ACDAvailableFromIdle	
ACDLogin	ACDLoginLogout	
ACDLogout	ACDLoginLogout	
ACDUnavailable	ACDAvailableFromIdle	
Answer	Answer	Call screen only
Applications	Main Browser	
ArrowDown	ArrowDown	
ArrowLeft	ArrowLeft	
ArrowRight	ArrowRight	
ArrowUp	ArrowUp	
BargeIn	BargInShowAppearances, BargeIn	Call screen only
BuddyStatus	Buddy Status	
Callers	Callers	
CallList	Call Lists	
CallPark	ParkEntry	Call screen only
CallPickup	CallPickupEntry	Call screen only
Conference	ConferenceCall	Call screen only
Delete	Delete	
Dialpad0	Dialpad0	
Dialpad1	Dialpad1	
Dialpad2	Dialpad2	

Label	Function	Notes
Dialpad3	Dialpad3	
Dialpad4	Dialpad4	
Dialpad5	Dialpad5	
Dialpad6	Dialpad6	
Dialpad7	Dialpad7	
Dialpad8	Dialpad8	
Dialpad9	Dialpad9	
DialpadPound	DialpadPound	
DialpadStar	DialpadStar	
DialpadURL	Dialname	Call screen only
DirectedPiclup	DirectedPickup	Call screen only
Directories	Directories	
Divert	Forward	
DoNotDisturb	Do Not Disturb menu	
Exit	Exist existing menu	Menu only
GroupPickup	GroupPickup	
Handsfree	Handsfree	
Headset	Headset	Desktop phones only
Hold	Toggle Hold	
Join	Join	Call screen only
LCR	LastCallReturn	
Line1	Line Key 1	
Line2	Line Key 2	
Line3	Line Key 3	
Line4	Line Key 4	
Line5	Line Key 5	
Line6	Line Key 6	
ListenMode	Turn on speaker to listen only	
Menu	Menu	
Messages	Messages menu	

Label	Function	Notes
MicMute	MicMute	
MyStatus	MyStatus	
NewCall	NewCall	Call screen only
Null	Do nothing	
Offline	Offline for presence	
QuickSetup	Quick Setup feature	Call screen only
EnterRecord	enterCallRecord	Call screen only
Redial	Redial	Call screen only
Release	EndCall or Cancel hot dial	SoundStation IP 7000 only
ParkedPickup	ParkedPickup	Call screen only
Select	Select	
ServerACDAgentAvailable	serverACDAgentAvailable	
ServerACDAgentUnavailable	serverACDAgentUnavailable	
ServerACDSignIn	serverACDSignIn	
ServerACDSignOut	serverACDSignOut	
Setup	Settings menu	
Silence	RingerSilence	Call screen only
SoftKey1	SoftKey1	
SoftKey2	SoftKey2	
SoftKey3	SoftKey3	
SoftKey4	SoftKey4	
SpeedDial	SpeedDial	
Split	Split	Call screen only
Transfer	Transfer	Call screen only
Video	Video	Polycom VVX 1500 only
VolDown	VolDown	
VolUp	VolUp	

Assigning a VLAN ID Using DHCP

To assign a VLAN ID to a phone using DHCP:

>> In the DHCP menu of the Main setup menu, set *VLAN Discovery* to "Fixed" or "Custom".

When set to "Fixed", the phone will examine DHCP options 128,144, 157 and 191 (in that order) for a valid DVD string.

When set to "Custom", the value set in "VLAN ID Option" will be examined for a valid DVD string.

DVD string in the DHCP option must meet the following conditions to be valid:

- Must start with ?VLAN-A=? (case-sensitive)
- Must contain at least one valid ID
- VLAN IDs range from 0 to 4095
- Each VLAN ID must be separated by a ?+? character
- The string must be terminated by a ?;?
- All characters after the ?;? will be ignored
- There must be no white space before the ?;?
- VLAN IDs may be decimal, hex, or octal

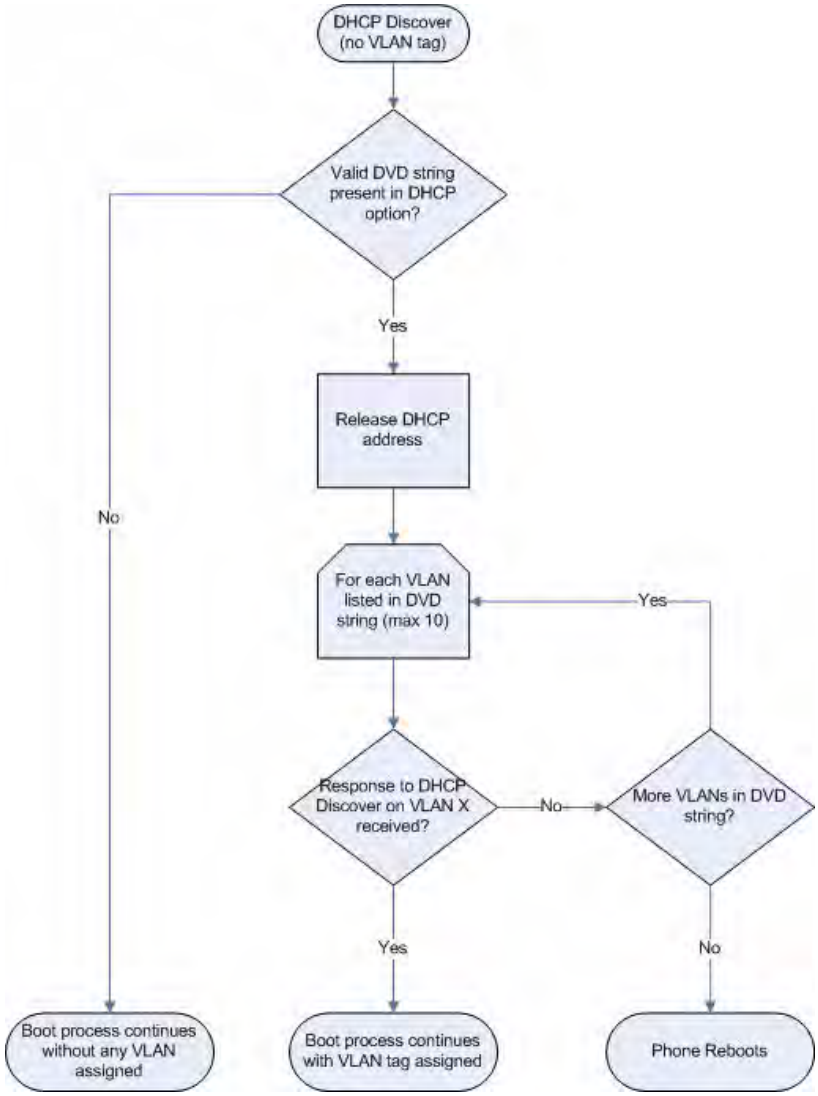
For example:

The following DVD strings will result in the phone using VLAN 10:

```
VLAN-A=10;
VLAN-A=0x0a;
VLAN-A=012;
```

Note If a VLAN tag is assigned by CDP, DHCP VLAN tags will be ignored.

The following figure shows the phone's processing to determine if the VLAN ID is valid:



Parsing Vendor ID Information

After the phone boots, it sends a DHCP Discover packet to the DHCP server. This is found in the Bootstrap Protocol/option "Vendor Class Identifier" section of the packet and includes the phone's part number and the BootROM version. The format of this option's data is not specified in RFC 2132, but is left to each vendor to define its own format. To be useful, every vendor's format must be distinguishable from every other vendor's format. To make our format uniquely identifiable, the format follows RFC 3925, which uses the

IANA Private Enterprise number to determine which vendor's format should be used to decode the remaining data. The private enterprise number assigned to Polycom is 13885 (0x0000363D).

This vendor ID information is not a character string, but an array of binary data. The steps for parsing are as follows:

1. Check for the Polycom signature at the start of the option:
   4 octet: 00 00 36 3d
2. Get the length of the entire list of sub-options:
   1 octet
3. Read the field code and length of the first sub-option, 1+1 octets
4. If this is a field you want to parse, save the data.
5. Skip to the start of the next sub-option.
6. Repeat steps 3 to 5 until you have all the data or you encounter the End-of-Suboptions code (0xFF).

For example, the following is a sample decode of a packet from an IP601:

```
3c 74
  - Option 60, length of Option data (part of the DHCP spec.)
00 00 36 3d
  - Polycom signature (always 4 octects)
6f
  - Length of Polycom data
01 07 50 6f 6c 79 63 6f 6d
  - sub-option 1 (company), length, "Polycom"
02 15 53 6f 75 6e 64 50 6f 69 6e 74 49 50 2d 53 50 49 50 5f 36 30 31
  - sub-option 2 (part), length, "SoundPointIP-SPIP_601"
03 10 32 33 34 35 2d 31 31 36 30 35 2d 30 30 31 2c 32
  - sub-option 3 (part number), length, "2345-11605-001,2"
04 1c 53 49 50 2f 54 69 70 2e 58 58 58 58 2f 30 38 2d 4a 75 6e 2d 30 37
20 31 30 3a 34 34
  - sub-option 4 (Application version), length, "SIP/Tip.XXXX/08-Jun-07
10:44"
05 1d 42 52 2f 33 2e 31 2e 30 2e 58 58 58 58 2f 32 38 2d 41 70 72 2d 30
35 20 31 33 3a 33 30
  - sub-option 5 (BootROM version), length, "BR/3.1.0.XXXX/28-Apr-05
13:30"
ff
  - end of sub-options
```

For the BootROM, sub-option 4 and sub-option 5 will contain the same string. The string is formatted as follows:

```
<apptype>/<buildid>/<date+time>
where:
<apptype> can be 'BR' (BootROM) or 'SIP' (SIP Application)
```

Product, Model, and Part Number Mapping

In SIP 2.1.2, enhancements to the master configuration file were made to allow you to direct phone upgrades to a software image and configuration files based on phone model number, firmware part number, or MAC address.

The part number specific version has precedence over the model number version, which has precedence over the original version. For example, CONFIG_FILES_2345-11560-001="phone1_2345-11560-001.cfg, sip_2345-11560-001.cfg" will override CONFIG_FILES_SPIP560="phone1_SPIP560.cfg, sip_SPIP560.cfg", which will override CONFIG_FILES="phone1.cfg, sip.cfg" for an SoundPoint IP 560.

You can also add variables to the master configuration file that are replaced when the phone reboots. The variables include PHONE_MODEL, PHONE_PART_NUMBER, and PHONE_MAC_ADDRESS.

The following table shows the product name, model name, and part number mapping for SoundPoint IP, SoundStation IP, and Polycom VVX 1500 phones:

Product Name	Model Name	Product Part Number
SoundPoint IP 300	SPIP300	2345-11300-001
SoundPoint IP 301	SPIP301	2345-11300-010
SoundPoint IP 320	SPIP320	2345-12200-002, 2345-12200-005
SoundPoint IP 321	SPIP321	2345-13600-001
SoundPoint IP 330	SPIP330	2345-12200-001, 2345-12200-004
SoundPoint IP 331	SPIP331	2345-12365-001
SoundPoint IP 335	SPIP335	2345-12375-001
SoundPoint IP 430	SPIP430	2345-11402-001
SoundPoint IP 450	SPIP450	2345-12450-001
SoundPoint IP 500	SPIP500	2345-11500-001, 2345-11500-010, 2345-11500-020
SoundPoint IP 501	SPIP501	2345-11500-030, 2345-11500-040
SoundPoint IP 550	SPIP550	2345-12500-001
SoundPoint IP 560	SPIP560	2345-12560-001
SoundPoint IP 600	SPIP600	2345-11600-001
SoundPoint IP 601	SPIP601	2345-11605-001

Product Name	Model Name	Product Part Number
SoundPoint IP 650	SPIP650	2345-12600-001
SoundPoint IP 670	SPIP670	2345-12670-001
SoundStation IP 4000	SSIP4000	2201-06642-001
SoundStation IP 6000	SSIP6000	3111-15600-001
SoundStation IP 7000	SSIP7000	3111-40000-001
Polycom VVX 1500	VVX1500	2345-17960-001

Disabling PC Ethernet Port

Certain SoundPoint IP phones have a PC Ethernet port. If it is unused, it can be disabled.

The PC Ethernet port can be disabled on the SoundPoint IP 33x, 430, 450, 550, 560, 601, 650, and 670, and Polycom VVX 1500 through the menu (shown below). The Ethernet port can also be disabled through the configuration files.

To disable the Ethernet port on a supported SoundPoint IP phone:

1. Press Menu.
2. Select **Settings > Advanced > Network Configuration > Ethernet Menu**.

   You must enter the administrator password to access the network configuration. The factory default password is 456.
3. Scroll down to PC Port Mode and select **Edit**.
4. Select **Disabled,** and then press the **OK** soft key.
5. Press the **Exit** soft key.
6. Select **Save Config**.

   The SoundPoint IP phone reboots. When the reboot is complete, the PC Ethernet port is disabled.

Modifying Phone's Configuration Using the Web Interface

You can make changes to the configuration files through the web interface to the phone.

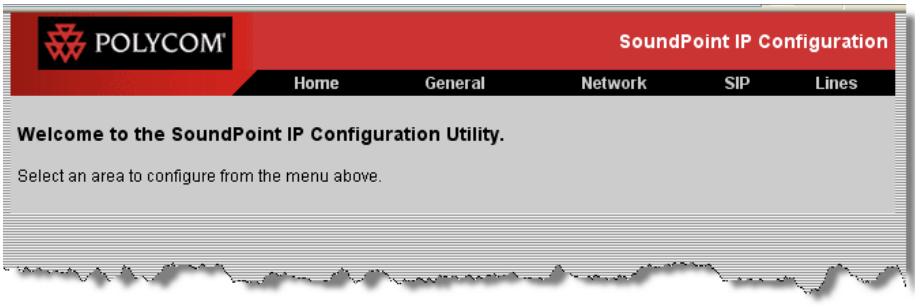
To configure your phone through the web interface:

>> Using your chosen browser, do the following:

a To get your phone's IP address, press the **Menu** key, and then selecting **Status > Platform > Phone**. Scroll down to see the IP address.

b Enter your phone's IP address as the browser address.

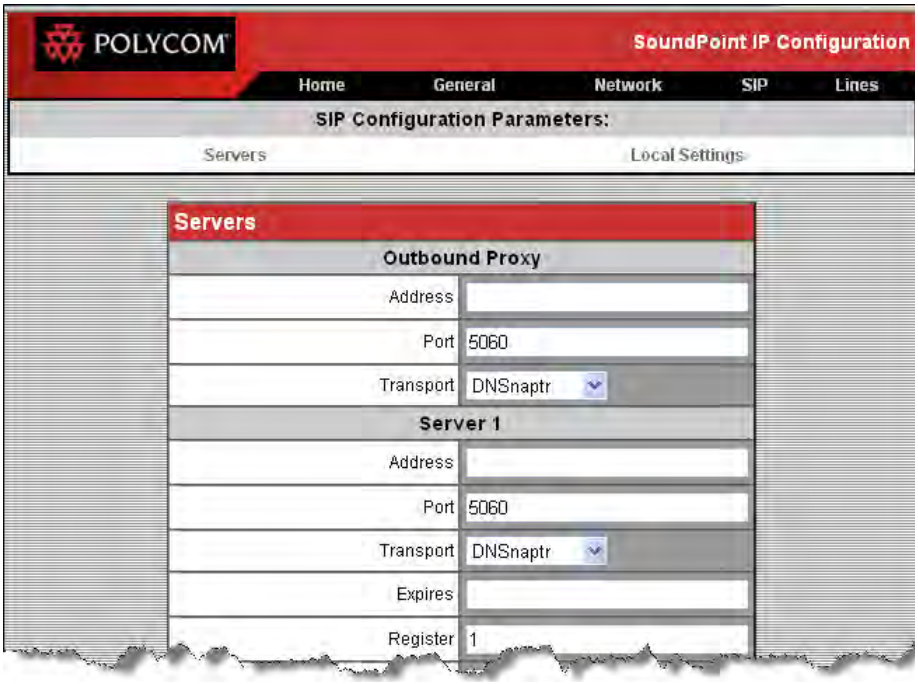
A web page similar to the one shown below appears.



c Select **SIP** from the menu tab.

You will be prompted for the SIP username and password.

A web page similar to the one shown below appears.

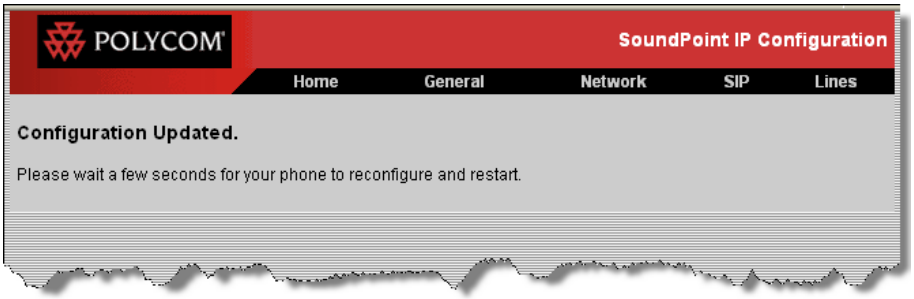


d Make the desired configuration changes.

e Scroll down to the bottom of the **Servers** section.

f Select the **Submit** button.

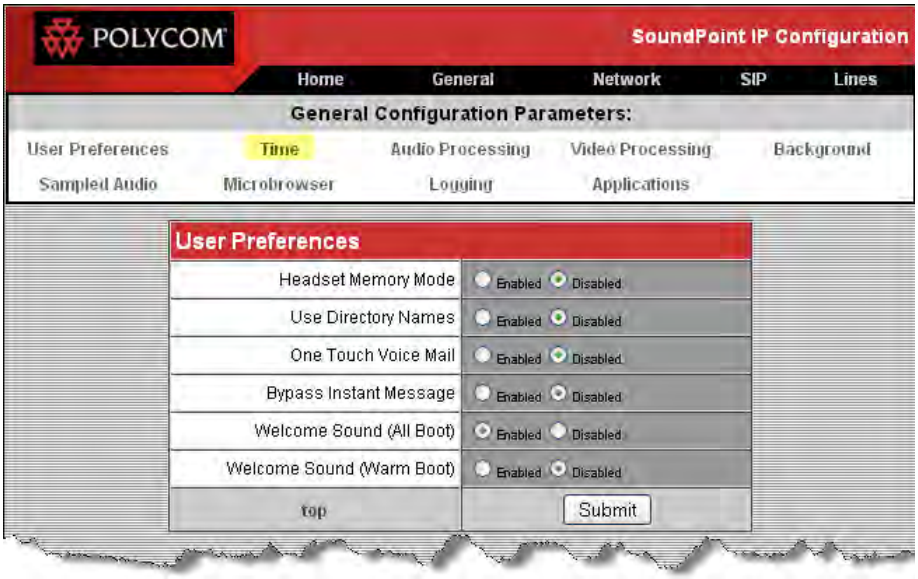
A web page similar to the one shown below appears.



Your phone will reboot.

g Select **General** from the menu tab.

A web page similar to the one shown below appears.



h If you make any changes, scroll down to the bottom of the section.

i Select the **Submit** button.

Your phone will reboot.

Capturing Phone's Current Screen

You can capture the current screen on a SoundPoint IP, SoundStation IP or Polycom VVX phone through the web interface to the phone.

To capture the phone's current screen:

1. Modify the **sip.cfg** configuration file as follows:
   a. Open **sip.cfg** in an XML editor.
   b. Locate the user preference parameter.
   c. Change `up.screenCapture.enabled` to **1**.
   d. Save the modified **sip.cfg** configuration file.
2. On the phone, do the following:
   a. Press the **Menu** key, and then select **Settings > Basic > Preferences > Screen Capture**.
   b. Using the arrow keys, select **Enabled**, and then press the **Select** soft key.

Note You need to re-enable the Screen Capture feature after every phone restart or reboot (repeat step 2).

1. Using your chosen browser, do the following:
   — To get your phone's IP address, press the **Menu** key, and then select **Status > Platform > Phone**. Scroll down to see the IP address.
   — As the browser address, enter **`http://<phone's IP address>/captureScreen .`**

   The current screen that is shown on the phone is shown in the browser window. The image can be saved as a **BMP** or **JPEG** file.

LLDP and Supported TLVs

The Link Layer Discovery Protocol (LLDP) is a vendor-neutral Layer 2 protocol that allows a network device to advertise its identity and capabilities on the local network. The protocol was formally ratified as IEEE standard 802.1AB- 2005 in May 2005. Refer to section 10.2.4.4 of the LLDP-MED standard at http://www.tiaonline.org/standards/technology/voip/documents/ANSI-TIA-1057_final_for_publication.pdf .

The LLDP feature (added in SIP 3.2.0) supports VLAN discovery and LLDP power management, but not power negotiation. LLDP has a higher priority than CDP and DHCP VLAN discovery.

The following Type Length Values (TLVs) are supported:

- Mandatory
  - Chassis ID—Must be first TLV
  - Port ID—Must be second TLV
  - Time-to-live—Must be third TLV, set to 120 seconds
  - End-of-LLDPDU—Must be last TLV
  - LLDP-MED Capabilities
  - LLDP-MED Network Policy—VLAN, L2 QoS, L3 QoS
  - LLDP-MED Extended Power-Via-MDI TLV—Power Type, Power Source, Power Priority, Power Value
- Optional
  - Port Description
  - System Name—Administrator assigned name
  - System Description—Includes device type, phone number, hardware version, and software version
  - System Capabilities—Set as "Telephone" capability
  - MAC / PHY config status—Detects duplex mismatch
  - Management Address—Used for network discovery
  - LLDP-MED Location Identification—Location data formats: Co-ordinate, Civic Address, ECS ELIN
  - LLDP-MED Inventory Management —Hardware Revision, Firmware Revision, Software Revision, Serial Number, Manufacturer's Name, Model Name, Asset ID

An LLDP frame shall contain all mandatory TLVs. The frame will be recognized as LLDP only if it contains mandatory TLVs. SoundPoint IP / SoundStation IP / VVX phones will support LLDP frames with both mandatory and optional TLVs. The basic structure of an LLDP frame and a table containing all TLVs along with each field is explained in Supported TLVs on page C-32.

Note

As per section 10.2.4.4 of the LLDP-MED standard, LLDP-MED endpoint devices need to transmit Location Identification TLVs if they are capable of either automatically determining their physical location by use of GPS or radio beacon or capable of being statically configured with this information.

At present, the SoundPoint IP / SoundStation IP / VVX phones do not have the capability to determine their physical location automatically or provision to a statically configured location. Because of these limitations, the SoundPoint IP / SoundStation IP / VVX phones will not transmit Location Identification TLV in the LLDP frame. However, the location information from the switch is decoded and displayed on the phone's menu.

For more information on configuration parameters, refer to Flash Parameter Configuration on page A-157.

Supported TLVs

This is the basic TLV format:

TLV Type (7 bits) [0-6]	TLV Length (9 bits) [7-15]	TLV Information (0-511 bytes)

The following is a list of supported TLVs:

No	Name	Type (7 bits) [0-6]	Length (9 bits) [7-15]	Type Length	Org. Unique Code (3 bytes)	Version	
						Sub-Type	Information
1	Chassis-Id[1]	1	6	0x0206	-	5	IP address of phone (4 bytes) ***Note**: 0.0.0.0 is sent until the phone has a valid IP address.*
2	Port-Id[1]	2	7	0x0407	-	3	MAC address of phone (6 bytes)
3	TTL	3	2	0x0602	-	-	TTL value is 120/0 sec
4	Port description	4	1	0x0801	-	-	Port description 1
5	System name	5	min len > 0, max len <= 255	-	-	-	Refer to System Names on page C-37

No	Name	Type (7 bits) [0-6]	Length (9 bits) [7-15]	Type Length	Org. Unique Code (3 bytes)	Version	
						Sub-Type	Information
6	System description	6	min len > 0, max len <= 255	-	-	-	Manufacturer's name - "Polycom"; Refer to Model Names on page C-37; Hardware version; Application version; BootROM version
7	Capabilities	7	4	0x0e04	-	-	System Capabilities: Telephone and Bridge if the phone has PC port support and it is not disabled. Enabled Capabilities: Telephone and Bridge if phone has PC port support, it is not disabled and PC port is connected to PC. ***Note**:* *PC port supported Phones: IP 330, IP 331, IP 335, IP 430, IP 450, IP 550, IP 560, IP 650, and IP 670.* *PC port not supported phones:* *IP6000, IP7000, IP320, and IP321*.
8	Management Address	8	12	0x100c	-	-	Address String Len - 5, IPV4 subtype, IP address, Interface subtype - "Unknown", Interface number - "0", ODI string Len - "0"
9	IEEE 802.3 MAC/PHY config/status[1]	127	9	0xfe09	0x00120f	1	Auto Negotiation Supported - "1", enabled/disabled, Refer to PMD Advertise and Operational MAU on page C-38

No	Name	Type (7 bits) [0-6]	Length (9 bits) [7-15]	Type Length	Org. Unique Code (3 bytes)	Version	
						Sub-Type	Information
10	LLDP-MED capabilities	127	7	0xfe07	0x0012bb	1	Capabilities - 0x33 (LLDP-Med capabilities, Network policy, Extended Power Via MDI-PD, Inventory) Class Type III ***Note**: Once support for configuring location Identification information is locally available:* *Capabilities - 0x37 (LLDP-Med capabilities, Network policy, Location Identification, Extended Power Via MDI-PD, Inventory)* *Class Type III*
11	LLDP-MED network policy[2]	127	8	0xfe08	0x0012bb	2	ApplicationType: Voice (1), Policy: (Unknown(=1)/Defined(=0) Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Networkpolicy TLV is received from the switch.), Tagged/Untagged, VlanId, L2 priority and DSCP

No	Name	Type (7 bits) [0-6]	Length (9 bits) [7-15]	Type Length	Org. Unique Code (3 bytes)	Version	
						Sub-Type	Information
12	LLDP-MED network policy[2]	127	8	0xfe08	0x0012bb	2	ApplicationType: Voice Signaling (2), Policy: (Unknown(=1)/Defined(=0) Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Networkpolicy TLV is received from the switch.),Tagged/Untagged, VlanId, L2 priority and DSCP. ***Note**: Voice signaling TLV is sent only if it contains configuration parameters that are different from voice parameters.*
13	LLDP-MED network policy[2]	127	8	0xfe08	0x0012bb	2	ApplicationType: Video Conferencing (6),Policy: (Unknown(=1)/Defined(=0) Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Networkpolicy TLV is received from the switch.),Tagged/Untagged, VlanId, L2 priority and DSCP. ***Note**: Video Conferencing TLV is sent only from Video capable phones (currently Polycom VVX 1500 only)*.
14	LLDP-MED location identification[3]	127	min len > 0, max len <= 511	-	0x0012bb	3	ELIN data format: 10 digit emergency number configured on the switch. Civic Address: physical address data such as city, street number, and building information.

No	Name	Type (7 bits) [0-6]	Length (9 bits) [7-15]	Type Length	Org. Unique Code (3 bytes)	Version	
						Sub-Type	Information
15	Extended power via MDI	127	7	0xfe07	0x0012bb	4	PowerType -PD device PowerSource-PSE&local Power Priority -Unknown PowerValue - Refer to Power Values on page C-39
16	LLDP-MED inventory hardware revision	127	min len > 0, max len <= 32	-	0x0012bb	5	Hardware part number and revision
17	LLDP-MED inventory firmware revision	127	min len > 0, max len <= 32	-	0x0012bb	6	BootROM revision
18	LLDP-MED inventory software revision	127	min len > 0, max len <= 32	-	0x0012bb	7	Application (SIP) revision
19	LLDP-MED inventory serial number	127	min len > 0, max len <= 32	-	0x0012bb	8	MAC Address (ASCII string)
20	LLDP-MED inventory manufacturer name	127	11	0xfe0b	0x0012bb	9	Polycom
21	LLDP-MED inventory model name	127	min len > 0, max len <= 32	-	0x0012bb	10	Refer to Model Names on page C-37
22	LLDP-MED inventory asset ID	127	4	0xfe08	0x0012bb	11	Empty (Zero length string)
23	End of LLDP DU	0	0	0x0000	-	-	-

Note

1. For other subtypes, refer to IEEE 802.1AB, March 2005 at http://www.ieee802.org/1/pages/802.1ab.html .
2. For other application types, refer to TIA Standards 1057, April 2006 at http://tia.nufu.eu/std/ANSI|TIA-1057 .
3. At this time, this TLV is not sent by the phone.

System Names

Model	System Name
IP 320	Polycom SoundPoint IP 320
IP 321	Polycom SoundPoint IP 321
IP 330	Polycom SoundPoint IP 330
IP 331	Polycom SoundPoint IP 331
IP 335	Polycom SoundPoint IP 335
IP 430	Polycom SoundPoint IP 430
IP 450	Polycom SoundPoint IP 450
IP 550	Polycom SoundPoint IP 550
IP 560	Polycom SoundPoint IP 560
IP 650	Polycom SoundPoint IP 650
IP 670	Polycom SoundPoint IP 670
IP 6000	Polycom SoundStation IP 6000
IP 7000	Polycom SoundStation IP 7000
VVX 1500	Polycom VVX 1500

Model Names

Model	Model Name
IP 320	SoundPointIP-SPIP_320
IP 321	SoundPointIP-SPIP_321
IP 330	SoundPointIP-SPIP_330
IP 331	SoundPointIP-SPIP_331
IP 335	SoundPointIP-SPIP_335
IP 430	SoundPointIP-SPIP_430

Model	Model Name
IP 450	SoundPointIP-SPIP_450
IP 550	SoundPointIP-SPIP_550
IP 560	SoundPointIP-SPIP_560
IP 650	SoundPointIP-SPIP_650
IP 670	SoundPointIP-SPIP_670
IP 6000	SoundStationIP-SSIP_6000
IP 7000	SoundStationIP-SSIP_7000
VVX 1500	VVX-VVX_1500

PMD Advertise and Operational MAU

Mode/Speed	PMD Advertise Capability Bit	Operational MAU Type
10BASE-T half duplex mode	1	10
10BASE-T full duplex mode	2	11
100BASE-T half duplex mode	4	15
100BASE-T full duplex mode	5	16
1000BASE-T half duplex mode	14	29
1000BASE-T full duplex mode	15	30
Unknown	0	0

Note By default, all phones have the PMD Advertise Capability set for 10HD, 10FD, 100HD and 100FD bits. For SoundPoint IP 560 and IP 670, and Polycom VVX 1500 phones that have Gigabit Ethernet support PMD Advertise Capability also contains set 1000FD bit.

Power Values

Model	Power Usage (Watts)	Power Value Sent in LLDP-MED Extended Power Via MDI TLV
IP 320/330	4.5	45
IP 321/331	4.5	45
IP 335	5.5	55
IP 430	4.5	45
IP 450	4.5	45
IP 550	6	60
IP 560	8	80
IP 650 with EM	12	120
IP 670 with EM	14	140
IP 6000	10.5	105
IP 7000	10.5	105
VVX 1500	14	140

Note By default, the power values for the SoundPoint IP 650 and 670 are sent for the phone and the Expansion Module(s). The values are not adjusted when the Expansion Module(s) are detached from the phone.

D

Third Party Software

This appendix provides the copyright statements for third party software products that are part of the application programs that run on Polycom SoundPoint IP, SoundStation IP, and VVX 1500 phones.

Product	License Location
c-ares	c-ares on page D-2
curl	curl on page D-3
eXpat	eXpat on page D-9
ILG JPEG	IJG JPEG on page D-9
libMng	libMng on page D-10
libPng	libPng on page D-11
libSRTP	libSRTP on page D-13
libssh2	libssh2 on page D-13
OpenLDAP	OpenLDAP on page D-14
OpenSSL	OpenSSL on page D-15
zlib	zlib on page D-18

This appendix provides the copyright statements for third party software products that are part of the application programs that run on Polycom VVX 1500 phones only.

Product	License Location
BusyBox	Refer to the “Polycom Voice OFFER of Source for GPL and LGPL Software”
dhcp	dhcp 4.0.0-14 on page D-3
droidfonts	droidfonts on page D-5
Dropbear	Dropbear on page D-4

Product	License Location
glibc	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
libstdc++	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
Linux kernel	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
module-init-tools	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
mtd-utils	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
ncurses	ncurses on page D-14
pmap	pmap-29092002 on page D-17
procps	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
tsattach	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
tslib	Refer to the "Polycom Voice OFFER of Source for GPL and LGPL Software"
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c-ares

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curl

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dhcp 4.0.0-14

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Dropbear

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=====

sshpty.c is taken from OpenSSH 3.5p1,

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=====

loginrec.c

loginrec.h

atomicio.h

atomicio.c

and strlcat() (included in util.c) are from OpenSSH 3.6.1p2, and are licensed under the 2 point BSD license.

loginrec is written primarily by Andre Lucas, atomicio.c by Theo de Raadt.

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jpegsrc.vN.tar.gz contains source code, documentation, and test files for release N in Unix format.

jpegsrN.zip contains source code, documentation, and test files for release N in Windows format.

jpegaltui.vN.tar.gz contains source code for an alternate user interface for cjpeg/djpeg in Unix format.

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wallace.ps.gz is a PostScript file of Greg Wallace's introductory article about JPEG. This is an update of the article that appeared in the April 1991 Communications of the ACM.

jpeg.documents.gz tells where to obtain the JPEG standard and documents about JPEG-related file formats.

jfif.ps.gz is a PostScript file of the JFIF (JPEG File Interchange Format) format specification.

jfif.txt.gz is a plain text transcription of the JFIF specification; it's missing a figure, so use the PostScript version if you can.

TIFFTechNote2.txt.gz is a draft of the proposed revisions to TIFF 6.0's JPEG support.

pm.errata.gz is the errata list for the first printing of the textbook "JPEG Still Image Data Compression Standard" by Pennebaker and Mitchell.

jdosaobj.zip contains pre-assembled object files for JMEMDOSA.ASM.

If you want to compile the IJG code for MS-DOS, but don't have an assembler, these files may be helpful.

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zlib

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Addendum to Administrator's Guide for SIP 3.2.3

This addendum addresses changes to the *Administrator's Guide for the SoundPoint IP / SoundStation IP / VVX Family* specific to the release of the Polycom® SoundStation IP® 5000 conference phone, which runs SIP 3.2.3 or later.

The SoundPoint IP 5000 conference phone is a next-generation phone with a modern industrial design and the latest advancements in Polycom voice technology. The SoundStation IP 5000 is similar to the SoundStation IP 6000 except for:

- Support for four soft keys
- Smaller footprint
- Support for six audio codecs

This addendum includes the following topics:

- Distribution Zip File
- Corporate Directory
- Single Registration
- Audio Codecs

Configuration file changes are described in Configuration File Changes on page 1-3.

The default SIP key layout for the SoundPoint IP 5000 conference phone is shown in Multiple Key Combinations and Default Key Layout on page 1-6.

For more information, refer to the Release Notes for the SIP Application, Version 3.2.3 .

Note For information on how to use the SoundPoint IP 5000 conference phone, refer to the *User Guide* at http://www.polycom.com/support/soundstation_ip5000 .

New or Changed Features

Distribution Zip File

As well as the **sip.ld** file in the archive, there is a separate file for the SoundPoint IP 5000 conference phone called **3111-30900-001.sip.ld** .

Corporate Directory

The SoundStation IP 5000 can be configured to interface with a corporate directory that supports the Lightweight Directory Access Protocol (LDAP) version 3. A license key is required and can purchased through a Certified Polycom Reseller.

Single Registration

The SoundStation IP 5000 phone supports a single registration.

When the phone is unable to register with the call control server, the icon ☎ is shown (outline). Once the phone is registered, the icon ☎ is shown (solid).

Audio Codecs

The following table summarizes the SoundStation IP 5000 phone's audio codec support:

Algorithm	MIME Type	Ref.	Bit Rate	Sample Rate	Frame Size	Effective audio bandwidth
G.711μ-law	PMCU	RFC 1890	64 Kbps	8 Ksps	10ms - 80ms	3.5KHz
G.711a-law	PCMA	RFC 1890	64 Kbps	8 Ksps	10ms - 80ms	3.5KHz
G.722	G722/8000	RFC 1890	64 Kbps	16 Ksps	10ms - 80ms	7 KHz
G.729AB	G729	RFC 1890	8 Kbps	8 Ksps	10ms - 80ms	3.5KHz
iLBC	iLBC	RFC 3951	13.33Kbps 15.2Kbps	8 Ksps	30ms - 60ms 20ms - 80ms	3.5KHz

Note The network bandwidth necessary to send the encoded voice is typically 5-10% higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48kbps consumes 5xkbps of network bandwidth (one-way audio). Two-way audio would take over 100kbps.

Configuration changes can performed centrally at the provisioning server or locally:

Central (provisioning server)	Configuration file: **sip.cfg**	Specify codec priority, preferred payload sizes, and jitter buffer tuning parameters. • For more information, refer to Codec Preferences <codecPref/> on page 1-3.

Configuration File Changes

The following **sip.cfg** configuration file changes were made to support the SoundStation IP 5000 conference phone:

- Voice Coding Algorithms <codecs/>
- Gains <gain/>
- Acoustic Echo Suppression <aes/>
- Background Noise Suppression <ns/>
- Receive Equalization <rxEq/>
- Transmit Equalization <txEq/>

Voice Coding Algorithms <codecs/>

The codecs include:

- Codec Preferences <codecPref/>

Codec Preferences <codecPref/>

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_5000.G711Mu	Null, 1-6	2	Specifies the codec preferences for the SoundStation IP 5000 platform. Interpretation as above.
voice.codecPref.IP_5000.G711A		3	
voice.codecPref.IP_5000.G722		1	
voice.codecPref.IP_5000.G729AB		4	
voice.codecPref.IP_5000. iLBC.13_33kbps		Null	
voice.codecPref.IP_5000. iLBC.15_2kbps		Null	

Gains <gain/>

The default gain settings have been carefully adjusted to comply with the TIA-810-A digital telephony standard.

Polycom recommends that you do not change these values.

Attribute	Default
voice.gain.rx.analog.chassis.IP_5000	0
voice.gain.rx.analog.ringer.IP_5000	0
voice.gain.rx.digital.chassis.IP_5000	11
voice.gain.rx.digital.ringer.IP_5000	-12
voice.gain.tx.analog.chassis.IP_5000	0
voice.gain.tx.digital.chassis.IP_5000	15

Acoustic Echo Suppression <aes/>

These settings control the performance of the speakerphone acoustic echo suppressor.

Polycom recommends that you do not change these values.

Attribute	Default
voice.aes.hs.enable	1
voice.aes.hf.duplexBalance.IP_5000.0	10
voice.aes.hf.duplexBalance.IP_5000.1	9
voice.aes.hf.duplexBalance.IP_5000.2	8
voice.aes.hf.duplexBalance.IP_5000.3	7
voice.aes.hf.duplexBalance.IP_5000.4	6
voice.aes.hf.duplexBalance.IP_5000.5	5

Attribute	Default
voice.aes.hf.duplexBalance.IP_5000.6	4
voice.aes.hf.duplexBalance.IP_5000.7	3
voice.aes.hf.duplexBalance.IP_5000.8	2

Background Noise Suppression <ns/>

These settings control the performance of the transmit background noise suppression feature.

Polycom recommends that you do not change these values.

Attribute	Default
voice.ns.hs.enable	1
voice.ns.hf.IP_5000.enable	1
voice.ns.hf.IP_5000.signalAttn	-6
voice.ns.hf.IP_5000.silenceAttn	-9

Receive Equalization <rxEq/>

These settings control the performance of the receive equalization feature.

Polycom recommends that you do not change these values.

Attribute	Default
voice.rxEq.hf.IP_5000.preFilter.enable	1
voice.rxEq.hf.IP_5000.postFilter.enable	0

Transmit Equalization <txEq/>

These settings control the performance of the hands-free transmit equalization feature.

Polycom recommends that you do not change these values.

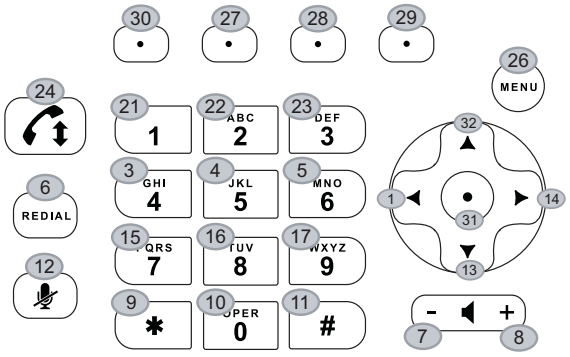
Attribute	Default
voice.txEq.hf.IP_5000.preFilter.enable	0
voice.txEq.hf.IP_5000.postFilter.enable	1

Multiple Key Combinations and Default Key Layout

The multiple key combinations on the SoundStation IP 5000 conference phone are as follows:

- Rebooting the Phone—Press and hold *, #, Volume-, and Volume+ simultaneously until a confirmation tone is heard or for about three seconds.
- Restoring Factory Defaults—Press and hold 1, 3, 5, and 7 dial pad keys simultaneously during the countdown process in the bootROM until the password prompt appears. Enter the administrator password to initiate the reset. Resetting to factory defaults will also reset the administrator password (factory default password is 456).
- Uploading Log Files—Press and hold Up, Down, Left, and Right arrow keys simultaneously until a confirmation tone is heard or for about three seconds.

The SoundStation IP 5000 conference phone default key layout is as follows:



Key ID	Function	Key ID	Function	Key ID	Function	Key ID	Function
1	ArrowLeft	12	MicMute	23	Dialpad3	34	n/a
2	n/a	13	ArrowDown	24	Handsfree	35	n/a
3	Dialpad4	14	ArrowRight	25	n/a	36	n/a
4	Dialpad5	15	Dialpad7	26	Menu	37	n/a
5	Dialpad6	16	Dialpad8	27	SoftKey2	38	n/a
6	Redial	17	Dialpad9	28	SoftKey3	39	n/a
7	VolDown	18	n/a	29	SoftKey4	40	n/a
8	VolUp	19	n/a	30	SoftKey1	41	n/a
9	DialpadStar	20	n/a	31	Select	42	n/a
10	Dialpad0	21	Dialpad1	32	ArrowUp		
11	DialpadPound	22	Dialpad2	33	n/a		

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