ADMINISTRATOR GUIDE

5.4.1 | December 2015 | 3725-49104-009A

# Polycom® UC Software 5.4.1

6001 America Center Drive
San Jose, CA 95002
USA

**Disclaimer** While Polycom uses reasonable efforts to include accurate and up-to-date information in this document, Polycom makes no warranties or representations as to its accuracy. Polycom assumes no liability or responsibility for any typographical or other errors or omissions in the content of this document.

**Limitation of Liability** Polycom and/or its respective suppliers make no representations about the suitability of the information contained in this document for any purpose. Information is provided "as is" without warranty of any kind and is subject to change without notice. The entire risk arising out of its use remains with the recipient. In no event shall Polycom and/or its respective suppliers be liable for any direct, consequential, incidental, special, punitive or other damages whatsoever (including without limitation, damages for loss of business profits, business interruption, or loss of business information), even if Polycom has been advised of the possibility of such damages.

**End User License Agreement** By installing, copying, or otherwise using this product, you acknowledge that you have read, understand and agree to be bound by the terms and conditions of the End User License Agreement for this product. The EULA for this product is available on the Polycom Support page for the product.

**Patent Information** The accompanying product may be protected by one or more U.S. and foreign patents and/or pending patent applications held by Polycom, Inc.

**Open Source Software Used in this Product** This product may contain open source software. You may receive the open source software from Polycom up to three (3) years after the distribution date of the applicable product or software at a charge not greater than the cost to Polycom of shipping or distributing the software to you. To receive software information, as well as the open source software code used in this product, contact Polycom by email at OpenSourceVideo@polycom.com.

**Customer Feedback** We are striving to improve our documentation quality and we appreciate your feedback. Email your opinions and comments to DocumentationFeedback@polycom.com.

**Polycom Support** Visit the Polycom Support Center for End User License Agreements, software downloads, product documents, product licenses, troubleshooting tips, service requests, and more.

# Contents

# Figures

# Tables

# Before You Begin

The Polycom® Unified Communications (UC) Software Administrator Guide provides general guidance on installing, provisioning, and managing Polycom phones. This guide helps you understand the Polycom Voice over Internet Protocol (VoIP) network, and helps you:

- Set up a large-scale provisioning environment
- Set up a VoIP network and provisioning server
- Configure phone features and user settings
- Troubleshoot, update, and maintain phones and UC Software

This Polycom UC Software Administrator Guide applies to the following Polycom devices except where noted:

- Polycom VVX business media phones, including the VVX 101, 201, 300 series (300/301/310/311), 400 series (400/401/410/411), 500 series (500/501) 600 series (600/601), and 1500
- Polycom Expansion Modules
- Polycom SoundStructure® VoIP Interface
- UC Software versions 5.3.0 and later do not support use of the VVX 1500 business media phone with Microsoft Skype for Business Server

## Audience and Purpose of This Guide

The primary audience for this guide is the person administering the session initiation protocol (SIP) server, provisioning servers, VoIP network, and Polycom UC Software that enable you to configure and manage phone features. This guide is not intended for end users. This guide provides information primarily for mid-level administrators with experience in networking who understand the basics of open SIP networks and VoIP endpoint environments. This guide indicates where information might be useful for novice administrators, and provides tips for advanced administrators where applicable.

Before reading this guide, you should be familiar with the following:

- Computer networking and driver administration for your operating system
- SIP networks
- VoIP environments and technologies
- An XML editor

In addition, this administrator guide provides guidance on the following Polycom-specific skills:

- Polycom provisioning methods
- Polycom UC Software and XML configuration files
- Configuration parameters and values for end-user device features

- Troubleshooting your Polycom devices
- Maintaining and updating devices and software

**Web Info: Latest Polycom UC Software**

To find out what's new for this release of UC Software, including enhanced features, and known and resolved issues, see the release notes at Latest Polycom UC Software Release.

# Phone Deployments

Because phone deployments vary, and administrators typically set up and maintain large-scale device deployments, Polycom cannot recommend a specific deployment scenario. For large-scale deployments, Polycom recommends setting up a provisioning server on the local area network (LAN) or on the Internet. For this reason, this administrator guide focuses on large-scale UC Software VoIP environments set up on a central SIP and provisioning server. Administrators typically use the administrator guide in three large-scale device deployment scenarios:

- **Enterprise deployment** An administrator sets up and maintains a deployment for a single organization and all users are in one physical location.
- **Multisite enterprise** An administrator sets up and maintains a deployment for an organization and users are spread out over several locations varying in size.
- **Service Provider Deployment** Service providers provide devices and service to a number of organizations and users spread out over several locations each varying in size.

## What You Need

You require the following to operate Polycom phones as SIP endpoints in large-scale deployments:

- A working IP network
- Routers configured for VoIP
- VoIP gateways configured for SIP
- The latest (or a compatible version) Polycom UC Software image
- An active, configured call server to receive and send SIP messages. For information on IP PBX and softswitch vendors, see Polycom ARENA - VoIP Interoperability Partner Matrix. With the call server you need:
  - ➢ A call server address that registers voice endpoints with the SIP server
  - ➢ SIP authentication user name and password the phone uses to respond to any SIP authentication challenges from the SIP server.
- An XML editor—such as XML Notepad—to create and edit configuration files

# Get Help

For more information about installing, configuring, and administering Polycom products, refer to Documents and Downloads at Polycom Support.

To access the latest Polycom UC Software Release Notes, refer to Polycom Voice Support.

To access the user guide for Polycom VVX business media phones, refer to the product support page for your phone at Polycom Voice Support.

Some Polycom products contain open source software. For details, refer to Polycom Voice Support.

To find help or technical support for your phones, you can search for Polycom documentation at the Polycom Unified Communications (UC) Software Resource Center.

You can find Request for Comments (RFC) documents by entering the RFC number at *http://www.ietf.org/rfc.html.*

For other references, look for the Web Info icon throughout this administrator guide.

## The Polycom Community

The Polycom Community gives you access to the latest developer and support information and enables you to participate in discussion forums to share ideas and solve problems with your colleagues. To register with the Polycom Community, create a Polycom online account. When logged in, you can access Polycom support personnel and participate in developer and support forums to find the latest information on hardware, software, and partner solutions topics.

Polycom | Support

For support or service, please contact your Polycom reseller or visit support.polycom.com for software downloads, product documents, product licenses, troubleshooting tips, service requests, and more.

We are constantly working to improve the quality of our documentation, and we would appreciate your feedback. Please send email to VoiceDocumentationFeedback@polycom.com.
Polycom recommends that you record the phone model numbers, software versions (for both the Updater and UC Software), and partner platform for future reference.

Phone models:

Updater version:

UC Software version:

Partner Platform:

# Polycom UC Software Provisioning Overview

This section provides a high-level overview of Polycom Unified Communications (UC) Software and the major tasks required to provision Polycom phones with UC Software. Because provisioning environments vary, Polycom cannot recommend a specific environment. However, Administrators typically set up and maintain large-scale device deployments and this administrator guide focuses on large-scale UC Software VoIP environments.

## What Is Polycom UC Software?

The Polycom phone software comprises four components:

- **Updater** The software that loads first when the phone is powered on. For information about the Updater, see the section About the Updater.
- **Polycom UC Software** The software that implements the phone functions and features. For information about Polycom UC Software, see the section About Polycom UC Software.

  To begin provisioning devices with Polycom UC Software, refer to Provision with Polycom UC Software.
- **Configuration files** The files included with the UC Software download that contain the phone's settings. Configuration files are for use with the centralized provisioning method. For information about the use of configuration files with centralized provisioning, see the section Use Centralized Provisioning.
- **Resource files** Optional configuration files that contain settings for advanced features. For information about the use of configuration files with centralized provisioning, see the section Use Centralized Provisioning.

### About the Updater

The Updater is a small application that resides in the flash memory on the phone. Polycom phones come installed with the Updater.

**Note: The Updater is also known as BootROM**

The Updater was referred to as the BootROM in previous versions of the UC Software, specifically UC Software 3.3.x and SIP 3.2.x and earlier.

When you start/boot/reboot the phone, the Updater automatically performs the following tasks:

1 The setup menu displays so you can set various network and provisioning options.

  The Updater requests IP settings and accesses the provisioning server (also called the boot server) to look for changes to the Updater software. If updates are found, they are downloaded and saved to flash memory, which overwrites itself after verifying the integrity of the download.

2 If new updates are downloaded, the Updater formats the file system, removes any application software and configuration files that were present.

3 The Updater downloads the master configuration file.

The Updater and the application use this file to acquire a list of other files that the phone needs.

4 The Updater 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 from the one stored in device settings, or there is no file stored in flash memory, the application file is downloaded.

5 The Updater extracts the Polycom UC Software from flash memory.

6 The Updater installs the application into RAM, and then uploads an event log file from the boot cycle.

7 The Updater completes the cycle, and the Polycom UC Software begins running the phone's operations.

## About Polycom UC Software

Polycom UC Software manages the protocol stack, the digital signal processor (DSP), the user interface, the network interaction, and implements the following functions and features on the phones:

- VoIP signaling for a wide range of voice and video telephony functions using SIP signaling for call setup and control.
- SIP and H.323 signaling for video telephony.
- Industry standard security techniques for ensuring that all provisioning, signaling, and media transactions are robustly authenticated and encrypted.
- Advanced audio signal processing for handset, headset, and speakerphone communications using a wide range of audio codecs.
- Flexible provisioning methods to support single phone, small business, and large multi-site enterprise deployments.
- The software is a binary file image and contains a digital signature that prevents tampering or the loading of rogue software images. There is a new image file in each release of software. Both the Updater and Polycom UC Software run on all Polycom device models.

## UC Software Deployment Scenarios

Administrators typically use the administrator guide in three large-scale device deployment scenarios:

- **Enterprise deployment** An administrator sets up and maintains a deployment for a single organization and all users are in one physical location.
- **Multisite enterprise** An administrator sets up and maintains a deployment for an organization and users are spread out over several locations varying in size.
- **Service Provider Deployment** Service providers provide devices and service to a number of organizations and users spread out over several locations each varying in size.

# Overview of Major Deployment Tasks

A typical large-scale deployment requires administrators to complete each of the following major tasks.

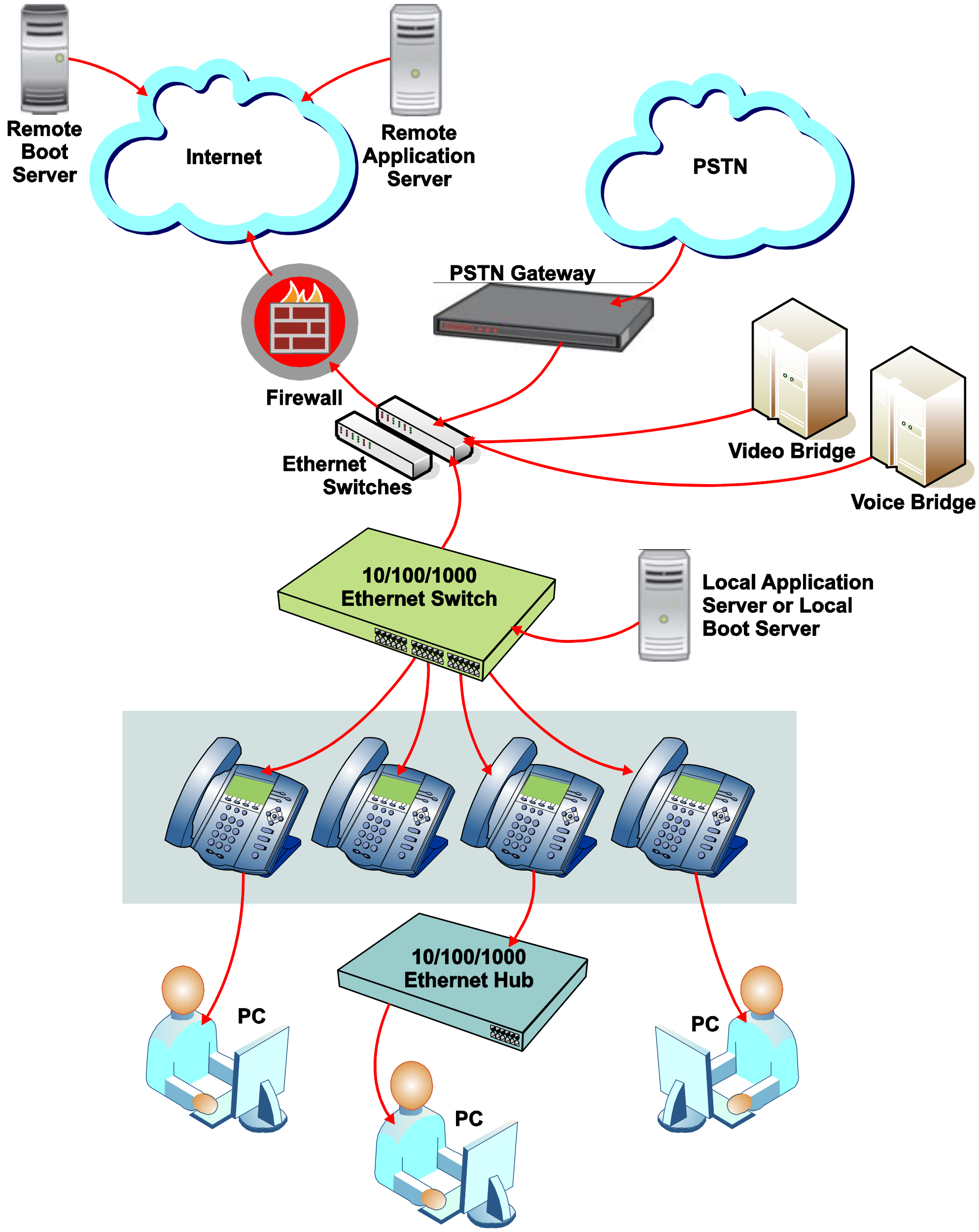
### To provision phones with UC Software:

1. Create user accounts on the SIP call server.
2. (Optional) Set up a provisioning server. In some cases a provisioning server is built into the SIP call server and if not, administrators must set up their own provisioning server.

   Polycom strongly recommends setting up a provisioning server for large-scale VoIP device deployments. A provisioning server maximizes the flexibility you have when installing, configuring, upgrading, and maintaining the phones, and enables you to store configuration, log, directory, and override files on the server.
3. (Optional) Configure security options on your network.
   - 802.1X
   - Virtual local area networks (VLANs)
   - File transfers using HTTPS
   - Configuration files
   - SIP signaling over Transport Layer Security (TLS)
4. Set up Dynamic Host Configuration Protocol (DHCP).
5. Set up Domain Name System (DNS). Polycom supports the following DNS records types:
   - DNS A record
   - Service (SRV) record for redundancy
   - Name Authority Pointer (NAPTR)
6. Connect the phones to the network.

The following figure illustrates one example of a device deployment.

**Polycom wired phones in a network**



# Provision with Polycom UC Software

After you create user accounts on the SIP call server you are using, provision devices with Polycom UC Software. This section explains methods you can use to provision devices and configure features on the phones. Available phone features are listed in the section Configure Devices and Call Controls. You can use one or multiple methods at the same time but note that features and settings vary by configuration method and by device.

It is important to be aware that there is a configuration priority among the methods when you use multiple methods at the same time—settings you make using a higher priority configuration method override settings made using a lower priority configuration method. When using multiple configuration methods, a setting you make using a lower-priority method does not apply to or override a duplicate setting made using a higher-priority method.

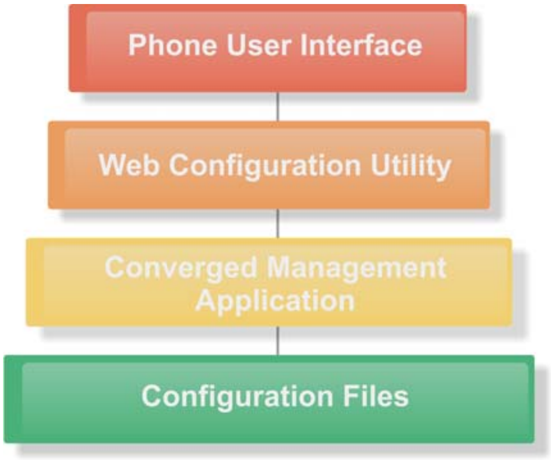
- **Local device interface** You can use the menu system or device interface to provision and configure one device at a time. Note that menu systems and interface settings vary among devices. Settings you make from the device menu or interface override settings you make using the Web Configuration Utility and settings you make on the central provisioning server.
- **Web Configuration Utility** This method provisions and configures features for one phone at a time and is recommended for device deployments of fewer than 20 devices. This method enables you to provision and configure phones using a web browser and enables you to manage phones remotely. However, note that the Web Configuration Utility contains a limited number of settings. Settings you make using the Web Configuration Utility override settings you make on the central provisioning server.
- **Centralized provisioning** Use this method for large-scale device deployments. This method requires you to set up your own provisioning server if your SIP call server does not provide one. Settings you make from a central provisioning server override default device and software settings.

**Web Info: Registering a single Polycom phone**

If you want to register a single Polycom phone, see *Quick Tip 44011: Registering Standalone Polycom SoundPoint IP, SoundStation IP, and VVX 1500 Phones* on Polycom Engineering Advisories and Technical Notifications.

The following figure illustrates the configuration method override priority among the configuration methods.

**Configuration method override priority**



# Provisioning Points to Consider

- If you are provisioning multiple phones, Polycom recommends that you set up a provisioning server to install and maintain your Polycom phones, as shown in the section Use Centralized Provisioning.
- A provisioning server maximizes the flexibility you have when installing, configuring, upgrading, and maintaining the phones, and enables you to store configuration, log, directory, and override files on the server. You can set up a provisioning server on the local area network (LAN) or anywhere on the Internet. If you allow the phone write access to your provisioning server, the phone can use the server to upload all of the file types and store administrator and user settings.
- Polycom phones boot up without the use of configuration files. You can specify a SIP server address and a registration address (the equivalent of a phone number) in a configuration file before or after the phone boots up or, after the phone boots up, from the phone’s interface or the Web Configuration Utility.
- If a phone cannot locate a provisioning server upon boot up, and has not been configured with settings from any other source, it operates with internally stored default values. If the phone that cannot locate a provisioning server has previously been configured with settings it operates with those previous settings.
- Each phone may open multiple connections to the server.
- Settings available only to administrators require a password and are not available to users. Non-administrative users cannot duplicate or override administrator-level settings.
- To view phone provisioning information, use the multikey shortcut by simultaneously pressing 1-4-7 to display:
  - Phone IP address
  - Phone MAC address
  - VLAN ID
  - Boot server type (FTP, TFTP, HTTP, HTTPS)
  - Boot Server Address

**Note: Use RFC-compliant servers**

Polycom recommends that you use RFC-compliant servers.

# Change Settings from the Device Interface

You can change provisioning settings and phone features locally from the menu system on the phone's user interface on a per-phone basis. Settings you make from the device menu or interface override settings you make using the Web Configuration Utility and settings you make on the central provisioning server. If you need to reset all settings made from the device interface to default, refer to the section Reset the Phone to Defaults. As with the Web Configuration Utility, the phone device interface makes settings available to users and administrators; settings available to administrators only can be accessed on the Advanced menu and require an administrator password. For information on setting passwords, see Set Local User and Administrator Passwords.

# Provision with the Web Configuration Utility

You can provision devices with UC software and control features and settings using the Web Configuration Utility, a web-based interface that is especially useful for remote configuration. Because features and settings can vary by device model and UC Software release, options available in the Web Configuration Utility can vary. The Web Configuration Utility makes settings available to users and administrators; settings available to administrators only can be accessed on the Advanced menu and require an administrator password. For information on setting passwords, see Set Local User and Administrator Passwords.

The Web Configuration Utility enables you to perform configuration changes on a per-phone basis. Note that the Web Configuration Utility contains a limited number of settings you can configure. You can use the Web Configuration Utility as the sole configuration method or in conjunction with centralized provisioning and the device interface. If you are provisioning more than ten or twenty phones, Polycom recommends using centralized provisioning as your primary configuration method.

Configuration changes made to individual phones using the Web Configuration Utility override configuration settings made with central provisioning. Configuration changes made using a phone's user interface override settings made using the Web Configuration Utility. If you want to remove settings applied from the Web Configuration Utility, click the Reset to Default button on any page in the Web Configuration Utility.

**Web Info: Using the Web Configuration Utility**

For more detailed help using the Web Configuration Utility, see the *Polycom Web Configuration Utility User Guide* on Polycom UC Software Support Center.

**Note: Updating UC Software on a single phone**

You can use the Software Upgrade tool in the Web Configuration Utility to update the UC Software version running on a single phone. For information, see *Feature Profile 67993: Using the Software Upgrade Tool in the Web Configuration Utility* on Polycom Profiled UC Software Features.

# Use Centralized Provisioning

This section explains the Polycom UC Software configuration files, and shows you how to set up a provisioning server.

The Polycom UC Software that you download contains template configuration files, valid XML files that you can modify using an XML editor. Use of the configuration files to provision the phones with UC Software and to modify features and settings is called the centralized provision method. The configuration files enable you to maintain a set of configuration files for all your devices on a central provisioning server and configure all of your phones to read the same set of files.

The template configuration files are flexible: you can rearrange the parameters within the template, move parameters to new files, or create your own configuration files from parameters you want. This flexibility is especially useful when you want to apply a set of features or settings to separate groups of phones. You can create and name as many configuration files as you want and your configuration files can contain any combination of parameters.

**Note: Configuration method priority**

Remember that settings made from the phone user interface and Web Configuration Utility override settings you make in configuration files using centralized provisioning.

Centralized provisioning requires that the phone be able to read files and directories you list in the master configuration file. In addition, the phone attempts to upload log files (log files provide a history of phone events), a configuration override file, and a provisioning directory file to the provisioning server. Though not required, Polycom recommends configuring a separate directory for each of these files to help organize: a log file directory, an override directory, a contact directory, and a license directory.

Each directory can have different access permissions, for example, you can allow log, contacts, and overrides to have full read and write access, and license to have read-only access. However, where the security environments permits, Polycom recommends that you allow these file uploads to the provisioning server which requires you to give delete, write, and read permissions for the phone’s server account. All other files that the phone needs to read, such as the application executable and the standard configuration files, should be made read-only using file server file permissions. Ensure that the file permissions you create provide the minimum required access and that the account has no other rights on the server. Without permissions, the phone cannot upload files.

**Note: Allow file uploads to your provisioning server**

Allowing file uploads can help Polycom provide customer support when diagnosing issues with the phone

In addition to the template configuration files, the UC Software download includes the following configuration files:

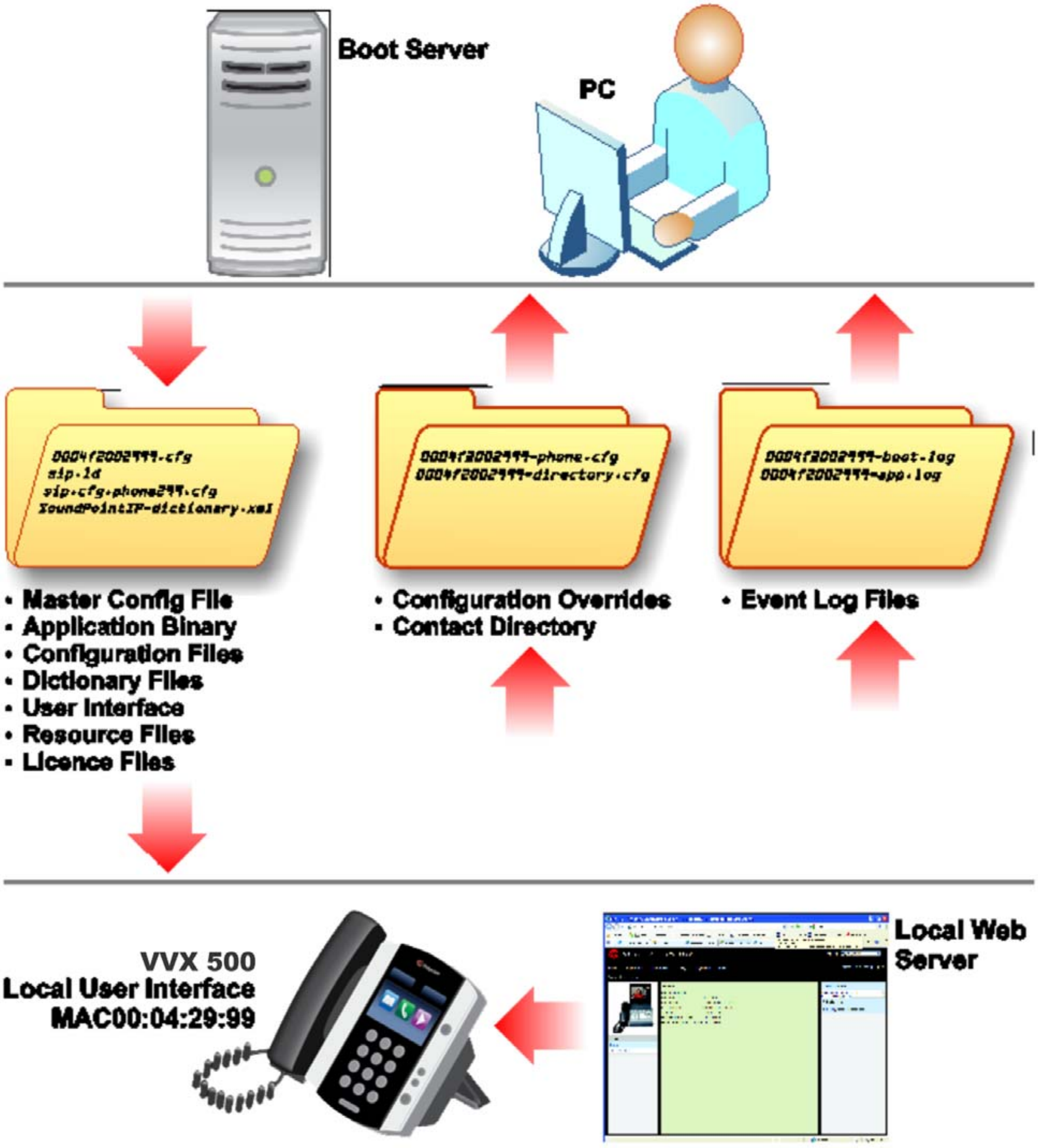
- Language dictionaries
- Ringtones
- Contact directories

If you need to change resource file settings, for example, if you are configuring a phone for a different user, you need to apply factory default settings to that phone as shown in the section Reset the Phone to Defaults

Note that as of Polycom UC Software 4.0.0, you can create user-specific configuration files that enable phone users to use their features and settings from any phone in an organization. For instructions, refer to the section Set User Profiles.

The following figure shows one example of a phone network layout when you use the centralized provisioning method.

**Network layout using centralized provisioning**



## Set Up the Provisioning Server

This section provides instructions for setting up a centralized provisioning server for your Polycom phones. Polycom phones support the FTP, TFTP, HTTP, and HTTPS protocols, and use FTP by default. The example shown in this section uses FTP and a personal computer (PC) as the provisioning server.

### Prerequisites

To begin, install and set up tools on your PC and gather some information:

- If using Power over Ethernet (PoE) with the phone, you need a PoE switch and network cable.
- Install an XML editor, such as XML Notepad 2007, on your computer.
- Install an FTP server application on your computer. FileZilla and *wftpd* are free FTP applications for windows and *vsftpd* is typically available with all standard Linux distributions.
- Take note of the following:
  - ➢ **SIP Server address** This is the hostname or IP address of the call server that handles VoIP services on your network.
  - ➢ **SIP account information** This may include SIP credentials such as a user name and password, and the phone's registration address. Although a user name and password are not required to get the phone working, Polycom strongly recommends using them for security reasons.
  - ➢ **MAC address** Each phone has a unique 12-digit serial number just above the phone's bar code on a label on the back of the phone. Collect the MAC address for each phone in your deployment.
  - ➢ **Your computer's IP address** To use your computer as the provisioning boot server, you need your computer's IP address. Jot this number down as you need it at the end of the provisioning process.

### To set up the provisioning server:

1. Provide power to the phone using a PoE switch, if available, or, if no PoE switch is available, using an external power adapter and a network cable to connect the phone to your network.
2. Install and set up an FTP application. FileZilla and *wftpd* are free FTP applications for windows and *vsftpd* is typically available with all standard Linux distributions.

   You must create a root FTP directory on the provisioning computer with full read and write access to all directories and files. You will be placing configuration files in this root directory.

   In your FTP server application, create a user account for the phone to use and take note of the user name and password as you will need these later in the provisioning process. Launch the FTP application and keep it running at all times so that the phones can communicate with the UC Software.
3. Download the UC software version(s) to your root directory from the *Polycom UC Software Support Center.* To match a phone model with a correct Polycom UC Software release, refer to the *Polycom UC Software Release Matrix for VVX Phones and SoundStructure.*

   You can choose the combined UC Software package or the split UC Software package, both in ZIP file format.

   - ➢ The combined version contains all files for all phone models.
   - ➢ The split software package is smaller, downloads more quickly, and contains **sip.ld** files for each phone model, enabling you to choose provisioning software for your phone model(s) and maintain software versions for each model in the same root directory.
4. To apply security settings to your configuration files, refer to the section Encrypt Configuration Files.

## Configure Multiple Servers

You can configure multiple (redundant) provisioning servers—one logical server with multiple addresses—by mapping the provisioning server DNS name to multiple IP addresses. The default number of provisioning servers is one and the maximum number is eight. For more information on the protocol used, see Supported Provisioning Protocols.

If you set up multiple provisioning servers, you must be able to reach all of the provisioning servers with the same protocol and the contents on each provisioning server must be identical. You can use the table  to configure the number of times each server is tried for a file transfer and also how long to wait between each attempt. You can configure the maximum number of servers to be tried. For more information, contact your certified Polycom reseller.

## Deploy Devices from the Provisioning Server

After setting up your provisioning server(s), you can deploy your devices. This section shows you how to deploy your Polycom devices from the provisioning server.

### To deploy phones with a provisioning server:

1 Using the list of MAC addresses of each phone you are deploying, create a per-phone **phone<MACaddress>.cfg** file.

Do not use the following file names as your per-phone file name: <*MACaddress*>-phone.cfg, <*MACaddress*>-web.cfg, <*MACaddress*>-app.log, <*MACaddress*>-boot.log, or <*MACaddress*>-license.cfg. These file names are used by the phone to store overrides and logging information.

2 Add the SIP server registration information and user account information to parameters in the per-phone file, for example `reg.1.address`, `reg.1.auth.userId`, `reg.1.auth.password`, `reg.1.label`, `reg.1.type`.

3 Create a per-site **site<*location*>.cfg** file.

For example, add the SIP server or feature parameters such as `voIpProt.server.1.address` and `feature.corporateDirectory.enabled`.

**Note: Configuring your phone for local conditions**

If SNTP settings are not available through DHCP, you need to edit the SNTP GMT offset, and possibly the SNTP server address for the correct local conditions. Changing the default daylight savings parameters might be necessary outside of North America. If the local security policy dictates you might need to disable the local Web (HTTP) server or change its signaling port (see<httpd/>). To change the default location settings for user interface language and time and date format (see <lcl/>)

4 Create a master configuration file by performing the following steps:

a Enter the name of each per-phone and per-site configuration file created in steps 2 and 3 in the `CONFIG_FILES` attribute of the master configuration file (**000000000000.cfg**). For help using the master configuration file, see the section Use the Master Configuration File.

For example, add a reference to **phone<*MACaddress*>.cfg** and **sipVVX500.cfg**.

b (Optional) Edit the `LOG_FILE_DIRECTORY` attribute of master configuration file to point to the log file directory.

   c (Optional) Edit the `CONTACT_DIRECTORY` attribute of master configuration file to point to the organization's contact directory.

   (Optional) Edit the `USER_PROFILES_DIRECTORY` attribute of master configuration file if you intend to enable the user login feature. For more information, see the section Set User Profiles.

   d (Optional) Edit the `CALL_LISTS_DIRECTORY` attribute of master configuration file to point to the user call lists.

5 Perform the following steps to configure the phone to point to the IP address of the provisioning server and set up each user:

   a On the phone's **Home** screen or idle display, select **Settings > Advanced > Admin Settings > Network Configuration > Provisioning Server**. When prompted for the administrative password, enter **456**.

   The Provisioning Server entry is highlighted.

   b Press the **Select** soft key.

   c Scroll down to **Server Type** and ensure that it is set to **FTP**.

   d Scroll down to **Server Address** and enter the IP address of your provisioning server.

   e Press the **Edit** soft key to edit the value and the **OK** soft key to save your changes.

   f Scroll down to **Server User** and **Server Password** and enter the user name and password of the account you created on your provisioning server, for example, `bill1234` and `1234`, respectively.

   g Press the **Back** soft key twice.

   h Scroll down to **Save & Reboot**, and then press the **Select** soft key.

   The phone reboots and the UC Software modifies the `APPLICATION APP_FILE_PATH` attribute of the master configuration file so that it references the appropriate sip.ld files.

   After this step, the UC Software reads the unmodified `APPLICATION APP_FILE_PATH` attribute. Then, the phone sends a DHCP Discover packet to the DHCP server. You can locate this in the Bootstrap Protocol/option 'Vendor Class Identifier' section of the packet which includes the phone's part number and the BootROM version. For more information, see the section Parse Vendor ID Information.

6 Ensure that the configuration process completed correctly:

   a On the phone, press **Settings** (**Menu** if using a VVX 1500) **> Status > Platform > Application > Main** to see the UC Software version and **Status > Platform > Configuration** to see the configuration files downloaded to the phone.

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

   The phone uploads two logs files to the LOG_DIRECTORY directory: **<*MACaddress*>-app.log** and **<*MACaddress*>-boot.log**.

   You can now instruct your users to begin making calls.

**Settings: View the phone's provisioning information**

To view phone provisioning information, use the multikey shortcut by simultaneously pressing 1-4-7 to display:

- Phone IP address
- Phone MAC address
- VLAN ID
- Boot server type (FTP, TFTP, HTTP, HTTPS)

## Override Files

When using a central provisioning server as part of your VoIP environment, you have the option to store the override file to the phone, or you can permit the phone to upload the override file to the provisioning server by giving the phone write access to the provisioning server. The advantage of allowing the phone write access to the provisioning server for override files is that user settings for a phone survive restarts, reboots, and software upgrades you apply to all phones using a provisioning server. You can also use the override files to save user custom preferences and to apply specific configurations to a device or device group. If you permit the phone to upload to the provisioning server, the override file is by default named either **<*MAC Address*>-phone.cfg** or **<*MAC Address*>-Web.cfg** depending on the whether the change was made from the phone or Web Configuration Utility respectively.

**Note: Priority of Configuration Methods**

Changes to settings using a configuration method having a higher priority than another create an override file that is uploaded to your provisioning server directory. The order of priority is as follows:

- `<`*`MAC Address`*`>-phone.cfg` overrides `<`*`MAC Address`*`>-Web.cfg`

Both override files override settings you make from the provisioning server. The phone uploads an override file each time a configuration change is made from the phone. If you reformat the phone's file system, the override file is deleted.

If you need to clear phone settings and features applied by override files, see the section Reset the Phone to Defaults.

# Use the Master Configuration File

The centralized provisioning method requires you to use a master configuration file, named **00000000000.cfg** in the UC Software download.

You can apply the master configuration file to phones in the following ways:

- **To all phones** If you are applying the same features and settings to all phones, you can use the default master configuration file to configure all the phones in a deployment. Note that the phones are programmed to look first for their own **<*MACaddress*>.cfg** file and if a phone does not find a matching file, it looks next for the default file named **000000000000.cfg**. If you do create and use a per-phone master configuration file, make a copy of the default file and rename it.
- **To a phone group or to a single phone** If you want to apply features or settings to a group of phones or to a single phone, make a copy of the default master configuration file and rename it. You can specify a device group by model or part number using the variable substitutions shown in the section Create Device Groups.

  For single phones, rename the file with a naming scheme that uses the phone's MAC address **<*MACaddress*>.cfg**. The MAC address, also known as the serial number (SN), is a unique a-f hexadecimal digit assigned to each phone. Note that you can use only lower-case letters, for example, **0004f200106c.cfg**. You can find the MAC address of a phone on a label on the back of the phone or on the phone's menu system at **Settings** (**Menu** if using a VVX 1500) **> Status > Platform > Phone > S/N:**. For more information about naming schemes and efficient provisioning with the master configuration file, refer to the section Use Variable Substitution.

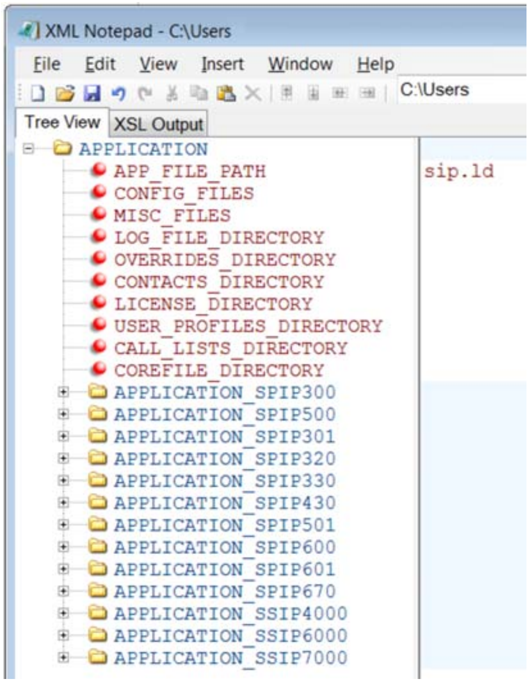
- **Specify a location** You can specify the location of a master configuration file you want the phones to use, for example, `http://usr:pwd@server/dir/example1.cfg`. The file name must be at least five characters long and end with **.cfg**. If the phone cannot find and download a location you specify, the phone searches for and uses a per-phone master configuration file and then the default master configuration file.

**Note: Pay attention to per-phone file names**

Do not use the following names as extensions for per-phone files: <*MACaddress*>-phone.cfg, <*MACaddress*>-Web.cfg, <*MACaddress*>-app.log, <*MACaddress*>-boot.log, or <*MACaddress*>-license.cfg. These filenames are used by the phone to store override files and logging information.

The master configuration file contains a number of default fields, as shown in the following figure.

**Default fields in the master configuration file**



```
APPLICATION
    APP_FILE_PATH                 sip.ld
    CONFIG_FILES
    MISC_FILES
    LOG_FILE_DIRECTORY
    OVERRIDES_DIRECTORY
    CONTACTS_DIRECTORY
    LICENSE_DIRECTORY
    USER_PROFILES_DIRECTORY
    CALL_LISTS_DIRECTORY
    COREFILE_DIRECTORY
  APPLICATION_SPIP300
  APPLICATION_SPIP500
  APPLICATION_SPIP301
  APPLICATION_SPIP320
  APPLICATION_SPIP330
  APPLICATION_SPIP430
  APPLICATION_SPIP501
  APPLICATION_SPIP600
  APPLICATION_SPIP601
  APPLICATION_SPIP670
  APPLICATION_SSIP4000
  APPLICATION_SSIP6000
  APPLICATION_SSIP7000
```

The following describes the XML field attributes in the master configuration file and the APPLICATION directories.

- **APP_FILE_PATH** The path name of the UC Software application executable. The default value is sip.ld. Note that the phone automatically searches for the sip.ld and <*part number*>.sip.ld. This field can have a maximum length of 255 characters. If you want the phone to search for a sip.ld file in a location other than the default or use a different file name, or both, modify the default. For example, you can specify a URL with its own protocol, user name, and password: http://usr:pwd@server/dir/sip.ld.

- **CONFIG_FILES** Enter the names of your configuration files here as a comma-separated list. Each file name has a maximum length of 255 characters and the entire list of file names has a maximum length of 2047 characters, including commas and white space. If you want to use a configuration file in a different location or use a different file name, or both, you can specify a URL with its own protocol, user name and password, for example: `ftp://usr:pwd@server/dir/phone2034.cfg`.

**Note: Order of the configuration files**

The order of the configuration files listed in CONFIG_FILES is significant:

- The files are processed in the order listed (left to right).
- If the same parameter is included in more than one file or more than once in the same file, the phone uses the first (left) parameter.

- **MISC_FILES** A comma-separated list of files. Use this to list volatile files that you want phones to download, for example, background images and ringtone .wav files. The phone downloads files you list here when booted, which can decrease access time.
- **LOG_FILE_DIRECTORY** An alternative directory for log files. You can also specify a URL. This field is blank by default.
- **CONTACTS_DIRECTORY** An alternative directory for user directory files. You can also specify a URL. This field is blank by default.
- **OVERRIDES_DIRECTORY** An alternative directory for configuration overrides files. You can also specify a URL. This field is blank by default.
- **LICENSE_DIRECTORY** An alternative directory for license files. You can also specify a URL. This field is blank by default.
- **USER_PROFILES_DIRECTORY** An alternative directory for the **<*user*>.cfg** files.
- **CALL_LISTS_DIRECTORY** An alternative directory for user call lists. You can also specify a URL. This field is blank by default.
- **COREFILE_DIRECTORY** An alternative directory for Polycom device core files to use to debug problems. This field is blank by default.

The directories labeled **APPLICATION_SPIP*XXX*** indicate phone models that are not compatible with the latest UC Software version. If you are using any of the phone models listed in these directories, open the directory for the phone model you are deploying, and use the available fields to provision and configure your phones.

## Use Variable Substitution

You can use the master configuration template file, by default named 000000000000.cfg in the UC Software files you download, to specify features and settings for single phones and phone groups. This section details two naming schemes you can use to efficiently provision with the master configuration file. The method you use depends on your deployment scenario and understanding both methods helps you to deploy and manage your phones efficiently.

### Method One: Define a Per-Phone *MACaddress*.cfg File

You can create a *MACaddress.cfg* file for each phone from the master configuration file template.

The advantage of using this method is a high degree of control over each phone. You can apply configuration files to phones by adding new files to the CONFIG_FILES field of each phone’s *MACaddress.cfg* file. If you want to modify or add settings, go to the configuration files for that phone and make your changes. If all of the phones in your deployment use the same settings, you can create a single configuration file for each phone.

This method can require some file management work as you need to create and edit at least two unique files for each phone in your deployment, namely, a *MACaddress.cfg* file and one or more configuration files unique to each phone. You can use the template files or you can create your own files from parameters in the template files. If you do not want to create a new file, add new parameters to any configuration file already in the CONFIG_FILES field of a phone's *MACaddress.cfg* file.

### To create a per-phone MAC address configuration files:

1. Create a *MACaddress.cfg* file for each phone, replacing `000000000000` with the unique MAC address of each phone you are configuring, for example `0004f2123456`.cfg.

   You can find the MAC address of your phone on a label on back of the phone.
2. Create a file for each phone containing information unique to each phone, for example, registration information. You can use the template files in the UC Software download, or you can create your own configuration file using parameters from the UC Software template files. Give your files a name that indicates the file contents or purpose. For example, you might use parameters from the `reg-basic.cfg` template file to create a registration file named `reg-basic_john_doe.cfg`.
3. Enter the name of the file you created in step two in the CONFIG_FILES field of the *MACaddress.cfg* file you created in step one for each phone.
4. Save the master configuration file.

## Method Two: Use a Variable Substitution

This method enables you to configure all phones using a single master configuration file instead of a MACaddress.cfg file for each phone. This method follows from the phone's programmed behavior: the phone looks first for a file containing its own MAC address and if it cannot find that, uses the default *000000000000.cfg* master configuration file.

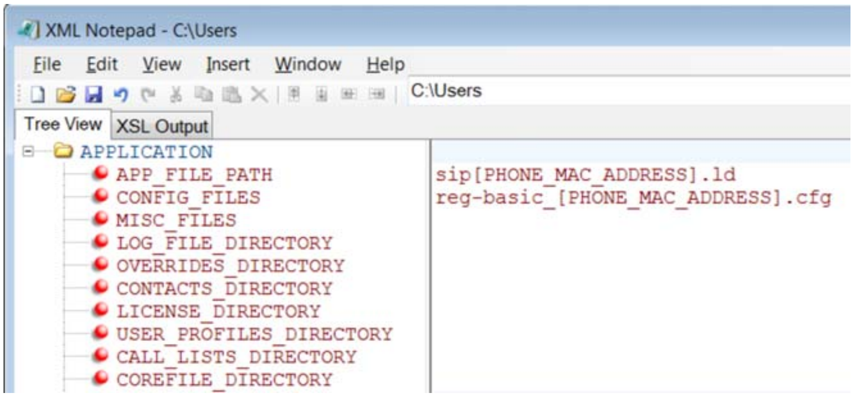
This method is useful if you need to maintain or modify settings common to all of the phones in your deployment. To apply a common configuration to all phones, you need only create one new configuration file and add it to the CONFIG_FILES field of the 000000000000.cfg master file. If you want to add a new phone to your deployment, you need only create one new file.

For more information on creating phone groups and using variable substitutions, see the section Create Device Groups.

### To configure using a variable substitution:

1. Create a file for each phone containing information unique to each phone, for example, registration information. The name of this file must contain the phone's unique MAC address, for example, `reg-basic_0004f2000001.cfg`. Each of these phone-specific configuration files must be named identically, varying only in the MAC address of each phone.
2. Enter the name of any one of your phone-specific files to the CONFIG_FILES field of the master configuration file.
3. Modify the file name in the CONFIG_FILES field by replacing the phone-specific MAC address with the variable [PHONE_MAC_ADDRESS] and include the square brackets. You must enter the variable in the same place you entered the phone's MAC address in the phone-specific file.

   For example, modify `reg-basic_0004f2000001.cfg` to `reg-basic_[`*`PHONE_MAC_ADDRESS`*`].cfg`.



```
APPLICATION
    APP_FILE_PATH
    CONFIG_FILES          sip[PHONE_MAC_ADDRESS].ld
    MISC_FILES            reg-basic_[PHONE_MAC_ADDRESS].cfg
    LOG_FILE_DIRECTORY
    OVERRIDES_DIRECTORY
    CONTACTS_DIRECTORY
    LICENSE_DIRECTORY
    USER_PROFILES_DIRECTORY
    CALL_LISTS_DIRECTORY
    COREFILE_DIRECTORY
```

**4** Save the master configuration file.

## Create Device Groups

You can create custom device groups by:

- Using a variable in the master configuration file
- Appending a device model or part number to a parameter.

### Use a Variable in the Master Configuration File

You can use any of the following variable strings to create custom device groups:

- [PHONE_MODEL]
- [PHONE_PART_NUMBER]
- [PHONE_MAC_ADDRESS]

You can find the MAC address of a phone on a label on the back of the phone or on the phone’s menu at **Menu > Status > Platform > Phone > S/N:**. To get the model number or part number of a device, see the table .

### Append a Device Model or Part Number to a Parameter

You can customize a set of parameters for a specific device model by appending a device model or part number descriptor to a parameter. The part number has precedence over the model number, which has precedence over the original firmware version.

For example, for a VVX 500, CONFIG_FILES_3111-44500-001=”phone1_3111-44500-001.cfg, sip_3111-44500-001” overrides CONFIG_FILES_ VVX500=”phone1_ VVX500.cfg, sip_ VVX500.cfg”, which overrides CONFIG_FILES=”phone1.cfg, sip.cfg”.

The following table lists the product name, model name, and part number mapping for Polycom devices.

**Product Name, Model Name, and Part Number**

| Product Name | Model Name | Part Number |
|---|---|---|
| SoundStructure VoIP Interface | SSTRVOIP | 3111-33215-001 |
| VVX 101 | VVX 101 | 3111-40250-001 |

**Product Name, Model Name, and Part Number**

| | | |
|---|---|---|
| VVX 201 | VVX 201 | 3111-40450-001 |
| VVX 300 | VVX 300 | 3111-46135-002 |
| VVX 301 | VVX 301 | 3111-48300-001 |
| VVX 310 | VVX 310 | 3111-46161-001 |
| VVX 311 | VVX 311 | 3111-48350-001 |
| VVX 400 | VVX 400 | 3111-46157-002 |
| VVX 401 | VVX 401 | 3111-48450-001 |
| VVX 410 | VVX 410 | 3111-46162-001 |
| VVX 411 | VVX 411 | 3111-48450-001 |
| VVX 500 | VVX 500 | 3111-44500-001 |
| VVX 501 | VVX 501 | 3111-48500-001 |
| VVX 600 | VVX 600 | 3111-44600-001 |
| VVX 601 | VVX 601 | 3111-48600-001 |
| VVX 1500 | VVX 1500 | 2345-17960-001 |

# Use the Template Configuration Files

The Polycom UC Software download includes a number of template configuration files containing configuration parameters. Most configuration parameters are located in only one template file; however, some are included in two or more files. Remember that configuration files you write to the CONFIG_FILES field of the master configuration file are read from left to right and the phone uses the file it reads first.

**Caution: Deprecated configuration parameters**

Polycom may deprecate configuration parameters that some organizations may still be using—deprecated parameters will not work. To view a list of deprecated parameters, see the latest Polycom UC Software Release Notes on Latest Polycom UC Software Release or check the Release Notes for earlier software versions on Polycom UC Software Support Center.

The following table lists each template file included with the UC Software download.

**Configuration File Templates**

| Name | Description | Deployment Scenarios |
|---|---|---|
| applications.cfg | For applications, browser, microbrowser, XMP-API | Typical Hosted Service Provider<br>Typical IP-PBX |
| device.cfg | Network Configuration parameters. Refer to the section Modify Ethernet Settings. | Troubleshooting<br>Administrative settings |

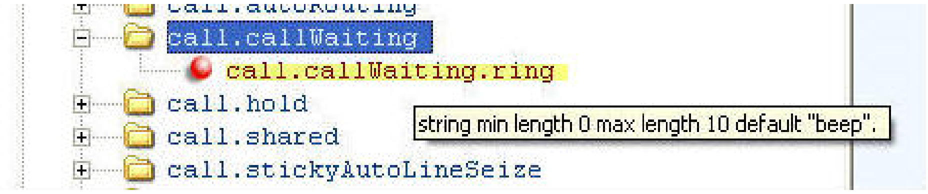
**Configuration File Templates**

| | | |
|---|---|---|
| features.cfg | Features including corporate directory, USB recording, presence, ACD | Typical Hosted Service Provider<br>Typical IP-PBX |
| firewall-nat.cfg | Firewall parameters | |
| H323.cfg | H.323 video use | Typical Hosted Service Provider using VVX 500/501, 600/601, and 1500 for video calls |
| lync.cfg | Microsoft Skype for Business parameters | Typical Microsoft Skype for Business environment |
| reg-advanced.cfg | Advanced call server, multi-line phones | Typical Hosted Service Provider<br>Typical IP-PBX |
| reg-basic.cfg | Basic registration | Simple SIP device<br>Typical Hosted Service Provider |
| region.cfg | Non-North American geographies | Typical Hosted Service Provider<br>Typical IP-PBX |
| sip-basic.cfg | Basic call server | Simple SIP device<br>Typical Hosted Service Provider |
| sip-interop.cfg | Advanced call server, multi-line phones | Typical Hosted Service Provider<br>Typical IP-PBX |
| site.cfg | Multi-site operations | Typical Hosted Service Provider<br>Typical IP-PBX |
| techsupport.cfg | Available by special request from Polycom Customer Support. | Use for troubleshooting and debugging only |
| video.cfg | VVX 500/501, 600/601, and 1500 video | Typical Hosted Service Provider if using VVX 500/501, 600/601, and 1500 for video calls |

Along with the template files, UC Software includes an XML schema file—polycomConfig.xsd—that provides information like parameters type (Boolean, integer, string, and enumerated type), permitted values, default values, and all valid enumerated type values. View this template file with an XML editor.

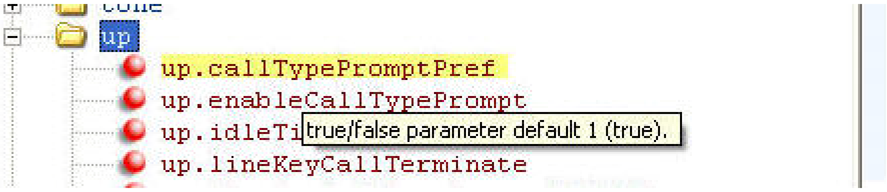
A string parameter, Boolean, and enumerated parameters are shown in the following figures: String parameter, Boolean parameter, and Enumerated parameter.

**String parameter**



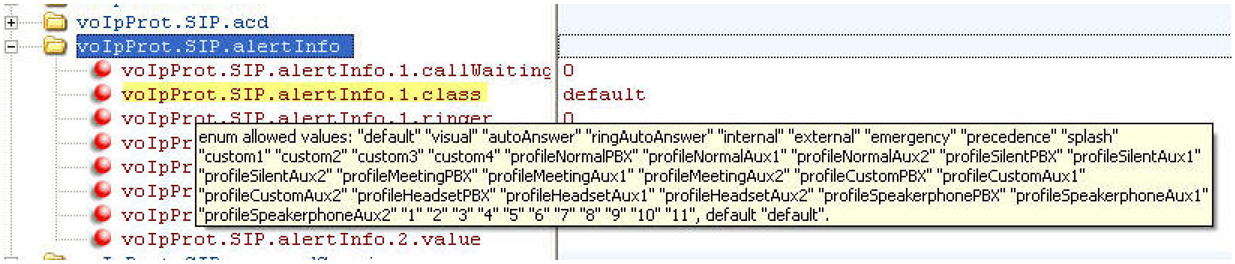
string min length 0 max length 10 default "beep".

**Boolean parameter**



```
up
  up.callTypePromptPref
  up.enableCallTypePrompt
  up.idleTi  true/false parameter default 1 (true).
  up.lineKeyCallTerminate
```

**Enumerated parameter**



```
voIpProt.SIP.acd
voIpProt.SIP.alertInfo
  voIpProt.SIP.alertInfo.1.callWaiting   0
  voIpProt.SIP.alertInfo.1.class         default
  voIpProt.SIP.alertInfo.1.ringer        0
  voIpPr
  voIpPr
  voIpPr
  voIpPr
  voIpProt.SIP.alertInfo.2.value
```

enum allowed values: "default" "visual" "autoAnswer" "ringAutoAnswer" "internal" "external" "emergency" "precedence" "splash" "custom1" "custom2" "custom3" "custom4" "profileNormalPBX" "profileNormalAux1" "profileNormalAux2" "profileSilentPBX" "profileSilentAux1" "profileSilentAux2" "profileMeetingPBX" "profileMeetingAux1" "profileMeetingAux2" "profileCustomPBX" "profileCustomAux1" "profileCustomAux2" "profileHeadsetPBX" "profileHeadsetAux1" "profileHeadsetAux2" "profileSpeakerphonePBX" "profileSpeakerphoneAux1" "profileSpeakerphoneAux2" "1" "2" "3" "4" "5" "6" "7" "8" "9" "10" "11", default "default".

## Change Parameter Values

The configuration parameters available in the UC Software use a variety of values, including Boolean, integer, enumerated types, and arrays (a table of values). Parameters available in the UC Software are listed in alphabetical order along with a description, the default value, and permissible values in the section Configuration Parameters.

Note that the values for Boolean configuration parameters are not case sensitive. The values `0, false`, and `off` are inter-changeable and supported. The values `1, true`, and `on` are interchangeable and supported. This Administrator Guide documents only `0` and `1`.

The following rules apply when you set a parameter with a numeric value outside of its valid range:

- If the configuration file's value is greater than the allowable range, the maximum value is used.
- If the configuration file's value is less than the allowable range, the minimum value is used.
- If you insert invalid parameter values into the configuration file, the value is ignored and the default value is used. Invalid parameters values can occur when enumerated type parameters do not match a pre-defined value, when numeric parameters are set to a non-numeric values, when string parameters are either too long or short, or when using null strings in numeric fields. Instances of invalid values are logged in the phone's log files.

**Note Using blank values and special characters in the configuration files**

The UC Software interprets Null as empty; that is, `attributeName=""`.

To enter special characters in a configuration file, enter the appropriate sequence using an XML editor:

- & as `&amp;`
- " as `&quot;`

- ’ as `&apos;`
- < as `&lt;`
- > as `&gt;`
- random numbers as `&0x12;`

# Configure Network Settings

The UC Software supports the deployment of Polycom phones for your device network:

- As a Session Initiation Protocol (SIP)-based endpoint interoperating with a SIP call server or softswitch. For more information on SIP, see the section Session Initiation Protocol (SIP).
- As an H.323 video endpoint (Polycom VVX 500/501, 600/601, and 1500 business media phones).

**Web Info: Using VVX 1500 phones in a strict H.323 environment**

For more information on using VVX 1500 phones in a strict H.323 environment, see *Deployment Guide for the Polycom VVX 1500D Business Media Phone* at Polycom VVX 1500 D on Polycom Support.

Polycom devices operate on an Ethernet local area network (LAN). Local area network design varies by organization and Polycom phones can be configured to accommodate a number of network designs. This section shows you several automated and manual ways to configure Polycom phones to operate on a LAN.

Connecting your Polycom phone to the LAN initiates the following startup sequence.

- Only step 1 is required and automatic.
- Steps 2, 3, and 4 are optional as these settings can be manually configured on the device. It is common to complete step 3 using a DHCP server within the LAN.

### Startup sequence:

1. The phone establishes network connectivity.

   Wired phones establish a 10M/100M/1000M network link with an Ethernet switch device and do not function until this link is established. If the phone cannot establish a link to the LAN, an error message '*Network link is Down'* displays.
2. (Optional) Apply appropriate security and Quality of Service (QoS) settings.
3. Assign the phone to a VLAN and/or 802.1X authentication.
4. Establish DHCP negotiation with the network and IP address, network addressing options, network gateway address, and time server.
5. Provisioning server discovery.

   This is commonly done using DHCP as part of the previous step. As of UC Software 4.0, the phone contacts the provisioning server after the phone is operational in order to speed up boot time. You can disable the provisioning server discovery process as a way of reducing load on a provision server, for example, after a power failure.

Each step is explained in more detail.

### Digest Authentication for Microsoft Internet Information Services

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 because the Microsoft Internet Information Server cannot recognize these extensions and will return a “File not found” error. To configure wildcard for MIME type, see IIS 6.0 does not serve unknown MIME types.

For more information, see Digest Authentication in IIS 6.0 on Microsoft TechNet.

# Establish Phone Connection to the Network

The phones are configured to automatically negotiate the Ethernet rate so that no special configuration is required. Typical network equipment supports one of the three following Ethernet line rates: 10Mbps, 100Mbps, and 1000Mbps. Though you have the option to change the line rates and/or duplex configuration, Polycom recommends keeping the default settings. If you do change the settings, make the changes before connecting your device to the network.

The phone supports two features to prevent Denial of Service (DoS):

- **Storm Filtering** To change this parameter, see the section Modify Ethernet Settings.
- **VLAN Filtering** To change this parameter, go to the section Modify VLAN Settings. VLAN filtering for the VVX business media phones is done by the Linux operating system and cannot be disabled.

Support for Storm and VLAN filtering varies by device.

# Apply Security and Quality of Service

You have the option of using several layer-2 mechanisms that increase network security and minimize audio latency. This section describes each of the network security options.

## Set Up VLANs and Wired Devices

You can use a virtual local area network (VLAN) to separate and assign higher priority to a voice LAN as a way of minimizing latency.

There are several methods you can use to configure the phone to work on a particular VLAN. If the phone receives a VLAN setting from more than one of the following methods, the priority is as follows (from highest to lowest):

- **LLDP** 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. To change these parameters, go to Modify VLAN Settings.
- **CDP Compatible** Cisco Discovery Protocol (CDP) is a proprietary Data Link Layer network protocol. CDP Compatible follows the same set of rules. To change this parameter, go to Modify VLAN Settings.
- **Static** The VLAN ID can be manually set from the phone UI or from a configuration file. To change this parameter, go to Modify VLAN Settings. This sets the device setting parameter only.

- **DHCP** Dynamic Host Configuration Protocol (DHCP) is an automatic configuration protocol used on IP networks. To change this parameter, go to Modify DHCP Settings. To use DHCP for assigning VLANs, see the section Assign a VLAN ID Using DHCP. Note that use of DHCP for assigning VLANs is not standardized and is recommended only if the switch equipment does not support LLDP or CDP Compatible methods.

## Set Up 802.1X Authentication

802.1X authentication is a technology that originated for authenticating Wi-Fi links. It has also been adopted for authenticating computers within fixed LAN deployments. Multiple Device Authentication is available for Polycom devices as of UC Software 4.0.0.

Note that when VoIP phones with a secondary Ethernet port are used to connect computers on a network, the 802.1X authentication process becomes more complex since the computer is not directly connected to the 802.1X switch. To configure 802.1X, see the section Modify 802.1X Settings.

**Web Info: Understand 802.1X**

For more information on 802.1X authentication, see Introduction to IEEE 802.1X and Cisco® Identity-Based Networking Services (IBNS) at Cisco 802.1X.
See also IEEE 802.1X Multi-Domain Authentication on Cisco Catalyst Layer 3 Fixed Configuration Switches Configuration Example.

There are three ways to configure 802.1X authentication of devices connected to the PC port of the phone:

- You can configure many switches to automatically *trust* or accept a VoIP phone based on its MAC address. This is sometimes referred to as MAC Address Bypass (MAB). Note that the maximum number of individuals MAC addresses supported on the VVX embedded internal switch is 4,000. Note that the maximum number of individual MAC addresses supported on the VVX embedded internal switch is 4,000.
- Some switches support a feature that automatically *trust* a device that requests a VLAN using the CDP protocol.
- Some deployments support Multiple Device Authentication (MDA). In this situation, both the phone and the PC separately authenticate themselves.

  In this scenario since the phone is closest to the 802.1X switch, the phone needs to notify the switch when the PC is disconnected. This can be achieved using an 802.1X EAPOL-Logoff message.

# Configure Network Settings Using DHCP

After the phone has established network connectivity it needs to acquire several IP network settings. These settings are typically obtained automatically from a Dynamic Host Configuration Protocol (DHCP) server. You have the option to configure IP network settings manually from the phone UI, or to pre-provision using a `device.set` capability. If you have never set up a provisioning server before, Polycom recommends reading the information provided in this section.

**Web Info: RFC information on DHCP options**

For more information on DHCP options, see RFC 2131 and RFC 2132.

When making the DHCP request, the phone includes information in Option 60 that can assist the DHCP server in delivering the appropriate settings to the device. For more information, see *Using DHCP Vendor Identifying Options with Polycom Phones: Technical Bulletin 54041* at Polycom Engineering Advisories and Technical Notifications.

**Timesaver: Reducing repetitive data**
Polycom recommends using DHCP where possible to eliminate repetitive manual data entry.

The following table details the settings supported through the DHCP menu.

**DHCP Network Parameters**

| Parameter | DHCP Option | DHCP | DHCP INFORM | Configuration File (application only) | Device Settings |
|---|---|---|---|---|---|
| IP address | - | • | - | - | • |
| Subnet mask | 1 | • | - | - | • |
| IP gateway | 3 | • | - | - | • |
| Boot server address | Refer to Modify DHCP Settings or Provisioning Server Discovery. | • | • | - | • |
| SIP server address | 151 **Note**: You can change this value by changing the device setting. Refer to <device/>. | • | - | - | • |
| SNTP server address | Look at option 42, then option 4. | • | - | • | • |
| SNTP GMT offset | 2 | • | - | • | • |
| Syslog | Refer to the section Modify Syslog Settings | | | | |
| DNS server IP address | 6 | • | - | - | • |
| DNS INFORM server IP address | 6 | • | - | - | • |

**DHCP Network Parameters**

| | | | | | |
|---|---|---|---|---|---|
| DNS domain | 15 | • | - | - | • |
| VLAN ID | Refer to the section Modify DHCP Settings | **Warning**: Link Layer Discovery Protocol (LLDP) overrides Cisco Discovery Protocol (CDP). CDP overrides Local FLASH which overrides DHCP VLAN Discovery. | | | |

**Note: Overriding the DHCP value**

You can configure parameter values for the **SNTP server address** and **SNTP GMT offset** to override the DHCP value. Refer tcpIpApp.sntp.address.overrideDHCP for more information.

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

## DHCP Option 43

DHCP Option 60 controls how the phone identifies itself to a DHCP server for Polycom-specific options that must be returned. If the format for Option 60 is set to RFC 3925, then all Option 43 returned values are ignored. If the format for Option 60 is set to ASCII string, then the Option 43 response should have a hexadecimal string value encapsulating sub-options that override any options received outside of DHCP Option 43.

The following table lists the individual sub-options and combination sub-options supported on VVX phones for DHCP Option 43.

**DHCP Option 43 Configuration Options**

| Option | Results |
|---|---|
| Option 1- subnet mask | The phone parses the value from Option 43 |
| Option 2 - Time offset | The phone parses the value. |
| Option 3 - Router | The phone parses the value. |
| Option 4 - Time server | The phone parses the value. |
| Option 6 - Domain Name Server | The phone parses the value. |
| Option 7 - Domain Log server | The phone parses the value. |
| Option 15 - Domain Name | The phone parses the value. |
| Option 42 - Network Time Protocol server | The phone parses the value. |
| Option 66 - TFTP Server Name | The phone parses the value. |
| Sub-options configured in Option 43 | |
| Options 1, 2, 3, 4, 5, 6, 7, 15, 42, and 66 | The phone parses the value. |

If you do not have control of your DHCP server or do not have the ability to set the DHCP options, enable the phone to automatically discover the provisioning server address. You can do this by connecting to a secondary DHCP server that responds to DHCP INFORM queries with a requested provisioning server value. For more information, see RFC 3361 and RFC 3925.

# Provisioning Server Discovery

After the phone has established network settings, the phone must discover a provisioning server that administrators typically use to obtain software updates and configuration settings. If you have never set up a provisioning server before, Polycom recommends reading the information provided in this section.

The phones support several methods to discover a provisioning server:

- **Static** You can manually configure the server address from the phone's user interface or the Web Configuration Utility, or you can pre-provision the phone. The server address is manually configured from the phone’s user interface, the Web Configuration Utility, or pre-provisioned using `device.set` in a configuration file.
- **DHCP** A DHCP option is used to provide the address or URL between the provisioning server and the phone.
- **DHCP INFORM** The phone makes an explicit request for a DHCP option (which can be answered by a server that is not the primary DHCP server). For more information, see RFC 3361 and RFC 3925.
- **Quick Setup** This feature offers a soft key that takes the user directly to a screen to enter the provisioning server address and information. This is simpler than navigating the menus to the relevant places to configure the provisioning parameters. For more information, see *Using Quick Setup with Polycom Phones: Technical Bulletin 45460 at* Polycom Engineering Advisories and Technical Notifications.

To change these parameters, refer to the section Modify Provisioning Server Settings.

## Supported Provisioning Protocols

By default, phones are shipped with FTP enabled as the provisioning protocol. Note that there are two types of FTP method—active and passive—and UC Software is not compatible with active FTP. You can change the provisioning protocol by updating the *Server Type* option. Or, you can specify a transfer protocol in the *Server Address*, for example, *http://usr:pwd@server*. The server address can be an IP address, domain string name, or URL, and can be obtained through DHCP. For more information, refer to the section Modify Provisioning Server Settings.

Configuration file names in the ***<MACaddress>*.cfg** file can include a transfer protocol, for example, https://usr:pwd@server/dir/file.cfg. If you specify a user name and password as part of the server address or file name, they are used only if the server supports them. If a user name and password are required but not specified, the device settings are sent to the server.

The Updater performs the provisioning functions of uploading log files, master configuration files, software updates, and device setting menu changes. To guarantee software integrity, the Updater downloads only cryptographically signed Updater or UC Software images. Though UC Software supports digest and basic authentication when using HTTP/HTTPS, the Updater supports only digest authentication when using HTTP. When using HTTPS, the phone trusts widely recognized certificate authorities and you can add custom certificates to the phone. Note that log files are not appended when using TTP or HTTPS.

**Settings: Choosing a valid URL**

A URL must 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 from device settings are used.

**Web Info: View trusted certificate authorities**

For more information, see *Certificate Updates for Polycom UC Software* and *Using Custom Certificates with Polycom Phones: EA 17877* at Polycom Engineering Advisories and Technical Notifications.

As of SIP 3.2, TLS authentication is available. For more information, refer to the section Support Mutual TLS Authentication.

As of UC Software 4.0.0, 802.1X authentication is available. For more information, refer to the section Set Up 802.1X Authentication.

## Digest Authentication for Microsoft Internet Information Services (IIS)

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 because the Microsoft Internet Information Server cannot recognize these extensions and will return a “File not found” error. To configure wildcard for MIME type, see IIS 6.0 does not serve unknown MIME types.

For more information, see Digest Authentication in IIS 6.0 on Microsoft TechNet.

# Modify Phone Network Settings

You have the option to modify phone network settings. This section lists network settings available from the device interface. Use the soft keys, the arrow keys, and the Select and Delete keys to make changes.

If you have never set up a provisioning server before, Polycom recommends reading the information provided in this section.

You can update the network configuration parameters at one of two stages:

- **During the Updater Phase.** The setup menu is accessible during the auto-boot countdown of the Updater phase of operation. While your phone boots up, press the **Cancel** soft key, and press the **Setup** soft key to launch the setup menu. To access the setup menu, you must enter the administrator’s password.
- **After your phone starts and is running UC Software.** The network configuration menu is accessible from the phone’s main menu. Select **Menu > Settings > Advanced > Admin Settings > Network Configuration**. To access the **Advanced** menu, you must enter the administrator’s password.

**Tip: Changing the default administrator password**

Polycom recommends that you change the default administrative password. Refer to the section Set Local User and Administrator Passwords.

Certain settings are read-only depending on the value of other settings. For example, if the **DHCP** client parameter is enabled, the **Phone IP Address** and **Subnet Mask** parameters are grayed out or not visible since the DHCP server automatically supplies these parameters and the statically assigned IP address and subnet mask is not used.

**Tip: Resetting network configurations**

The phone default network configuration referred to in subsequent sections can be reset to factory default settings using the phone's main menu: **Menu > Settings > Advanced > Admin Settings > Reset to Defaults > Reset Device Settings**. You can also use a multiple key combination, as shown in the section Use Multiple Key Combinations.

## Modify Main Menu Settings

You can modify the configuration settings shown in the following table from the setup menu while the phone boots, or from the phone Administrative Settings menu.

**Main Menu**

| Name | Possible Values |
|---|---|
| **Provisioning Menu**<br>Refer to the section Modify Provisioning Server Settings. | |
| **Network Interfaces Menu or Ethernet Menu**<br>Refer to the Modify Ethernet Settings. | |
| **TLS Security Menu**<br>Refer to the section Modify TLS Security Settings. | |
| **SNTP Address** | **IP address or domain name string** |
| The Simple Network Time Protocol (SNTP) server the phone obtains the current time from. | |
| **GMT Offset** | **-13 through +12** |
| The offset of the local time zone from Greenwich Mean Time (GMT) in half hour increments. | |
| **DNS Server** | **IP address** |
| The primary server the phone directs Domain Name System (DNS) queries to. | |
| **DNS AltServer** | **IP address** |

**Main Menu**

| | |
|---|---|
| The secondary server to which the phone directs DNS queries. | |
| **DNS Domain**<br>The phone’s DNS domain. | **Domain name string** |
| **Hostname**<br>The DHCP client hostname. | **hostname** |
| **Syslog Menu**<br>Refer to the section Modify Syslog Settings. | |
| **Quick Setup**<br>If enabled, a QSetup soft key displays on the idle screen when you are in Lines View. When you press this soft key, a menu displays enabling you to configure the parameters required to access the provisioning server.<br>Note: The Quick Setup option is not available in the Updater. | **Enabled, Disabled** |
| **Reset to Defaults**<br>There are five ways to reset or clear phone features and settings, including settings from web or local override files. For details, see the Reset the Phone to Defaults. | |
| **Base Profile**<br>Use this to enable Skype for Business-compatible phones to register with Skype for Business Server. When set to Lync, the phone automatically provisions with the minimum parameters required to register with Skype for Business Server. You cannot modify or customize the Base Profile.<br>By default, the Base Profile for normal SKUs is set to Generic.The Base Profile for Lync and Skype for Business SKUs is Lync. | **Generic, Lync** |

## Modify Provisioning Server Settings

You can modify the configuration settings shown in the table Provisioning Server Menu from the Provisioning Server menu on the phone.

**Note: Change the server user and server password parameters**
For security reasons, always change the Server User and Server Password fields from their default values.

**Provisioning Server Menu**

| Name | Possible Values |
|---|---|
| **DHCP Menu**<br>Refer to the section Modify DHCP Settings. Note: This menu is disabled when the DHCP client is disabled. | |

**Provisioning Server Menu**

<table>
<tr><td><b>Server Type</b></td><td><b>0=FTP, 1=TFTP, 2=HTTP, 3=HTTPS, 4=FTPS</b></td></tr>
<tr><td colspan="2">The protocol that the phone uses to obtain configuration and phone application files from the provisioning server.<br><b>Note</b>: Active FTP is not supported for BootROM version 3.0 or later. Passive FTP is supported. Only implicit FTPS is supported.</td></tr>
<tr><td><b>Server Address</b></td><td><b>IP address or URL</b></td></tr>
<tr><td colspan="2">Domain name string or a URL. All addresses can be followed by an optional directory. The address can also be followed by the file name of a <b>.cfg</b> master configuration file, which the phone uses instead of the default <b><MACaddress>.cfg</b> file. The provisioning server to use if the DHCP client is disabled, if the DHCP server does not send a boot server option, or if the <b>Boot Server</b> parameter is set to <b>Static</b>.<br>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. For information on how to specify a directory and use the master configuration file, see the section Use the Master Configuration File.<br><b>Note</b>: ":", "@", or "/" can be used in the user name or password if they are correctly escaped using the method specified in RFC 1738.</td></tr>
<tr><td><b>Server User</b></td><td><b>String</b></td></tr>
<tr><td colspan="2">The user name requested when the phone logs into the server (if required) for the selected <b>Server Type</b>.<br><b>Note</b>: If the Server Address is a URL with a user name, this is ignored.</td></tr>
<tr><td><b>Server Password</b></td><td><b>String</b></td></tr>
<tr><td colspan="2">The password requested when the phone logs in to the server if required for the selected <b>Server Type</b>.<br><b>Note</b>: If the Server Address is a URL with user name and password, this is ignored.</td></tr>
<tr><td><b>File Transmit Tries</b></td><td><b>1 to 10 Default 3</b></td></tr>
<tr><td colspan="2">The maximum 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.)</td></tr>
<tr><td><b>Retry Wait</b></td><td><b>0 to 300 seconds Default 1</b></td></tr>
<tr><td colspan="2">The minimum amount of time that must elapse before retrying a file transfer. 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. If the set of transactions in an attempt is equal to or greater than the Retry Wait value, then there is no further delay before the next attempt is started.</td></tr>
<tr><td><b>Tag SN to UA</b></td><td><b>Disabled, Enabled</b></td></tr>
<tr><td colspan="2">If enabled, the phone’s serial number (MAC address) is included in the User-Agent header of HTTP/HTTPS transfers and communications to the browser.<br>The default value is Disabled.</td></tr>
<tr><td><b>Upgrade Server</b></td><td><b>String</b></td></tr>
</table>

**Provisioning Server Menu**

| | |
|---|---|
| The address/URL that is accessed for software updates requested from the phone’s Web Configuration Utility. | |
| **ZTP** | **Disabled, Enabled** |
| See Zero-Touch Provisioning Solution on Polycom Support. Also see ZTP Frequently Asked Questions. | |

## Modify DHCP Settings

The DHCP menu is accessible only when the DHCP client is enabled. You can update DHCP configuration settings shown in the table DHCP Menu.

**Note: Multiple DHCP INFORM servers**

If multiple DHCP INFORM servers respond, the phone should gather the responses from these DHCP INFORM servers. If configured for Custom+Option66, the phone selects the first response that contains a valid *custom* option value. If none of the responses contain a *custom* option value, the phone selects the first response that contains a valid *option66* value.

**DHCP Menu**

| Name | Possible Values |
|---|---|
| **Boot Server** | **0=Option 66, 1=Custom, 2=Static, 3=Custom+Option 66** |
| • **Option 66** The phone looks 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 section Modify Provisioning Server Settings.<br>• **Custom** The phone looks 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.<br>• **Static** The phone uses the boot server configured through the *Server Menu*. For more information, see the section Modify Provisioning Server Settings.<br>• **Custom + Option 66** The phone uses the custom option first or use Option 66 if the custom option is not present.<br>Note: If the DHCP server sends nothing, the following scenarios are possible:<br>• 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.<br>• Otherwise the phone sends out a DHCP INFORM query.<br>– If a single DHCP INFORM server responds, this is functionally equivalent to the scenario where the primary DHCP server responds with a valid boot server value.<br>– If no DHCP INFORM server responds, the INFORM query process retries and eventually times out.<br>• If the server address is not discovered using DHCP INFORM then the phone contacts the ZTP server if the ZTP feature is enabled. See ZTP Frequently Asked Questions. | |
| **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 looks for its boot server. | |

**DHCP Menu**

| | |
|---|---|
| **Boot Server Option Type** | **0=IP Address, 1=String** |
| When the *Boot Server* parameter is set to Custom, this parameter specifies the type of DHCP option in which the phone looks for its provisioning server. The IP Address provided must specify the format of the provisioning server. The string provided must match one of the formats described for *Server Address* in the section Modify Provisioning Server Settings. | |
| **Option 60 Format** | **0=RFC 3925 Binary, 1=ASCII String** |
| RFC 3925 Binary: Vendor-identifying information in the format defined in RFC 3925.<br>ASCII String: Vendor-identifying information in ASCII.<br>For more information, see *Using DHCP Vendor Identifying Options with Polycom Phones: Technical Bulletin 54041* at Polycom Engineering Advisories and Technical Notifications.<br>**Note**: DHCP option 125 containing the RFC 3295 formatted data is sent whenever option 60 is sent. DHCP option 43 data is ignored. | |

## Modify Ethernet Settings

The Ethernet Menu is available only if there are multiple network interfaces to the phone.

**Note: LAN port mode**

You can set the LAN Port Mode on all phones. The PC Port Mode parameters are available only on phones with a second Ethernet port.

The following phones support the LAN Port Mode and PC Port Mode setting of 1000FD: VVX 310/311, 410/411, 500/501, 600/601, and 1500.

**Ethernet Menu**

| Name | Possible Values |
|---|---|
| **DHCP** | **Enabled, Disabled** |
| If enabled, DHCP is used to obtain the parameters discussed in the section Modify DHCP Settings. | |
| **IP Address** | **IP address** |
| The phone's IP address. Note: This parameter is disabled when DHCP is enabled. | |
| **Subnet Mask** | **Subnet mask** |
| The phone's subnet mask. Note: This parameter is disabled when DHCP is enabled. | |
| **IP Gateway**<br>The phone's default router. | **IP address** |
| **VLAN**<br>See the section Modify VLAN Settings. | |
| **802.1X Authentication** | **Enabled, Disabled** |

**Ethernet Menu**

| | |
|---|---|
| If enabled, the phone uses the 802.1 Authentication parameters to satisfy the negotiation requirements for each EAP type. | |
| **802.1X Menu**<br>See the section Modify 802.1X Settings. | |
| **Storm Filtering** | **Enabled, Disabled** |
| If enabled, received Ethernet packets are filtered so that the TCP/IP stack does not process bad data or too much data. The default value is Enabled. | |
| **LAN Port Mode** | **0 = Auto, 1 = 10HD, 2 = 10FD, 3 = 100HD, 4 = 100FD, 5 = 1000FD** |
| The network speed over Ethernet. The default value is Auto. HD means half duplex and FD means full duplex.<br>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 Ethernet. The default value is Auto. HD means half duplex and FD means full duplex.<br>Note: Polycom recommends that you do not change this setting unless you want to disable the PC port. | |

## Modify VLAN Settings

You can modify the settings listed in the following table.

**VLAN Menu**

| Name | Possible Values |
|---|---|
| **VLAN ID** | **Null, 0 through 4094** |
| The phone’s 802.1Q VLAN identifier. The default value is Null. Note: Null = no VLAN tagging | |
| **LLDP** | **Enabled, Disabled** |
| If enabled, the phone uses the LLDP protocol to communicate with the network switch for certain network parameters. Most often this is 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.<br>For more information on how to set VLAN and LLDP, refer to the section LLDP and Supported TLVs. | |
| **CDP Compatibility** | **Enabled, Disabled** |
| If enabled, the phone uses CDP-compatible signaling to communicate with the network switch for certain network parameters. Most often this is 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 Discovery** | **0=Disabled, 1=Fixed (default), 2=Custom** |

| | |
|---|---|
| **VLAN Menu** | |
| • **Disabled:** No VLAN discovery through DHCP.<br>• **Fixed:** Use predefined DHCP vendor-specific option values of 128, 144, 157 and 191. If one of these is used, VLAN ID Option is ignored**.**<br>• Custom: Use the number specified for VLAN ID Option as the DHCP private option value.<br>For a detailed description, refer to the section Assign a VLAN ID Using DHCP. | |
| **VLAN ID Option** | **128 through 254 (Cannot be the same as Boot Server Option) (default is 129)** |
| The DHCP private option (when VLAN Discovery is set to Custom).<br>For a detailed description, refer to the section Assign a VLAN ID Using DHCP. | |

## Modify 802.1X Settings

The 802.1X Menu displays on the VVX 500/501 and 600/601 when 802.1X authentication is enabled. You can modify configuration parameters shown in the following table.

**802.1X Menu**

| Name | Possible Values |
|---|---|
| **EAP Method** | **0 = None, 1=EAP-TLS, 2=EAP-PEAPv0/MSCHAPv2, 3=EAP-PEAPv0/GTC, 4=EAP-TTLS/EAP-MSCHAPv2, 5=EAP-TTLS/EAP-GTC, 6=EAP-FAST, 7=EAP-MD5** |
| The selected EAP type to be used for authentication. For more information, see the section Support 802.1X Authentication. | |
| **Identity** | **UTF-8 encoded string** |
| The identity (or user name) required for 802.1X authentication. | |
| **Password** | **UTF-8 encoded string** |
| The password required for 802.1X authentication. The minimum length is 6 characters. | |
| **Anonymous ID** | **UTF-8 encoded string** |
| The anonymous user name for constructing a secure tunnel for tunneled authentication and FAST authentication. | |
| **PAC File Info**<br>See the section Modify PAC File Settings. | |
| **EAP-FAST Inband Provisioning** | **Enabled, Disabled** |
| A flag to determine whether EAP-FAST inband provisioning is enabled. This parameter is used only if EAP method is EAP-FAST. | |

## Modify PAC File Settings

You can modify Protected Access Credential (PAC) File Information settings shown in the following table.

**PAC File Information Menu**

| Name | Possible Values |
|---|---|
| **PAC File Password** | **UTF-8 encoded string** |
| The password required to decrypt the PAC file. | |
| **PAC File Name** | **UTF-8 encoded string** |
| The path or URL of the PAC file for download. | |
| **Remove PAC File** | **UTF-8 encoded string** |
| A flag to determine whether or not to delete the PAC file from the phone. | |

## Modify Login Credential Settings

You can modify settings shown in the following table.

**Login Credentials Menu**

| Name | Possible Values |
|---|---|
| **Domain**<br>The domain name used by a server. | **UTF-8 encoded string** |
| **User**<br>The user name used to authenticate to a server. | **UTF-8 encoded string** |
| **Password**<br>The password used to authenticate to a server. | **UTF-8 encoded string** |

## Modify TLS Security Settings

This section refers to the TLS Security menu available in the Updater and UC Software. You can modify the settings shown in the following table.

**TLS Security Menu**

| Name | Possible Values |
|---|---|
| **OCSP** | **Enabled, Disabled** |
| The Online Certificate Status Protocol checks the revocation status of X.509 digital certificates downloaded during negotiation of a TLS connection. | |

**TLS Security Menu**

| | |
|---|---|
| **FIPS**<br>The Federal Information Processing Standard enables the validation and usage of FIPS-140 certified encryption algorithms. | **Enabled, Disabled** |
| **Install Custom CA Cert**<br>A CA certificate that is installed on the phone to be used for TLS authentication. | **URL** |
| **Install Custom Device Cert**<br>A device certificate installed on the phone to be used for Mutual TLS authentication. | **URL** |
| **Clear Certificate**<br>A flag to determine whether or not the device certificate can be removed from the phone. | **Yes, No** |
| **TLS Profile x**<br>There are currently two TLS Platform profiles. See the section Modify TLS Profile Settings. | |
| **Web Services**<br>See the section Modify Applications Settings. | |

## Modify TLS Profile Settings

You can modify settings shown in the following table.

**TLS Profile Menu**

| Name | Possible Values |
|---|---|
| **SSL Cipher Suite**<br>The global cipher suite. | **String** |
| **Custom SSL Cipher Suite**<br>A custom cipher suite. | **String** |
| **CA Cert List**<br>The CA certificate sources that are valid for this profile. | **String** |
| **Device Cert List**<br>The device certificate sources that are valid for this profile. | **String** |

## Modify Applications Settings

You can modify settings shown in the following table.

**Applications Menu**

| Name | Possible Values |
|---|---|
| **802.1X**<br>The TLS Profile to use for 802.1X authentication. | **1 or 2** |
| **Provisioning**<br>The TLS Profile to use for provisioning authentication. | **1 or 2** |
| **Provisioning**<br>The TLS Profile to enable or disable common name validation. | **Enable or Disable** |
| **Syslog**<br>The TLS Profile to use for syslog authentication. | **1 or 2** |

## Modify Syslog Settings

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 simple 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.

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

**Web Info: Information on Syslog**
For more information on the syslog protocol, see RFC 3164.

You can modify settings shown in the following table.

**Syslog Menu**

| Name | Possible Values |
|---|---|
| **Server Address**<br>The syslog server IP address. The default value is Null. | **IP address or domain name string** |
| **Server Type**<br>The protocol that the phone uses to write to the syslog server. If set to None (or 0), transmission is turned off, but the server address is preserved. | **None=0, UDP=1, TCP=2, TLS=3** |

**Syslog Menu**

| | |
|---|---|
| **Facility** | **0 to 23** |
| A description of what generated the log message. For more information, see section 4.1.1 of RFC 3164.<br>The default value is 16, which maps to local 0. | |
| **Render Level** | **0 to 6** |
| Specifies the lowest class of event that rendered to syslog. It is based on `log.render.level` and can be a lower value. See <log/>. Note: Use left and right arrow keys to change values. | |
| **Prepend MAC Addres** | **Enabled, Disabled** |
| If enabled, the phone’s MAC address is prepended to the log message sent to the syslog server. | |

# Configure Devices and Call Controls

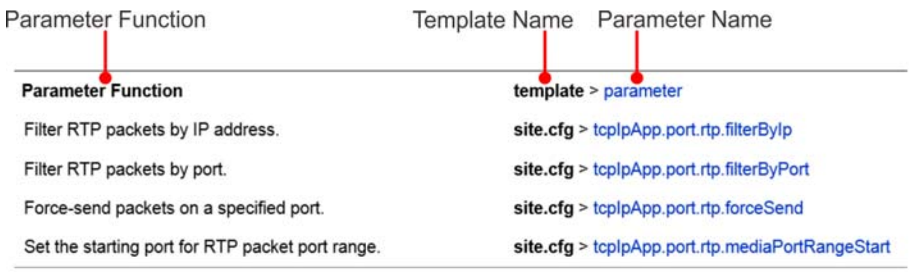
This section provides information to successfully configure the UC Software configuration files and parameters that controls device features and settings.

## How to Read the Feature Parameter Tables

Each of the feature descriptions includes a table of parameters that you configure to make the features work. The following section explains the conventions used in the feature parameter tables.

The feature parameter tables indicate the parameters you can configure for a feature when using the centralized provisioning method. The following illustration shows you how to read the feature parameter tables to locate the function of the parameter, the name of the template file the parameter is in, and the name of the parameter you configure. Each parameter listed in the feature parameter tables is linked to the reference section Configuration Parameters in this guide, which provides a full description, permissible, and default values for each parameter.

**Feature parameter table format**


Parameter Function
Template Name
Parameter Name

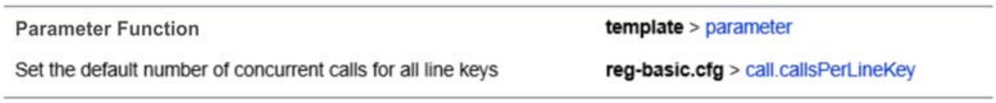
| Parameter Function | template > parameter |
|---|---|
| Filter RTP packets by IP address. | **site.cfg** > tcpIpApp.port.rtp.filterByIp |
| Filter RTP packets by port. | **site.cfg** > tcpIpApp.port.rtp.filterByPort |
| Force-send packets on a specified port. | **site.cfg** > tcpIpApp.port.rtp.forceSend |
| Set the starting port for RTP packet port range. | **site.cfg** > tcpIpApp.port.rtp.mediaPortRangeStart |

To locate a parameter in your UC Software download, open the template name indicated. Then use the parameter name to navigate the folders in the XML tree structure. The parameter name contains the XML folder path. The two following examples explain this convention in more detail.

### Example One: Feature Parameter Tables

The following example is taken from the section Call Forward on Shared Lines.

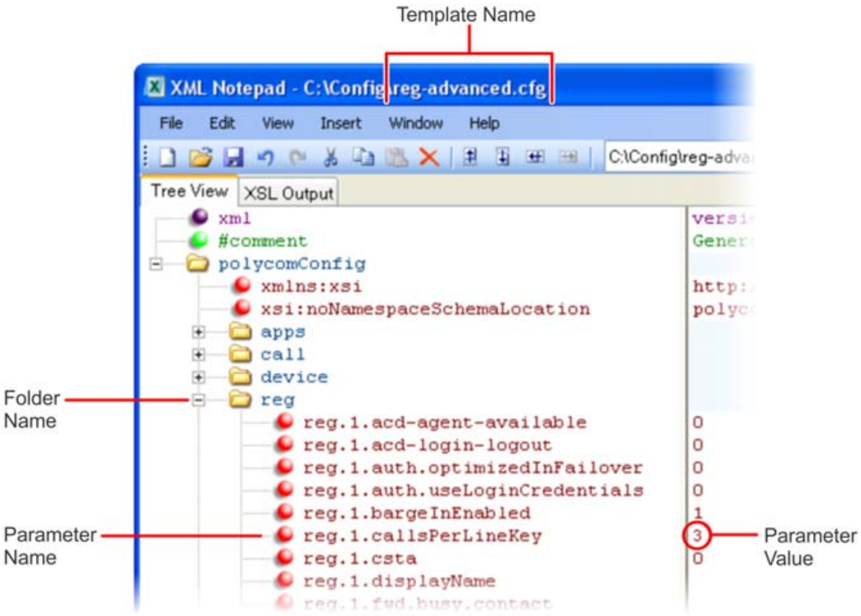
**Feature parameter table for multiple call appearances**



| Parameter Function | template > parameter |
|---|---|
| Set the default number of concurrent calls for all line keys | **reg-basic.cfg** > call.callsPerLineKey |

This example indicates that the **reg-basic.cfg** template file contains the `calls.callsPerLineKey` parameter, which sets the number of concurrent calls for a phone's line keys. Because the default value varies by device, click on the blue parameter name to go to the parameter description, which includes default and permissible values. If you want to change the parameter value, locate and open the reg.basic template, expand the reg folder, and locate the parameter `calls.callsPerLineKey`. Set the parameter value to a permissible number, as shown in the following illustration.

**Example Multiple Call Appearances**



```
xml                                      versi
#comment                                 Gener
polycomConfig
  xmlns:xsi                              http:
  xsi:noNamespaceSchemaLocation          polyc
  apps
  call
  device
  reg
    reg.1.acd-agent-available            0
    reg.1.acd-login-logout               0
    reg.1.auth.optimizedInFailover       0
    reg.1.auth.useLoginCredentials       0
    reg.1.bargeInEnabled                 1
    reg.1.callsPerLineKey                3
    reg.1.csta                           0
    reg.1.displayName
    reg.1.fwd.busy.contact
```

Some file paths in the templates are long and you may have to expand several folders in the XML tree structure to locate a specific parameter.

Note also that some feature parameters are located in more than one template file. In these cases, the parameter tables list all related template files.

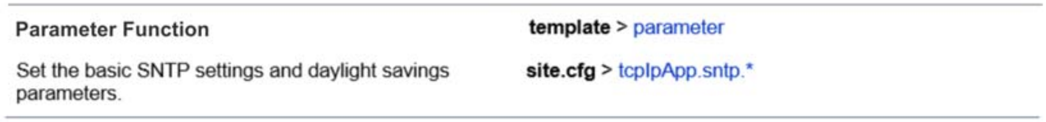
**Note: Each parameter is linked**
Each parameter listed in the tables in various sections is linked to its definition in the section Configuration Parameters. This section defines each parameter and list the permissible values, including the default value, of each parameter. If you want to find out more about a parameter you see listed in the tables, click the blue parameter name.

## Example Two: Configuring Grouped Parameters

Some of the features have several related parameters that you need to configure to get the feature working. In these cases, instead of listing every parameter, the table specifies a group of related parameters with an abbreviated XML path name ending with (.*), which indicates you can configure a group of related parameters.

Abbreviated XML paths, like full parameter names, are linked to their definitions in the reference sections in the section Configuration Parameters. Specifically, since the reference sections lists parameters alphabetically, abbreviated XML path are linked to the first of a group of parameters listed alphabetically in the reference section. The next example shows you that in the **site.cfg** template, the `tcpIpApp.sntp` folder contains several related parameters that configure basic SNTP settings.

**Feature parameter table for time and date SNTP settings**

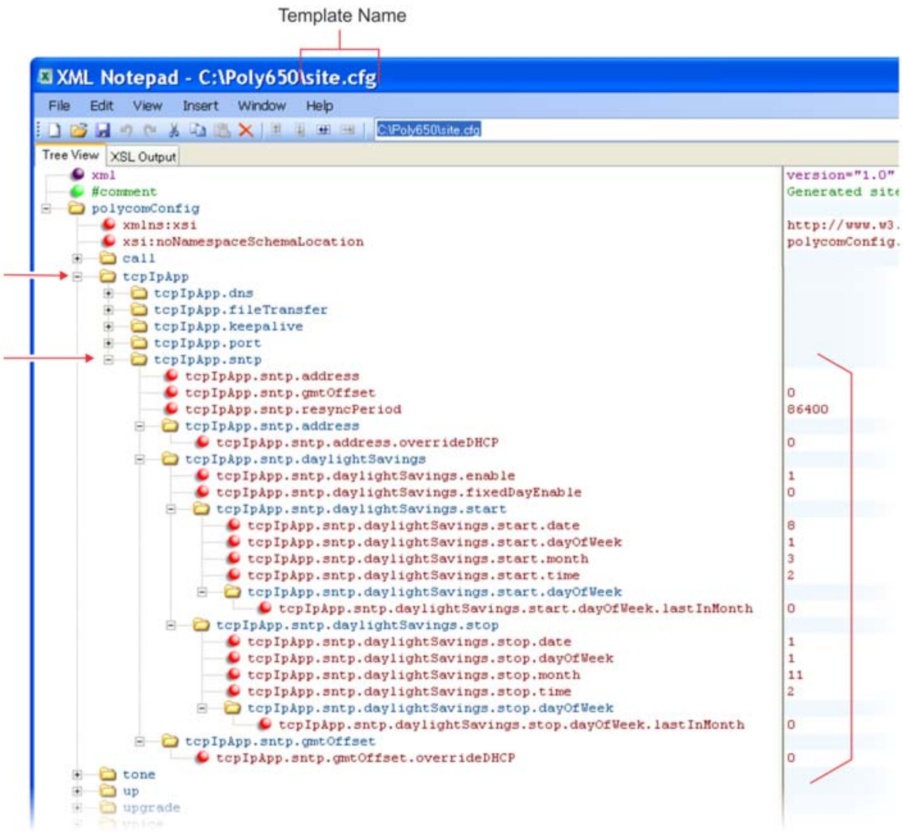


| Parameter Function | template > parameter |
|---|---|
| Set the basic SNTP settings and daylight savings parameters. | **site.cfg** > tcpIpApp.sntp.* |

This example indicates that there is a group of SNTP parameters you can configure in the **site.cfg** template file. The abbreviated parameter name `tcpIpApp.sntp.*` indicates that you can configure parameters in the `tcpIpApp.sntp` folder as well as parameters in `tcpIpApp.sntp` subfolders.

To locate these parameters in the XML file, use the parameter name. The parameter name contains the XML folder path, as shown in the following illustration.

**Locating parameters in the templates**



```
xml                                                              version="1.0"
#comment                                                         Generated site
polycomConfig
  xmlns:xsi                                                      http://www.w3.
  xsi:noNamespaceSchemaLocation                                  polycomConfig.
  call
  tcpIpApp
    tcpIpApp.dns
    tcpIpApp.fileTransfer
    tcpIpApp.keepalive
    tcpIpApp.port
    tcpIpApp.sntp
      tcpIpApp.sntp.address
      tcpIpApp.sntp.gmtOffset                                    0
      tcpIpApp.sntp.resyncPeriod                                 86400
      tcpIpApp.sntp.address
        tcpIpApp.sntp.address.overrideDHCP                       0
      tcpIpApp.sntp.daylightSavings
        tcpIpApp.sntp.daylightSavings.enable                     1
        tcpIpApp.sntp.daylightSavings.fixedDayEnable             0
        tcpIpApp.sntp.daylightSavings.start
          tcpIpApp.sntp.daylightSavings.start.date               8
          tcpIpApp.sntp.daylightSavings.start.dayOfWeek          1
          tcpIpApp.sntp.daylightSavings.start.month              3
          tcpIpApp.sntp.daylightSavings.start.time               2
          tcpIpApp.sntp.daylightSavings.start.dayOfWeek
            tcpIpApp.sntp.daylightSavings.start.dayOfWeek.lastInMonth   0
        tcpIpApp.sntp.daylightSavings.stop
          tcpIpApp.sntp.daylightSavings.stop.date                1
          tcpIpApp.sntp.daylightSavings.stop.dayOfWeek           1
          tcpIpApp.sntp.daylightSavings.stop.month               11
          tcpIpApp.sntp.daylightSavings.stop.time                2
          tcpIpApp.sntp.daylightSavings.stop.dayOfWeek
            tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth    0
      tcpIpApp.sntp.gmtOffset
        tcpIpApp.sntp.gmtOffset.overrideDHCP                     0
  tone
  up
  upgrade
```

**Note: Using an XML editor**

Polycom recommends using an XML editor such as XML Notepad 2007 to open and edit the configuration template files.

# Configure Phone Signaling

This section provides information on configuring phone signaling.

## Quick Setup of Polycom Phones

A quick setup feature simplifies 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 VVX business media phones.

When you enable this feature, a QSetup soft key displays. When you press the QSetup soft key, a new menu displays. The menu enables you to access the provisioning server and quickly configure the phone to work. After configuring the quick setup, you can disable display of the QSetup soft key using a configuration file setting. The following table indicates the parameter that enables this feature.

**Web Info: Configure quick setup**

For details on how to configure quick setup, see *Technical Bulletin 45460: Using Quick Setup with Polycom Phones*.

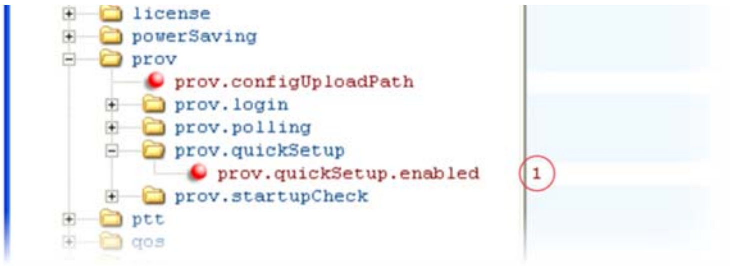
**Quick Setup of Polycom Phones**

| Parameter Function | **template** > parameter |
|---|---|
| To enable or disable Quick Setup. | **site.cfg** > prov.quickSetup.enabled |

## Example Quick Setup Configuration

To display a QSetup soft key on the phone screen and access the Quick Setup menu, enable the `prov.quickSetup.enabled` parameter in the site.cfg template file, as shown next.

**Quick Setup Configuration**



## Configure Real-Time Transport Protocol (RTP) Ports

You can configure the phone to filter incoming RTP packets. You can filter the packets by IP address, or by port. For greater security, you can also configure RTP settings to reject packets arriving from a non-negotiated IP address or from an unauthorized source. You can reject packets that the phone receives from a non-negotiated IP address or a non-negotiated port.

You can configure the phone to enforce symmetric port operation for RTP packets. When the source port is not set to the negotiated remote sink port, arriving packets can be rejected.

You can also fix the phone's destination transport port to a specified value regardless of the negotiated port. This can be useful for communicating through firewalls. When you use a fixed transport port, all RTP traffic is sent to and arrives on that specified port. Incoming packets are sorted by the source IP address and port, which allows multiple RTP streams to be multiplexed.

You can specify the phone's RTP port range. Since the phone supports conferencing and multiple RTP streams, the phone can use several ports concurrently. Consistent with RFC 1889, the next-highest odd-numbered port is used to send and receive RTP. The following table provides a link to the reference section.

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 endpoints 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.

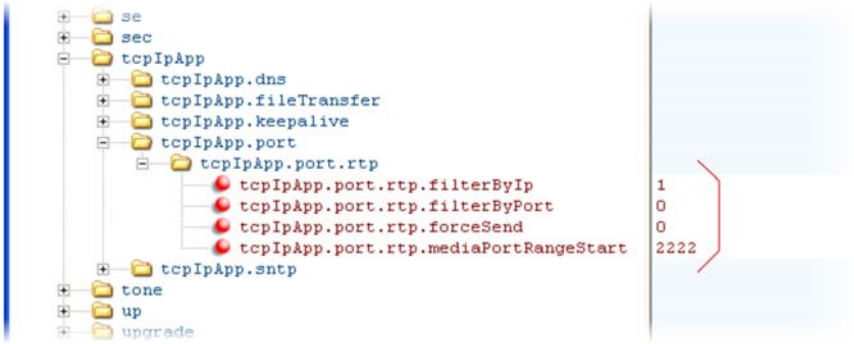
**Configure Real-Time Transport Protocol Ports**

| **Parameter Function** | **template** > parameter |
|---|---|
| Filter RTP packets by IP address. | **site.cfg** > tcpIpApp.port.rtp.filterByIp |
| Filter RTP packets by port. | **site.cfg** > tcpIpApp.port.rtp.filterByPort |
| Force-send packets on a specified port. | **site.cfg** > tcpIpApp.port.rtp.forceSend |
| Set the starting port for RTP packet port range. | **site.cfg** >tcpIpApp.port.rtp.mediaPortRangeStart |

### Example Real-Time Transport Protocol Configuration

The following illustration shows the default real-time transport protocol settings in the **site.cfg** template file. The parameter `tcpIpApp.port.rtp.filterByIp` is set to 1 so that the phone rejects RTP packets sent from non-negotiated IP addresses. The parameter `tcpIpApp.port.rtp.filterByPort` is set to 0 so that RTP packets sent from non-negotiated ports are not rejected. Enter a value in the `tcpIpApp.port.rtp.forceSend` parameter to specify the port that all RTP packets are sent to and received from. The parameter `tcpIpApp.port.rtp.mediaPortrangeStart` shows the default starting port 2222 for RTP packets. The starting port must be entered as an even integer.

**Default real-time transport protocol**



```
se
sec
tcpIpApp
  tcpIpApp.dns
  tcpIpApp.fileTransfer
  tcpIpApp.keepalive
  tcpIpApp.port
    tcpIpApp.port.rtp
      tcpIpApp.port.rtp.filterByIp               1
      tcpIpApp.port.rtp.filterByPort             0
      tcpIpApp.port.rtp.forceSend                0
      tcpIpApp.port.rtp.mediaPortRangeStart      2222
  tcpIpApp.sntp
tone
up
upgrade
```

# Configure Network Address Translation

The phone can work with certain types of Network Address Translation (NAT). NAT enables a local area network (LAN) to use one set of IP addresses for internal traffic and another set for external traffic. The phone's signaling and RTP traffic use symmetric ports. You can configure the external IP address and ports used by the NAT on the phone's behalf on a per-phone basis. The following table lists each of the parameters you can configure. Note that the source port in transmitted packets is the same as the associated listening port used to receive packets.

**Network Access Translation**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the external NAT IP address. | **sip-interop.cfg** > nat.ip |
| Specify the external NAT keepalive interval. | **sip-interop.cfg** > nat.keepalive.interval |
| Specify the external NAT media port start. | **sip-interop.cfg** > nat.mediaPortStart |
| Specify the external NAT signaling port. | **sip-interop.cfg** > nat.signalPort |

## Example Network Address Translation Configuration

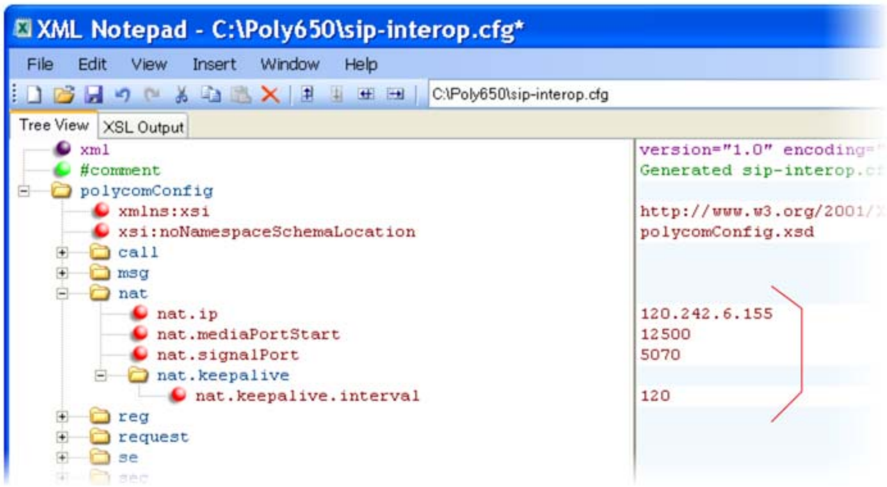
The following illustration shows the default NAT parameter settings. The parameter `nat.ip` is the public IP that you want to advertise in SIP signaling. The default IP is 120.242.6.155.

The parameter `nat.mediaPortStart` is the RTP used to send media. If non-Null, this attribute is set the initially allocated RTP port and overrides the value set in `tcpIpApp.port.rtp.mediaPortRangeStart`. In the example, the starting port is 12500 and the phone cycles through start-port + 47 for phones that support audio only or start-port + 95 for phones that support video.

The parameter `nat.signalPort` specifies the port that the phone uses for SIP signaling. This parameter overrides `voIpProt.local.Port`. In the example below, the phone uses port 5070 for SIP traffic.

Use the `nat.keepalive.interval` to specify the keepalive interval in seconds. This parameter sets the interval at which phones sends a keepalive packet to the gateway/NAT device. The keepalive packet keeps the communication port open so that NAT can continue to function as initially set up. In the example below, the phone sends the keepalive every 120 seconds.

**Default NAT parameter settings**


XML Notepad - C:\Poly650\sip-interop.cfg*
File Edit View Insert Window Help
C:\Poly650\sip-interop.cfg
Tree View
XSL Output

```
xml                                  version="1.0" encoding=
#comment                             Generated sip-interop.c
polycomConfig
    xmlns:xsi                        http://www.w3.org/2001/
    xsi:noNamespaceSchemaLocation    polycomConfig.xsd
    call
    msg
    nat
        nat.ip                       120.242.6.155
        nat.mediaPortStart           12500
        nat.signalPort               5070
        nat.keepalive
            nat.keepalive.interval   120
    reg
    request
    se
    sec
```

## DNS SIP Server Name Resolution

If a DNS name is given for a proxy/registrar address, the IP addresses associated with that name is discovered as specified in RFC3263. If a port is given, the only lookup is an A record. If no port is given, NAPTR and SRV records are 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, port 5060 is used. If the registration type is TLS, port 5061 is used.

**Caution: No DNS resolution causes failover**

Failure to resolve a DNS name is treated as signaling failure that causes a failover.

The following configuration causes the phone to build an SRV request based on the address you provide, including all subdomains. Use the format:

- voIpProt.SIP.outboundProxy.address="<*sip.example.com*>”
- voIpProt.SIP.outboundProxy.port="0"

This SRV request produces a list of servers ordered by weight and priority, enabling you to specify subdomains for separate servers, or you can create partitions of the same system. Please note that while making SRV queries and transport is configured as TCP, the phone adds the prefix `<_service._proto.>` to the configured address/FQDN but does not remove the subdomain prefix, for example `sip.example.com` becomes `_sip._tcp.sip.example.com`. A single SRV query can be resolved into many different servers, session border controllers (SBCs), or partitions ordered by weight and priority, for example, `voice.sip.example.com` and ***video.sip.example.com***. Alternatively, use DNS NAPTR to discover what services are available at the root domain.

## Behavior When the Primary Server Connection Fails

### For Outgoing Calls (INVITE Fallback)

When the user initiates a call, the phone completes the following steps to connect the call:

1. The phone tries to call the working server.
2. If the working server does not respond correctly to the INVITE, the phone tries and makes 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 tries 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 fails.

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

- 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, see <server/> and <reg/>.

**Caution: Use long TTLs to avoid DNS timeout delays**

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 attempts to contact the DNS server to resolve the address of all servers in its list before initiating a call. These attempts timeout, but the timeout mechanism can cause long delays (for example, two minutes) before the phone call proceeds using the working server. To prevent this issue, long TTLs should be used. Polycom recommends deploying an on-site DNS server as part of the redundancy solution.

## Phone Configuration

The phones at the customer site are configured as follows:

- Server 1 (the primary server) is configured with the address of the service provider call server. The IP address of the server(s) is provided by the DNS server, for example: `reg.1.server.1.address=voipserver.serviceprovider.com`.
- Server 2 (the fallback server) is 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: Caution when using multiple servers per registration**

It is possible to configure the phone for more than two servers per registration but ensure that the phone and network load generated by registration refresh of multiple registrations does not become excessive. This is of particular concern when a phone has multiple registrations with multiple servers per registration and some of these servers are unavailable.

## Phone Operation for Registration

After the phone has booted up, it registers to all configured servers.

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

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

**Note: Failover to servers that are not registered**

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

## Recommended Practices for Fallback Deployments

In situations where server redundancy for fallback purpose is used, the following measures should be taken to optimize the solution:

- Deploy an on-site DNS server to avoid long call initiation delays that can result if the DNS server records expire.
- Do not use OutBoundProxy configurations on the phone if the OutBoundProxy could be unreachable when the fallback occurs.
- Avoid using too many servers as part of the redundancy configuration as each registration generates more traffic.
- Educate users as to the features that are not available when in fallback operating mode.

**Note: Compatibility with Microsoft Skype for Business**

The concurrent/registration failover/fallback feature is not compatible with Microsoft environments.

# Configure the Static DNS Cache

Failover redundancy can only be used when the configured IP server hostname resolves (through SRV or A record) to multiple IP addresses. Unfortunately, the DNS cache cannot always be configured to take advantage of failover redundancy.

The solution in SIP 3.1 is to enable you to statically configure a set of DNS NAPTR SRV and/or A records into the phone. See the table Configuring the Static DNS Cache for configurable parameters.

Phones configured with a DNS server behave by default as follows:

- The phone makes an initial attempt to resolve a hostname that is within the static DNS cache. For example, a query is made to the DNS if the phone registers with its SIP registrar.
- 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 again results 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.

Phones not configured with a DNS server behave by default as follows:

- When the phone attempts to resolve a hostname within the static DNS cache, it always returns 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, see RFC2308.

**Configuring the Static DNS Cache**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the line registration. | **sip-interop.cfg** > reg.x.address |
| Specify the call server used for this registration. | **sip-interop.cfg** > reg.x.server.y.* |
| Specify the DNS A address, hostname, and cache time interval. | **site.cfg** > dns.cache.A.x.* |
| Specify the DNS NAPTR parameters, including: name, order, preference, regexp, replacement, service, and ttl. | **site.cfg** > dns.cache.NAPTR.x.* |
| Specify DNS SRV parameters, including: name, port, priority, target, ttl, and weight. | **site.cfg** > dns.cache.SRV.x.* |
| Specify whether to use DNS primary and secondary address set using the parameters `tcpIpApp.dns.server` and `tcpIpApp.dns.altServer`. | **site.cfg** > tcpIpApp.dns.address.overrideDHCP |
| Specify whether to use the DNS domain name set using the parameter `tcpIpApp.dns.domain`. | **site.cfg** > tcpIpApp.dns.domain.overrideDHCP |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

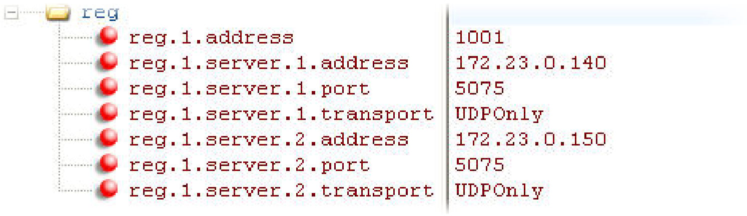
## Example Static DNS Cache Configuration

The following examples show you how to configure the static DNS cache.

### Example 1

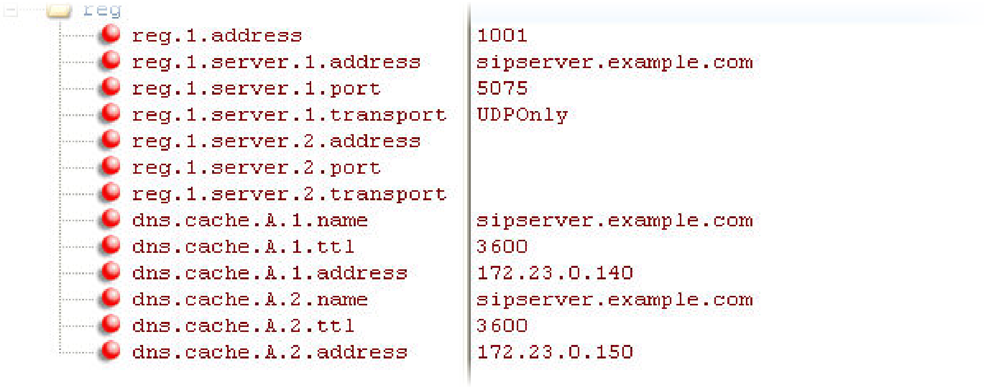
This example shows how to configure static DNS cache using A records IP addresses in SIP server address fields.

When the static DNS cache is not used, the **site.cfg** configuration looks as follows:



```
reg
  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 **site.cfg** configuration looks as follows:



```
reg
  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
```

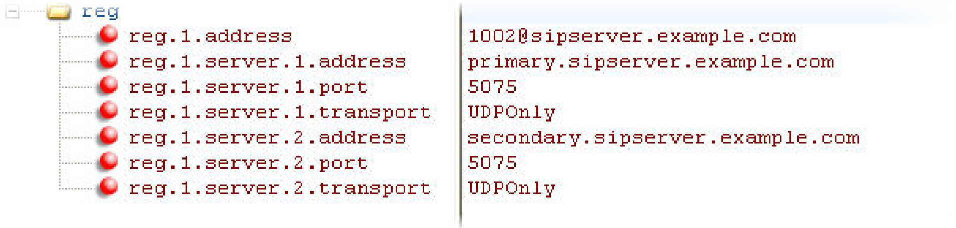
**Note: Order of addresses**

The addresses listed in this example are read by Polycom UC Software in the order listed.

## Example 2

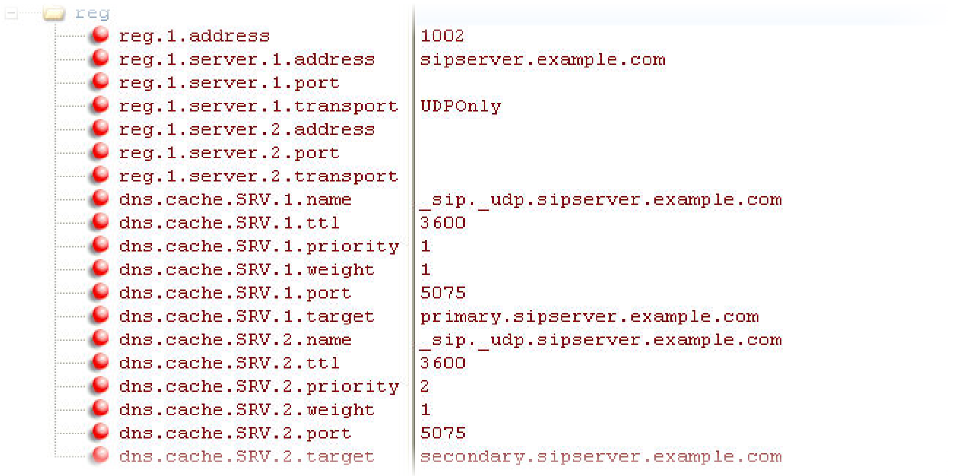
This example shows how to configure static DNS cache where your DNS provides A records for `reg.x.server.x.address` but not SRV. In this case, the static DNS cache on the phone provides SRV records. For more information, see RFC 3263.

When the static DNS cache is not used, the **site.cfg** configuration looks as follows:



```
reg
  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 **site.cfg** configuration looks as follows:



```
reg
  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
```

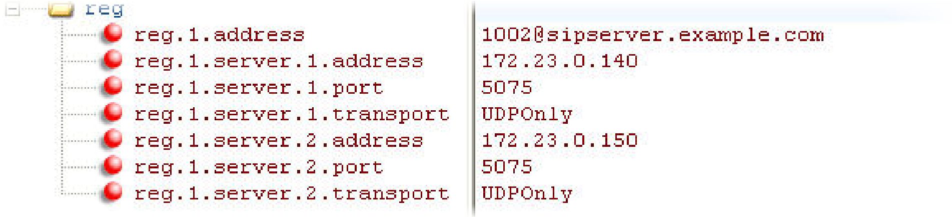
**Note: Port value settings**

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

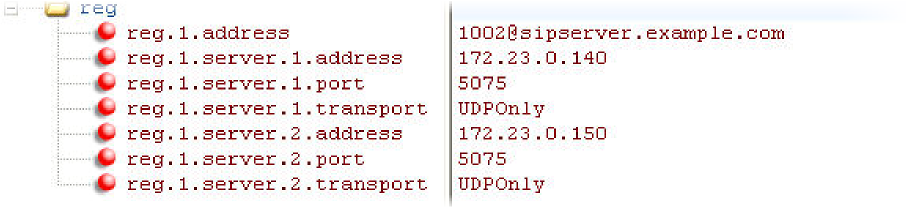
## Example 3

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

When the static DNS cache is not used, the **site.cfg** configuration looks as follows:

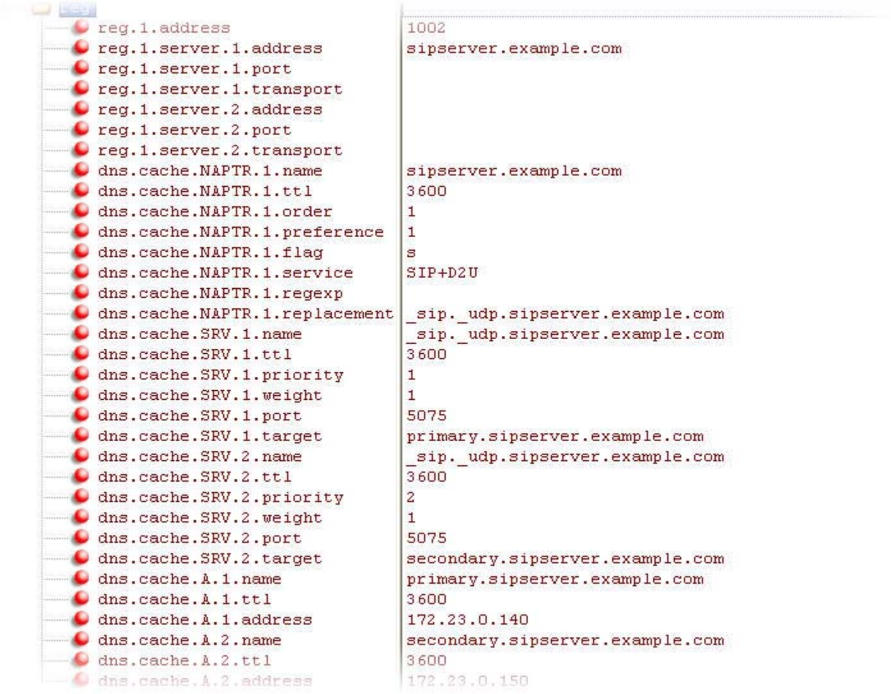


```
reg
  reg.1.address                 1002@sipserver.example.com
  reg.1.server.1.address        172.23.0.140
  reg.1.server.1.port           5075
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address        172.23.0.150
  reg.1.server.2.port           5075
  reg.1.server.2.transport   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



```
reg
  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 **site.cfg** configuration looks as follows:



```
reg
  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
```

**Settings: Forcing NAPTR lookups**

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.

## Enable Access URL in SIP Messages

When this feature is enabled, the server can attach a URL to incoming and active calls. The phone's browser or microbrowser can read this URL and render it as web content that displays on the phone screen. This feature can be enabled on VVX 500/501 and 1500 phones as shown in the table Enable Access URL in SIP Messages.

This feature is flexible and can be used in the following ways.

- A Call Center
  - A URL is attached to an incoming call and displays extended information about a customer before the agent takes the call.
  - The phone can display a script of questions for an agent to ask a caller, and a different script can be provided to different agent groups.
- A Restaurant menu on a hotel phone
  - A guest dials a number for the restaurant or room service and a voice indicates that the menu is available for viewing on the phone.

There are three user interface aspects to this feature:

- **Web Content Status Indication**—When valid web content is available on the phone, an icon displays beside the call information. In the examples shown next, a lightning bolt icon indicates web content is available for a call appearance. The phone user can press the Select key to display the web content.
- **Web Content Retrieval**—Phone users can choose to retrieve web content in Active Mode (spontaneously) or in Passive Mode (by request).
  - **Active Mode**—There are two ways to configure spontaneous web content retrieval: you can set the web content retrieval parameter in the configuration file to 'active' or, if your call server supports access URL, you can specify active retrieval in the SIP heading. If parameters in the SIP signal conflict with the file configuration, parameters in the SIP signaling takes precedence. Note that incoming active web content interrupts web content currently being viewed.
  - **Passive Mode**—There are two ways to configure web content retrieval by request: you can set the web content retrieval parameter in the configuration file to 'passive' or, if your call server supports access URL, you can specify passive retrieval in the SIP heading. When passive mode is enabled, an icon displays beside a call appearance indicating that web content is available. For more information about the web content icon, see Web Content Status Indication earlier in this section. When an icon shows that web content is available, the phone user can press the Select key to view the content. If the web content has expired, no icon displays and the Select key performs no function. Note that incoming active web content interrupts web content currently being viewed. Passive mode is recommended when the microbrowser is used for other applications.
- **Settings Menu**—You can enable new web content to be added to the phone's menu. Using the phone's menu, users can set the default display mode for individual URLs to active or passive.

You must use the following standards if you want to set the retrieval display mode of web content in the SIP headers:

- A new SIP header must be used to report web content associated with SIP phone calls (the SSAWC header follows 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 that begins with the scheme identifier. Currently only the HTTP scheme is supported.

The following is an example of a valid SIP header:

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

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

You may also define two optional parameters:

- An `expires` parameter is defined to indicate the lifespan of the URL itself. Or, if the URL is permanent, you can set how long the web content displays with the call. An absent or invalid parameter is interpreted to mean that the content or the URL itself is persistent. A value, if present, indicates the lifespan of the content in seconds (zero has special significance—see the next example). When the lifespan expires, the phone removes 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. An absent or invalid parameter is interpreted the same as passive, meaning that the web content is retrievable on-demand but is not be spontaneously displayed. If the value is set to active, the web content is spontaneously displayed, subject to the rules discussed under Active Mode in Web Content Retrieval earlier in this section.

  For example:

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

In this case, an icon indicates that web content is available for a period of 60 seconds.

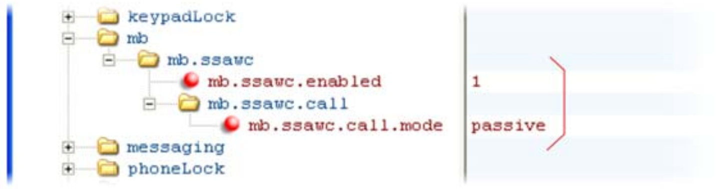
**Enable Access URL in SIP Messages**

| **Parameter Function** | **template** > parameter |
|---|---|
| To turn this feature on or off. | **features.cfg** > mb.ssawc.enabled |
| To retrieve content. | **features.cfg** > mb.ssawc.call.mode |

## Example Access URL in SIP Messages Configuration

In the following example, in the **features.cfg** template, the access URL in SIP message feature is enabled in `mb.ssawc.enabled`. The parameter `mb.ssawc.call.mode` is set to passive, which means web content does not display spontaneously; web content displays when activated by the phone user.

**Access URL in SIP messages example**



```
keypadLock
mb
  mb.ssawc
    mb.ssawc.enabled                1
    mb.ssawc.call
      mb.ssawc.call.mode            passive
messaging
phoneLock
```

# Display SIP Header Warnings

The warning field from a SIP header may be configured to display a three second pop-up message on the phone, for example, that a call transfer failed due to an invalid extension number. For more information, refer to the section Supported SIP Request Headers.

You can display these pop-up messages in any language supported by the phone. The messages display for three seconds unless overridden by another message or action. To turn the warning display on or off or specify which warnings are displayable, you can configure the parameters in the following table.

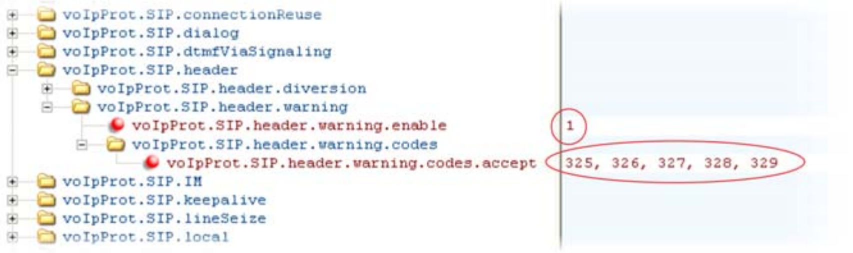
**SIP Header Warnings**

| **Parameter Function** | **template** > parameter |
|---|---|
| Turn this feature on or off. | **sip-interop.cfg** > voIpProt.SIP.header.warning.enable |
| Specify which warnings can be displayed. | **sip-interop.cfg** > voIpProt.SIP.header.warning.codes.accept |

## Example Display of Warnings from SIP Headers Configuration

To enable the display of warnings from SIP headers, set the `voIpProt.SIP.header.warning.enable` parameter in the **features.cfg** template to 1. Enter the warning codes as a comma-separated string. The strings associated with the values 325 to 329 that display on the phone screen, as shown in the next illustration, have been entered automatically by the call server and are not entered by the administrator in the configuration file.

The following illustration shows a sample configuration from the sip-interop.cfg template file:



```
voIpProt.SIP.connectionReuse
voIpProt.SIP.dialog
voIpProt.SIP.dtmfViaSignaling
voIpProt.SIP.header
  voIpProt.SIP.header.diversion
  voIpProt.SIP.header.warning
    voIpProt.SIP.header.warning.enable              1
    voIpProt.SIP.header.warning.codes
      voIpProt.SIP.header.warning.codes.accept      325, 326, 327, 328, 329
voIpProt.SIP.IM
voIpProt.SIP.keepalive
voIpProt.SIP.lineSeize
voIpProt.SIP.local
```

# Set Up Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of phone service if, for example, where the call server needs to be taken offline for maintenance, the server fails, or the connection between the phone and the server fails. The table Set Up Server Redundancy lists parameters you can configure.

Two types of redundancy are possible:

- **Failover**—In this mode, full phone system functionality is preserved by having a second call server of equivalent capability take over from the server that went down/off-line. Use this mode of operation with DNS mechanisms or ‘IP Address Moving’ from the primary to the back-up server.

**Caution: Old failover behavior is not supported**

Prior to SIP 2.1, the `reg.x.server.y parameters` in <reg/> could be used for failover 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 only supports advanced SIP features such as shared lines, missed calls, and presence with the primary server (y=1).

- **Fallback**—In this mode, a second call server of lesser capability (router or gateway device) takes over call control to provide basic calling capability 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. Consult your SIP server provider for recommended methods of configuring phones and servers for failover configuration.

**Note: Compatibility with Microsoft environments**

The concurrent failover/fallback feature is not compatible with Microsoft environments.

**Set Up Server Redundancy**

| Parameter Function | **template** > parameter |
|---|---|
| Specify server redundancy options including failback mode, failback timeout, and failover registration behavior. | **sip-interop.cfg** > voIpProt.server.x.failOver.* |
| Specify which server to contact if failover occurs. | **reg-advanced.cfg** > reg.x.auth.optimizedInFailover |
| Override the default server redundancy options for a specific registration. | **reg-advanced.cfg** > reg.x.outboundProxy.failOver.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

**Web Info: Failover configuration details**

For more information, see *Technical Bulletin 5844: SIP Server Fallback Enhancements on Polycom Phones* and *Technical Bulletin 66546: Configuring Optional Re-Registration on Failover Behavior*.

## Use Presence

The presence feature enables you to monitor the status of other remote users and phones. By adding remote users to your buddy list, you can monitor changes in the status of remote users in real time or you can monitor remote users as speed-dial contacts. You can also manually specify your status in order to override or mask automatic status updates to others and you can receive notifications when the status of your a remote line changes.

VVX phones support a maximum of 64 buddies for Open SIP server platforms and 200 contacts on Skype for Business server. For information on the Skype for Business contacts, refer to the *Deploying Polycom UC Software for Use with Microsoft Skype for Business Server - Deployment Guide*.

The following table lists the parameters you can configure. Note that other phone users can block you from monitoring their phones.

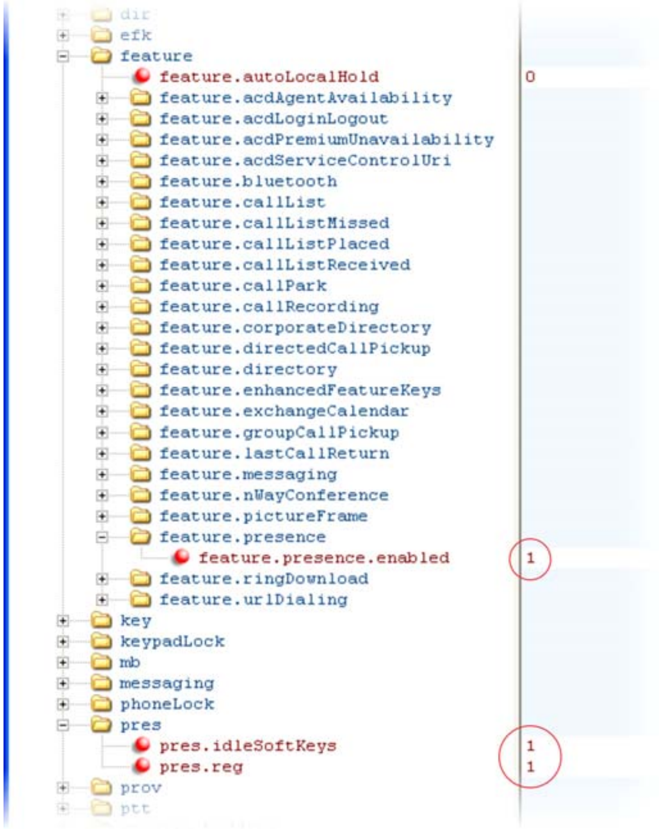
For more information about the Microsoft Skype for Business or BroadSoft UC-One presence features, see the *Polycom VVX Business Media Phones User Guide*.

**Use the Presence Feature**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the line/registration number used to send SUBSCRIBE for presence. | **features.cfg** > pres.reg |
| Specify if the MyStatus and Buddies soft keys display on the Home screen. | **features.cfg** > pres.idleSoftkeys |
| Turn the presence feature on or off. | **features.cfg** > feature.presence.enabled |

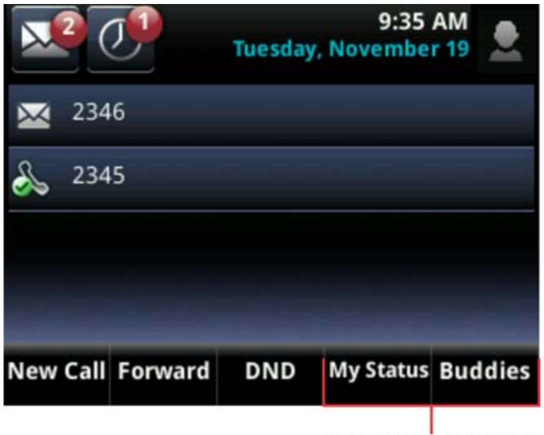
## Example Presence Configuration

In the following illustration, the presence feature has been enabled in `feature.presence.enabled`. The **MyStatus** and **Buddies** soft keys both display on the phone’s home screen when you enable the `pres.idleSoftkeys` parameter. The `pres.reg` parameter uses the address of phone line 1 for the presence feature.



```
dir
efk
feature
    feature.autoLocalHold                         0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
    feature.exchangeCalendar
    feature.groupCallPickup
    feature.lastCallReturn
    feature.messaging
    feature.nWayConference
    feature.pictureFrame
    feature.presence
        feature.presence.enabled                  1
    feature.ringDownload
    feature.urlDialing
key
keypadLock
mb
messaging
phoneLock
pres
    pres.idleSoftKeys                             1
    pres.reg                                      1
prov
ptt
```

This configuration enables the presence feature and display the **MyStatus** and **Buddies** soft keys on the phone. When you press the Buddies soft key, contacts you have entered to your buddy list displays.



Presence Soft Keys

## Provisional Polling of Polycom Phones

You can configure how your phone provisioning automatically by configuring the parameters in the table Provisional Polling of Polycom Phones.

You can set the phone's automatic provisioning behavior to be:

- **Absolute**—The phone polls at the same time every day.
- **Relative**—The phone polls every x seconds, where x is a number greater than 3600.
- **Random**—The phone polls randomly based on a time interval you set.
  - If the time period is less than or equal to one day, the first poll is at a random time, x, between the phone starting up and the polling period. Afterwards, the phone polls every x seconds.
  - If you set the polling period to be greater than one day with the period rounded up to the nearest day, the phone polls on a random day based on the phone's MAC address, and within a random time set by the start and end polling time.

For example:

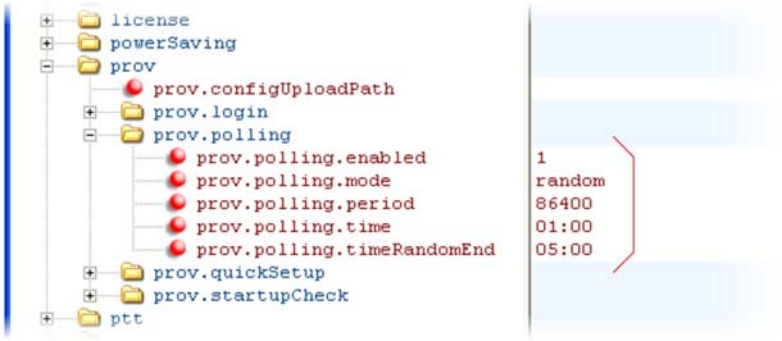
- If `prov.polling.mode` is set to rel and `prov.polling.period` is set to *7200*, the phone polls every two hours.
- If `prov.polling.mode` is set to abs and `prov.polling.timeRandomEnd` is set to *04:00*, the phone polls at 4am every day.
- If `prov.polling.mode` is set to random, prov.polling.period is set to *604800 (7 days)*, `prov.polling.time` is set to `01:00`, `prov.polling.timeRandomEnd` is set to *05:00*, and you have 25 phones, a random subset of those 25 phones, as determined by the MAC address, polls randomly between 1am and 5am every day.
- If `prov.polling.mode` is set to abs and `prov.polling.period` is set to *2328000*, the phone polls every 20 days.

**Provisional Polling of Polycom Phones**

| **Parameter Function** | **template** > parameter |
|---|---|
| To enable polling and set the mode, period, time, and time end parameters. | **site.cfg** > prov.polling.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

## Example Provisional Polling Configuration

The following illustration shows the default sample random mode configuration for the provisional polling feature in the **site.cfg** template file. In this setup, every phone polls once per day, between 1 and 5 am.



```
license
powerSaving
prov
    prov.configUploadPath
    prov.login
    prov.polling
        prov.polling.enabled         1
        prov.polling.mode            random
        prov.polling.period          86400
        prov.polling.time            01:00
        prov.polling.timeRandomEnd   05:00
    prov.quickSetup
    prov.startupCheck
ptt
```

**Note: Only provision files when polling**

If `prov.startupCheck.enabled`='0' then Polycom phones do not look for the sip.ld or the configuration files when they are rebooted, lose power, or restarted. Instead, they look only when receiving a checksync message, a polling trigger, or a manually started update from the menu or web UI.

Some files such as bitmaps, .wav, the local directory, and any custom ringtones are downloaded each time as they are stored in RAM and lost with every reboot.

## Configure SIP Subscription Timers

This feature enables you to configure a subscription expiry independently of the registration expiry. You can also configure an overlap period for subscription independently of the overlap period for the registration, and a subscription expiry and subscription overlap for global SIP servers and per-registration SIP servers. Note that per-registration configuration parameters override global parameters. If you have not explicitly configured values for any user features, the default subscription values are used.

**SIP Subscription Timers**

| **Parameter Function** | **template** > parameter |
|---|---|
| A global parameter that sets the phone’s requested subscription period. | **reg-advanced.cfg** ><br>voIpProt.server.x.subscribe.expires |
| A global parameter that sets the number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. | **reg-advanced-cfg** ><br>voIpProt.server.x.subscribe.expires.overlap |
| A per-registration parameter that sets the phone’s requested subscription period. | **reg-advanced-cfg** ><br>reg.x.server.y.subscribe.expires |
| A per-registration parameter that sets the number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. | **reg-advanced-cfg** ><br>reg.x.server.y.subscribe.expires.overlap |

# Default Features

This section lists phone features for which no configuration is required.

## Call Timer

A call timer displays on the phone's screen. A separate call duration timer displays the hours, minutes, and seconds of each call in progress.

## Called Party Identification

By default, the phone displays and logs the identity of parties called from the phone. The phone obtains called party identity from the network signaling. Because called party identification is a default state, the phone displays caller IDs matched to the call server and does not match IDs to entries in the local contact directory or corporate directory.

## Connected Party Identification

By default, the phone displays and logs the identity of remote parties you connect to if the call server can derive the name and ID from the network signaling. Note that in cases where remote parties have set up certain call features, the remote party you connect to—and the caller ID that displays on the phone—may be different than the intended party. For example, Bob places a call to Alice, but Alice has call diversion configured to divert Bob's incoming calls to Fred. In this case, the phone logs and displays the connection between Bob and Fred. Note that the phone does not match party IDs to entries in the contact directory or the corporate directory.

## Microphone Mute

All phones have a microphone mute button. When you activate microphone mute, a red LED glows or a mute icon displays on the phone screen, depending on the phone model you are using.

No configuration changes can be made to the microphone mute feature.

## Automatic Gain Control

Automatic Gain Control (AGC) is applicable to conference phone models and is used to boost the transmit gain of the local talker in certain circumstances. This increases the effective user-phone radius and helps you to hear all participants equally.

## Background Noise Suppression

Background noise suppression is designed primarily for handsfree operation and reduces background noise, such as from fans, projectors, or air conditioners, to enhance communication.

## Synthesized Comfort Noise Fill

This feature is an integral part of handsfree echo reduction; it is unrelated to Comfort Noise packets generated if Voice Activity Detection is enabled. Synthesized Comfort Noise fill is designed to help provide a consistent noise level to the remote user of a handsfree 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.

### 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 or 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.

# Configure Phone Alerts

This section shows you how to configure phone and call alert features.

## Enable Persistent Mute

This feature enables you to have the mute state of your phone persist across calls. Default mute behavior allows you to activate the mute state only if the phone is in an active call and ends when the active call ends. When you enable this feature and press Mute, the phone stays in the mute state until you press Mute again or until the phone restarts. When you mute the phone in an idle state, the mute LED glows but no icon displays on the screen. When you initiate a new active call with mute on, the mute LED glows and a Mute icon displays on the phone screen.

**Persistent Mute Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the persistent mute feature. | **features.cfg** > feature.persistentMute.enabled |

## Configure the Headset and Speakerphone

All VVX phones come with a handset and a dedicated connector for a headset and include support for a USB headset; all Polycom phones have built-in speakerphones. You can enable and disable each of these options, as shown in the following table. Note that although handsets are shipped with your phones, headsets are not provided.

VVX phones have a dedicated key to switch between speakerphone and headset. You can enable or disable the handsfree speakerphone mode.

**Web Info: Configuring an external electronic hookswitch**

You can configure all supported Polycom desktop phones with an external electronic hookswitch. For more information, see *Technical Bulletin 35150: Using an Electronic Hookswitch with SoundPoint IP and Polycom VVX 1500 Phones*.

**Configure the Headset and Speakerphone**

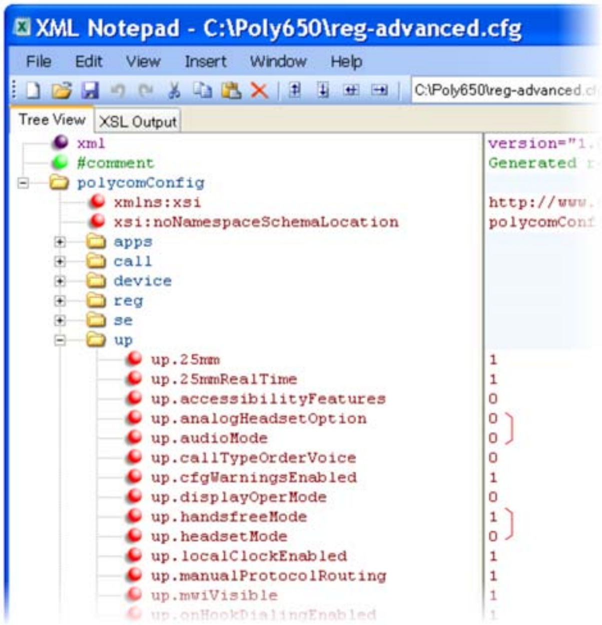
| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable headset memory mode. | **reg-advanced.cfg** and **site.cfg** > up.headsetMode |

**Configure the Headset and Speakerphone  (continued)**

| | |
|---|---|
| Enable or disable handsfree speakerphone mode. | **reg-advanced.cfg** and **site.cfg** > up.handsfreeMode |
| Specify if the electronic hookswitch is enabled and what type of headset is attached. | **reg-advanced.cfg** and **site.cfg** > up.analogHeadsetOption |
| Specify if the handset or a headset should be used for audio. | **reg-advanced.cfg** and **site.cfg** > up.audioMode |
| Specify how phone and the USB headset interact. | **site.cfg** >up.headset.phoneVolumeControl |
| Specify if the USB headset volume persists between calls. | **site.cfg** > voice.volume.persist.headset |

## Example Handset, Headset, and Speakerphone Configuration

The following illustration shows the default settings in the reg-advanced.cfg template. In this example, handsfree mode is enabled and headset memory mode and electronic hookswitch are disabled.


XML Notepad - C:\Poly650\reg-advanced.cfg
File
Edit
View
Insert
Window
Help
C:\Poly650\reg-advanced.c
Tree View
XSL Output

```
xml                                  version="1.
#comment                             Generated 
polycomConfig
    xmlns:xsi                        http://www.
    xsi:noNamespaceSchemaLocation    polycomConf
    apps
    call
    device
    reg
    se
    up
        up.25mm                      1
        up.25mmRealTime              1
        up.accessibilityFeatures     0
        up.analogHeadsetOption       0
        up.audioMode                 0
        up.callTypeOrderVoice        0
        up.cfgWarningsEnabled        1
        up.displayOperMode           0
        up.handsfreeMode             1
        up.headsetMode               0
        up.localClockEnabled         1
        up.manualProtocolRouting     1
        up.mwiVisible                1
        up.onHookDialingEnabled      1
```

## Apply Distinctive Ringing

The distinctive ringing feature enables you to apply a distinctive ringtone to a registered line, a specific contact, or type of call.

There are three ways to set distinctive ringing, and the following table shows you the parameters for each. If you set up distinctive ringing using more than one of the following methods, the phone uses the highest priority method.

- You can assign ringtones to specific contacts in the contact directory. For more information, see Apply Distinctive Incoming Call Treatment. This option is first and highest in priority.
- You can select a ringtone for each registered line on the phone. Press the **Menu** key, and select **Settings > Basic > Ring Type**. This option has the lowest priority.

- You can use the `voIpProt.SIP.alertInfo.x.value` and `voIpProt.SIP.alertInfo.x.class` parameters in the **sip-interop.cfg** template to map calls to specific ringtones. The value you enter depends on the call server. This option requires server support and is second in priority.

**Note:Using the SIP Alert-Info header to delay autoanswer**

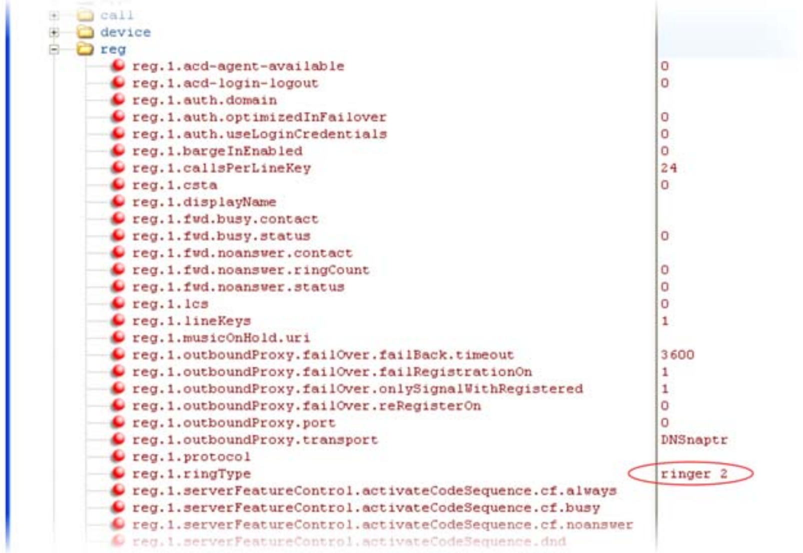
If you set **delay=0** in the **SIP.alert-Info** header, the phone immediately auto-answers incoming calls without ringing. If you set **delay=x** where x=time in seconds, the phone rings for that duration of time before auto-answering incoming calls.

**Apply Distinctive Ringing**

| Parameter Function | **template** > parameter |
|---|---|
| Map the Alert-Info string in the SIP header to ringtones. | **sip-interop.cfg** > voIpProt.SIP.alertInfo.x.class<br>**sip-interop.cfg** > voIpProt.SIP.alertInfo.x.value |
| Specify a ringtone for a specific registered line. | **reg-advanced.cfg** > reg.x.ringType |
| Specify ringtones for contact directory entries. | 000000000000-directory~.xml |

## Example Distinctive Ringing Configuration

The following illustration shows that the ring type ringer2 has been applied to incoming calls to line 1.



```
call
device
reg
  reg.1.acd-agent-available                                     0
  reg.1.acd-login-logout                                        0
  reg.1.auth.domain
  reg.1.auth.optimizedInFailover                                0
  reg.1.auth.useLoginCredentials                                0
  reg.1.bargeInEnabled                                          0
  reg.1.callsPerLineKey                                         24
  reg.1.csta                                                    0
  reg.1.displayName
  reg.1.fwd.busy.contact
  reg.1.fwd.busy.status                                         0
  reg.1.fwd.noanswer.contact
  reg.1.fwd.noanswer.ringCount                                  0
  reg.1.fwd.noanswer.status                                     0
  reg.1.lcs                                                     0
  reg.1.lineKeys                                                1
  reg.1.musicOnHold.uri
  reg.1.outboundProxy.failOver.failBack.timeout                 3600
  reg.1.outboundProxy.failOver.failRegistrationOn               1
  reg.1.outboundProxy.failOver.onlySignalWithRegistered         1
  reg.1.outboundProxy.failOver.reRegisterOn                     0
  reg.1.outboundProxy.port                                      0
  reg.1.outboundProxy.transport                                 DNSnaptr
  reg.1.protocol
  reg.1.ringType                                                ringer 2
  reg.1.serverFeatureControl.activateCodeSequence.cf.always
  reg.1.serverFeatureControl.activateCodeSequence.cf.busy
  reg.1.serverFeatureControl.activateCodeSequence.cf.noanswer
  reg.1.serverFeatureControl.activateCodeSequence.dnd
```

For a list of all parameters and their corresponding ringtones, see Ringtone Pattern Names.

# Configure Do Not Disturb

You can use the do not disturb (DND) feature to temporarily stop incoming calls. You can also turn off audio alerts and receive visual call alerts only, or you can make your phone appear busy to incoming callers. Incoming calls received while DND is turned on are logged as missed.

DND can be enabled locally through the phone or through a server. The table Configure Do Not Disturb lists parameters for both methods. The local DND feature is enabled by default, and you have the option of disabling it. When local DND is enabled, you can turn DND on and off using the Do Not Disturb button on the phone. Local DND can be configured only on a per-registration basis. If you want to forward calls while DND is enabled, see Configure Call Forwarding.

**Note: Using do not disturb on shared lines**

A phone that has DND enabled and activated on a shared line visually alerts you to an incoming call, but the phone does not ring.

If you want to enable server-based DND, you must enable the feature on both a registered phone and on the server. The benefit of server-based DND is that if a phone has multiple registered lines, you can apply DND to all line registrations on the phone; however, you cannot apply DND to individual registrations on a phone that has multiple registered lines. Note that although server-based DND disables the local Call Forward and DND features, if an incoming is not routed through the server, you still receive an audio alert.

Server-based DND behaves the same way as the pre-SIP 2.1 per-registration feature with the following exceptions:

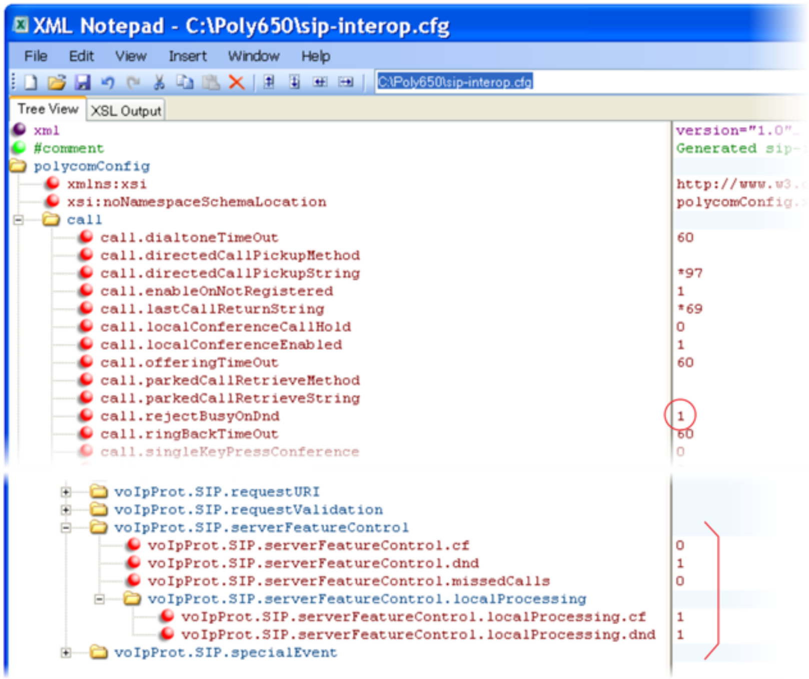
- You cannot enable server-based DND if the phone is configured as a shared line.
- If server-based DND is enabled but not turned on, and you press the DND key or select DND on the phone’s Features menu, the “Do Not Disturb” message displays on the phone and incoming calls continue to ring.

**Configure Do Not Disturb**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable server-based DND. | **sip-interop.cfg** > voIpProt.SIP.serverFeatureControl.dnd |
| Enable or disable local DND behavior when server-based enabled. | **sip-interop.cfg** > voIpProt.SIP.serverFeatureControl.localProcessing.dnd |
| Specify whether, when DND is turned on, the phone rejects incoming calls with a busy signal or gives you a visual and no audio alert. | **sip-interop.cfg** > call.rejectBusyOnDnd |
| Enable DND as a per-registration feature or use it as a global feature for all registrations. | **reg-advanced.cfg** > call.donotdisturb.perReg |

## Example Do Not Disturb Configuration

In the following example, taken from the sip-interop.cfg template, server-based DND has been enabled in `serverFeatureControl.dnd`, and `rejectBusyOnDnd` has been set to 1 so that when you turn on DND on the phone, incoming callers receive a busy signal.


XML Notepad - C:\Poly650\sip-interop.cfg
File Edit View Insert Window Help
C:\Poly650\sip-interop.cfg
Tree View
XSL Output

```
xml                                        version="1.0"...
#comment                                   Generated sip-...
polycomConfig
  xmlns:xsi                                http://www.w3...
  xsi:noNamespaceSchemaLocation            polycomConfig...
  call
    call.dialtoneTimeOut                   60
    call.directedCallPickupMethod
    call.directedCallPickupString          *97
    call.enableOnNotRegistered             1
    call.lastCallReturnString              *69
    call.localConferenceCallHold           0
    call.localConferenceEnabled            1
    call.offeringTimeOut                   60
    call.parkedCallRetrieveMethod
    call.parkedCallRetrieveString
    call.rejectBusyOnDnd                   1
    call.ringBackTimeOut                   60
    call.singleKeyPressConference          0

      voIpProt.SIP.requestURI
      voIpProt.SIP.requestValidation
      voIpProt.SIP.serverFeatureControl
        voIpProt.SIP.serverFeatureControl.cf                        0
        voIpProt.SIP.serverFeatureControl.dnd                       1
        voIpProt.SIP.serverFeatureControl.missedCalls               0
        voIpProt.SIP.serverFeatureControl.localProcessing
          voIpProt.SIP.serverFeatureControl.localProcessing.cf      1
          voIpProt.SIP.serverFeatureControl.localProcessing.dnd     1
      voIpProt.SIP.specialEvent
```

**Note: DND LED alerts on the VVX**

The LED on the Do Not Disturb key on the VVX 1500 is red when pressed or when server-based DND is enabled.

## Configure Call Waiting Alerts

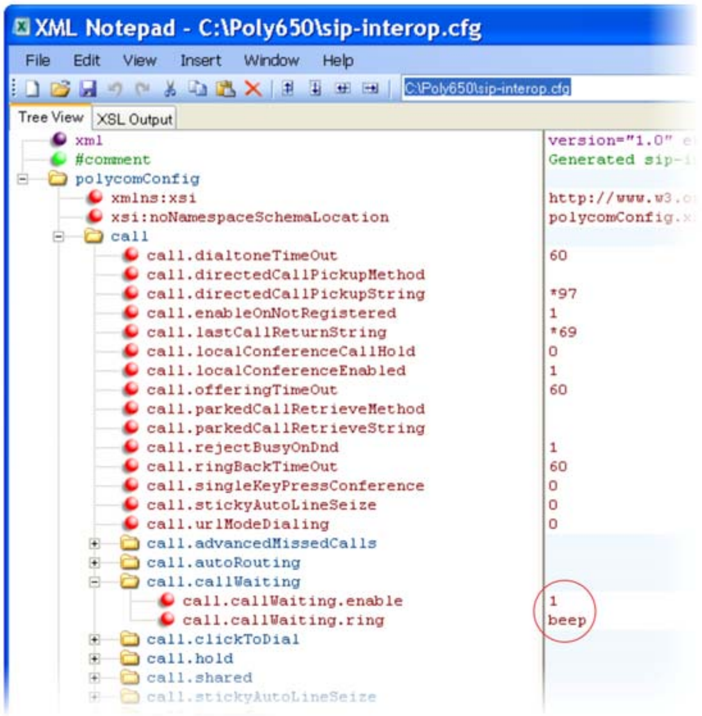
By default, the phone alerts you to incoming calls while you are in an active call. As shown in the table Configuring Call Waiting Alerts, you can disable call waiting alerts and you can specify the ringtone of incoming calls.

**Configuring Call Waiting Alerts**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable call waiting. | **sip-interop.cfg** > call.callWaiting.enable |
| Specify the ringtone of incoming calls when you are in an active call. | **sip-interop.cfg** > call.callWaiting.ring |

### Example Call Waiting Configuration

The following illustration shows you where to disable call waiting alerts and how to change the ringtone of incoming calls in the **sip-interop.cfg** template.


XML Notepad - C:\Poly650\sip-interop.cfg
File
Edit
View
Insert
Window
Help
C:\Poly650\sip-interop.cfg
Tree View
XSL Output

```
xml                                          version="1.0"
#comment                                     Generated sip-
polycomConfig
    xmlns:xsi                                http://www.w3.
    xsi:noNamespaceSchemaLocation            polycomConfig.
    call
        call.dialtoneTimeOut                 60
        call.directedCallPickupMethod
        call.directedCallPickupString        *97
        call.enableOnNotRegistered           1
        call.lastCallReturnString            *69
        call.localConferenceCallHold         0
        call.localConferenceEnabled          1
        call.offeringTimeOut                 60
        call.parkedCallRetrieveMethod
        call.parkedCallRetrieveString
        call.rejectBusyOnDnd                 1
        call.ringBackTimeOut                 60
        call.singleKeyPressConference        0
        call.stickyAutoLineSeize             0
        call.urlModeDialing                  0
        call.advancedMissedCalls
        call.autoRouting
        call.callWaiting
            call.callWaiting.enable          1
            call.callWaiting.ring            beep
        call.clickToDial
        call.hold
        call.shared
        call.stickyAutoLineSeize
```

## Configure Calling Party Identification

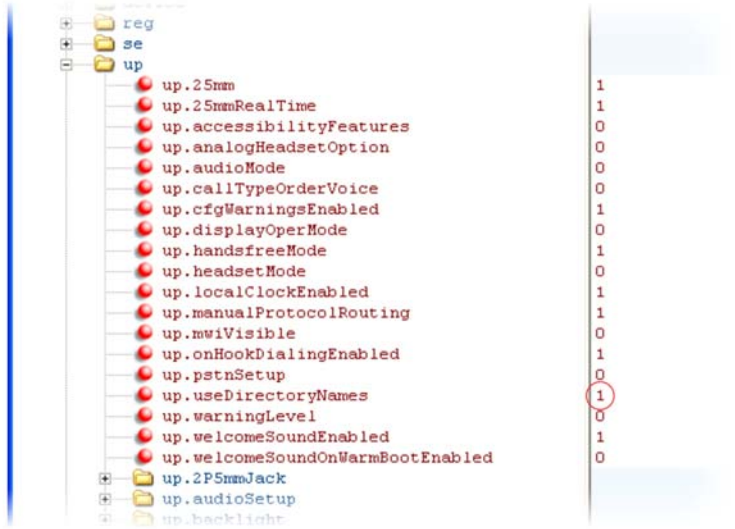
By default, the phone displays the identity of incoming callers if available to the phone through the network signal. If the incoming call address has been assigned to the contact directory, you can choose to display the name you assigned there. Note that the phone cannot match the identity of calling parties to entries in the corporate directory.

### Configure Calling Party Identification

| **Parameter Function** | **template** > parameter |
|---|---|
| Substitute the network address ID with the Contact Directory name. | **reg-advanced.cfg** > up.useDirectoryNames |
| Override the default number of calls per line key for a specific line. | **reg-advanced.cfg** > reg.x.callsPerLineKey |

## Example Calling Party Configuration

The following illustration shows you how to substitute the network address caller ID with the name you assigned to that contact in the contact directory. The ID of incoming call parties displays on the phone screen.



```
reg
se
up
    up.25mm                               1
    up.25mmRealTime                       1
    up.accessibilityFeatures              0
    up.analogHeadsetOption                0
    up.audioMode                          0
    up.callTypeOrderVoice                 0
    up.cfgWarningsEnabled                 1
    up.displayOperMode                    0
    up.handsfreeMode                      1
    up.headsetMode                        0
    up.localClockEnabled                  1
    up.manualProtocolRouting              1
    up.mwiVisible                         0
    up.onHookDialingEnabled               1
    up.pstnSetup                          0
    up.useDirectoryNames                  1
    up.warningLevel                       0
    up.welcomeSoundEnabled                1
    up.welcomeSoundOnWarmBootEnabled      0
    up.2P5mmJack
    up.audioSetup
    up.backlight
```

## Enable Missed Call Notification

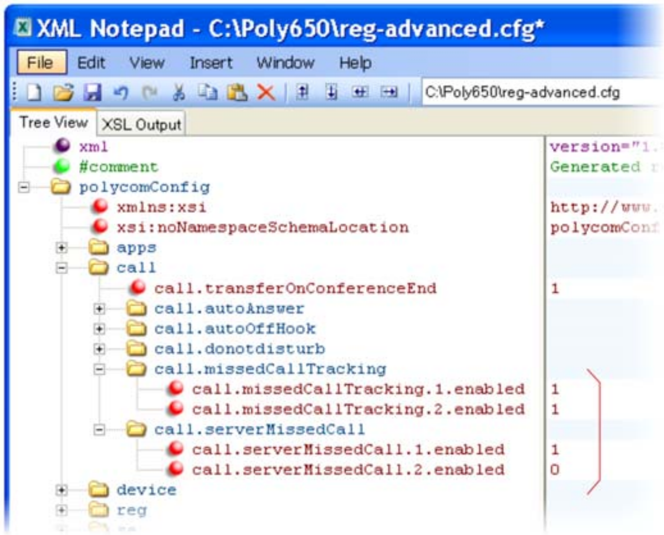
You can display on the phone’s screen a counter that shows the number of missed calls. To reset the counter, view the Missed Calls list on the phone. As the following table indicates, you can also configure the phone to record all missed calls or to display only missed calls that arrive through the SIP server. You can enable missed call notification for each registered line on a phone.

**Enable Missed Call Notification**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the missed call counter for a specific registration. | **reg-advanced.cfg** > call.missedCallTracking.x.enabled |
| Specify, on a per-registration basis, whether to display all missed calls or only server-generated missed calls. | **reg-advanced.cfg** > call.serverMissedCall.x.enabled |

### Example Missed Call Notification Configuration

In the following example, the missed call counter is enabled by default for registered lines 1 and 2, and only server-generated missed calls display on line 1.


XML Notepad - C:\Poly650\reg-advanced.cfg*
File
Edit
View
Insert
Window
Help
C:\Poly650\reg-advanced.cfg
Tree View
XSL Output

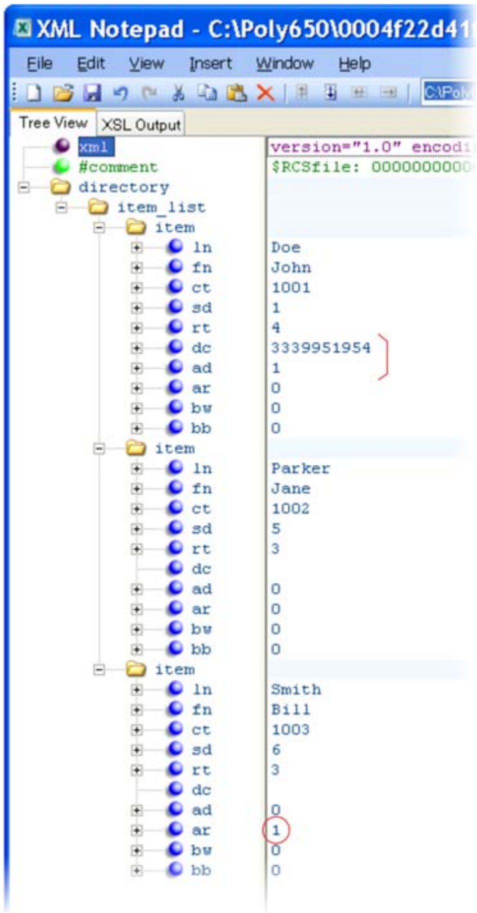
```
xml                                         version="1.
#comment                                    Generated
polycomConfig
    xmlns:xsi                               http://www.
    xsi:noNamespaceSchemaLocation           polycomConf
    apps
    call
        call.transferOnConferenceEnd        1
        call.autoAnswer
        call.autoOffHook
        call.donotdisturb
        call.missedCallTracking
            call.missedCallTracking.1.enabled   1
            call.missedCallTracking.2.enabled   1
        call.serverMissedCall
            call.serverMissedCall.1.enabled     1
            call.serverMissedCall.2.enabled     0
    device
    reg
```

## Apply Distinctive Incoming Call Treatment

You can apply distinctive treatment to specific calls and contacts in your contact directory. You can set up distinctive treatment for each of your contacts by specifying a Divert Contact, enabling Auto-Reject, or by enabling Auto-Divert for a specific contact in the local contact directory (see Use the Local Contact Directory). You can also apply distinctive treatment to calls and contacts through the phone’s user interface.

### Example Call Treatment Configuration

In the following example, the auto divert feature has been enabled in ad so that incoming calls from John Doe are diverted to SIP address *3339951954* as specified in `dc`. Incoming calls from Bill Smith have been set to auto reject in `ar` and are sent to voicemail.


XML Notepad - C:\Poly650\0004f22d41
File Edit View Insert Window Help
C:\Pol
Tree View
XSL Output

```
xml                  version="1.0" encodi
#comment             $RCSfile: 0000000000
directory
  item_list
    item
      ln             Doe
      fn             John
      ct             1001
      sd             1
      rt             4
      dc             3339951954
      ad             1
      ar             0
      bw             0
      bb             0
    item
      ln             Parker
      fn             Jane
      ct             1002
      sd             5
      rt             3
      dc
      ad             0
      ar             0
      bw             0
      bb             0
    item
      ln             Smith
      fn             Bill
      ct             1003
      sd             6
      rt             3
      dc
      ad             0
      ar             1
      bw             0
      bb             0
```

Note that if you enable both the auto divert and auto reject features, auto divert has precedence over auto reject. For a list of all parameters you can use in the contact directory, see the table Understanding the Local Contact Directory.

## Apply Distinctive Call Waiting

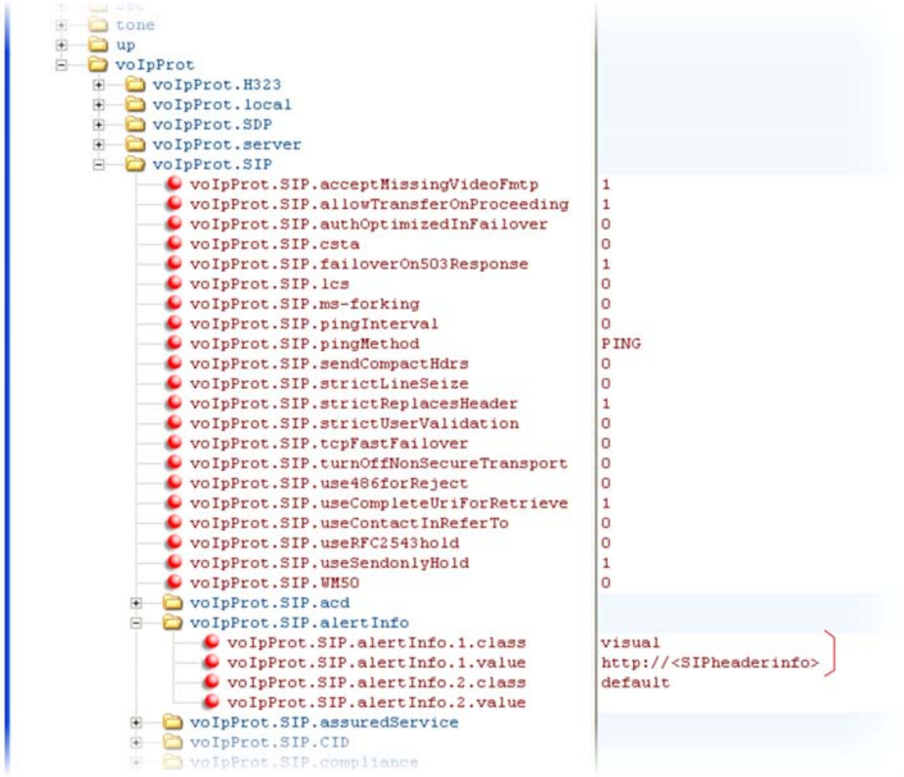
You can use the alert-info values and class fields in the SIP header to map calls to distinct call-waiting types. You can apply three call waiting types: beep, ring, and silent. The following table shows you the parameters you can configure for this feature. This feature requires call server support.

### Apply Distinctive Call Waiting

| **Parameter Function** | **template** > parameter |
|---|---|
| Enter the string which displays in the SIP Alert-Info header. | **sip-interop.cg** > voIpProt.SIP.alertInfo.x.value |
| Enter the ring class name. | **sip-interop.cfg** > voIpProt.SIP.alertInfo.x.class |

## Example Distinctive Call Waiting Configuration

In the following illustration, `voIpProt.SIP.alertInfo.1.value` is set to *http://<SIP headerinfo>*. An incoming call with this value in the SIP Alert-Info header cause the phone to ring in a manner specified by voIpProt.SIP.alertInfo.x.class. In this example, the phone displays a visual LED notification, as specified by the value visual.



```
tone
up
voIpProt
    voIpProt.H323
    voIpProt.local
    voIpProt.SDP
    voIpProt.server
    voIpProt.SIP
        voIpProt.SIP.acceptMissingVideoFmtp          1
        voIpProt.SIP.allowTransferOnProceeding       1
        voIpProt.SIP.authOptimizedInFailover         0
        voIpProt.SIP.csta                            0
        voIpProt.SIP.failoverOn503Response           1
        voIpProt.SIP.lcs                             0
        voIpProt.SIP.ms-forking                      0
        voIpProt.SIP.pingInterval                    0
        voIpProt.SIP.pingMethod                      PING
        voIpProt.SIP.sendCompactHdrs                 0
        voIpProt.SIP.strictLineSeize                 0
        voIpProt.SIP.strictReplacesHeader            1
        voIpProt.SIP.strictUserValidation            0
        voIpProt.SIP.tcpFastFailover                 0
        voIpProt.SIP.turnOffNonSecureTransport       0
        voIpProt.SIP.use486forReject                 0
        voIpProt.SIP.useCompleteUriForRetrieve       1
        voIpProt.SIP.useContactInReferTo             0
        voIpProt.SIP.useRFC2543hold                  0
        voIpProt.SIP.useSendonlyHold                 1
        voIpProt.SIP.WM50                            0
        voIpProt.SIP.acd
        voIpProt.SIP.alertInfo
            voIpProt.SIP.alertInfo.1.class           visual
            voIpProt.SIP.alertInfo.1.value           http://<SIPheaderinfo>
            voIpProt.SIP.alertInfo.2.class           default
            voIpProt.SIP.alertInfo.2.value
        voIpProt.SIP.assuredService
        voIpProt.SIP.CID
        voIpProt.SIP.compliance
```

## Synthesized Call Progress Tones

Polycom phones play call signals and alerts, called call progress tones, such as busy signals, ringback sounds, and call waiting tones. The built-in call progress tones on your phone match standard North American tones. If you would like to customize the phone's call progress tones to match the standard tones in your region, contact Polycom Support.

# Configure the Phone Display

This section provides information on setting up features available on the phone display screen.

## Set the Time and Date Display

A clock and calendar are enabled by default. You can display the time and date for your time zone in several formats, or you can turn it off altogether. You can also set the time and date format to display differently when the phone is in certain modes. For example, the display format can change when the phone goes from idle mode to an active call. You have to synchronize the phone to the Simple Network Time Protocol (SNTP) time server. Until a successful SNTP response is received, the phone continuously flashes the time and date to indicate that they are not accurate.

The time and date display on phones in PSTN mode are set by an incoming call with a supported caller ID standard, or when the phone is connected to Ethernet and you enable the turn on the date and time display.

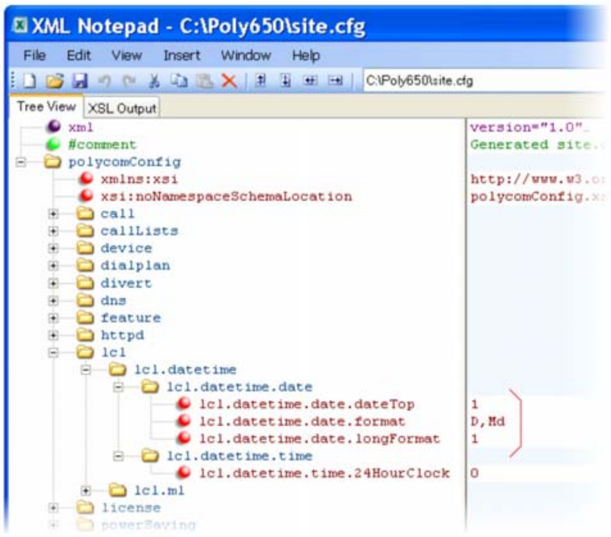
See the following table for basic time and display parameters.

**Set the Time and Date Display**

| **Parameter Function** | **template** > parameter |
|---|---|
| Turn the time and date display on or off. | **reg-advanced.cfg** and **site.cfg** > up.localClockEnabled |
| Set the time and date display format. | **site.cfg** > lcl.datetime.date.* |
| Display time in the 24-hour format. | **site.cfg** > lcl.datetime.time.24HourClock |
| Set the basic SNTP settings and daylight savings parameters. | **site.cfg** > tcpIpApp.sntp.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

### Example Configuration

The following illustration shows an example configuration for the time and date display format. In this illustration, the date is set to display over the time and in long format. The D, Md indicates the order of the date display, in this case, day of the week, month, and day. In this example, the default time format is used, or you can enable the 24-hour time display format.


XML Notepad - C:\Poly650\site.cfg
File Edit View Insert Window Help
C:\Poly650\site.cfg
Tree View
XSL Output

```
xml                                         version="1.0"
#comment                                    Generated site.
polycomConfig
    xmlns:xsi                               http://www.w3.o
    xsi:noNamespaceSchemaLocation           polycomConfig.x
    call
    callLists
    device
    dialplan
    divert
    dns
    feature
    httpd
    lcl
        lcl.datetime
            lcl.datetime.date
                lcl.datetime.date.dateTop       1
                lcl.datetime.date.format        D,Md
                lcl.datetime.date.longFormat    1
            lcl.datetime.time
                lcl.datetime.time.24HourClock   0
        lcl.ml
    license
    powerSaving
```

Use the table Date Formats to choose values for the `lcl.datetime.date.format` and `lcl.datetime.date.longformat` parameters. The table shows values for Friday, August 19, 2011.

**Date Formats**

| lcl.datetime.date.format | lcl.datetime.date.longformat | Date Displayed on Phone |
|---|---|---|
| dM,D | 0 | 19 Aug, Fri |
| dM,D | 1 | 19 August, Friday |
| Md,D | 0 | Aug 19, Fri |
| Md,D | 1 | August 19, Friday |
| D,dM | 0 | Fri, 19 Aug |
| D,dM | 1 | Friday, August 19 |
| DD/MM/YY | n/a | 19/08/11 |
| DD/MM/YYYY | n/a | 19/08/2011 |
| MM/DD/YY | n/a | 08/19/11 |
| MM/DD/YYYY | n/a | 08/19/2011 |
| YY/MM/DD | n/a | 11/08/19 |
| YYYY/MM/DD | n/a | 2011/08/11 |

## Set a Graphic Display Background

You can display a graphic image on the background of the all VVX business media phones and connected VVX Color Expansion Modules. The table Set a Graphic Display Background lists parameters you must configure to set the graphic display background on VVX business media phones and connected expansion modules. Note that the background image you configure displays across the entire phone screen; the time and date, and line key and soft key labels display over the background. If you want the background image to display more visibly from behind line key labels, use `up.transparentLines` to render line key labels transparent - this option is available only on the VVX 500/501 and 600/601 business media phones.

For VVX phones:

- The VVX phones display a default background picture. You can select your own background picture or design, or you can import a custom image. You can also select images from a USB using the picture frame feature as shown in the section Configure the Digital Picture Frame.
- All Polycom devices support JPEG, BMP, and PNG image file formats; progressive/multiscan JPEG images are not supported.
- The maximum image size varies with the phone LCD screen size:

**Phone Screen Sizes**

| Phone | Screen Size |
|---|---|
| VVX 300 series | 208x104 pixels (Grayscale) |
| VVX 400 series | 320x240 pixels |
| VVX 500 series | 320x240 pixels |
| VVX 600 series | 480x272 pixels |
| VVX 1500 | 800x480 pixels |
| VVX Color Expansion Module | 272x480 pixels |

**Web Info: Adding a graphic display background**
For detailed instructions on adding a graphic display to a VVX phone, see the *Polycom VVX Business Media Phones User Guide*.

**Set a Graphic Display Background**

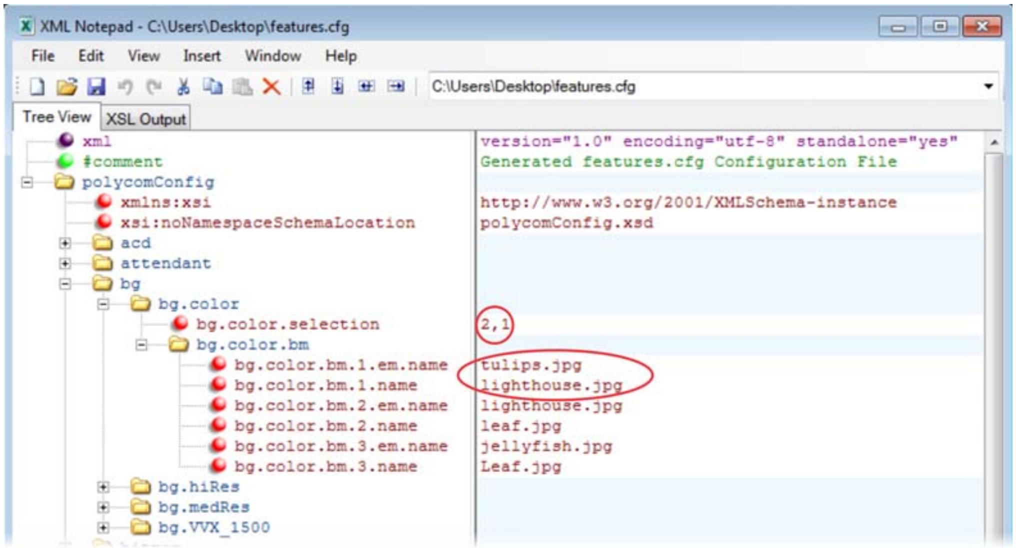
| **Parameter Function** | **template** > parameter |
|---|---|
| Specify a background to display for your phone type. | **features.cfg** > bg.* |
| Enable or disable transparent line key labels on the VVX 500/501 and 600/601. | **features.cfg** > up.transparentLines |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Example Graphic Display Background Configuration

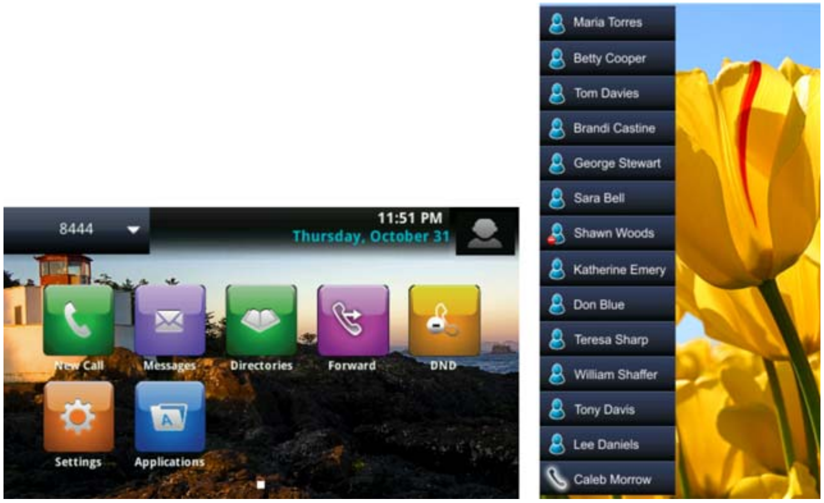
This example configuration shows a background image applied to a VVX phone and expansion module. The default background in the **features.cfg** template file, specified in the `bg.color.selection` parameter, is set to 2,1, where 2 enables background images and 1 selects the image. For example, 1=

bg.color.bm.1.em.name and bg.color.bm.1.name. The phone displays the background image, in this case lighthouse.jpg, and the expansion module displays the tulips.jpg.


XML Notepad - C:\Users\Desktop\features.cfg
File
Edit
View
Insert
Window
Help
C:\Users\Desktop\features.cfg
Tree View
XSL Output

```
xml                                  version="1.0" encoding="utf-8" standalone="yes"
#comment                             Generated features.cfg Configuration File
polycomConfig
  xmlns:xsi                          http://www.w3.org/2001/XMLSchema-instance
  xsi:noNamespaceSchemaLocation      polycomConfig.xsd
  acd
  attendant
  bg
    bg.color
      bg.color.selection             2,1
      bg.color.bm
        bg.color.bm.1.em.name        tulips.jpg
        bg.color.bm.1.name           lighthouse.jpg
        bg.color.bm.2.em.name        lighthouse.jpg
        bg.color.bm.2.name           leaf.jpg
        bg.color.bm.3.em.name        jellyfish.jpg
        bg.color.bm.3.name           Leaf.jpg
    bg.hiRes
    bg.medRes
    bg.VVX_1500
```

This example configuration results in the following graphic display background on the phone and expansion module. Note that line and soft key labels display over the background image.



## Enable Background Image Lock

Administrators can disable the user option, available in the digital picture frame feature, to set images as a background when viewing images on a USB attached to the phone. By default, users can set the background image. Disabling this feature removes the following options when logged into the phone as a user:

- On the phone, the Background menu at **Home > Settings > Basic > Preferences > Background**.
- On the phone, the icon to set image as background at S**ettings > Features > Removable Storage Media > Picture Frame**. On VVX 1500 phones, at **Menu > Features > Removable Storage**.
- On the Web Configuration Utility, the Background option in the Preferences menu.

**Background Image Lock Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the option to set an image as background. | **features.cfg** > bg.background.enabled |

# Configure the Digital Picture Frame

On the VVX phones, you can display a slide show of images on the phone's idle screen. Images must be saved in JPEG, BMP, or PNG format on a directory on a USB device that is attached to the phone. The parameters you can configure are listed in the table Configure the Digital Picture Frame. The phone can display a maximum image size of 9999x9999 pixels and a maximum of 1000 images.

**Note: Maximum image size**

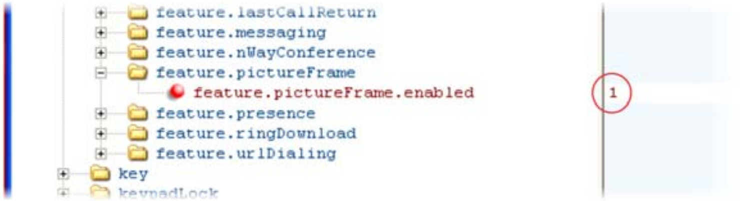
Although 9999x9999 images and progressive/multiscan JPEG images are supported, the maximum image size that can be downloaded is restricted by the available memory in the phone.

**Configure the Digital Picture Frame**

| Parameter Function | **template** > parameter |
|---|---|
| To enable or disable the digital picture frame. | **features.cfg** > feature.pictureFrame.enabled |
| Specify the name of the folder on the USB device containing the images. | **reg-advanced.cfg** > up.pictureFrame.folder |
| Set how long each picture displays. | **reg-advanced.cfg** > up.pictureFrame.timePerImage |

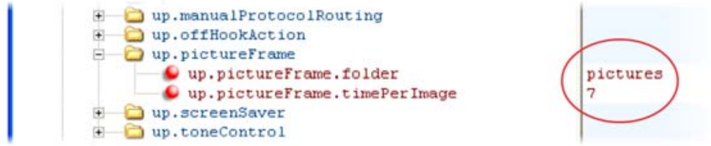
## Example Digital Picture Frame Configuration

In the following illustration, the digital picture frame feature is enabled in the **features.cfg** template file.



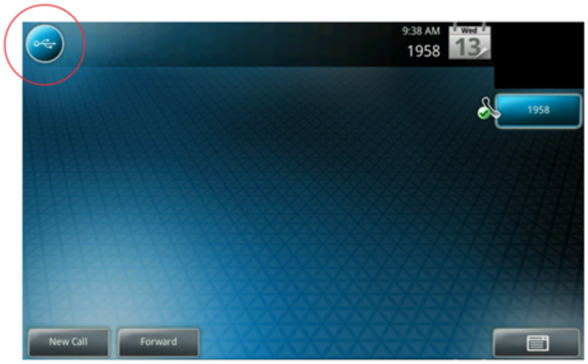
```
feature.lastCallReturn
feature.messaging
feature.nWayConference
feature.pictureFrame
    feature.pictureFrame.enabled        1
feature.presence
feature.ringDownload
feature.urlDialing
key
keypadLock
```

In the **reg-advanced.cfg** template file, the phone looks on the USB device for images in the folder named pictures and each picture displays for seven seconds.



```
up.manualProtocolRouting
up.offHookAction
up.pictureFrame
    up.pictureFrame.folder              pictures
    up.pictureFrame.timePerImage        7
up.screenSaver
up.toneControl
```

After the configuration is complete, restart the phone, insert the USB device to the phone. A Removable Storage Media icon displays on the phone's screen, shown next on the VVX 1500.



To show your pictures, press the icon and press **Picture Frame**.

**Note: Accessing the digital picture frame**

You can access the digital picture frame using *PicFrame:// URL*.

## Set the Phone Language

You can select the language that displays on the phone using the parameters in the table Set the Phone Language. Each language is stored as a language file in the **VVXLocalization** folder. This folder is included with the Polycom UC Software you downloaded to your provisioning server. If you want to edit the language files, you must use a Unicode-compatible XML editor such as XML Notepad 2007 and familiarize yourself with the guidelines on basic and extended character support, see <ml/>.

All phones support the following the following languages: Arabic, Simplified Chinese, Traditional Chinese, Danish, Dutch, English, French, German, Italian, Japanese, Korean, Norwegian, Polish, Brazilian Portuguese, Russian, Slovenian, International Spanish, and Swedish.

**Note: Multilingual support for the updater**

At this time, the updater is available in English only.

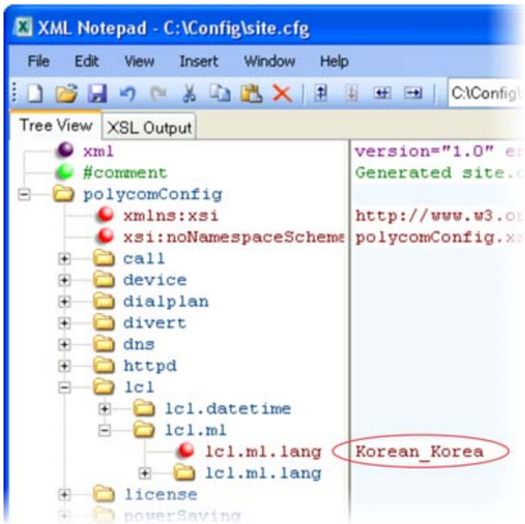
**Set the Phone Language**

| Parameter Function | **template** > parameter |
|---|---|
| Obtain the parameter value for the language you want to display on the phone. | **site.cfg** > lcl.ml.lang.menu.* |
| Specify the language used on the phone's display screen. | **site.cfg** > lcl.ml.lang |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

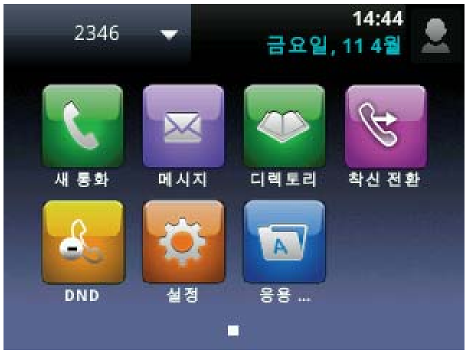
### Example Phone Language Configuration

The following illustration shows you how to change the phone language. Locate the language you want the phone to display in the site.cfg template in `lcl.ml.lang.*` menu. From the list, select the language you want to use and enter it in `lcl.ml.lang`. In the following example, the phone is set to use the Korean language.



```
xml                           version="1.0" e
#comment                      Generated site.
polycomConfig
  xmlns:xsi                   http://www.w3.o
  xsi:noNamespaceScheme       polycomConfig.x
  call
  device
  dialplan
  divert
  dns
  httpd
  lcl
    lcl.datetime
    lcl.ml
      lcl.ml.lang             Korean_Korea
      lcl.ml.lang
  license
  powerSaving
```

After configuration, the phone uses Korean characters.



## Use Pinyin Text Input

Pinyin is the phonetic system used to transcribe Mandarin pronunciation of Chinese into Latin characters. The pinyin text input feature on Polycom VVX business media phones uses Nuance XT9® Smart Input to enable you to enter Chinese characters into text input fields using the phone's dial pad keys or through the onscreen keyboard.

The pinyin text input feature is available on the following Polycom phones:

- VVX 1500 running Polycom UC Software 4.0.2B or later
- VVX 300 series, 400 series, 500 series, and 600 series phones running UC Software 5.0.0 or later

You can enter pinyin text using the pinyin text input widget or the onscreen keyboard. To use the pinyin text input feature on Polycom phones, administrators must download a license key to the user's phone.

**Web Info: Pinyin text input**

For complete information on the pinyin text input feature, see the *Polycom VVX Business Media Phones User Guide*.

## Unique Line Label for Registration Lines

Administrators can configure line registrations to display for multiple line keys. When using this feature with the parameter reg.x.label.y and x=2 or higher, multiple line keys display for the registered line address. If you configure the line to display on multiple line keys without a unique label assigned to each line, the lines are labeled automatically in numeric order. For example, if you have four line keys for line 4144 labeled Polycom, the line keys are labeled as 1_Polycom, 2_ Polycom, 3_ Polycom, and 4_ Polycom. This also applies to lines without labels.

### Configure Unique Line Labels

| **Parameter Function** | **template** > parameter |
|---|---|
| Configure a unique line label for multiple line keys. | **reg-advanced.cfg, site.cfg** > reg.x.line.y.label |
| Determines the label that displays on the line key. | **features.cfg** > up.cfgLabelElide |
| Determines the label that displays on the line key. | **features.cfg** > up.cfgUniqueLineLabel |

## Set Patterns for LED Indicators

The LED indicators on VVX phones and expansion modules alert users to the different states of the phone and remote contacts. You can turn LED indicators on or off, and set the pattern, color, and duration of a pattern for all physical keys on the phones.

You can set the pattern, color, and duration for the following LED indicators on VVX phones:

- Line keys
- Message Waiting Indicator (MWI)
- Headset key (excluding VVX 101 and 201)

The following table lists parameters that enable you to set the pattern state, color, and duration of the LED indicators on VVX phones and expansion modules.

### Configure the LED Indicator Patterns

| **Parameter Function** | **template** > parameter |
|---|---|
| Turns the LED indicator on or off depending on the pattern's state. | **features.cfg** > ind.pattern.x.step.y.state |
| Indicates the color of the line key LED indicators. | **features.cfg** > ind.pattern.x.step.y.color |
| Sets the duration of the pattern of the LED indicator. | **features.cfg** > ind.pattern.x.step.y.duration |

In the LED indicator pattern parameters, *x* is the pattern type and *y* is the pattern number. For y, enter a value of 1-20 to indicate the pattern number. For x, you can enter one of the values in the following table to indicate the LED indicator pattern type.

**LED Indicator Pattern Type**

| Pattern Type | Function |
|---|---|
| powerSaving | Sets the behavior for Message Waiting Indicator when the phone is in Power Saving mode. |
| active | Sets the pattern for line keys during active calls. |
| on | Turns on the LED indicator pattern. |
| off | Turns off the LED indicator pattern. |
| offering | Sets the pattern for line keys during incoming calls. |
| flash | Sets the pattern for line keys during held calls and the Message Waiting Indicator when there are unread voicemail messages. |
| lockedOut | Sets the pattern for line keys when a remote party is busy on a shared line. |
| FlashSlow | Sets the pattern for the Headset key when Headset Memory Mode is enabled. |
| held | Sets the pattern for line keys during a held call. |
| remoteBusyOffering | Sets the pattern for line keys for monitored BLF contacts. |

## LED Pattern Examples

This section includes example configurations you can use to set the patterns of LED indicators for VVX phones and expansion modules.

### Disable the Headset Key LED in Headset Memory Mode

By default, the Headset key on all VVX phones, excluding VVX 101 and 201, glows green for analog headsets and blue for USB headsets. The Headset key also flashes by default if Headset Memory Mode is enabled. You can disable and turn off the flash pattern for the Headset key when Headset Memory Mode is enabled.

By default, the following parameters set the behavior of the MWI during Power Saving mode:

- `ind.pattern.flashSlow.step.1.state =` 1—turns on the LED indicator.
- `ind.pattern.flashSlow.step.1.duration` = 100—sets the duration of the pattern.
- `ind.pattern.flashSlow.step.2.state` = 0—turns off the LED indicator for the second step.
- `ind.pattern.flashSlow.step.2.duration` = 2900 —sets the duration for how long the LED indicator is off before the pattern repeats.

### To disable the flash pattern for the Headset key:

» Set the parameter `ind.pattern.flashSlow.step.1.state` to 0.

### Turn Off the Message Waiting Indicator in Power Saving Mode

When Power Saving mode is enabled, the screen darkens, and the MWI flashes red. By default, the powerSaving pattern has two steps before the pattern is repeated: a quick on period and then a long off period. You can turn off the MWI or change the duration of the pattern steps.

By default, the following parameters set the behavior of the MWI during Power Saving mode:

- `ind.pattern.powerSaving.step.1.state =` 1 —turns on the LED indicator.
- `ind.pattern.powerSaving.step.1.duration` = 100—sets the duration of the pattern.
- `ind.pattern.powerSaving.step.2.state` = 0— turns off the LED indicator for the second step.
- `ind.pattern.powerSaving.step.2.duration` = 2900 —sets the duration for how long the LED indicator is off before the pattern repeats.

### To disable the pattern for the MWI in Power Saving mode:

» Set the parameter `ind.pattern.powerSaving.step.1.state` to 0.

### Change the Color of Line Key Indicators for Incoming Calls

When a phone receives an incoming call, the line key LED indicator flashes green. You can change the color of the indicator to Yellow or Red for incoming calls.

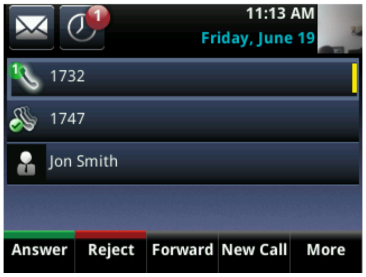
By default, the following parameters set the behavior of the line key LED indicators for incoming calls:

- `ind.pattern.offering.step.1.state` = 1—turns on the LED indicator.
- `ind.pattern.offering.step.1.duration` = 250—sets the duration of the pattern.
- `ind.pattern.offering.step.1.color` = Green—sets the color of the LED indicator for the pattern.
- `ind.pattern.offering.step.2.state`= 0— turns off the LED indicator for the second step
- `ind.pattern.offering.step.2.duration`= 250—sets the duration for how long the LED indicator is off before the pattern repeats.

### To change the color of the line key indicator:

» Set the parameter `ind.pattern.offering.step.1.color` to Yellow.

The following figure shows a Yellow line key LED indicator for an incoming call on a VVX 500/501 phone.



# Configure Call Controls

This section provides information on configuring general phone call controls.

## Enable Last Call Return

The phone supports redialing of the last received call. The following table shows you the parameters to enable this feature. This feature requires 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.

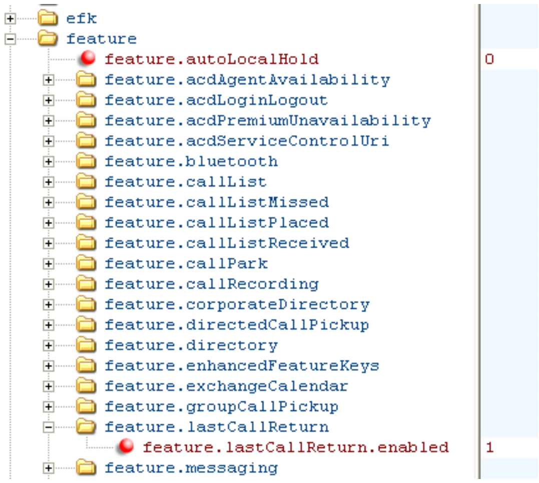
**Enable Last Call Return**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable last call return. | **features.cfg** > feature.lastCallReturn.enabled |
| Specify the string sent to the server for last-call-return. | **sip-interop.cfg** > call.lastCallReturnString |

### Example Configuration for Last Call Return

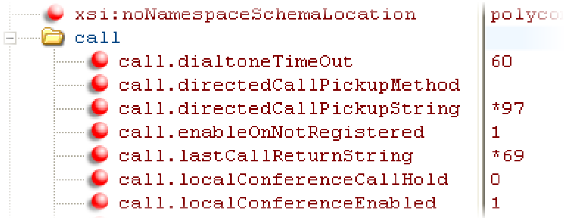
The configuration parameters for last call return feature are located in two template files. You can enable the feature using the **features.cfg** template file and configure the feature using the **sip-interop.cfg** file.

In the following configuration example, the last call return feature has been enabled in the features.cfg template file:



```
efk
feature
    feature.autoLocalHold                    0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
    feature.exchangeCalendar
    feature.groupCallPickup
    feature.lastCallReturn
        feature.lastCallReturn.enabled       1
    feature.messaging
```

When last call return is enabled, you can configure the feature using parameters located in the **sip-interop.cfg** template file. The following shows the default value for the `call.lastCallReturnString` parameter. The last call return string value depends on the call server you use. Consult with your call server provider for the last call return string.



```
xsi:noNamespaceSchemaLocation    polyco
call
    call.dialtoneTimeOut           60
    call.directedCallPickupMethod
    call.directedCallPickupString  *97
    call.enableOnNotRegistered     1
    call.lastCallReturnString      *69
    call.localConferenceCallHold   0
    call.localConferenceEnabled    1
```

When you enable the last call return feature, the phone displays an **LCR** soft key when it goes off-hook. When you press the **LCR** soft key, you place a call to the phone address that last called you.

When you select **Last Call Return**, you place a call to the phone address that last called you.

## Configure Call Hold

The purpose of call hold is to pause activity on one call so that you can use the phone for another task, for example, to place or receive another call or to search your phone’s menu for information. See the following table for a list of available parameters you can configure for this feature. When you place an active call on hold, a message informs the held party that they are on hold. You can also configure a call hold alert to remind you after a period of time that a call is still on hold.

As of SIP 3.1, if supported by the call server, you can enter a music-on-hold URI. For more information, see RFC Music on Hold draft-worley-service-example.

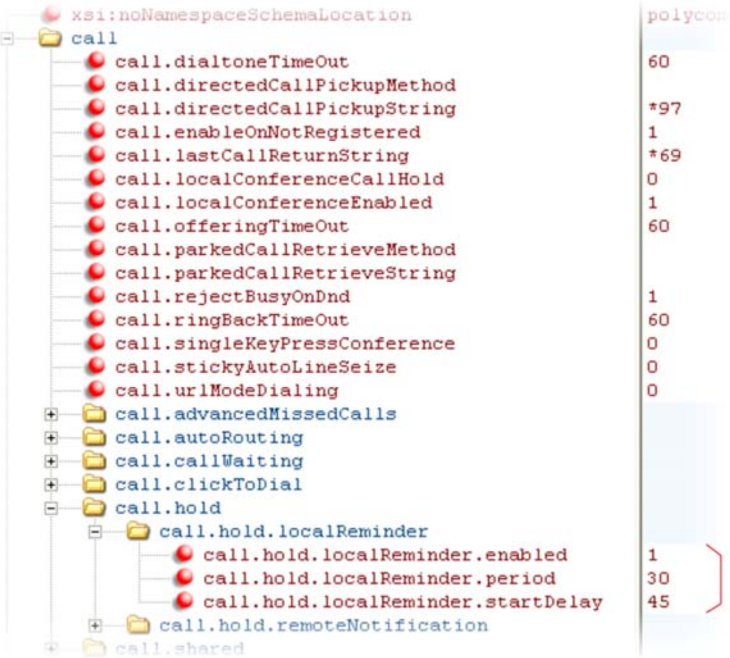
**Enable Call Hold**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify whether to use RFC 2543 (c=0.0.0.0) or RFC 3264 (a=sendonly or a=inactive) for outgoing hold signaling. | **sip-interop.cfg** > voIpProt.SIP.useRFC2543hold |
| Specify whether to use sendonly hold signaling. | **sip-interop.cfg** > voIpProt.SIP.useSendonlyHold |
| Configure local call hold reminder options. | **sip-interop.cfg** > call.hold.localReminder.* |
| Specify the music-on-hold URI. | **sip-interop.cfg** > voIpProt.SIP.musicOnHold.uri |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

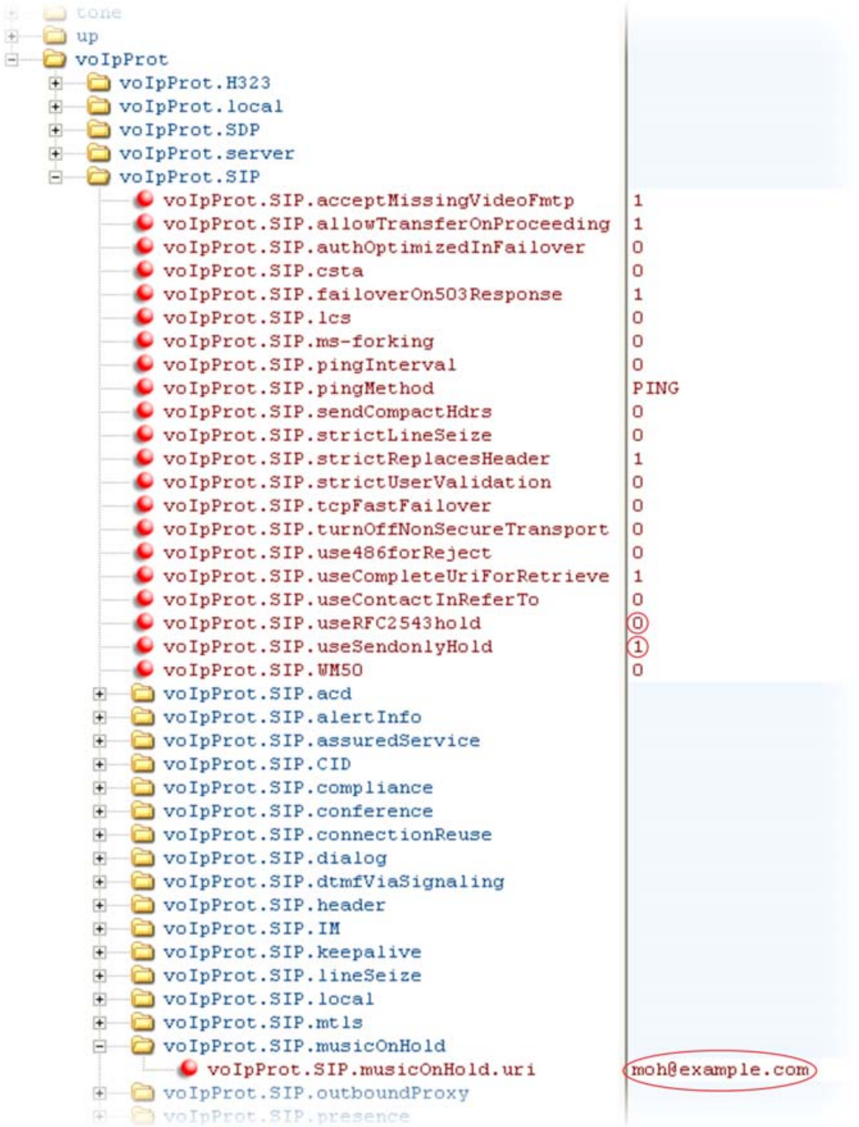
### Example Call Hold Configuration

The following two illustrations show a sample configuration for the call hold feature. Both illustrations are taken from the **sip-interop.cfg** template. In the first illustration, the three `localReminder.*` parameters have been configured to play a tone to remind you of a party on hold, that the tone begins to play 45 seconds after you put a party on hold, and that the tone repeats every 30 seconds.



```
xsi:noNamespaceSchemaLocation                 polycom
call
    call.dialtoneTimeOut                      60
    call.directedCallPickupMethod
    call.directedCallPickupString             *97
    call.enableOnNotRegistered                1
    call.lastCallReturnString                 *69
    call.localConferenceCallHold              0
    call.localConferenceEnabled               1
    call.offeringTimeOut                      60
    call.parkedCallRetrieveMethod
    call.parkedCallRetrieveString
    call.rejectBusyOnDnd                      1
    call.ringBackTimeOut                      60
    call.singleKeyPressConference             0
    call.stickyAutoLineSeize                  0
    call.urlModeDialing                       0
    call.advancedMissedCalls
    call.autoRouting
    call.callWaiting
    call.clickToDial
    call.hold
        call.hold.localReminder
            call.hold.localReminder.enabled       1
            call.hold.localReminder.period        30
            call.hold.localReminder.startDelay    45
        call.hold.remoteNotification
    call.shared
```

In the second illustration, the `musicOnHold.uri` parameter has been configured so the party on hold hears music played from SIP URI *moh@example.com*.



```
tone
up
voIpProt
  voIpProt.H323
  voIpProt.local
  voIpProt.SDP
  voIpProt.server
  voIpProt.SIP
    voIpProt.SIP.acceptMissingVideoFmtp          1
    voIpProt.SIP.allowTransferOnProceeding       1
    voIpProt.SIP.authOptimizedInFailover         0
    voIpProt.SIP.csta                            0
    voIpProt.SIP.failoverOn503Response           1
    voIpProt.SIP.lcs                             0
    voIpProt.SIP.ms-forking                      0
    voIpProt.SIP.pingInterval                    0
    voIpProt.SIP.pingMethod                      PING
    voIpProt.SIP.sendCompactHdrs                 0
    voIpProt.SIP.strictLineSeize                 0
    voIpProt.SIP.strictReplacesHeader            1
    voIpProt.SIP.strictUserValidation            0
    voIpProt.SIP.tcpFastFailover                 0
    voIpProt.SIP.turnOffNonSecureTransport       0
    voIpProt.SIP.use486forReject                 0
    voIpProt.SIP.useCompleteUriForRetrieve       1
    voIpProt.SIP.useContactInReferTo             0
    voIpProt.SIP.useRFC2543hold                  0
    voIpProt.SIP.useSendonlyHold                 1
    voIpProt.SIP.WM50                            0
    voIpProt.SIP.acd
    voIpProt.SIP.alertInfo
    voIpProt.SIP.assuredService
    voIpProt.SIP.CID
    voIpProt.SIP.compliance
    voIpProt.SIP.conference
    voIpProt.SIP.connectionReuse
    voIpProt.SIP.dialog
    voIpProt.SIP.dtmfViaSignaling
    voIpProt.SIP.header
    voIpProt.SIP.IM
    voIpProt.SIP.keepalive
    voIpProt.SIP.lineSeize
    voIpProt.SIP.local
    voIpProt.SIP.mtls
    voIpProt.SIP.musicOnHold
      voIpProt.SIP.musicOnHold.uri               moh@example.com
    voIpProt.SIP.outboundProxy
    voIpProt.SIP.presence
```

## Configure Call Park and Retrieve

This feature is available as open SIP. If you want to use the call park feature available with Skype for Business Server, see the *Polycom VVX Business Media Phones User Guide*. You can park an active call on a separate call orbit and retrieve parked calls from the call orbit on any phone. Whereas call hold keeps the held call on the same line, call park moves the call to a separate address where the call can be retrieved by any phone. This feature requires support from a SIP server and setup of this feature depends on the SIP server. For example, while some SIP servers implement group call pick-up using a particular star-code sequence, others implement the feature using network signaling. See the table Configure Call Park and Retrieve for parameters you can configure.

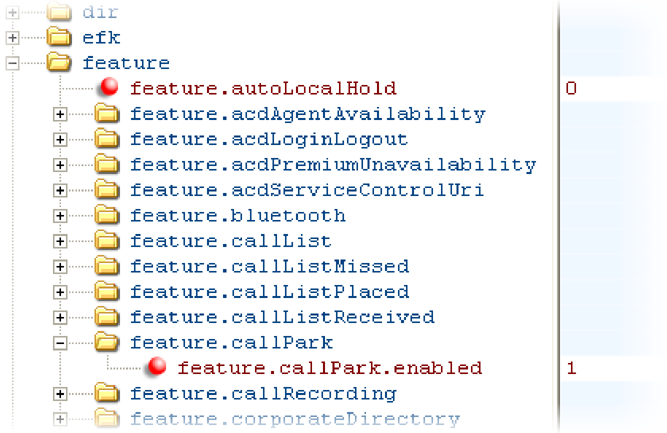
**Configure Call Park and Retrieve for Open SIP**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable call park and retrieve. | **features.cfg** > feature.callPark.enabled |
| Specify the method the phone uses to retrieve a BLF call. | **sip-interop.cfg** > call.parkedCallRetrieveMethod |
| Specify the star code used to retrieve a parked call. | **sip-interop.cfg** > call.parkedCallRetrieveString |

## Example Call Park and Retrieve Configuration

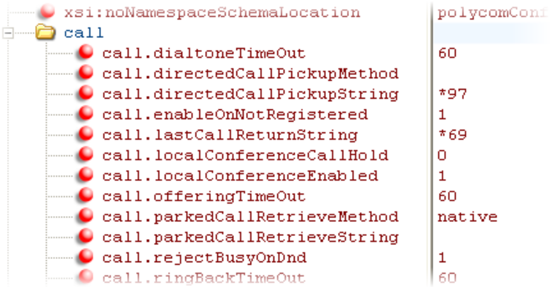
The configuration parameters for the call park and retrieve feature are located in two template files. You can enable the feature using the **features.cfg** template file and configure the feature using the **sip-interop.cfg** file.

In the following configuration example, the call park feature has been enabled in the **features.cfg** template file.



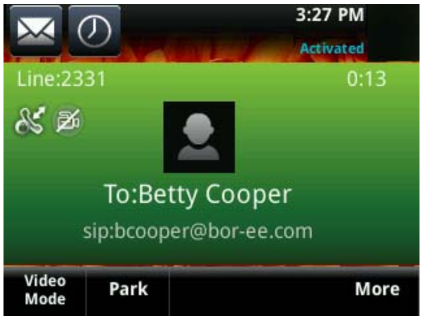
```
+ dir
+ efk
- feature
    feature.autoLocalHold                   0
  + feature.acdAgentAvailability
  + feature.acdLoginLogout
  + feature.acdPremiumUnavailability
  + feature.acdServiceControlUri
  + feature.bluetooth
  + feature.callList
  + feature.callListMissed
  + feature.callListPlaced
  + feature.callListReceived
  - feature.callPark
      feature.callPark.enabled              1
  + feature.callRecording
  + feature.corporateDirectory
```

You can configure the call park and call retrieve feature using parameters located in the **sip-interop.cfg** template file. The following illustration shows that the parked call retrieve method has been set to `native`, meaning that the phone uses SIP INVITE with the Replaces header. The method can also be set to `legacy`, meaning that the phone uses the `call.parkedCallRetrieveString` star code to retrieve the parked call.

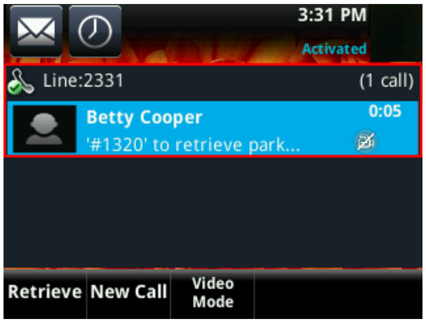


```
  xsi:noNamespaceSchemaLocation          polycomConf
- call
    call.dialtoneTimeOut                 60
    call.directedCallPickupMethod
    call.directedCallPickupString        *97
    call.enableOnNotRegistered           1
    call.lastCallReturnString            *69
    call.localConferenceCallHold         0
    call.localConferenceEnabled          1
    call.offeringTimeOut                 60
    call.parkedCallRetrieveMethod        native
    call.parkedCallRetrieveString
    call.rejectBusyOnDnd                 1
    call.ringBackTimeOut                 60
```

When the call park and retrieve feature is enabled, the **Park** soft key displays when you are in a connected call. To park the call, press the **Park** soft key and enter the number of the call orbit and park the call.



To retrieve a parked call, go off-hook and press the **Pickup** soft key. Enter the number of the call orbit and press the **More** and **Retrieve** soft keys, shown next.



## Use Call Transfer

The call transfer feature enables you to transfer an existing active call to a third-party address using a Transfer soft key. For example, if party A is in an active call with party B, party A can transfer party B to party C (the third party). In this case, party B and party C begin a new call and party A disconnects. You can also set the default transfer type. The following table shows you how to specify call transfer behavior.

You can perform two types of call transfers:

- **Blind Transfer**—Party A transfers the call without speaking to party C.
- **Consultative Transfer**—Party A speaks to party C before party A transfers the call.

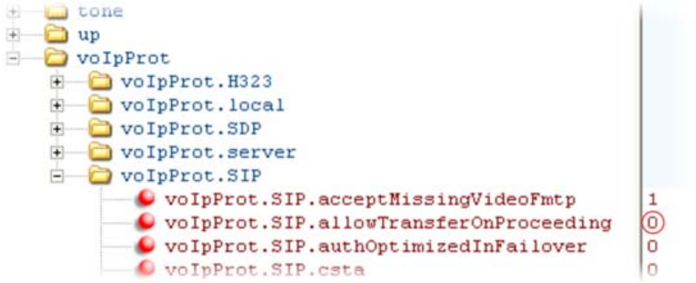
  By default, a Transfer soft key displays when party A calls Party C and Party C's phone is ringing, the proceeding state. In this case, party A has the option to complete the transfer before party C answers, which ends party A's connection to party B and C. You can disable this option so that the Transfer soft key does not display during the proceeding state. In this case, party A can either wait until party C answers or press the Cancel soft key and return to the original call.

**Use Call Transfer**

| Parameter Function | **template** > parameter |
|---|---|
| Specify whether to allow transfers while calls are in a proceeding state. | **sip-interop.cfg** > voIpProt.SIP.allowTransferOnProceeding |
| Set the default transfer type the phone uses when transferring a call. | **features.cfg** > call.DefaultTransferType |

### Example Call Transfer Configuration

In the following example configuration, the parameter `allowTransferOnProceeding` has been disabled so that the Transfer soft key does not display while the third-party phone is ringing, the proceeding state. After you have connected to the third-party, the Transfer soft key displays. If the third-party does not answer, you can press the Cancel soft key to return to the active call.



```
tone
up
voIpProt
    voIpProt.H323
    voIpProt.local
    voIpProt.SDP
    voIpProt.server
    voIpProt.SIP
        voIpProt.SIP.acceptMissingVideoFmtp        1
        voIpProt.SIP.allowTransferOnProceeding     0
        voIpProt.SIP.authOptimizedInFailover       0
        voIpProt.SIP.csta                          0
```

## Configure Call Forwarding

The phone provides a flexible call forwarding feature that enables you to forward incoming calls to another destination. You can apply call forwarding in the following ways:

- To all calls
- To incoming calls from a specific caller or extension
- When your phone is busy
- when do not disturb is enabled
- When the phone has been ringing for a specific period of time
- You can have incoming calls forwarded automatically to a predefined destination you choose or you can manually forward calls to a destination.

You can find parameters for all of these options in the table Configure Call Forwarding.

To enable server-based call forwarding, you must enable the feature on both a registered phone and on the server and the phone is registered. If you enable server-based call forwarding on one registration, other registrations are not affected.

Server-based call forwarding behaves the same as pre-SIP 2.1 feature with the following exception:

- If server-based call forwarding is enabled, but inactive, and you press the Forward soft key, the ‘moving arrow’ icon does not display on your phone and incoming calls are not forwarded.

**Troubleshooting: Call forwarding does not work on my phone**

The server-based and local call forwarding features do not work with the shared call appearance (SCA) and bridged line appearance (BLA) features. If you have SCA or BLA enabled on your phone, you must disable the feature before you can use call forwarding.

The call server uses the Diversion field with a SIP header to inform the phone of a call’s history. For example, when you enable call forwarding, the Diversion header allows the receiving phone to indicate who the call was from, and the phone number it was forwarded from.

If you are registering your Polycom phones with Skype for Business Server, the following types of call forwarding are available on Skype for Business-enabled Polycom phones:

- Disable Call Forwarding

- Forward to a contact
- Forward to voicemail

No parameters are needed to enable call forwarding on Skype for Business-enabled phones.

**Configure Call Forwarding**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable server-based call forwarding. | **sip-interop.cfg** > voIpProt.SIP.serverFeatureControl.cf |
| Enable or disable local call forwarding behavior when server-based call forwarding is enabled. | **sip-interop.cfg** > voIpProt.SIP.serverFeatureControl.localProcessing.cf |
| Enable or disable the display of the Diversion header and the order in which to display the caller ID and number. | **sip-interop.cfg** > voIpProt.SIP.header.diversion.* |
| 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. | **site.cfg** > divert.* |
| Enable or disable server-based call forwarding as a per-registration feature. | **reg-advanced.cfg** > reg.x.fwd.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

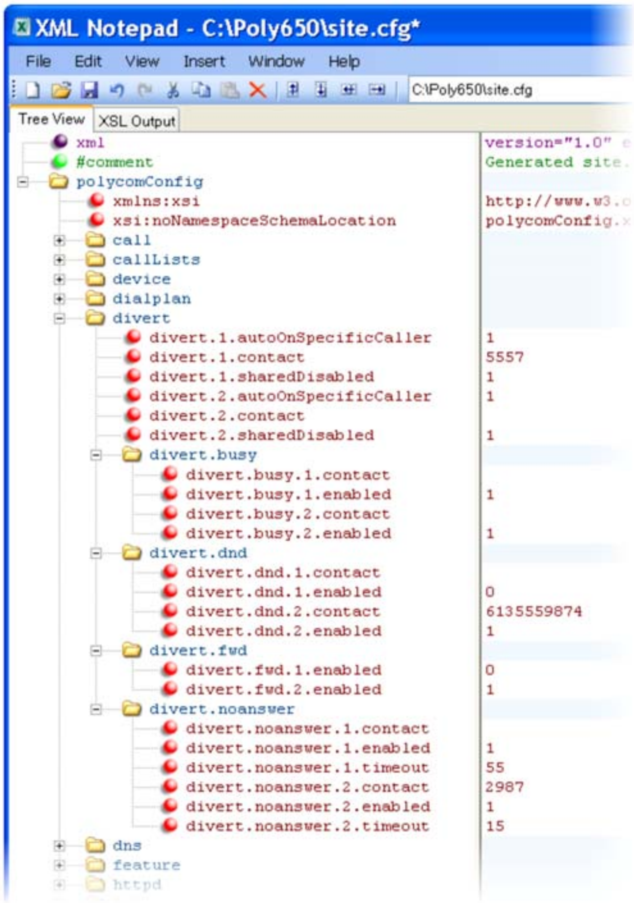
## Example Call Forwarding Configuration

In the example configuration shown next, the call forwarding parameters for registration 1 have been changed from the default values. The forward-always contact for registration 1 is *5557* and this number is used if the parameters `divert.busy`, `divert.dnd`, or `divert.noanswer` are not set. Parameters you set in those fields override `divert.1.contact`.

To enable these three divert options for each registration, you must enable the `divert.fwd.x.enabled` parameter and the `.enabled` parameter for each of the three forwarding options you want to enable.

In this example, `divert.fwd.1.enabled` has been disabled; all calls to registration 1 are diverted to *5557* and you do not have the option of enabling any of the three forwarding options on the phone. The three divert options are enabled for registration 2 in the `divert.fwd.2.enabled` parameter, giving you the option to enable or disable any one of the three forwarding options on the phone.

When do not disturb (DND) is turned on, you can set calls to registration 2 to be diverted to *6135559874* instead of *5557*. The parameter `divert.noanswer.2.enabled` is enabled so that, on the phone, you can set calls to registration 2 that ring for more than 15 seconds, specified in `divert.noanswer.2.timeout`, to be diverted to 2987, as set in `divert.noanswer.2.contact`.


XML Notepad - C:\Poly650\site.cfg*
File Edit View Insert Window Help
C:\Poly650\site.cfg
Tree View
XSL Output

```
xml                                   version="1.0"
#comment                              Generated site.
polycomConfig
  xmlns:xsi                           http://www.w3.c
  xsi:noNamespaceSchemaLocation       polycomConfig.x
  call
  callLists
  device
  dialplan
  divert
    divert.1.autoOnSpecificCaller     1
    divert.1.contact                  5557
    divert.1.sharedDisabled           1
    divert.2.autoOnSpecificCaller     1
    divert.2.contact
    divert.2.sharedDisabled           1
    divert.busy
      divert.busy.1.contact
      divert.busy.1.enabled           1
      divert.busy.2.contact
      divert.busy.2.enabled           1
    divert.dnd
      divert.dnd.1.contact
      divert.dnd.1.enabled            0
      divert.dnd.2.contact            6135559874
      divert.dnd.2.enabled            1
    divert.fwd
      divert.fwd.1.enabled            0
      divert.fwd.2.enabled            1
    divert.noanswer
      divert.noanswer.1.contact
      divert.noanswer.1.enabled       1
      divert.noanswer.1.timeout       55
      divert.noanswer.2.contact       2987
      divert.noanswer.2.enabled       1
      divert.noanswer.2.timeout       15
  dns
  feature
  httpd
```

## Enable Automatic Off-Hook Call Placement

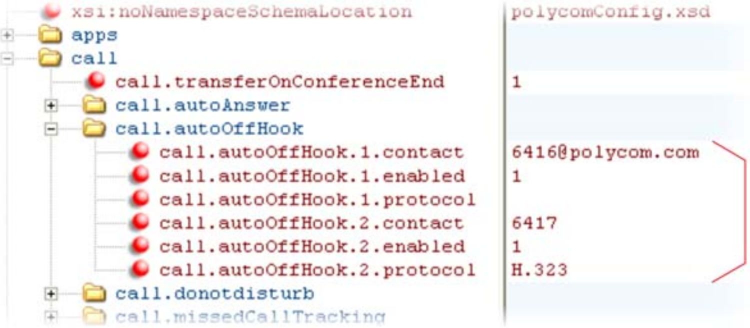
You can configure the phone to automatically place a call to a specified number when you go off-hook. This feature is sometimes referred to as hot dialing. The phone goes off-hook when you lift the handset, press the New Call soft key, or press the headset or speakerphone buttons on the phone. As shown in the following table, you can specify an off-hook call contact and enable or disable the feature for specific line registrations. If you are using the VVX 500 series, 600 series, or 1500 phones, you can specify whether the automatic call uses the SIP (audio only) protocol or the H.323 (video) protocol.

**Enable Automatic Off-Hook Call Placement**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the contact to dial when the phone goes off-hook. | **reg-advanced** > call.autoOffHook.x.contact |
| Enable or disable automatic off-hook call placement on registration x. | **reg-advanced** > call.autoOffHook.x.enabled |
| Specify the call protocol to use for the VVX 500/501, 600/601, and 1500. | **reg-advanced** > call.autoOffHook.x.protocol |

### Example Automatic Off-Hook Placement Configuration

In the example configuration shown next, the automatic off-hook call placement feature has been enabled for registration 1 and registration 2. If registration 1 goes off-hook, a call is automatically placed to *6416@polycom.com*, the contact that has been specified for registration 1 in `call.autoOffHook.1.contact`. Similarly, if registration 2 goes off-hook, a call is automatically placed to *6417*. On VVX 500 series, 600 series, and 1500 phones, registration 2 automatically places a call using the H.323 protocol instead of the SIP protocol. Other phones ignore the protocol parameter.



```
xsi:noNamespaceSchemaLocation            polycomConfig.xsd
apps
call
    call.transferOnConferenceEnd         1
    call.autoAnswer
    call.autoOffHook
        call.autoOffHook.1.contact       6416@polycom.com
        call.autoOffHook.1.enabled       1
        call.autoOffHook.1.protocol
        call.autoOffHook.2.contact       6417
        call.autoOffHook.2.enabled       1
        call.autoOffHook.2.protocol      H.323
    call.donotdisturb
    call.missedCallTracking
```

## Configure Directed Call Pickup

Directed call pickup enables you to pick up incoming calls to another phone by dialing the extension of that phone. This feature requires support from a SIP server and setup of this feature depends on the SIP server. For example, while some SIP servers implement directed call pick-up using a star-code sequence, others implement the feature using network signaling.

To enable or disable this feature for Sylantro call servers, set:

- `feature.directedCallPickup.enabled=1`

To configure this feature for all other call servers, use the parameters:

- `call.directedCallPickupMethod`
- `call.directedCallPickupString`

The following table lists the configuration parameters for the directed call pick-up feature.

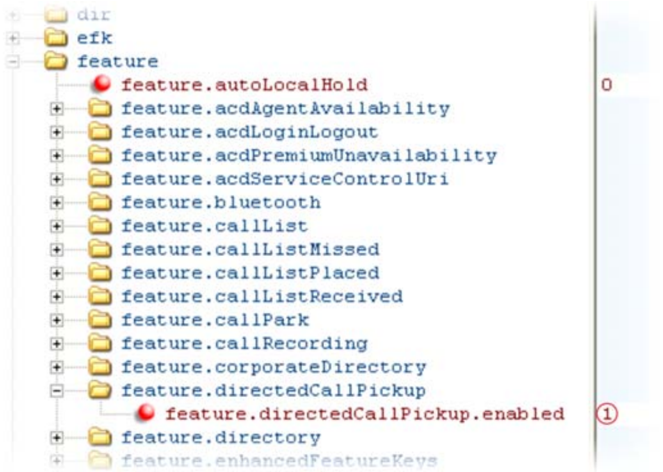
**Configure Directed Call Pickup**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable Directed Call Pickup for Sylantro call Servers. | **features.cfg** > feature.directedCallPickup.enabled |
| Specify the type of directed call pick-up. | **sip-interop.cfg** > call.directedCallPickupMethod |
| Specify the star code to initiate a directed call pickup. | **sip-interop.cfg** > call.directedCallPickupString |
| Determine the type of SIP header to include. | **sip-interop.cfg** > voIpProt.SIP.strictReplacesHeader |

## Example Directed Call Pickup Configuration

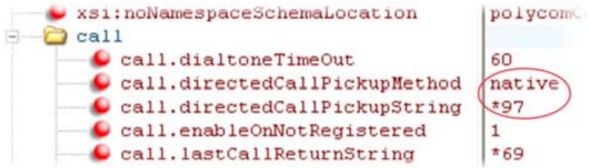
The configuration parameters for the directed call pickup feature are located in two template files. You enable directed call pickup in the **features.cfg** template file and configure the feature using the **sip-interop.cfg** file.

In the following configuration example, the directed call pickup feature has been enabled in the **features.cfg** template file:



```
dir
efk
feature
    feature.autoLocalHold                         0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
        feature.directedCallPickup.enabled        1
    feature.directory
    feature.enhancedFeatureKeys
```

Once directed call pickup is enabled, you can configure the feature using parameters located in the **sip-interop.cfg** template file. In the following illustration, the pickup method has been set to `native`, which means that the server is used for directed call pickup instead of the `PickupString`. If the pickup method was set to `legacy`, the pickup string `*97` would be used by default. The pickup string can be different for different call servers, check with your call server provider if you configure legacy mode directed call pickup.



```
xsi:noNamespaceSchemaLocation        polycom
call
    call.dialtoneTimeOut             60
    call.directedCallPickupMethod    native
    call.directedCallPickupString    *97
    call.enableOnNotRegistered       1
    call.lastCallReturnString        *69
```

When you enable directed call pickup, the phone displays a **Pickup** soft key when you go off-hook. When you press the **Pickup** soft key, the **Directd** soft key displays.

# Enable Multiple Registrations

Polycom phones can have multiple registrations; each registration requires an address, or phone number. Polycom phones registered with Microsoft Skype for Business Server support one Skype for Business registration. The maximum number of registrations vary by phone and are listed in the following table. The maximum registrations listed are supported with UC Software 4.0.1 and later.

**Maximum Number of Registrations Per Phone**

| Phone Model Name | Maximum Registrations |
|---|---|
| VVX 101 | One (1) |

**Maximum Number of Registrations Per Phone**

| | |
|---|---|
| VVX 201 | Two (2) |
| VVX 300/301/310/311 | Six (6) |
| VVX 400/401/410/411 | Eight (8) |
| VVX 500/501 | Twelve (12) |
| VVX 600/601 | Sixteen (16) |
| VVX 1500 | Twenty four (24) |

You can also add up to three VVX Expansion Modules to a single VVX 300 series, 400 series, 500 series, or 600 series phone to increase the total number of registrations to 34. For more information, see Connect Polycom VVX Expansion Modules.

Each registration can be mapped to one or more line keys. Note that 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. Note that this feature is one of several features associated with Flexible Call Appearances. For definitions of all features associated with flexible call appearances, see the following table.

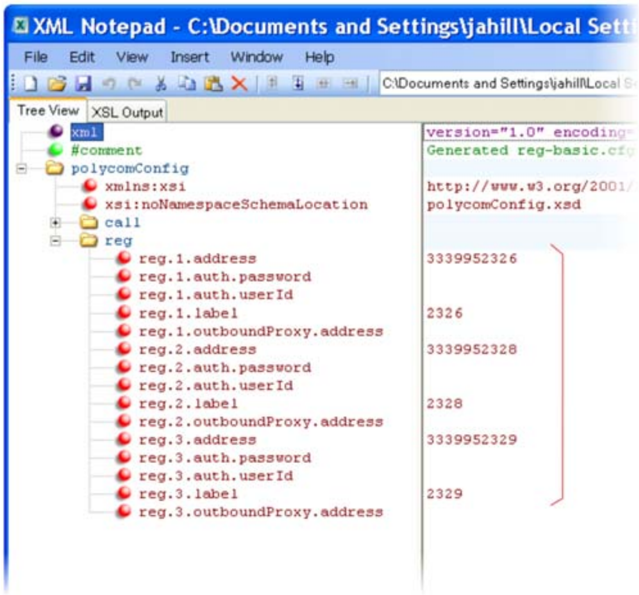
**Enable Multiple Registrations**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the local SIP signaling port and several optional SIP servers to register to. For each server specify the registration period and the signaling failure behavior. | **sip-interop.cfg** > voIpProt.SIP.*<br>and voIpProt.server.x.* |
| 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 are used instead. The optional array of servers and their parameters override the servers specified in <voIpProt.server/> if non-Null. | **reg-basic.cfg**, **reg-advanced.cfg** > reg.x.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

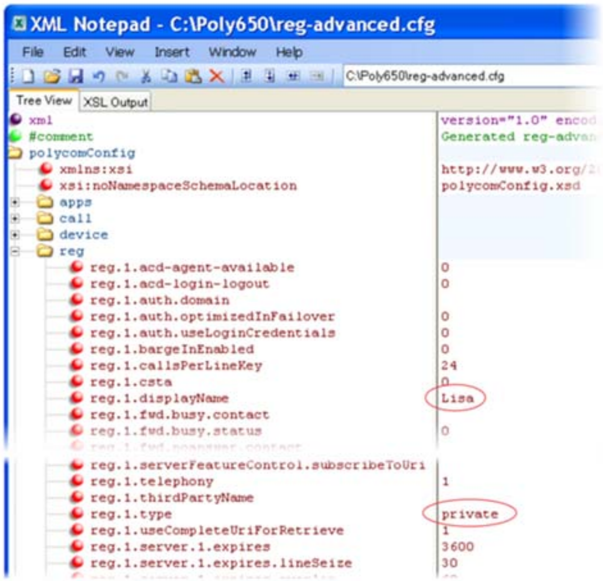
## Example Multiple Registration Configuration

In the next illustration, in the **reg-basic.cfg** template, multiple line registrations and a label for each registration has been enabled for lines 1, 2, and 3.


XML Notepad - C:\Documents and Settings\jahill\Local Setti
File Edit View Insert Window Help
C:\Documents and Settings\jahill\Local S
Tree View
XSL Output

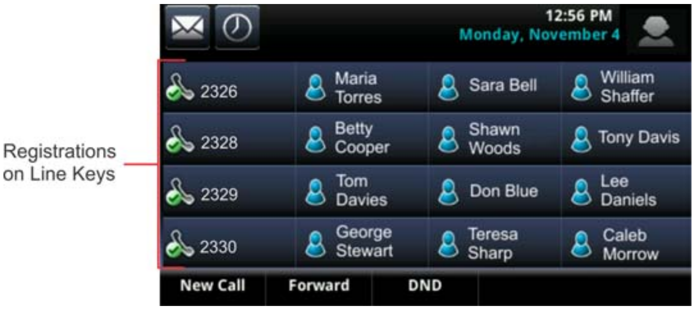
```
xml                                        version="1.0" encoding
#comment                                   Generated reg-basic.cfg
polycomConfig
    xmlns:xsi                              http://www.w3.org/2001/
    xsi:noNamespaceSchemaLocation          polycomConfig.xsd
  call
  reg
      reg.1.address                        3339952326
      reg.1.auth.password
      reg.1.auth.userId
      reg.1.label                          2326
      reg.1.outboundProxy.address
      reg.2.address                        3339952328
      reg.2.auth.password
      reg.2.auth.userId
      reg.2.label                          2328
      reg.2.outboundProxy.address
      reg.3.address                        3339952329
      reg.3.auth.password
      reg.3.auth.userId
      reg.3.label                          2329
      reg.3.outboundProxy.address
```

In the **reg-advanced.cfg** template shown next, when you make a call using line 1, the name you enter in `reg.1.displayname` displays as your caller ID, in this case *Lisa*. The parameter `reg.x.type` is left in the default `private`, which indicates that the registration uses standard call signaling.


XML Notepad - C:\Poly650\reg-advanced.cfg
File Edit View Insert Window Help
C:\Poly650\reg-advanced.cfg
Tree View
XSL Output

```
xml                                            version="1.0" encod
#comment                                       Generated reg-advan
polycomConfig
    xmlns:xsi                                  http://www.w3.org/2
    xsi:noNamespaceSchemaLocation              polycomConfig.xsd
  apps
  call
  device
  reg
      reg.1.acd-agent-available                0
      reg.1.acd-login-logout                   0
      reg.1.auth.domain
      reg.1.auth.optimizedInFailover           0
      reg.1.auth.useLoginCredentials           0
      reg.1.bargeInEnabled                     0
      reg.1.callsPerLineKey                    24
      reg.1.csta                               0
      reg.1.displayName                        Lisa
      reg.1.fwd.busy.contact
      reg.1.fwd.busy.status                    0
      reg.1.fwd.noanswer.contact
      reg.1.serverFeatureControl.subscribeToUri
      reg.1.telephony                          1
      reg.1.thirdPartyName
      reg.1.type                               private
      reg.1.useCompleteUriForRetrieve          1
      reg.1.server.1.expires                   3600
      reg.1.server.1.expires.lineSeize         30
```

his configuration results in the following registrations on a VVX 600/601 phone:



## Assign Multiple Line Keys Per Registration

You can assign a single registered phone line address to multiple line keys on VVX business media phones. This feature is not supported on Polycom phones registered with Microsoft Skype for Business Server.

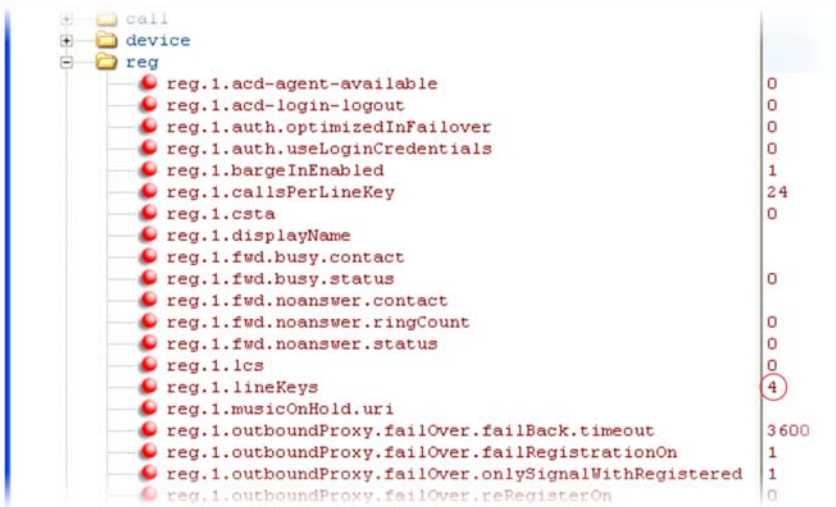
This feature can be useful for managing a high volume of calls to a line. This feature is one of several features associated with flexible call appearances. For the maximum number of line keys per registration for each phone model, and for definitions of all features associated with flexible call appearances, refer to the following table.

### Multiple Line Keys Per Registration

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the number of line keys to use for a single registration. | **reg-advanced.cfg** > reg.x.lineKeys |

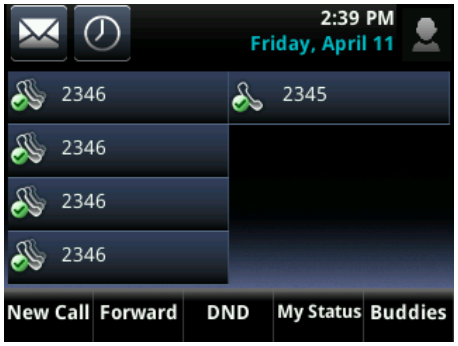
### Example Configuration

The following illustration shows you how to enable four line keys with the same registered line address. In this example, four line keys are configured with registration address *2346*.



```
call
device
reg
    reg.1.acd-agent-available                              0
    reg.1.acd-login-logout                                 0
    reg.1.auth.optimizedInFailover                         0
    reg.1.auth.useLoginCredentials                         0
    reg.1.bargeInEnabled                                   1
    reg.1.callsPerLineKey                                  24
    reg.1.csta                                             0
    reg.1.displayName
    reg.1.fwd.busy.contact
    reg.1.fwd.busy.status                                  0
    reg.1.fwd.noanswer.contact
    reg.1.fwd.noanswer.ringCount                           0
    reg.1.fwd.noanswer.status                              0
    reg.1.lcs                                              0
    reg.1.lineKeys                                         4
    reg.1.musicOnHold.uri
    reg.1.outboundProxy.failOver.failBack.timeout          3600
    reg.1.outboundProxy.failOver.failRegistrationOn        1
    reg.1.outboundProxy.failOver.onlySignalWithRegistered  1
    reg.1.outboundProxy.failOver.reRegisterOn              0
```

The phone displays the registered line address *2346* on four line keys, as shown next.



## Configure Shared Call Appearances

Shared call appearance enables an active call to display simultaneously on multiple phones in a group. By default, the answering phone has sole access to the incoming call, called line seize. If the answering phone places the call on hold, that call becomes available to all phones of that group. You can enable another phone in the group the ability to enter a conversation, called a barge in. All call states of a call —active, inactive, on hold—are displayed on all phones of a group. The parameters you can configure are listed in the following table.

This feature is dependent on support from a SIP call server. To enable shared call appearances on your phone, you must obtain a shared line address from your SIP service provider. For more details on SIP signaling with shared call appearances, see the section Shared Call Appearance (SCA) Signaling.

**Tip: Shared call and bridged line appearance are distinct**

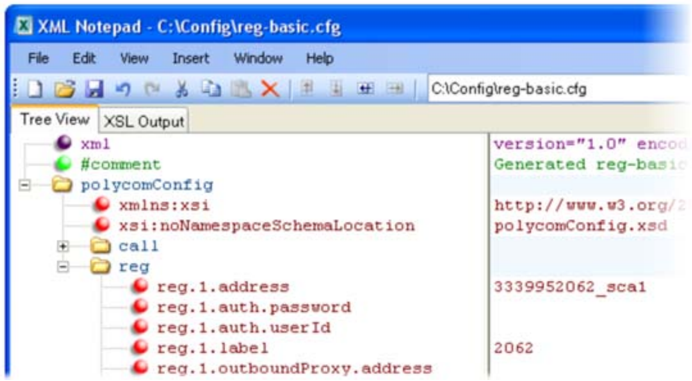
Shared call appearances and bridged line appearances are similar signaling methods that enable more than one phone to share the same line or registration. The method you use varies with the SIP call server you are using.

**Configure Shared Call Appearances**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the shared line address. | **reg-basic.cfg** > reg.x.address |
| Specify the line type as shared. | **reg-advanced.cfg** > reg.x.type |
| To disable call diversion, expose auto-holds, resume with one touch, or play a tone if line-seize fails. | **sip-interop.cfg** > call.shared.* |
| Specify standard or non-standard behavior for processing a line-seize subscription for mutual exclusion. | **sip-interop.cfg** > voIpProt.SIP.specialEvent.lineSeize.nonStandard |
| Specify barge-in capabilities and line-seize subscription period if using per-registration servers. A shared line subscribes to a server providing call state information. | **reg-advanced.cfg** > reg.x.* |
| Specify per-registration whether diversion should be disabled on shared lines. | **sip-interop.cfg** > divert.x.sharedDisabled |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

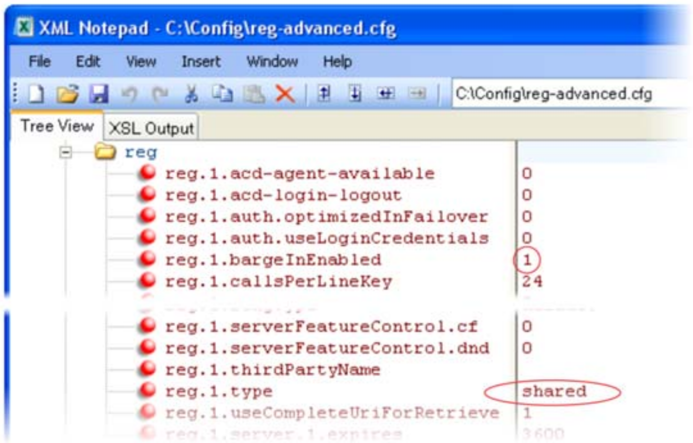
## Example Configuration

The following illustration shows the address of a registered phone line and the label that displays beside the line key, as specified in the **reg-basic.cfg** template.


XML Notepad - C:\Config\reg-basic.cfg
File Edit View Insert Window Help
C:\Config\reg-basic.cfg
Tree View XSL Output

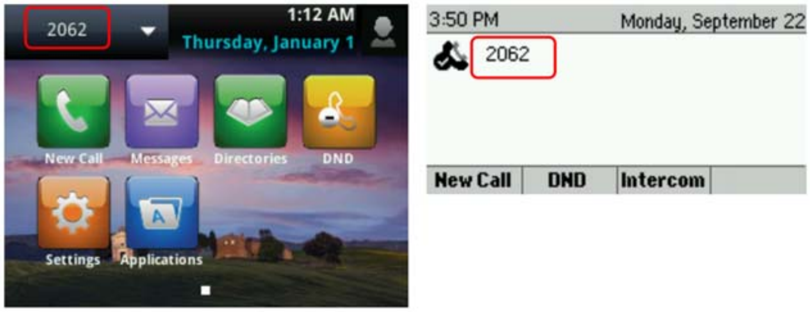
```
xml                                   version="1.0" encod
#comment                              Generated reg-basic
polycomConfig
  xmlns:xsi                           http://www.w3.org/2
  xsi:noNamespaceSchemaLocation       polycomConfig.xsd
  call
  reg
    reg.1.address                     3339952062_sca1
    reg.1.auth.password
    reg.1.auth.userId
    reg.1.label                       2062
    reg.1.outboundProxy.address
```

If you want to configure this line to be shared, in the **reg-advanced.cfg** template, specify shared in `reg.1.type`. All phones that specify shared for registration 1 have shared call appearance enabled for this line. In the following example, the `reg.1.bargeInEnabled` parameter is set to ‘1’ to enable phones of this group to barge in on active calls.


XML Notepad - C:\Config\reg-advanced.cfg
File Edit View Insert Window Help
C:\Config\reg-advanced.cfg
Tree View XSL Output

```
reg
  reg.1.acd-agent-available            0
  reg.1.acd-login-logout               0
  reg.1.auth.optimizedInFailover       0
  reg.1.auth.useLoginCredentials       0
  reg.1.bargeInEnabled                 1
  reg.1.callsPerLineKey                24
  reg.1.serverFeatureControl.cf        0
  reg.1.serverFeatureControl.dnd       0
  reg.1.thirdPartyName
  reg.1.type                           shared
  reg.1.useCompleteUriForRetrieve      1
  reg.1.server.1.expires               3600
```

After setting these parameters, activity on line *2062* displays on all phones that configure a shared call appearance for line *2062*, as shown in the following illustration.



## Call Forward on Shared Lines

You can enable server-based call forwarding on shared lines for VVX phones. If using BroadWorks R20 server, note the following

- Local call-forwarding is not supported on shared lines on the BroadWorks R20 server.
- Dynamic call forwarding-forwarding incoming calls without answering the call-is not supported on BroadWorks R20 server.

### Enable Call Forward on Shared Lines

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable server-based call forwarding per-registration. This parameter overrides `voIpProt.SIP.serverFeatureControl.cf`. | **reg-advanced.cfg** > reg.x.serverFeatureControl.cf |
| Enable or disable per-registration diversion on shared lines. | **sip-interop.cfg** > divert.x.sharedDisabled |
| Enable or disable server-based call forwarding. This parameter overrides `reg.x.serverFeatureControl.cf`. | **sip-interop.cfg** > voIpProt.SIP.serverFeatureControl.cf |
| This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.cf`. | **sip-interop.cfg** > voIpProt.SIP.serverFeatureControl.localProcessing.cf |
| Enable or disable call forwarding behavior on all calls received. This parameter overrides `voIpProt.SIP.serverFeatureControl.localProcessing.cf`. | **sip-interop.cfg** > reg.x.serverFeatureControl.localProcessing.cf |
| Enable or disable the diversion feature for shared lines. This feature is disabled on most call servers. | **sip-interop.cfg** > call.shared.disableDivert |

## Private Hold on Shared Lines

Enable the private hold feature to display the PvtHold soft key on a shared line. When users in an active call on a shared line press the soft key, the active call is placed on hold and displays the shared line as busy to others sharing the line. The shared line also shows as busy when users transfer a call or initiate a conference call during an active call. When you enable the feature, users can hold a call, transfer a call, or initiate a conference call and the shared line displays as busy to others sharing the line.

You can configure private hold only using configuration files; you cannot configure the feature on the Web Configuration Utility or from the local phone interface.

Note that `call.shared.exposeAutoHolds` is an existing parameter updated for private hold and `reg.X.enablePvtHoldSoftKey` is a new parameter for this feature.

### Configure Private Hold

| **Parameter Function** | **template** > parameter |
|---|---|

**Configure Private Hold**

| | |
|---|---|
| Enable or disable the private hold feature for all lines. | **sip-interop.cfg** > call.shared.exposeAutoHolds |
| Enable or disable the Private Hold soft key for a specific shared line. | **reg-advanced.cfg** > reg.x.enablePvtHoldSoftKey |

## Enable Multiple Call Appearances

You can enable each registered phone line to support multiple concurrent calls and have each concurrent call display on the phone's user interface. For example, you can place one call on hold, switch to another call on the same registered line, and have both calls display. As shown in the following table, you can set the maximum number of concurrent calls per registered line and the default number of calls per line key.

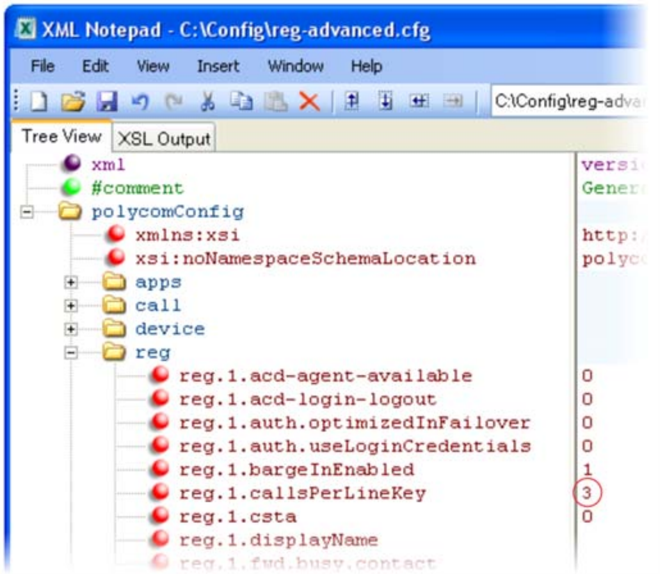
This feature is one of several features associated with flexible call appearances. If you want to enable multiple line keys per registration, see the section Assign Multiple Line Keys Per Registration. Note that if you assign a registered line to multiple line keys, the default number of concurrent calls applies to all line keys. If you want use multiple registrations on a phone, and for definitions of all features associated with flexible call appearances, see the following table. Use this table to customize the number of registrations, line keys per registration, and concurrent calls.

**Enable Multiple Call Appearances**

| **Parameter Function** | **template** > parameter |
|---|---|
| Set the default number of concurrent calls for all line keys. | **reg-basic.cfg** > call.callsPerLineKey |
| Override the default number of calls per line key for a specific line. | **reg-advanced.cfg** > reg.x.callsPerLineKey |

### Example Multiple Call Appearances Configuration

The following illustration shows that in the **reg-advanced.cfg** template you can enable line 1 on your phone with three call appearances.



```
xml                                     versi
#comment                                Gener
polycomConfig
    xmlns:xsi                           http:/
    xsi:noNamespaceSchemaLocation       polyc
    apps
    call
    device
    reg
        reg.1.acd-agent-available       0
        reg.1.acd-login-logout          0
        reg.1.auth.optimizedInFailover  0
        reg.1.auth.useLoginCredentials  0
        reg.1.bargeInEnabled            1
        reg.1.callsPerLineKey           3
        reg.1.csta                      0
        reg.1.displayName
        reg.1.fwd.busy.contact
```

After you have set the `reg.1.callsPerLineKey` parameter to `3`, you can have three call appearances on line 1. By default, additional incoming calls are automatically forwarded to your voicemail. If you have more than two call appearances, a call appearance counter displays at the top-right corner of your phone's screen.

A number of features are associated with flexible call appearances. Use the following table to understand how you can organize registrations, line keys per registration, and concurrent calls per line key.

In the following table,

- **Registrations**—The maximum number of user registrations
- **Line Keys**—The maximum number of line keys
- **Line Keys Per Registration**—The maximum number of line keys per user registration
- **Calls Per Line Key**—The maximum number of concurrent calls per line key
- **Concurrent Calls (includes Conference Legs)**—The runtime maximum number of concurrent calls. (The number of conference participants minus the moderator.)

**Flexible Call Appearances**

| *Phone Model* | *Registrations* | *Line Keys* | *Line keys Per Registration* | *Calls Per Line Key* | *Concurrent Calls** |
|---|---|---|---|---|---|
| VVX 300/301/310/311 | 34 | 48 | 34 | 24 | 24 (2) |
| VVX 400/401/410/411 | 34 | 48 | 34 | 24 | 24 (2) |
| VVX 500/501 | 34 | 48 | 12 | 24 | 24 (2) |
| VVX 600/601 | 34 | 48 | 12 | 24 | 24 (2) |
| VVX 1500 | 34 | 48 | 24 | 24 | 24 (2) |
| SoundStructure VOIP Interface ** | 12 | 12 | 12 | 24 | 24 (2) |
| * Note that each conference leg counts as one call. The total number of concurrent calls in a conference indicated in this table includes all conference participants minus the moderator.<br>** For more information on using line and call appearances with the SoundStructure VOIP Interface, refer to the SoundStructure Design Guide, available at Polycom Support. | | | | | |

* Note that each conference leg counts as one call. The total number of concurrent calls in a conference indicated in this table includes all conference participants minus the moderator.

## Enable Bridged Line Appearance

Bridged line appearance connects calls and lines to multiple phones. See the following table for a list of the parameters you can configure. With bridged line appearance enabled, an active call displays simultaneously on multiple phones in a group. By default, the answering phone has sole access to the incoming call—line seize. If the answering phone places the call on hold, that call becomes available to all phones of that group. All call states—active, inactive, on hold—are displayed on all phones of a group. For more information, see the section Bridged Line Appearance Signaling.

**Tip: Bridged line and shared call appearance are distinct**

Shared call appearances and bridged line appearances are similar signaling methods that enable more than one phone to share the same line or registration. The methods you use vary with the SIP call server you are using. In the configuration files, bridged lines are configured by shared line parameters. The barge-in feature is not available with bridged line appearances; it is available with shared call appearances.

**Enable Bridged Line Appearance**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify whether call diversion should be disabled by default on all shared lines. | **sip-interop.cfg** > call.shared.disableDivert |
| Specify the per-registration line type (private or shared). | **reg-advanced.cfg** > reg.x.type |

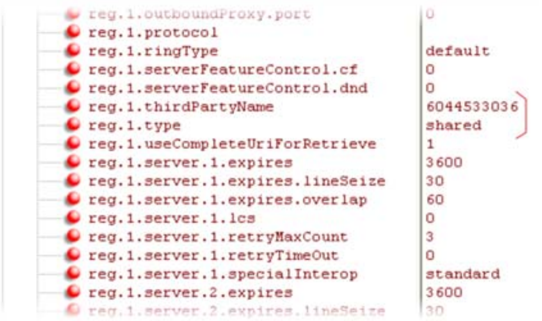
**Enable Bridged Line Appearance**

| | |
|---|---|
| Specify the shared line third-party name. | **reg-advanced.cfg** > reg.x.thirdPartyName |
| Specify whether call diversion should be disabled on a specific shared line (overrides default). | **reg-advanced.cfg** > divert.x.sharedDisabled |

## Example Bridged Line Appearance Configuration

To begin using bridged line appearance, you must get a registered address dedicated for use with bridged line appearance from your call server provider. This dedicated address must be assigned to a phone line in the `reg.x.address` parameter of the **reg-basic.cfg** template.

Next, in the **reg-advanced.cfg** template, enter the dedicated address in `thirdPartyName` for all phones of the BLA group and set the line type to `shared`. In this example, two or more phones can use the same dedicated address *6044533036* as the BLA address, and the line `type` has been set to `shared` from the default `private`.



```
reg.1.outboundProxy.port                    0
reg.1.protocol
reg.1.ringType                              default
reg.1.serverFeatureControl.cf               0
reg.1.serverFeatureControl.dnd              0
reg.1.thirdPartyName                        6044533036
reg.1.type                                  shared
reg.1.useCompleteUriForRetrieve             1
reg.1.server.1.expires                      3600
reg.1.server.1.expires.lineSeize            30
reg.1.server.1.expires.overlap              60
reg.1.server.1.lcs                          0
reg.1.server.1.retryMaxCount                3
reg.1.server.1.retryTimeOut                 0
reg.1.server.1.specialInterop               standard
reg.1.server.2.expires                      3600
reg.1.server.2.expires.lineSeize            30
```

For example, two phones *6044533036* and *6044533037* are configured with the *3036* BLA address. There is an incoming call to *6044533036* from *3038* that causes *3036* and *3037* phones to show the incoming call.

# Enable Voicemail Integration

The phone is compatible with voicemail servers. You can configure each phone or line registration per phone to subscribe with a SIP URL to a voicemail server contact. You can also configure the phone to access voicemail with a single key, for example, the **Messages** key on the VVX 300 series and 400 series phones, the **MSG** key on the VVX 1500 phone, and the **Messages** icon on the VVX 500 series and 600 series phones. When you access the voicemail server, the phone gives a visual and audio alert; you can also configure a message waiting alert to indicate that you have unread voicemail messages. The following table shows you the parameters you can configure.

**Voicemail Integration**

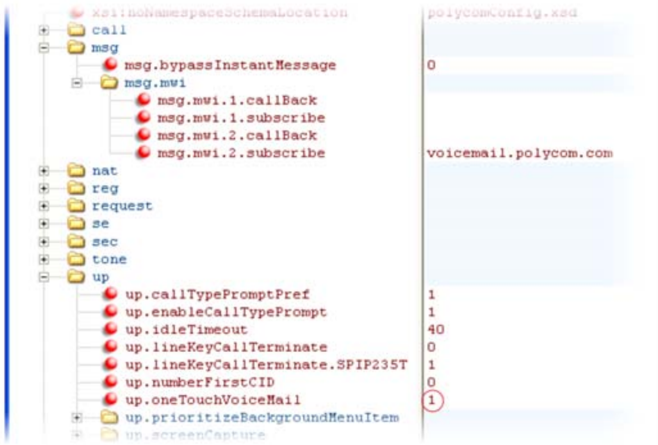
| **Parameter Function** | **template** > parameter |
|---|---|
| To turn one-touch Voicemail on or off. | **sip-interop.cfg** > up.oneTouchVoiceMail |
| Specify the URI of the message center server. | **sip-interop.cfg** > msg.mwi.x.subscribe |
| Set the mode of message retrieval. | **sip-basic.cfg** > msg.mwi.x.callBackMode |

**Voicemail Integration**

| | |
|---|---|
| Specify a contact number for the phone to call to retrieve messages, `callBackMode` must be set to Contact. | **sip-interop.cfg** > msg.mwi.x.callBack |
| Specify if message waiting notifications should display or not. | **site.cfg** > up.mwiVisible |
| Specify if the phone screen backlight illuminates when you receive a new voicemail message. | **site.cfg** > mwi.backLight.disable |

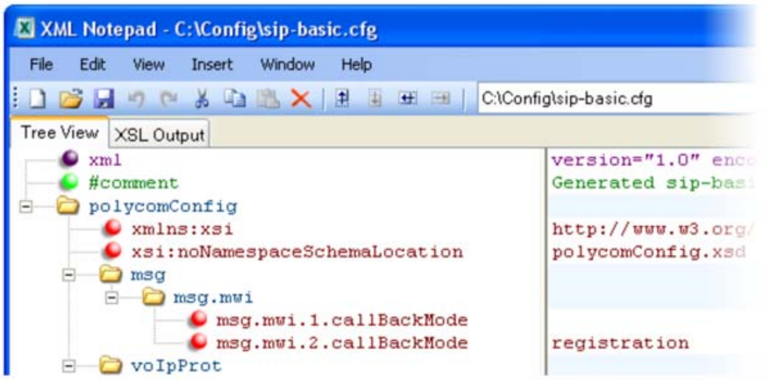
## Example Voicemail Configuration

The following illustration shows you how to enable one-touch access to the voicemail server. In the next illustration, line 2 is configured to `subscribe` to the voicemail server at *voicemail.polycom.com*.



```
xsi:noNamespaceSchemaLocation          polycomConfig.xsd
call
msg
  msg.bypassInstantMessage             0
  msg.mwi
    msg.mwi.1.callBack
    msg.mwi.1.subscribe
    msg.mwi.2.callBack
    msg.mwi.2.subscribe                voicemail.polycom.com
nat
reg
request
se
sec
tone
up
  up.callTypePromptPref                1
  up.enableCallTypePrompt              1
  up.idleTimeout                       40
  up.lineKeyCallTerminate              0
  up.lineKeyCallTerminate.SPIP235T     1
  up.numberFirstCID                    0
  up.oneTouchVoiceMail                 1
  up.prioritizeBackgroundMenuItem
  up.screenCapture
```

The following illustration shows that, in the **sip-basic.cfg** template, the default `callBackMode` setting for line 2 is set to `registration`. The phone uses the address assigned to line 2 to subscribe to the voicemail server you entered in `msg.mwi.2.subscribe`.



XML Notepad - C:\Config\sip-basic.cfg

```
xml                                  version="1.0" enc
#comment                             Generated sip-bas
polycomConfig
  xmlns:xsi                          http://www.w3.org/
  xsi:noNamespaceSchemaLocation      polycomConfig.xsd
  msg
    msg.mwi
      msg.mwi.1.callBackMode
      msg.mwi.2.callBackMode         registration
  voIpProt
```

After this is enabled in the sip-interop.cfg template, on the phone, press the Messages key and select Message Center to access your voicemail.

## Record and Play Audio Calls Locally

You can configure the VVX phones to record audio calls to a USB device that you plug into the phone. You can play back recorded audio on the phone as well as on other devices that run applications like Windows Media Player® or iTunes® on a Windows®- or Apple®-based computer.

To enable this feature, the USB device must be compatible with Polycom phones.

**Web Info: Supported USB devices**

For a list of supported USB devices, see S*upported USB Devices for Polycom SoundPoint IP 650 and VVX Phones: Technical Bulletin 38084* and *Supported USB Headsets for Polycom VVX 500 Business Media Phones: Engineering Advisories 62760* at Polycom Engineering Advisories and Technical Notitfications.

You can enable call recording with the parameter shown in the table Record and Play Audio Calls. Audio calls are recorded in **.wav** format and include a date/time stamp, for example, **20Apr2007_190012.wav** was created on April 20, 2007, at 19:00:12. The phone displays the recording time remaining on the attached USB device and you can browse all recorded files using the phone's menu.

**Note: Inform parties when you are recording calls**

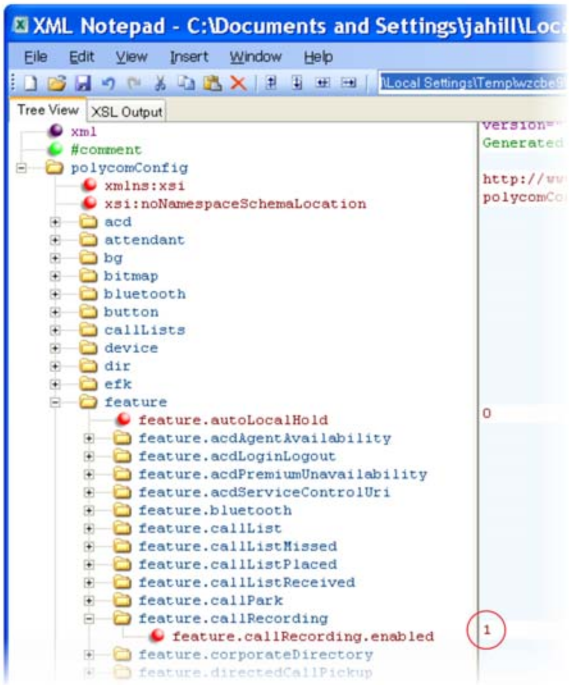
Federal, state, and/or local laws may legally require that you to notify some or all of the call parties that you are recording.

**Record and Play Audio Calls**

| Parameter Function | **template** > parameter |
|---|---|
| To enable or disable call recording. | **features.cfg** > feature.callRecording.enabled |

### Example Call Recording Configuration

To record audio from the phone, you need a USB device plugged into the phone, and you need to enable the call recording feature in the **features.cfg** template file. In **features.cfg**, locate `feature.callRecording.enabled` and enter `1`, as shown next.


XML Notepad - C:\Documents and Settings\jahill\Loc
File
Edit
View
Insert
Window
Help
\Local Settings\Temp\wzcbe
Tree View
XSL Output

```
xml                                         version="
#comment                                    Generated
polycomConfig
    xmlns:xsi                               http://ww
    xsi:noNamespaceSchemaLocation           polycomCo
    acd
    attendant
    bg
    bitmap
    bluetooth
    button
    callLists
    device
    dir
    efk
    feature
        feature.autoLocalHold               0
        feature.acdAgentAvailability
        feature.acdLoginLogout
        feature.acdPremiumUnavailability
        feature.acdServiceControlUri
        feature.bluetooth
        feature.callList
        feature.callListMissed
        feature.callListPlaced
        feature.callListReceived
        feature.callPark
        feature.callRecording
            feature.callRecording.enabled   1
        feature.corporateDirectory
        feature.directedCallPickup
```

Press the **Pause** soft key to pause recording and press the **Stop** soft key to stop recording.

You can browse recorded audio files by navigating on the phone to **Menu** or **Settings > Removable Storage Media > Browse Recordings**.

## Enable Centralized Call Recording

This feature, available on Polycom VVX phones, enables users to record audio and video calls and control call recording directly from phones registered with BroadSoft BroadWorks r20 server. Administrators must enable this feature on the BroadSoft BroadWorks r20 server and on the phones using the configuration parameters listed in the table Centralized Call Recording Parameters. On the BroadSoft server, administrators assign phone users one of several call recording modes listed in **Call Recording Modes**. You can manage your recorded audio and video files on a third-party call recording server.

**Caution: Enable only one recording mechanism on the phone, not both**

You can record calls using a central server or locally using the phone's USB call recording feature – you cannot use both at the same time. By default, both features are disabled. If you enable one call recording feature, ensure that the other is disabled. Use either centralized or the local call recording; do not use both.

By default, far-side participants are not alerted that calls are being recorded; BroadWorks server r20 provides administrators the option to enable an announcement at the beginning of a call that the call is being recorded. If a call being recorded is transferred, the new call continues to be recorded.

This feature can be enabled using the configuration parameters in the following table.

**Centralized Call Recording Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the BroadSoft BroadWorks v20 call recording feature for all lines on a phone. | **sip-interop.cfg** ><br>voIpProt.SIP.serverFeatureControl.callRecording |
| Enable or disable BroadSoft BroadWorks v20 call recording feature for a specific line on a phone. | **reg-advanced.cfg** ><br>reg.x.serverFeatureControl.callRecording |

Set the phone call recording modes on BroadSoft BroadWorks R20 server. You can set the following call recording modes:

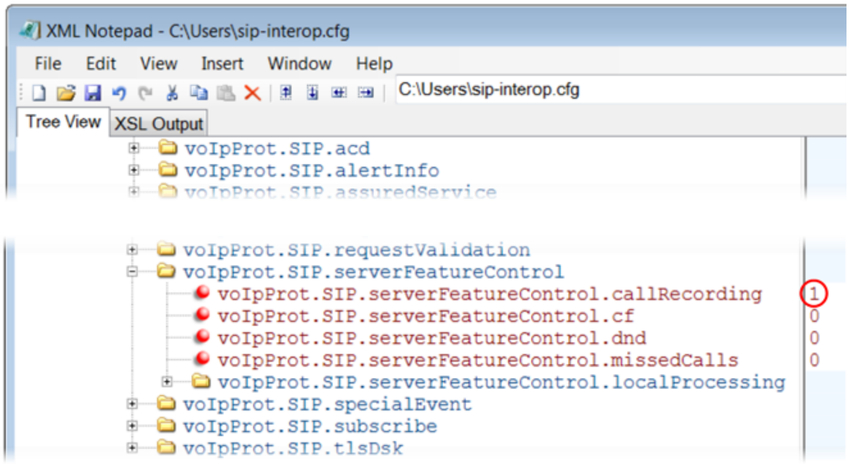
- **Never Mode** Call recording is never initiated and the phone never displays call recording soft keys.
- **Always Mode** In Always mode, the entire incoming or outgoing call is recorded and no control options are available to users. During active calls, the phone displays a Record symbol. Call recording stops when the call ends and the call is stored on the server.
- **Always with Pause/Resume Support Mode** Call recording starts automatically when the call connects and a Pause soft key displays. When you press the Pause soft key, a Resume soft key displays. The phone display indicates the status of the call recording state. Call recording stops when the call ends and the recorded part of the call is stored on the server.
- **On Demand Mode** In On Demand mode, recording mode starts on the server when the call connects but the recorded file is not saved until you press the **Record > Start** soft key. When you press the Start soft key, the recording is saved to the server and the phone displays the Pause and Resume soft keys
- **On Demand Mode with User-Initiated Start Mode** In On Demand with User-Initiated Start Mode, recording of a call does not begin automatically and a Record soft key displays. If you want to record during an active call, press Record to enter the recording sub-menu. Press the **Start** soft key to start recording and save to the server. While recording, the phone displays the Pause, Resume, and Stop soft keys.
- **Recording two separate calls and creating a conference** This mode enables you to record two participants as separate calls sessions when connected in a conference call. The server stores the conference call as two separate recording sessions.

**Troubleshooting: Record soft key does not display**

When you select the Start or Pause soft key while recording an active call and the server sends an error processing your selection, the Record soft key does not display for the duration of the call. To display the Record soft key again, end and then reconnect the call.

## Example Call Recording Configuration

This section provides an example configuration for the call recording feature.



```
voIpProt.SIP.acd
voIpProt.SIP.alertInfo
voIpProt.SIP.assuredService

voIpProt.SIP.requestValidation
voIpProt.SIP.serverFeatureControl
    voIpProt.SIP.serverFeatureControl.callRecording      1
    voIpProt.SIP.serverFeatureControl.cf                 0
    voIpProt.SIP.serverFeatureControl.dnd                0
    voIpProt.SIP.serverFeatureControl.missedCalls        0
    voIpProt.SIP.serverFeatureControl.localProcessing
voIpProt.SIP.specialEvent
voIpProt.SIP.subscribe
voIpProt.SIP.tlsDsk
```

## Use Busy Lamp Field

The busy lamp field (BLF) feature enables users to monitor the status of lines on remote phones, display remote party information, and answer incoming calls to remote phones (called directed call pickup). The BLF feature must be supported by a call server and the specific functions vary with the call server you use. You may need to consult your SIP server partner or Polycom channel partner to find out how to configure BLF.

Prior to UC Software 3.2 (Polycom SIP versions 2.1-3.1), a basic version of BLF was available on VVX 1500 phones. This basic BLF enables you to be monitored and to monitor idle and active phone states. As of UC Software 3.2, Polycom added enhanced BLF to all Polycom phones except VVX 1500 phones; this enhanced version also notifies you of the ringing state of a monitored phone. Currently, the enhanced version is available on all Polycom phones except VVX 1500 phones. You can monitor VVX 1500 phones and use VVX 1500 phones to monitor other phones; however, VVX 1500 phones monitoring other phones notify you of the idle state and active states of monitored phones and do not notify you of the ringing state. Note that BLF is not available with Polycom phones registered with Skype for Business Server.

The table Busy Lamp Field lists the parameters you may need to set. You can set up multiple BLF lines and monitor remote phones in active, ringing, and idle state. When BLF is enabled and you are monitoring a remote user, a BLF line key icon displays on the phone's screen. You can configure the line key label, and how call appearances and caller ID information are displayed. As of SIP 3.2.0, you can configure one-touch call park and retrieve and one-touch directed call pickup. Specifying the type of monitored resource as normal or automata changes the default actions of key presses. As the resource type, enter normal if the monitored resource type is a phone and automata if the monitored resource type is, for example, a call orbit. If you select normal, pressing the BLF line key places an active call on hold before dialing the selected BLF phone. If you select automata, pressing the BLF line key immediately transfers active calls to that resource. To learn how to configure a park orbit and for examples, refer to the section Configure Enhanced Feature Keys.

Note that how you manage calls on BLF lines depends on the state of your phone—whether it is in the idle, active, or alerting state.

**Web Info: Managing monitored lines**

For information on how to manage calls to monitored phones, see the section Handling Remote Calls on Attendant Phones in *Using Statically Configured Busy Lamp Field with Polycom SoundPoint IP and VVX Phones: Technical Bulletin 62475 at Polycom Engineering Advisories and Technical Notifications.*

As of the SIP 3.1.0 release, the BLF feature was updated in the following ways:

- The phone gives a visual and audible indication when monitored BLF lines have incoming calls.
- The phone displays the caller ID of incoming calls to a remote monitored phones.

  BLF lines display a Pickup soft key that you can press to answer incoming calls to that monitored resource.

As of the SIP 3.2 release, the BLF feature was updated in the following ways:

- You can create a list of monitored parties to a maximum of 47 and configure the line key labels.
- You can configure key functions.
- You can disable spontaneous call appearances from incoming calls on monitored lines.

The following call servers are known to support this feature:

- Back to Back2 User Agent (B2BUA) Architecture
  - Metaswitch Metasphere Call Feature Server (CFS)
  - Asterisk® v1.6 or later
  - BroadSoft® BroadWorks
- Proxy Architecture
  - Avaya® SipX Enterprise Communications Server (ECS)
  - eZuce openUC™

These proxy architectures may support the full range of statically configured BLF features. However, they do not provide configuration control through their web management console.

The following call servers may support this feature, depending on the call server software variation and deployment:

- Proxy Architecture
  - OpenSIPS (formerly OpenSER)
  - Repro ReSIProcate

These proxy architectures or any other proxy server that allows the phone end-to-end communications with the monitored phone should be supported. However, these solutions have not been specifically tested by Polycom nor does Polycom guarantee their full interoperability.

**Note: Use BLF with TCPpreferred transport**

Use this feature with TCPpreferred transport (see <server/>).

**Busy Lamp Field**

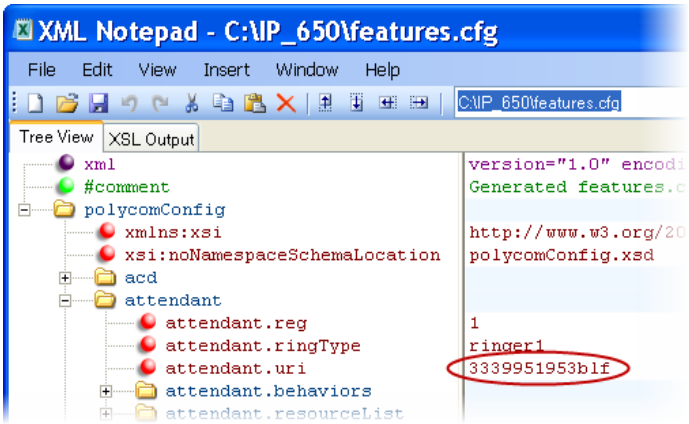
| **Parameter Function** | **template** > parameter |
|---|---|
| Specify an index number for the BLF resource. | **features.cfg** > attendant.reg |
| Specify the ringtone to play when a BLF dialog is in the offering state. | **features.cfg** > attendant.ringType |
| Specify the SIP URI of the call server resource list. | **features.cfg** > attendant.uri |
| Specify how call appearances and remote party caller ID display on the attendant phone. | **features.cfg** > attendant.behaviours.display.* |
| Specify the address of the monitored resource, a label for the resource, and the type of resource. | **features.cfg** > attendant.resourceList.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Example BLF Configuration

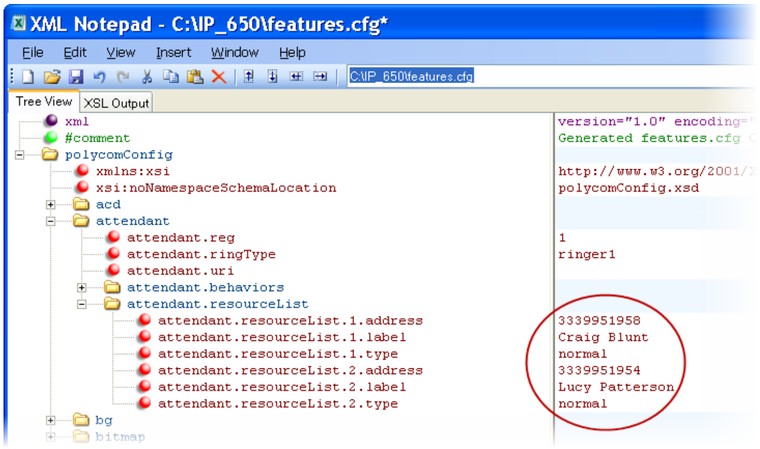
Typically, call servers support one of two methods of BLF configuration. Using the first method, you subscribe to a BLF resource list that is set up on your call server. Using the second method, you enter BLF resources to a configuration file and the call server directs the requests to those BLF resources. If you are unsure which method to use, consult your SIP server partner or Polycom Channel partner. This section shows you how to set up BLF using both methods.

To subscribe to a BLF list on a call server, you must access the call server and set up a list of monitored resources. The call server provides you with an address for that BLF resource list. To subscribe to that list, enter the address and any other information specific to your call server in the `attendant.uri` field located in the **features.cfg** template file, as shown next.



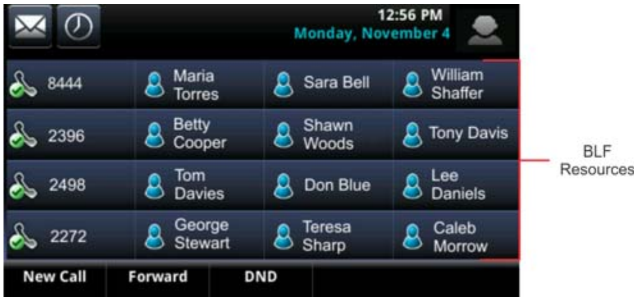
```
xml                                  version="1.0" encodi
#comment                             Generated features.c
polycomConfig
  xmlns:xsi                          http://www.w3.org/20
  xsi:noNamespaceSchemaLocation      polycomConfig.xsd
  acd
  attendant
    attendant.reg                    1
    attendant.ringType               ringer1
    attendant.uri                    3339951953blf
    attendant.behaviors
    attendant.resourceList
```

To specify BLF resources in the configuration file, open the **features.cfg** template file and enter the address (phone number) of the BLF resource you want to monitor, the label that displays beside the line key on the phone, and the type of resource you are monitoring. Multiple registrations are available for a single SIP server. Your call server must support static BLF in order to configure BLF using the static method. In the following example, the phone is monitoring *Craig Blunt* and *Lucy Patterson*.


XML Notepad - C:\IP_650\features.cfg*
File Edit View Insert Window Help
C:\IP_650\features.cfg
Tree View
XSL Output

```
xml                                             version="1.0" encoding="
#comment                                        Generated features.cfg 
polycomConfig
  xmlns:xsi                                     http://www.w3.org/2001/
  xsi:noNamespaceSchemaLocation                 polycomConfig.xsd
  acd
  attendant
    attendant.reg                               1
    attendant.ringType                          ringer1
    attendant.uri
    attendant.behaviors
    attendant.resourceList
      attendant.resourceList.1.address          3339951958
      attendant.resourceList.1.label            Craig Blunt
      attendant.resourceList.1.type             normal
      attendant.resourceList.2.address          3339951954
      attendant.resourceList.2.label            Lucy Patterson
      attendant.resourceList.2.type             normal
  bg
  bitmap
```

Both configuration methods result in the following BLF contacts—called BLF resources—beside line keys on the phone:



The following table shows the BLF key icons.

**BLF Line Key Icons**

| *States* | *Line Icons* |
|---|---|
| Line monitoring is active | 👤 |
| Monitored line is busy | 👤⛔ |
| Monitored line is ringing | ☎ |

# Configure Group Call Controls

This section provides information on configuring group call controls.

## Enable Instant Messaging

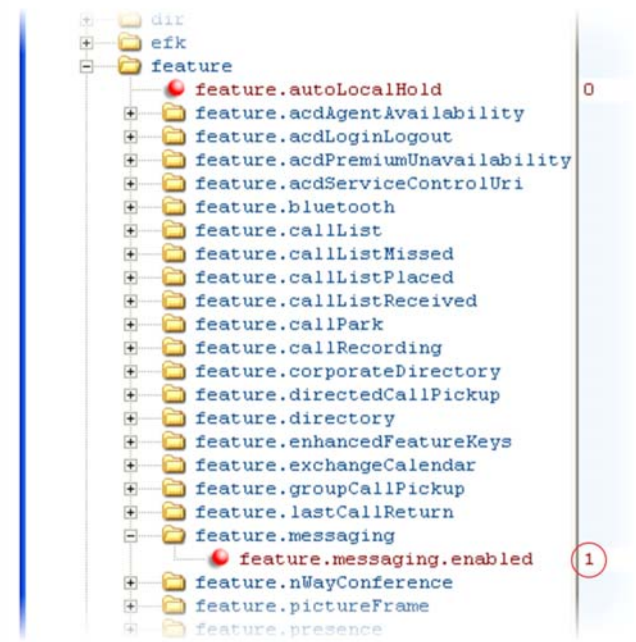
All phones can send and receive instant text messages. Phones registered with Microsoft Skype for Business Server cannot send or receive instant messages. See the following table for the parameter you need to set to enable instant messaging. Once the feature is enabled, the phone’s message waiting indicator (MWI) LED alerts you to incoming text messages visually; you can also set audio alerts. When you want to send an instant message, you can use the phone’s dial pad to type your messages or you can choose a short message from a preset list. You can send instant messages by initiating a new dialogue or by replying to a received message. In addition, you can choose the message destination manually or you can select a contact from your local contact directory; see the section Use the Local Contact Directory.

**Enable Instant Messaging**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable instant messaging. | **features.cfg** > feature.messaging.enabled |

### Example Instant Messaging Configuration

The following illustration shows you how to enable instant messaging in the features.cfg template.



```
dir
efk
feature
    feature.autoLocalHold                    0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
    feature.exchangeCalendar
    feature.groupCallPickup
    feature.lastCallReturn
    feature.messaging
        feature.messaging.enabled            1
    feature.nWayConference
    feature.pictureFrame
    feature.presence
```

After setting this parameter, press the Messages key on the phone’s keypad to display the Instant Messages option. Select the Instant Messages menu to send and receive instant messages.

# Enable Group Call Pickup

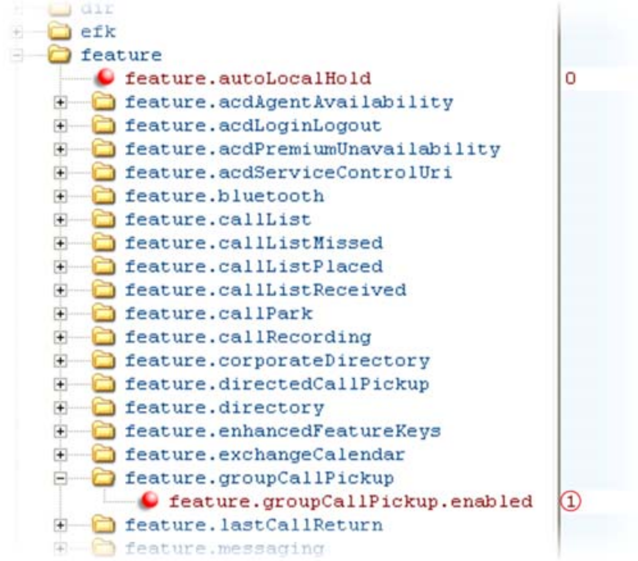
This feature enables you to pick up incoming calls to any phone within a predefined group of phones, without dialing the extension of another phone. The parameter to enable this feature is shown in the table Enable Group Call Pickup. This feature requires support from a SIP server and setup of this feature depends on the SIP server. For example, while some SIP servers implement group call pick-up using a particular star-code sequence, others implement the feature using network signaling.

**Enable Group Call Pickup**

| **Parameter Function** | **template** > parameter |
|---|---|
| Turn this feature on or off. | **features.cfg** > feature.groupCallPickup.enabled |

## Example Group Call Pickup Configuration

The following illustration shows you how to enable the group call pickup feature in the **features.cfg** template.



```
dir
efk
feature
    feature.autoLocalHold                          0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
    feature.exchangeCalendar
    feature.groupCallPickup
        feature.groupCallPickup.enabled            1
    feature.lastCallReturn
    feature.messaging
```

When you enable the group call pickup, the phone displays a **Pickup** soft key when you go off-hook. If you select **Pickup**, the **Group** soft key displays.

After you press the Group soft key, the phone performs a just-in-time subscription request to the fixed address `<groupcallpickup@<yourCallServerDomain>` for dialog details with which it can pick up the original caller using a replaces header in a new INVITE.

# Create Local and Centralized Conferences

You can set up local or centralized audio and video conferences. Local conferences require a host phone to process the audio and video of all parties. Alternatively, you can use an external audio bridge, available via a central server, to create a centralized conference call. All Polycom phones support local- and server-based centralized conferencing. Polycom recommends using centralized conferencing for conferences with four or more parties. The availability of centralized conferencing and features can vary by the call platform you use.

The maximum number of callers you can host in a local conference varies by phone:

- 
- **VVX phones.** Support three-way calls
- **SoundStructure VoIP Interface.** Supports three-way calls

See the parameters in the following table to set up a conference type and the options available for each type of conference. You can specify whether, when the host of a three-party local conference leaves the conference, the other two parties remain connected or disconnected. If you want the other two parties remain connected, the phone performs a transfer to keep the remaining parties connected. If the host of four-party local conference leaves the conference, all parties are disconnected and the conference call ends. If the host of a centralized conference leaves the conference, each remaining party remains connected. For more ways to manage conference calls, see Enable Conference Management.

**Create Local and Centralized Conferences**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify whether, during a conference call, the host can place all parties or only the host on hold. | **sip-interop.cfg** > call.localConferenceCallHold |
| Specify whether or not the remaining parties can communicate after the conference host exits the conference. | **sip-interop.cfg** > call.transferOnConferenceEnd |
| Specify whether or not all parties hear sound effects while setting up a conference. | **sip-interop.cfg** > call.singleKeyPressConference |
| Specify which type of conference to establish and the address of the centralized conference resource. | **sip-interop.cfg** > voIpProt.SIP.conference.address |

## Enable Conference Management

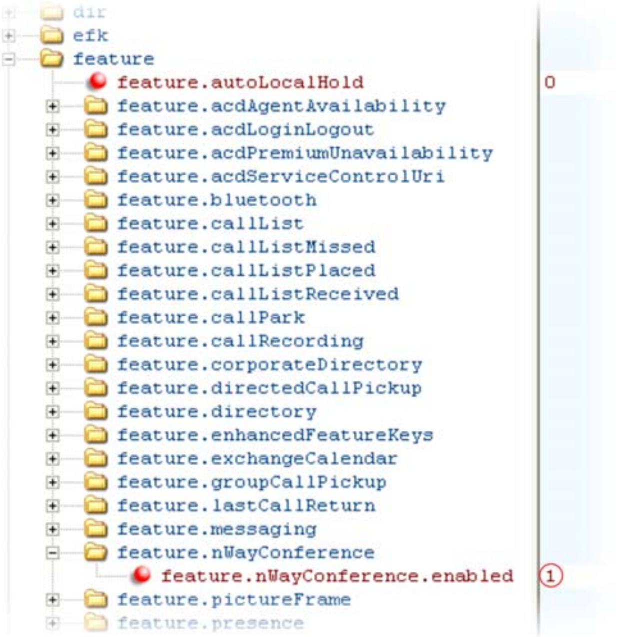
This feature enables you to add, hold, mute, and remove conference participants, as well as obtain additional information about participants. Use the parameters listed in the table Manage Conferences to configure how you want to manage conferences. VVX phone users can choose which conference call participants to exchange video with.

**Manage Conferences**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the conference management feature. | **features.cfg** > feature.nWayConference.enabled |

### Example Conference Management Configuration

The following example shows you how to enable the conference management feature in the features.cfg file.



```
dir
efk
feature
    feature.autoLocalHold                      0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
    feature.exchangeCalendar
    feature.groupCallPickup
    feature.lastCallReturn
    feature.messaging
    feature.nWayConference
        feature.nWayConference.enabled         1
    feature.pictureFrame
    feature.presence
```

When you enable conference management, a **Manage** soft key displays on the phone during a conference. When you press the **Manage** soft key, the **Manage Conference** screen displays with soft keys you can use to manage conference participants.

## Configure Intercom Calls

The Intercom feature enables users to place an intercom call that is answered automatically on the dialed contact's phone. This is a server-independent feature provided the server does not alter the Alert-Info header sent in the INVITE. You can configure the behavior of the answering phone using `voIpProt.SIP.alertInfo.x.class`.

The following procedure tells you how to place an intercom call when the intercom feature is enabled using configuration parameters.

### To place an intercom call:

1 Press the **Intercom** soft key.

The New Call screen displays.

2 In the **New Call** screen, enter a number or select a contact from the directory or call lists.

You have the option to initiate intercom calls using enhanced feature keys (EFKs). For information on configuring EFK functions, see the section Configure Enhanced Feature Keys. You do not need to disable the default Intercom soft key to create a custom soft key. For example, you can create an intercom action string:

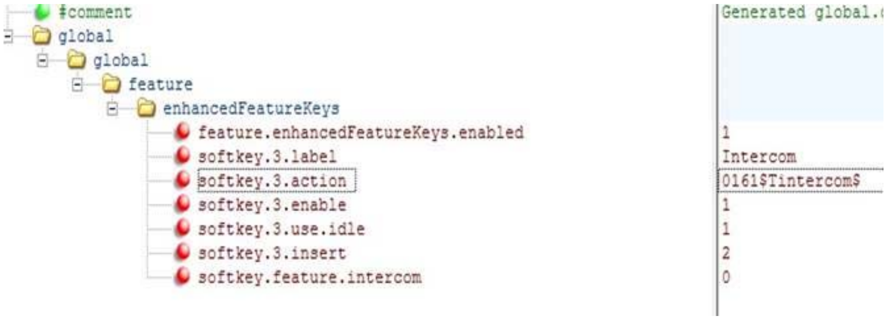
- $FIntercom$

  This is an F type macro that behaves as a custom Intercom soft key. Pressing the soft key opens the Intercom dial prompt you can use to place an Intercom call by entering the destination’s digits and using a speed dial or BLF button.

- <number>$Tintercom$

  This is a T type macro enables you to specify a Direct intercom button that always calls the number you specify in <number>. No other input is necessary.

In the following illustration, the action given as *0161$Tintercom*. When the Intercom soft key is pressed, an intercom call is placed to 0161. Because `softkey.3.insert` is set to 2, the Intercom soft key displays at the second position. However, for some features, soft key positions are fixed and in this example, PTT is enabled, which means a PTT soft key occupies a fixed second position and the Intercom soft key displays at the third position, as shown next.



```
#comment                                   Generated global.(
global
  global
    feature
      enhancedFeatureKeys
        feature.enhancedFeatureKeys.enabled  1
        softkey.3.label                      Intercom
        softkey.3.action                     0161$Tintercom$
        softkey.3.enable                     1
        softkey.3.use.idle                   1
        softkey.3.insert                     2
        softkey.feature.intercom             0
```

**Intercom Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the intercom feature. | **features.cfg** > feature.intercom.enable |
| Enable or disable the Intercom icon on the device home screen. | **features.cfg** > homescreen.intercom.enable |
| Enable or disable the intercom soft key. | **features.cfg** > softkey.feature.intercom |
| The string you want to use in the Alert-Info header. | **sip-interop.cfg** > voIpProt.SIP.intercom.alertInfo |
| A string to match the Alert-Info header in the incoming INVITE. | **sip-interop.cfg** > voIpProt.SIP.alertInfo.x.value |
| Specify a ring class name. | **sip-interop.cfg** > voIpProt.SIP.alertInfo.x.class |

## Configure Push-to-Talk and Group Paging

The push-to-talk (PTT) and group paging features are supported on all Polycom phone models installed with UC Software 4.0.0 or later.

The group paging feature enables you to make pages —one-way audio announcements—to users subscribed to a page group. The PTT feature is a collaborative tool that enables you to exchange broadcasts to users subscribed to a PTT channel, much like a walkie-talkie. You can transmit pages and PTT broadcasts using your handset, headset, or speakerphone and you can reject them, place them on hold, and end them at any time. PTT broadcasts can be received on the speakerphone, handset, and headset, and pages can be received only through the speakerphone. Both features are available on all phones that use UC Software 4.0.0 or later.

You can enable one of these features or you can operate both simultaneously. There are 25 groups/channels you can subscribe to.

- **PTT Mode**—PTT mode is intended primarily for Wi-Fi phones. In PTT mode, the phone behaves like a walkie-talkie; you can broadcast audio to a PTT channel and recipients subscribed to that channel can respond to your message. To configure PTT, see the table Configure Push-to-Talk.
- **Paging Mode**—Paging mode is intended primarily for desktop phones. in paging mode, you can send announcements to recipients subscribed to a page group. In page mode, announcements play only through the phone’s speakerphone. To configure paging, see the table Configure Group Paging.

Administrators must enable paging and PTT before users can subscribe to a page group or PTT channel.

**Web Info: Using a different IP multicast address**

The push-to-talk and group paging features use an ip multicast address. if you want to change the default IP multicast address, ensure that the new address does not already have an official purpose as specified in the IPv4 Multicast Address Space Registry.

## Push-to-Talk

You specify the same IP multicast address in the parameter ptt.address for both PTT and paging mode. PTT administrator settings are located in the site.cfg template file. The parameters shown in the following table are located in the features.cfg template file.

## Group Paging

### Configure Push-to-Talk

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the IP multicast address used for the PTT and paging features. | **site.cfg** > ptt.address |
| Enable PTT mode. | **site.cfg** > ptt.pageMode.enable |
| Specify the name to display (per phone). | **site.cfg** > ptt.pageMode.displayName |
| Change default settings for PTT mode. | **site.cfg** > ptt.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

You specify the same IP multicast address in the parameter ptt.address for both PTT and paging mode. Paging administrator settings shown in the following table are located in the site.cfg template file. Page group settings are located in the features.cfg template file.

### Configure Group Paging

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the IP multicast address used for the PTT and paging features. | **site.cfg** > ptt.address |

### Configure Group Paging

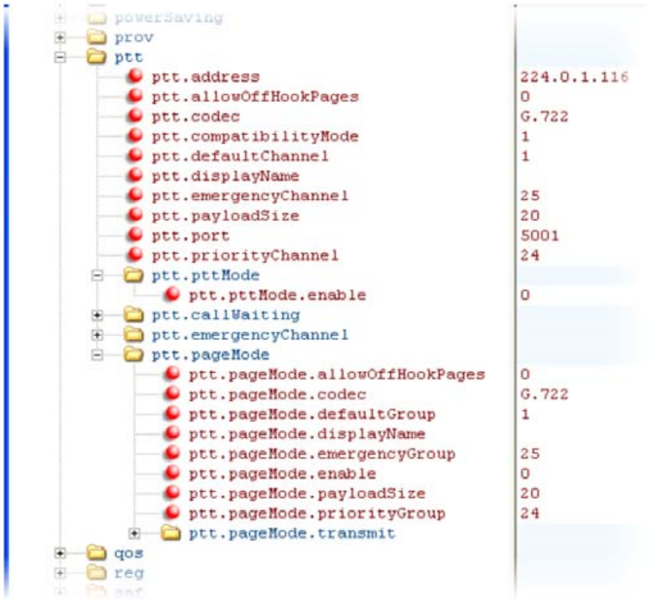
| | |
|---|---|
| Enable paging mode. | **site.cfg** > ptt.pageMode.enable |
| Specify the display name. | **site.cfg** > ptt.pageMode.displayName |
| Specify settings for all page groups. | **features.cfg** > ptt.pageMode.group.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

**Web Info: Configuring push-to-talk and group paging**
Though the example configurations in this section get you started, Polycom recommends that you read how to use PTT and paging in the *Polycom VVX Business Media Phones User Guide* before configuring settings.

## Example PTT/Paging Configuration

The following illustration shows the default PTT and paging administrator settings in the site.cfg template file.

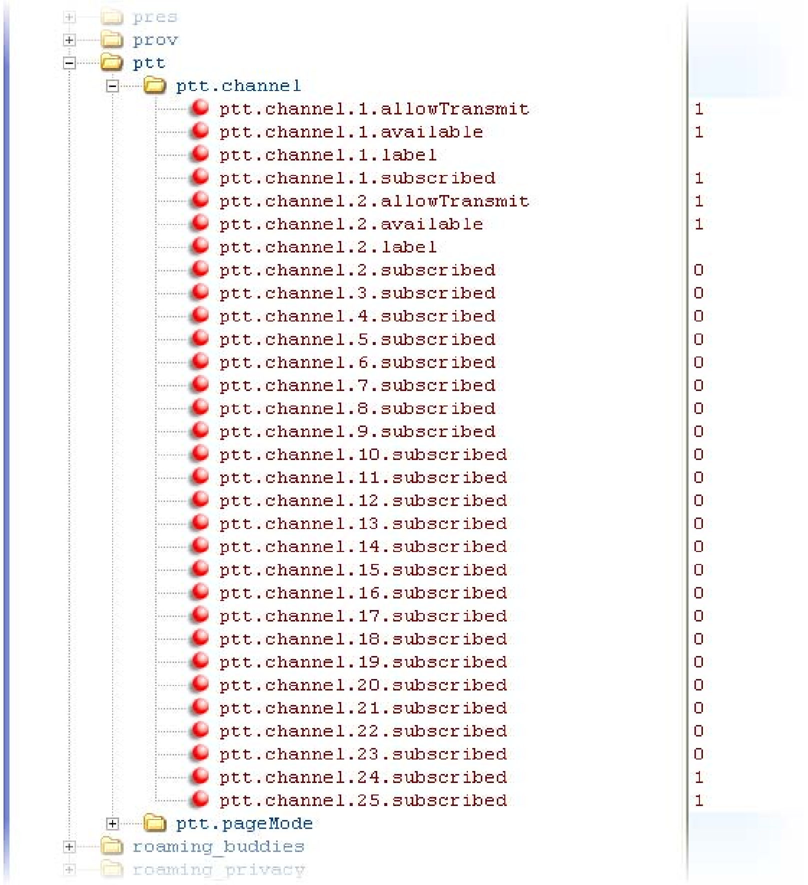


```
powerSaving
prov
ptt
    ptt.address                          224.0.1.116
    ptt.allowOffHookPages                0
    ptt.codec                            G.722
    ptt.compatibilityMode                1
    ptt.defaultChannel                   1
    ptt.displayName
    ptt.emergencyChannel                 25
    ptt.payloadSize                      20
    ptt.port                             5001
    ptt.priorityChannel                  24
    ptt.pttMode
        ptt.pttMode.enable               0
    ptt.callWaiting
    ptt.emergencyChannel
    ptt.pageMode
        ptt.pageMode.allowOffHookPages   0
        ptt.pageMode.codec               G.722
        ptt.pageMode.defaultGroup        1
        ptt.pageMode.displayName
        ptt.pageMode.emergencyGroup      25
        ptt.pageMode.enable              0
        ptt.pageMode.payloadSize         20
        ptt.pageMode.priorityGroup       24
        ptt.pageMode.transmit
qos
reg
```

Note that you can enter a display name for sent PTT broadcasts in ptt.displayName and for sent page announcements in ptt.pageMode.displayName. The two following illustrations show the range of PTT channels and page groups you can subscribe to.

You can subscribe to the following PTT channels. Note that channels one and two are enabled by default, and that channels 24 and 25, the priority and emergency channels respectively, are also enabled by default.

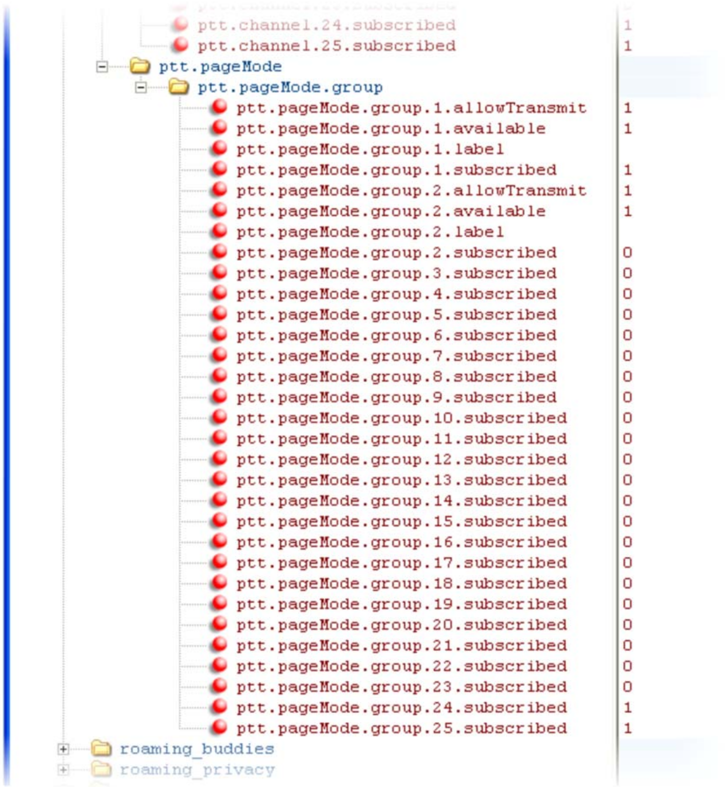
### PTT Mode Channels



```
pres
prov
ptt
  ptt.channel
    ptt.channel.1.allowTransmit    1
    ptt.channel.1.available        1
    ptt.channel.1.label
    ptt.channel.1.subscribed       1
    ptt.channel.2.allowTransmit    1
    ptt.channel.2.available        1
    ptt.channel.2.label
    ptt.channel.2.subscribed       0
    ptt.channel.3.subscribed       0
    ptt.channel.4.subscribed       0
    ptt.channel.5.subscribed       0
    ptt.channel.6.subscribed       0
    ptt.channel.7.subscribed       0
    ptt.channel.8.subscribed       0
    ptt.channel.9.subscribed       0
    ptt.channel.10.subscribed      0
    ptt.channel.11.subscribed      0
    ptt.channel.12.subscribed      0
    ptt.channel.13.subscribed      0
    ptt.channel.14.subscribed      0
    ptt.channel.15.subscribed      0
    ptt.channel.16.subscribed      0
    ptt.channel.17.subscribed      0
    ptt.channel.18.subscribed      0
    ptt.channel.19.subscribed      0
    ptt.channel.20.subscribed      0
    ptt.channel.21.subscribed      0
    ptt.channel.22.subscribed      0
    ptt.channel.23.subscribed      0
    ptt.channel.24.subscribed      1
    ptt.channel.25.subscribed      1
  ptt.pageMode
roaming_buddies
roaming_privacy
```

You can subscribe to the following paging groups. Note that groups one and two are enabled by default, and that groups 24 and 25, the priority and emergency channels respectively, are also enabled by default.

### Paging Mode Groups



```
ptt.channel.24.subscribed              1
ptt.channel.25.subscribed              1
ptt.pageMode
  ptt.pageMode.group
    ptt.pageMode.group.1.allowTransmit   1
    ptt.pageMode.group.1.available       1
    ptt.pageMode.group.1.label
    ptt.pageMode.group.1.subscribed      1
    ptt.pageMode.group.2.allowTransmit   1
    ptt.pageMode.group.2.available       1
    ptt.pageMode.group.2.label
    ptt.pageMode.group.2.subscribed      0
    ptt.pageMode.group.3.subscribed      0
    ptt.pageMode.group.4.subscribed      0
    ptt.pageMode.group.5.subscribed      0
    ptt.pageMode.group.6.subscribed      0
    ptt.pageMode.group.7.subscribed      0
    ptt.pageMode.group.8.subscribed      0
    ptt.pageMode.group.9.subscribed      0
    ptt.pageMode.group.10.subscribed     0
    ptt.pageMode.group.11.subscribed     0
    ptt.pageMode.group.12.subscribed     0
    ptt.pageMode.group.13.subscribed     0
    ptt.pageMode.group.14.subscribed     0
    ptt.pageMode.group.15.subscribed     0
    ptt.pageMode.group.16.subscribed     0
    ptt.pageMode.group.17.subscribed     0
    ptt.pageMode.group.18.subscribed     0
    ptt.pageMode.group.19.subscribed     0
    ptt.pageMode.group.20.subscribed     0
    ptt.pageMode.group.21.subscribed     0
    ptt.pageMode.group.22.subscribed     0
    ptt.pageMode.group.23.subscribed     0
    ptt.pageMode.group.24.subscribed     1
    ptt.pageMode.group.25.subscribed     1
roaming_buddies
roaming_privacy
```

## Use Hoteling

The hoteling feature enables users to use any available shared phone by logging in to a guest profile. The following table shows you the parameters you can configure. After logging in, users have access to their own guest profile and settings on the shared phone. This feature is available on Polycom VVX 300 series, 400 series, 500 series, 600 series, and 1500 phones. To use Hoteling, you must configure Polycom phones with the BroadSoft BroadWorks R17 platform and use UC Software 4.0.2 or later.

**Web Info: Use the hoteling feature**

For details on configuring the hoteling feature, see *Using Hoteling on Polycom Phones*: *Feature Profile 76554* at Polycom Engineering Advisories and Technical Notifications.

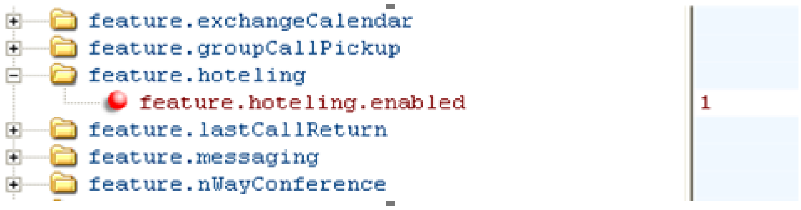
You can use hoteling in conjunction with the feature-synchronized automatic call distribution (ACD) feature. For information, see the section Configure Feature-Synchronized Automatic Call Distribution (ACD).

**Use Hoteling**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable hoteling. | **features.cfg** > feature.hoteling.enabled |
| Choose a line registration index. | **features.cfg** > hoteling.reg |

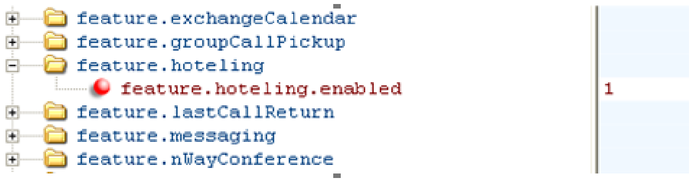
## Example Hoteling Configuration

This example configuration shows the hoteling feature enabled and uses registration line 1. In the features.cfg template, the feature.hoteling.enabled parameter is set to 1 to enable.



```
feature.exchangeCalendar
feature.groupCallPickup
feature.hoteling
    feature.hoteling.enabled        1
feature.lastCallReturn
feature.messaging
feature.nWayConference
```

The hoteling feature is applied to phone line 1.



```
feature.exchangeCalendar
feature.groupCallPickup
feature.hoteling
    feature.hoteling.enabled        1
feature.lastCallReturn
feature.messaging
feature.nWayConference
```

When hoteling is enabled, the line 1 index key *2326* has hoteling enabled and the **GuestIn** soft key displays.

# Configure SIP-B Automatic Call Distribution

You can use your VVX business media phones in a call center agent/supervisor role on a supported call server.

**Configure SIP-B Automatic Call Distribution**

| Parameter Function | **template** > parameter |
|---|---|
| To turn Automatic Call Distribution on or off. | **features.cfg** > feature.acdLoginLogout.enabled |
| To enable or disable Automatic Call Distribution for a specific registration. | **reg-advanced.cfg** > reg.x.acd-login-logout |
| To enable or disable Feature Synchronized ACD. | **sip-interop.cfg** > voIpProt.SIP.acd.signalingMethod |

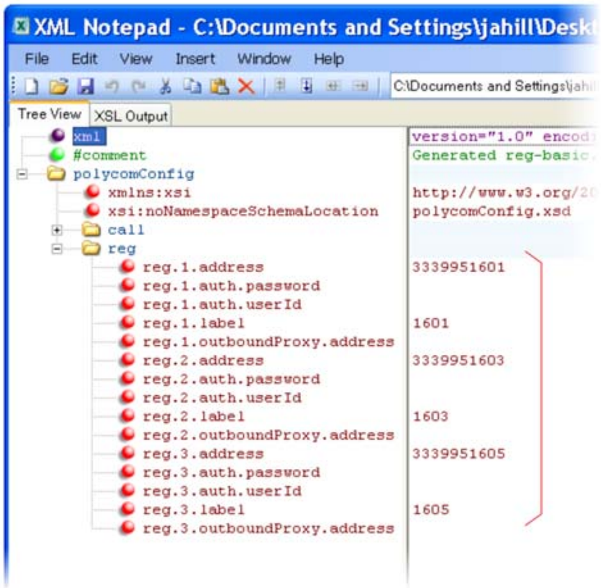
The also support ACD agent availability. This feature depends on support from a SIP server.

**ACD Agent Availability**

| **Parameter Function** | **template** > parameter |
|---|---|
| To turn ACD Agent Availability on or off. | **features.cfg** > feature.acdAgentAvailable.enabled |
| To enable or disable ACD Agent Availability feature for a specific registration. | **reg-advanced.cfg** > reg.x.acd-agent-available |

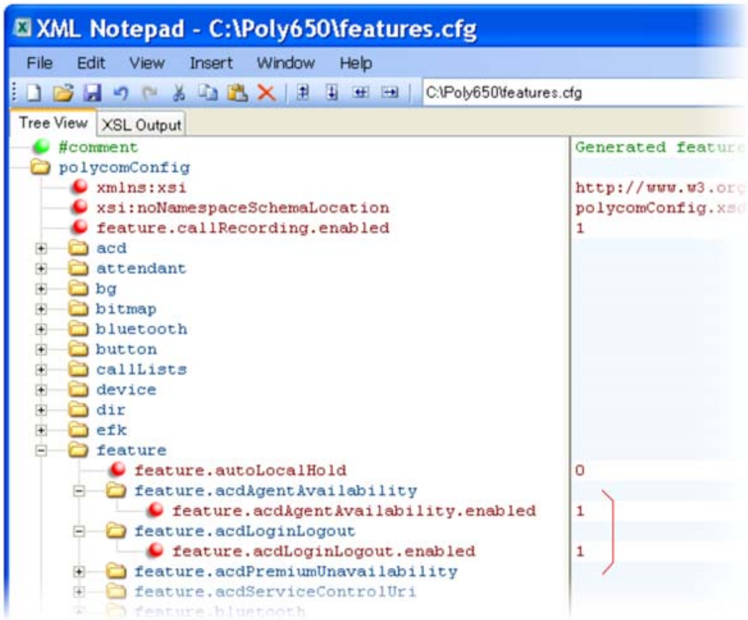
## Example SIP-B Automatic Call Distribution Configuration

In the following illustration, in the **reg-basic.cfg** template file, three line registrations and labels have been set up.


XML Notepad - C:\Documents and Settings\jahill\Deskt
File
Edit
View
Insert
Window
Help
C:\Documents and Settings\jahil
Tree View
XSL Output

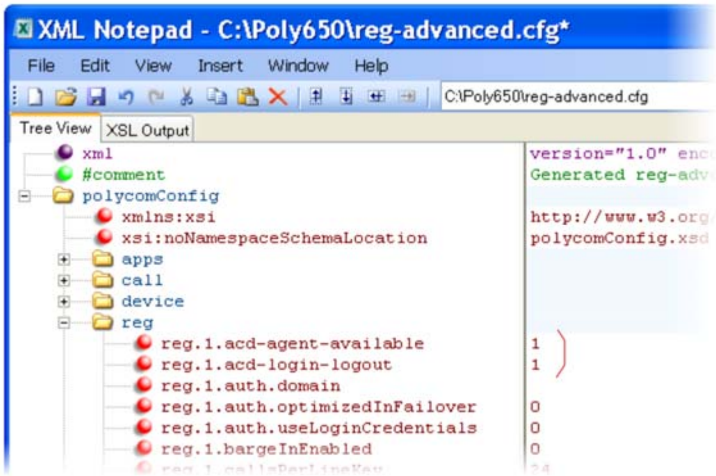
```
xml                                 version="1.0" encod
#comment                            Generated reg-basic.
polycomConfig
    xmlns:xsi                       http://www.w3.org/20
    xsi:noNamespaceSchemaLocation   polycomConfig.xsd
    call
    reg
        reg.1.address               3339951601
        reg.1.auth.password
        reg.1.auth.userId
        reg.1.label                 1601
        reg.1.outboundProxy.address
        reg.2.address               3339951603
        reg.2.auth.password
        reg.2.auth.userId
        reg.2.label                 1603
        reg.2.outboundProxy.address
        reg.3.address               3339951605
        reg.3.auth.password
        reg.3.auth.userId
        reg.3.label                 1605
        reg.3.outboundProxy.address
```

In this example, SIP-B ACD is enabled in **features.cfg** using the parameters feature.acdAgentAvailability.enabled and feature.acdLoginLogout.enabled, as shown next.



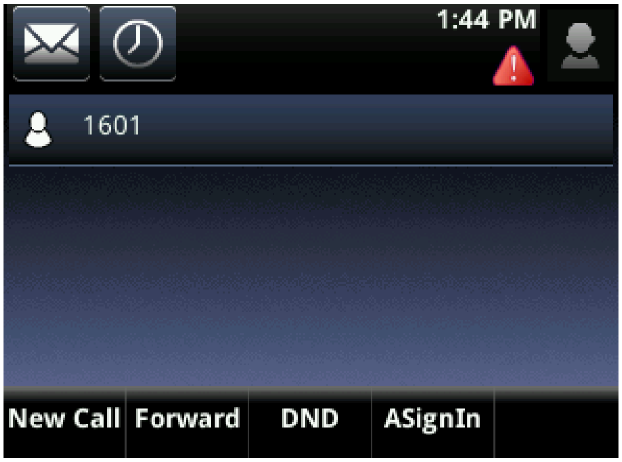
```
#comment                                    Generated featur
polycomConfig
  xmlns:xsi                                 http://www.w3.or
  xsi:noNamespaceSchemaLocation             polycomConfig.xs
  feature.callRecording.enabled             1
  acd
  attendant
  bg
  bitmap
  bluetooth
  button
  callLists
  device
  dir
  efk
  feature
    feature.autoLocalHold                   0
    feature.acdAgentAvailability
      feature.acdAgentAvailability.enabled  1
    feature.acdLoginLogout
      feature.acdLoginLogout.enabled        1
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
```

You must also enable SIP-B ACD in the **reg-advanced.cfg** template file. The next illustration shows the two parameters you need to enable to display the ACD soft keys on the phone screen.


XML Notepad - C:\Poly650\reg-advanced.cfg*
File
Edit
View
Insert
Window
Help
C:\Poly650\reg-advanced.cfg
Tree View
XSL Output

```
xml                                  version="1.0" enc
#comment                             Generated reg-adv
polycomConfig
  xmlns:xsi                          http://www.w3.org
  xsi:noNamespaceSchemaLocation      polycomConfig.xsd
  apps
  call
  device
  reg
    reg.1.acd-agent-available        1
    reg.1.acd-login-logout           1
    reg.1.auth.domain
    reg.1.auth.optimizedInFailover   0
    reg.1.auth.useLoginCredentials   0
    reg.1.bargeInEnabled             0
    reg.1.callsPerLineKey            24
```

Once SIP-B ACD is enabled, the following soft keys display on the phone.



The ACD agent 1601 displays on phone line 1 and the agent can log in and out of the ACD feature.

## Configure Feature-Synchronized Automatic Call Distribution (ACD)

You can use your VVX phones in a call center agent/supervisor role on a supported call server. Feature-synchronized ACD is distinct from and provides more advanced ACD functions than the Hoteling feature (see Use Hoteling).

Feature-synchronized automatic call distribution (ACD) enables organizations that handle a large number of incoming phone calls to use in a call center role. Feature-synchronized ACD is available as a standard or a premium service. The premium ACD service has been enhanced in two ways: *Hoteling* and *Queue Status Notification*. Hoteling enables agents to use their agent credentials to log in to any available phone. If you want to use the hoteling feature with feature-synchronized ACD, see the section Use Hoteling. Queue status notification enables agents to view the queue status of a call center so that agents can adjust their call response.

**Web Info: Further information on ACD enhancements**

For more information on standard and premium ACD as well as the hoteling and queue status notification enhancements, see *Using Premium Automatic Call Distribution for Call Centers: Feature Profile 76179* at Polycom Engineering Advisories and Technical Notifications.

See the following table for parameters you can configure. When standard functions are enabled, the phone indicates it is in the ACD call center agent state. Phone users can sign in and sign out of the ACD state as a call center agent using soft keys or the phone’s menu. When ACD is enabled and a user is signed in as an agent, the phone can display the current state of the agent, for example, whether the agent is available or unavailable to take new calls.

The capabilities of this feature vary with the SIP call server. Please consult your call server provider for information and for documentation. The SIP signaling used for this implementation is described in the BroadSoft BroadWorks document *Device Key Synchronization Requirements Document; Release R14 sp2; Document version 1.6*.

The following phones support the feature-synchronized ACD feature:

- VVX 300 series, 400 series, 500 series, 600 series, and 1500 business media phones. Note that you must use UC Software 5.0 or later to use this feature with the VVX business media phones.

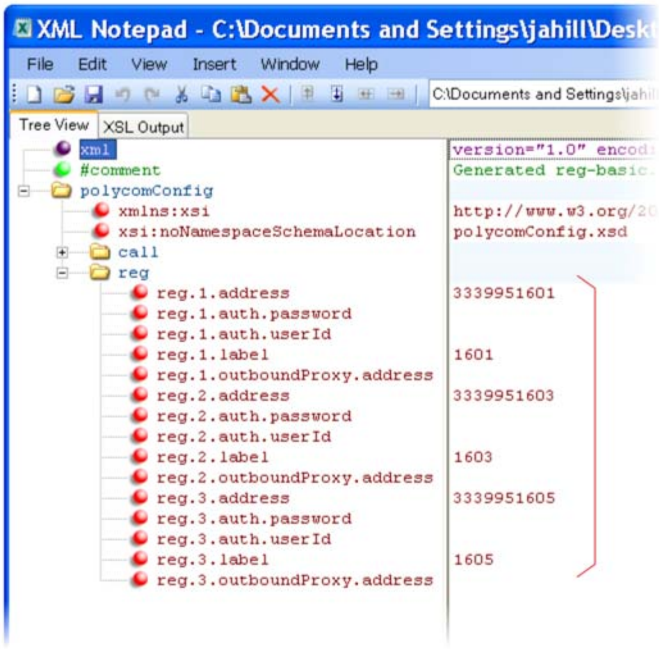
**Configure Feature Synchronized Automatic Call Distribution**

| Parameter Function | **template** > parameter |
|---|---|
| To turn Feature Synchronized ACD on or off. | **features.cfg** > feature.acdLoginLogout.enabled |
| To turn ACD Agent Availability on or off. | **features.cfg** > feature.acdAgentAvailable.enabled |
| To turn Premium Feature Synchronized ACD on or off. | **features.cfg** > feature.acdPremiumUnavailability.enabled |
| To turn Feature Synchronized ACD Control URI on or off. | **features.cfg** > feature.acdServiceControlUri.enabled |
| To set the registration to be used for Feature Synchronized ACD and the users' sign-in state. | **features.cfg** > acd.* |
| To enable or disable Feature Synchronized ACD. | **sip-interop.cfg** > voIpProt.SIP.acd.signalingMethod |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

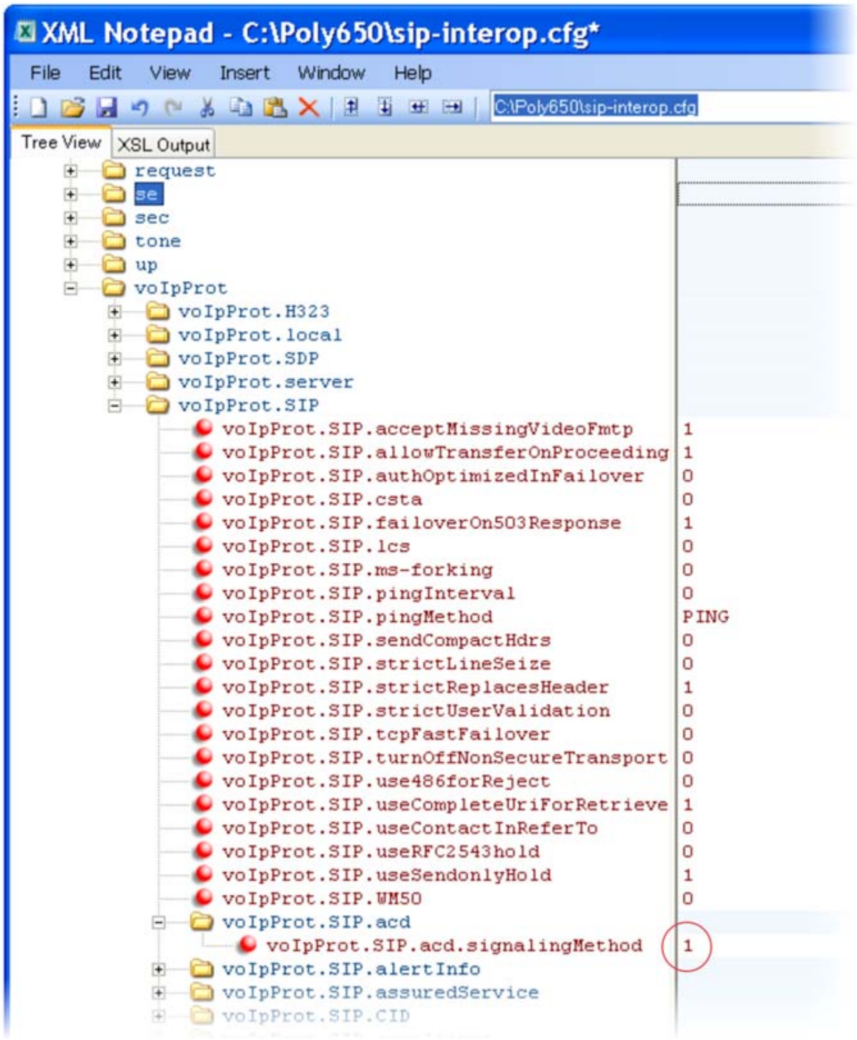
## Example Feature Synchronized ACD Configuration

In the following illustration, in the **reg-basic.cfg** template file, three line registrations and labels have been set up.


XML Notepad - C:\Documents and Settings\jahill\Deskt
File
Edit
View
Insert
Window
Help
C:\Documents and Settings\jahil
Tree View
XSL Output

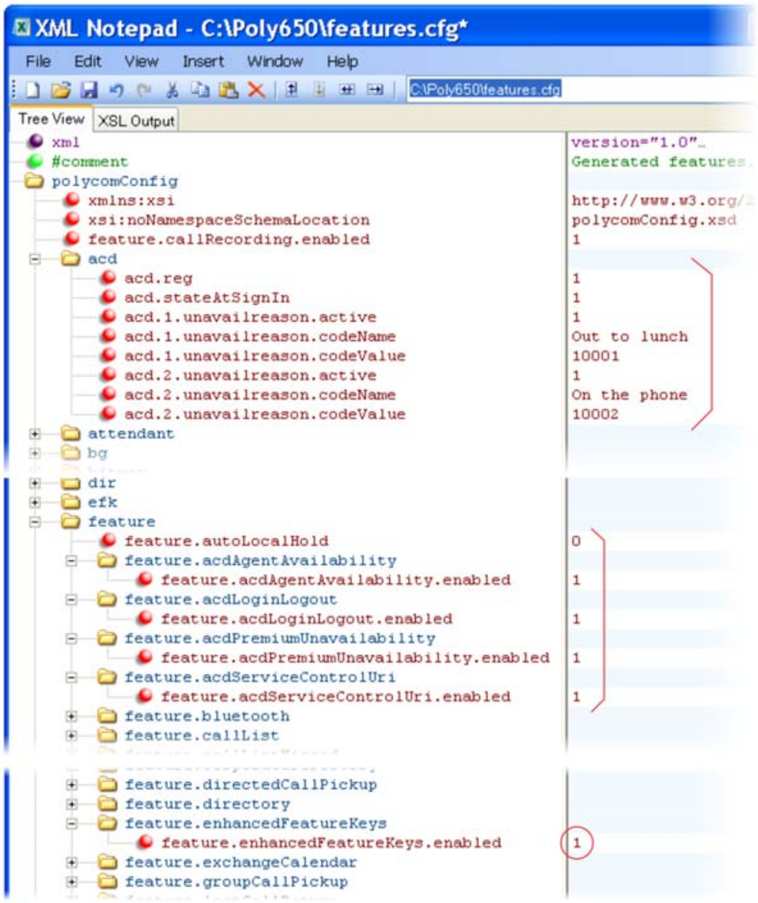
| | |
|---|---|
| xml | version="1.0" encodi |
| #comment | Generated reg-basic. |
| polycomConfig | |
| xmlns:xsi | http://www.w3.org/20 |
| xsi:noNamespaceSchemaLocation | polycomConfig.xsd |
| call | |
| reg | |
| reg.1.address | 3339951601 |
| reg.1.auth.password | |
| reg.1.auth.userId | |
| reg.1.label | 1601 |
| reg.1.outboundProxy.address | |
| reg.2.address | 3339951603 |
| reg.2.auth.password | |
| reg.2.auth.userId | |
| reg.2.label | 1603 |
| reg.2.outboundProxy.address | |
| reg.3.address | 3339951605 |
| reg.3.auth.password | |
| reg.3.auth.userId | |
| reg.3.label | 1605 |
| reg.3.outboundProxy.address | |

To enable feature-synchronized ACD for these registrations, in the sip-interop.cfg template file, set voIpProt.SIP.acd.signalingMethod to 1, as shown next.


XML Notepad - C:\Poly650\sip-interop.cfg*
File Edit View Insert Window Help
C:\Poly650\sip-interop.cfg
Tree View
XSL Output

```
request
se
sec
tone
up
voIpProt
  voIpProt.H323
  voIpProt.local
  voIpProt.SDP
  voIpProt.server
  voIpProt.SIP
    voIpProt.SIP.acceptMissingVideoFmtp        1
    voIpProt.SIP.allowTransferOnProceeding     1
    voIpProt.SIP.authOptimizedInFailover       0
    voIpProt.SIP.csta                          0
    voIpProt.SIP.failoverOn503Response         1
    voIpProt.SIP.lcs                           0
    voIpProt.SIP.ms-forking                    0
    voIpProt.SIP.pingInterval                  0
    voIpProt.SIP.pingMethod                    PING
    voIpProt.SIP.sendCompactHdrs               0
    voIpProt.SIP.strictLineSeize               0
    voIpProt.SIP.strictReplacesHeader          1
    voIpProt.SIP.strictUserValidation          0
    voIpProt.SIP.tcpFastFailover               0
    voIpProt.SIP.turnOffNonSecureTransport     0
    voIpProt.SIP.use486forReject               0
    voIpProt.SIP.useCompleteUriForRetrieve     1
    voIpProt.SIP.useContactInReferTo           0
    voIpProt.SIP.useRFC2543hold                0
    voIpProt.SIP.useSendonlyHold               1
    voIpProt.SIP.WM50                          0
    voIpProt.SIP.acd
      voIpProt.SIP.acd.signalingMethod         1
    voIpProt.SIP.alertInfo
    voIpProt.SIP.assuredService
    voIpProt.SIP.CID
```

A shown next, you must enable the feature.enhancedFeaturekeys.enabled parameter, four parameters in feature.acd*, and the acd.reg and acd.stateAtSignIn parameters. If you want to use reason codes, enable acd.unavailreason.active and enter the reason codes in the acd.x.unavailreason.codeName and acd.x.unavailreason.codeValue parameters. You can define up to 100 reason codes. In the following example, two reason codes have been enabled and set to Out to lunch and On the phone.



```
xml                                                   version="1.0"
#comment                                              Generated features
polycomConfig
  xmlns:xsi                                           http://www.w3.org/
  xsi:noNamespaceSchemaLocation                       polycomConfig.xsd
  feature.callRecording.enabled                       1
  acd
    acd.reg                                           1
    acd.stateAtSignIn                                 1
    acd.1.unavailreason.active                        1
    acd.1.unavailreason.codeName                      Out to lunch
    acd.1.unavailreason.codeValue                     10001
    acd.2.unavailreason.active                        1
    acd.2.unavailreason.codeName                      On the phone
    acd.2.unavailreason.codeValue                     10002
  attendant
  bg
  dir
  efk
  feature
    feature.autoLocalHold                             0
    feature.acdAgentAvailability
      feature.acdAgentAvailability.enabled            1
    feature.acdLoginLogout
      feature.acdLoginLogout.enabled                  1
    feature.acdPremiumUnavailability
      feature.acdPremiumUnavailability.enabled        1
    feature.acdServiceControlUri
      feature.acdServiceControlUri.enabled            1
    feature.bluetooth
    feature.callList
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
      feature.enhancedFeatureKeys.enabled             1
    feature.exchangeCalendar
    feature.groupCallPickup
```

The ACD agent number displays on the screen and the available status soft keys display: New Call, Forward, Unavailable. When the agent presses the Unavailable soft key, the reason codes you entered display for the agent to select.

**Web Info: Configuration details for feature-synchronized ACD**

For details on how to configure phones for feature-synchronized ACD, see *Using Feature-Synchronized Automatic Call Distribution with Polycom Phones: Feature Profile 57216* at Polycom Engineering Advisories and Technical Notifications.

# Customize Phone Functions

This section shows you how to set up a number of custom phone functions.

## Lock the Basic Settings Menu

By default, all users can access the Basic settings menu available on VVX phones. Using this menu, users can customize non-administrative features on their phone. You can choose to lock the Basic settings menu and only allow certain users access to the menu to customize the phone.

If enabled, you can use the default user password (123) or administrator password (456) to access the Basic settings menu, unless the default passwords are not in use.

**Lock the Basic Settings Menu**

| **Parameter Function** | **template** > parameter |
|---|---|
| Require a password to access the Basic settings menu on the phone. | **features.cfg** ><br>up.basicSettingsPasswordEnabled |

## Use the Microbrowser and Web Browser

The VVX business media phones support a full web browser. The microbrowser and browser parameters you can configure are listed in the following table. Note that the exact functions and performance of the microbrowser and web browser vary with the model of phone you are using.

You can configure the microbrowser and web browser to display a non-interactive web page on the phone's idle screen, and you can specify an interactive home web page that you can launch in a web browser by pressing the **Applications** key on the phone or by navigating to **Menu > Applications**. On the VVX 1500 phone, you can launch the web browser by pressing the **App** key on the phone or by navigating to **Menu > Applications**. On the VVX 300 series, 400 series, 500 series, and 600 series phones, go to **Home > Applications**. On the VVX only, when you tap on a link that displays on the idle browser the phone launches that link in the web browser.

Polycom provides a default microbrowser and browser feature for the phone's idle screen. My Info Portal is a Polycom-developed application that gives you access to the latest news, sports, weather, stock, and other news. You can sign up for access to My Info Portal through the Polycom VVX 1500 phone or through a computer. Note that the first time you sign in to My Info Portal, you are asked to accept the Polycom End User Licensing Agreement (EULA).

**Note: My Info Portal might require browser setting changes**

To get the My Info Portal to appear in the VVX phones' idle browser, set `mb.idleDisplay.home to http://idle.myinfoportal.apps.polycom.com/idle` and `mb.idleDisplay.refresh to 600, or set mb.main.home =http://myinfoportal.apps.polycom.com`.

**Note: Web browser restarts**

If the 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, the browser restarts. After the browser has restarted, the last displayed web page is restored.

For more information, see the P*olycom Web Application Developer's Guide* at Polycom UC Software Support Center.

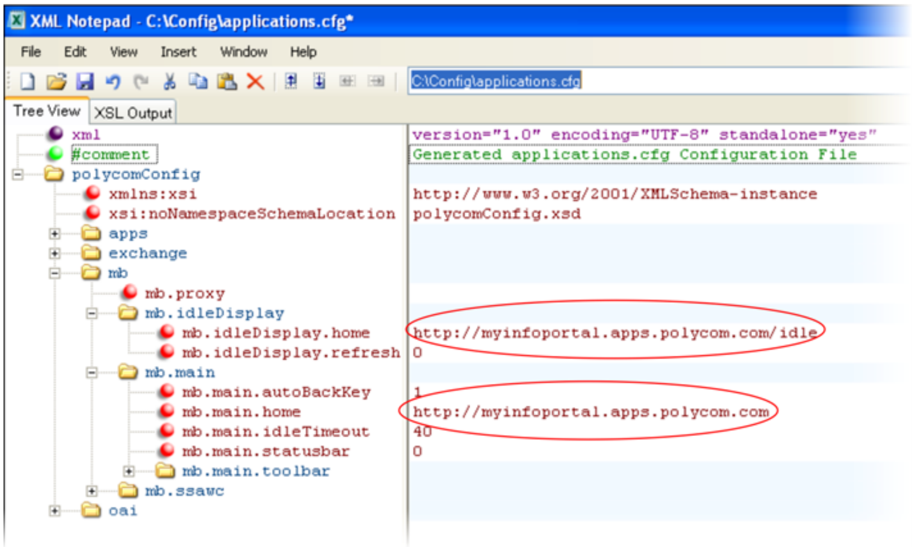
### Use the Microbrowser and the Web Browser

| Parameter Function | **template** > parameter |
|---|---|
| Specify the Application browser home page, a proxy to use, and size limits. | **applications.cfg** > mb.* |
| Specify the Telephony Event Notification events to be recorded and the URL where notifications are sent. | **applications.cfg** > apps.telNotification.* |
| Specify phone state polling settings, such as response mode, the poll URL, and a user name and password. | **applications.cfg** > apps.statePolling.* |
| Specify the push server settings, including message type, port, tunnel, and a user name and password. | **applications.cfg** > apps.push.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

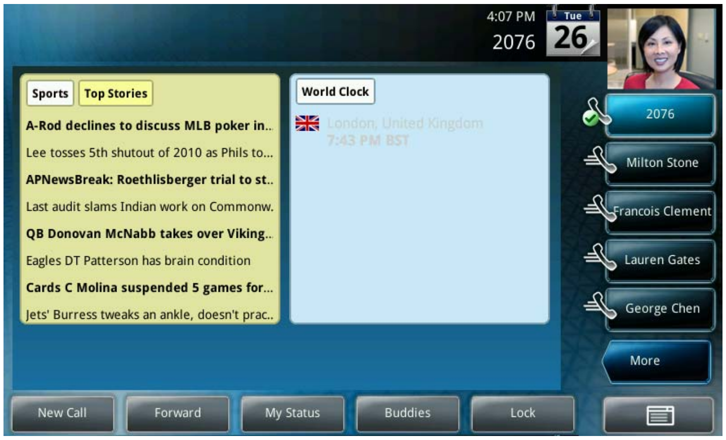
## Example Microbrowser and Web Browser Configuration

The following example shows you how to set a web page on the idle screen of the VVX phone and how to set the interactive web browser's home page on the VVX 1500 phone.

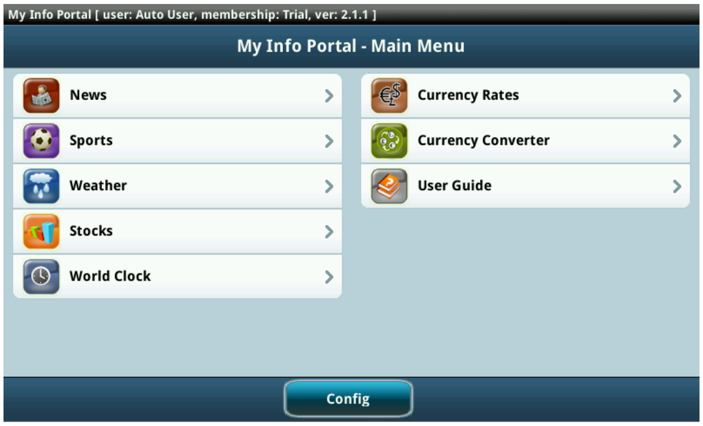

XML Notepad - C:\Config\applications.cfg*
File
Edit
View
Insert
Window
Help
C:\Config\applications.cfg
Tree View
XSL Output

```
xml                                   version="1.0" encoding="UTF-8" standalone="yes"
#comment                              Generated applications.cfg Configuration File
polycomConfig
  xmlns:xsi                           http://www.w3.org/2001/XMLSchema-instance
  xsi:noNamespaceSchemaLocation       polycomConfig.xsd
  apps
  exchange
  mb
    mb.proxy
    mb.idleDisplay
      mb.idleDisplay.home             http://myinfoportal.apps.polycom.com/idle
      mb.idleDisplay.refresh          0
    mb.main
      mb.main.autoBackKey             1
      mb.main.home                    http://myinfoportal.apps.polycom.com
      mb.main.idleTimeout             40
      mb.main.statusbar               0
      mb.main.toolbar
    mb.ssawc
  oai
```

The following illustration shows a non-interactive idle web browser on the VVX 1500 phone.


4:07 PM
2076
Tue
26
Sports
Top Stories
A-Rod declines to discuss MLB poker in..
Lee tosses 5th shutout of 2010 as Phils to...
APNewsBreak: Roethlisberger trial to st..
Last audit slams Indian work on Commonw.
QB Donovan McNabb takes over Viking..
Eagles DT Patterson has brain condition
Cards C Molina suspended 5 games for...
Jets' Burress tweaks an ankle, doesn't prac..
World Clock
London, United Kingdom
7:43 PM BST
2076
Milton Stone
Francois Clement
Lauren Gates
George Chen
More
New Call
Forward
My Status
Buddies
Lock

The following illustration shows the web browser’s interactive home page on the VVX 1500 phone.


My Info Portal [ user: Auto User, membership: Trial, ver: 2.1.1 ]
My Info Portal - Main Menu
News
Sports
Weather
Stocks
World Clock
Currency Rates
Currency Converter
User Guide
Config

## Configure Soft Keys

You can customize the functions of the phone’s soft keys. This feature is typically used to access frequently used functions, to create menu shortcuts to frequently used phone settings; or, if your phone does not have a particular hard key, you can create a soft key. For example, if the phone does not have a Do Not Disturb hard key, you can create a Do Not Disturb soft key. The parameters that configure soft keys are shown in the table Configure Soft Keys. As with EFK line keys, you assign functions to soft keys using macros. For a list of the available macros, see Understand Macro Definitions. You can configure soft keys on all VVX phones.

You can configure the soft keys to display functions depending on the phone’s menu level or call state. For example, you can make a Call Park soft key available when the phone is in an active call state.

Custom soft keys can be added for 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 a 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.

New soft keys can be created as:

- An enhanced feature key sequence
- A speed dial contact directory entry
- An enhanced feature key macro
- A URL
- A chained list of actions

Note that if you are using UC Software 5.1.0 on VVX phones, you can disable the display of any default soft key to make room for custom soft keys; you cannot disable default soft keys with any other UC Software release. The default soft keys that can be disabled include:

- New Call
- End Call
- Split
- Join
- Forward
- Directories
- MyStatus and buddies
- Hold, transfer, and conference

**Note: Inserting soft keys between the Hold, Transfer, and Conference soft keys**

The Hold, Transfer, and Conference soft keys are grouped together to avoid usability issues. You may experience errors if you try to insert a soft key between these three grouped soft keys.

If you want your phone to display both default and custom soft keys, you can configure them in any order. However, the order in which soft keys display depends on the phone's menu level and call state. If you have configured custom soft keys to display with the default soft keys, the order of the soft keys may change.

Up to 10 custom soft keys can be configured. If more soft keys are configured than fit on the phone's screen, a **More** soft key displays. Press the **More** soft key to view the remaining soft keys.

The following table shows you the parameters for configuring soft keys. However, this feature is part of enhanced feature keys (EFK) and you must enable the enhanced feature keys parameter to configure soft keys. See the section Configure Enhanced Feature Keys for details about configuring soft keys and line keys on the phone.

**Configure Soft Keys**

| Parameter Function | **template** > parameter |
|---|---|
| To turn enhanced feature keys on (required). | **features.cfg** > feature.enhancedFeatureKeys.enabled |

**Configure Soft Keys  (continued)**

| | |
|---|---|
| Specify the macro for a line key or soft key function. | **features.cfg** > softkey.x.action |
| To enable a custom soft key. | **features.cfg** > softkey.x.enable |
| Specify the position of the soft key on the phone screen. | **features.cfg** > softkey.x.insert |
| Specify the text to display on the soft key label. | **features.cfg** > softkey.x.label |
| To position the custom soft key before the default soft keys. | **features.cfg** > softkey.x.precede |
| Specify which call states the soft key displays in. | **features.cfg** > softkey.x.use.* |
| To display soft keys for various phone features, including default soft keys. | **features.cfg** > softkey.feature.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Example Soft Key Configurations

This section provides a few examples of available soft key configurations.

**Web Info: Using configurable soft keys**

For more example configurations, see the two following documents at Polycom Engineerig Advisories and Technical Notifications:

- *Using Enhanced Feature Keys and Configurable Soft Keys on Polycom Phones: Technical Bulletin 42250*
- *Using Enhanced Feature Keys (EFK) Macros to Change Soft Key Functions on Polycom Community: Feature Profile 42250*

### Example 1

Use the following example to automatically transfer an active call to BroadSoft voicemail. In this example, *55 is the star code for BroadSoft voicemail, and 8545 is the extension of the voicemail line the call transfers to. Note that enabling the parameter `softkey.1.use.active` causes the soft key to display when a call becomes active on the line. When you press the soft key—labelled ‘VMail’ in this example—the call is placed on hold and automatically transferred to BroadSoft voicemail.

### To map a send-to-voicemail enhanced feature key sequence to a soft key:

1 Update the configuration file as follows:

```
➢ softkey.1.label="VMail"
➢ softkey.1.action="$FTransfer$$Cpause1$$FDialpadStar$$FDialpad5$$FDialpa
  d5$$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$$FSoftKey1$"
➢ softkey.1.enable="1"
➢ softkey.1.use.active="1"
```

2 Reboot the phone.

When an incoming call connects and becomes active, the **VMail** soft key displays.

3 Press the **VMail** soft key to transfer an incoming call to voicemail.

### Example 2

The following example enables you to enter a voicemail extension to transfer an active call to BroadSoft voicemail. In this example, *55 is the star code for BroadSoft voicemail. Note that enabling the parameter `softkey.1.use.active` causes the soft key to display when a call becomes active on the line. When you press the soft key, the call is placed on hold and a field prompts you to enter the extension of the voicemail line you want to transfer the call to. The `efk.prompt*` parameters control the numeric prompt field you enter the extension to. Note that this example works only on line 1 of your phone.

### To create a send-to-voicemail prompt that allows a mailbox number entry:

1 Update the configuration file as follows:

   - ➢ `softkey.1.label="VMail"`
   - ➢ `softkey.1.action="^*55$P1N10$$Tinvite$"`
   - ➢ `softkey.1.enable="1"`
   - ➢ `softkey.1.use.active="1"`
   - ➢ `efk.efkprompt.1.label="Voice Mail"`
   - ➢ `efk.efkprompt.1.status="1"`
   - ➢ `efk.efkprompt.1.type="numeric"`

2 Reboot the phone.

   When an incoming call connects and becomes active, the **VMail** soft key displays.

3 Press the **VMail** soft key.

   A field displays prompting you to enter an extension.

4 Dial *55 and the extension you want to transfer the call to.

### Example 3

This section provides an example of a speed dial soft key and an example of a speed dial line key linked to a directory file. In both example, the macro action includes a pause in the dialing sequence.

Use the following example to configure a soft key to automatically dial a number with a pause in the dialing sequence. In this example, use `$CpauseX$` where `X` is the number of seconds to pause—7 in this example. Adding this pause function enables you to automatically dial into a conference ID that requires an entry code after the conference call is connected.

### To program a pause into a soft key dial number:

1 Update the configuration file as follows:

   - ➢ `softkey.1.label="VMail"`
   - ➢ `softkey.1.action="$S1$$Tinvite$$Cwc$$Cpause7$$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$"`
   - ➢ `softkey.1.enable="1"`
   - ➢ `softkey.1.use.idle="1"`
   - ➢ `feature.enhancedFeatureKeys.enabled="1"`

The values for this example are explained as follows:

- `$S1$` Speed dial line 1
- `$S1$$Tinvite$$` The phone sends an invite to $S1$
- `$Cwc$` The phone waits for the call to connect
- `$Cpause7$` The phone waits for 7 seconds before dialing the remaining numbers
- `$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$` The phone enters the entry code 8545

Use the following example to add a speed dial line key from a directory file. The speed dial includes a pause.

### To program a pause into a directory-linked speed dial line key:

1. Update the configuration file as follows:
   - `feature.enhancedFeatureKeys.enabled="1"`
   - `efk.efklist.1.action.string="501$Tinvite$$Cwc$$Cpause7$123 4#$Tdtmf$"`
     `efk.efklist.1.label="number"`
   - `efk.efklist.1.mname="number"`
   - `efk.efklist.1.status="1"`
2. In a contact directory file or speed dial file (000000000000-directory.xml or <*MACaddress*>-directory.xml), add the following:
   - `<fn>Call Number</fn>`
   - `<ct>!number</ct>`
   - `<sd>99</sd>`

In the action string: `<ct>"501$Tinvite$$Cwc$$Cpause7$123 4#$Tdtmf$"`:

- `501$Tinvite$` Dial 501
- `$Cwc$` Wait for the call to connect
- `$Cpause7$` A seven second pause
- `1234#$Tdtmf$` Send 1234 dual-tone multi-frequency

The EFK commands are linked to the directory file as follows:

- The parameter `efk.efklist.1.mname="number"` is linked to the speed dial contact `<ct>!number</ct>` of the directory file
- Use `<fn>Call Number</fn>` to define the name that displays on the key
- Use `<sd>99</sd>` to identify which directory entry to link to the key

**Tip: Active call transfer star codes depend on your call server**

The exact star code to transfer the active call to voicemail depends on your call server.

## Configure Enhanced Feature Keys

Enhanced feature keys (EFK) enables you to customize the functions of a phone's line and soft keys and, as of UC Software 4.0.1, hard keys. You can use EFK to assign frequently used functions to line keys, soft keys, and hard keys or to create menu shortcuts to frequently used phone settings.

See the following table for the parameters you can configure and a brief explanation of how to use the contact directory to configure line keys. Enhanced feature key functionality is implemented using star code sequences (like *69) and SIP messaging. Star code sequences that define EFK functions are written as macros that you apply to line and soft keys. The EFK macro language was designed to follow current configuration file standards and to be extensible. The macros are case sensitive.

The rules for configuring EFK for line keys, soft keys, and hard keys are different. Before using EFK, you are advised to become familiar with the macro language shown in this section and in the reference section <efk/>.

**Web Info: Using enhanced feature keys**

For instructions and details on how to use enhanced feature keys, see *Using Enhanced Feature Keys and Configurable Soft Keys on Polycom Phones: Technical Bulletin 42250.* at Polycom Engineerig Advisories and Technical Notifications.

Note that the configuration file changes and the enhanced feature key definitions can be included together in one configuration file. Polycom recommends creating a new configuration file in order to make configuration changes.

**Tip: EFK compatibility**

The EFK feature from SIP 3.0 is compatible with the EFK feature from SIP 3.1. However, improvements have been made and Polycom recommends that existing configuration files be reviewed and updated.

**Enhanced Feature Keys**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify at least two calls per line key. | **reg-basic.cfg** > reg.x.callsPerLineKey |
| Enable or disable enhanced feature keys. | **features.cfg** > feature.enhancedFeatureKeys.enabled |
| Specify the EFK List parameters. | **features.cfg** > efk.efklist.x.* |
| Specify the EFK Prompts. | **features.cfg** > efk.efkprompt.x.* |

Because line keys and their functions are linked to fields in the contact directory file you need to match the contact field (ct) in the directory file to the macro name field (mname) in the configuration file that contains the EFK parameters. When you enter macro names to the contact field (ct) in the directory file, add the '!' prefix to the macro name. The template directory configuration file is named **000000000000-directory~.xml**. To use this file, remove the tilde (~) from the file name. For more detailed information on using the contact directory, see Use the Local Contact Directory.

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

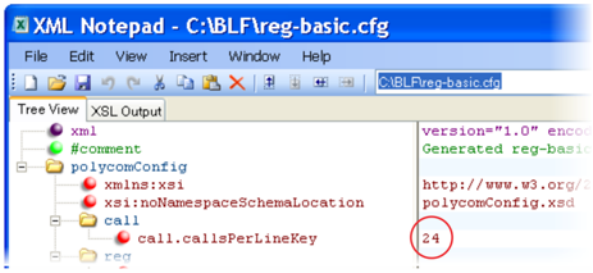
## Some Guidelines for Configuring Enhanced Feature Keys

The following guidelines help you to configure enhanced feature keys (EFKs) efficiently:

- Activation of EFK functions requires valid macro construction.
- All failures are logged at level 4 (minor).
- If two macros have the same name, the first one is used and the subsequent ones is ignored.
- 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.
- “!” and “^” macro prefixes cannot be mixed in the same macro line.
- The sequence of characters must be prefixed by either “!” or “^” to 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.
- 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 after you reach 50 macro substitutions).

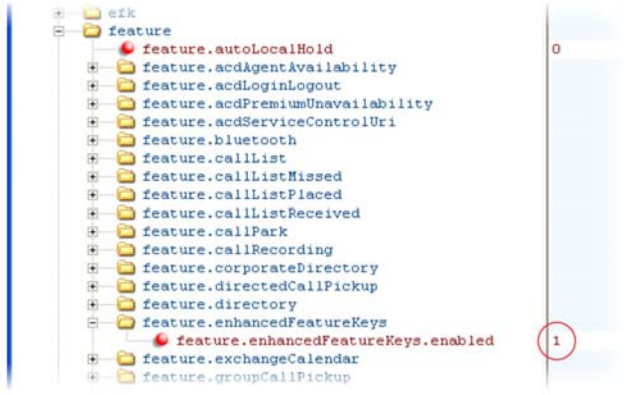
## Enhanced Feature Key Examples

The following illustration shows the default value 24 calls per line key. Ensure that you specify at least two calls per line key.



```
xml                                  version="1.0" enco
#comment                             Generated reg-basi
polycomConfig
  xmlns:xsi                          http://www.w3.org/
  xsi:noNamespaceSchemaLocation      polycomConfig.xsd
  call
    call.callsPerLineKey             24
  reg
```

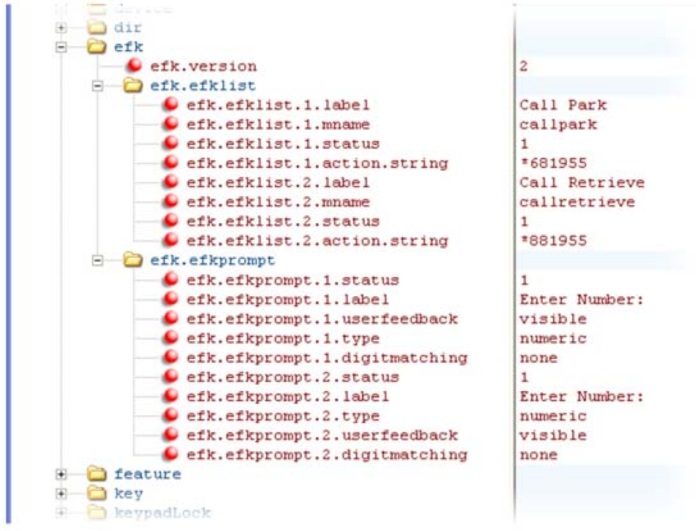
Enable the enhanced feature keys feature in the features.cfg template file, as shown next.



```
efk
feature
    feature.autoLocalHold                          0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
        feature.enhancedFeatureKeys.enabled        1
    feature.exchangeCalendar
    feature.groupCallPickup
```

In the following illustration, the EFK parameters are located in the **features.cfg** template file. In the `efk.efklist.x.*` parameters, line key 1 has been assigned a Call Park address (1955) and line key 2 a call retrieve function. The parameter `acton.string` shows you the macro definition for these two functions. In addition, status is enabled and a label has been specified to display next to the line key. The entry in the `mname` parameter corresponds to the `contact (ct)` field in the contact directory.

In the `efk.prompt.*` parameters, `status` has been enabled. The label on the user prompt has been defined as *Enter Number:* and this prompt displays on the phone screen. The `type` parameter has been set to `numeric` to allow only numbers and because `userfeedback` has been specified as `visible`, you are able to see the numbers you enter into the prompt.



```
dir
efk
    efk.version                                2
    efk.efklist
        efk.efklist.1.label                    Call Park
        efk.efklist.1.mname                    callpark
        efk.efklist.1.status                   1
        efk.efklist.1.action.string            *681955
        efk.efklist.2.label                    Call Retrieve
        efk.efklist.2.mname                    callretrieve
        efk.efklist.2.status                   1
        efk.efklist.2.action.string            *881955
    efk.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          none
        efk.efkprompt.2.status                 1
        efk.efkprompt.2.label                  Enter Number:
        efk.efkprompt.2.type                   numeric
        efk.efkprompt.2.userfeedback           visible
        efk.efkprompt.2.digitmatching          none
feature
key
keypadLock
```

## Understand Macro Definitions

The efk.efklist.x.action.string can be defined by one of the following:

- Macro Actions
- Prompt Macro Substitution
- Expanded Macros

## Macro Actions

The action string is executed in the order it displays. User input is collected before any action is taken. The action string can contain the fields shown in the following table.

**Macro Actions and Descriptions**

**$L<label>$**

This is the label for the entire operation. The value can be any string including the null string (in this case, no label displays). This label is used if no other operation label collection method worked (up to the point where this field is introduced). Make this the first entry in the 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 parameter depends on the action string.

**$C<command>$**

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`

`intercom`

Note: Polycom recommends that you always define this field. If it is not defined, the supplied digits are 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 (see Prompt Macro Substitution)

.

**$S<speed dial index>$**

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, see Map Internal Key Functions.

**Macro Actions and Descriptions**

| URL |
|---|
| A URL. Only one per action string is supported. |

## Prompt Macro Substitution

The efk.efklist.x.action.string can be defined by a macro substitution string, PnNn where:

- *Pn* is the prompt x as defined by `efk.efkprompt.x`.
- *Nn* is the number of digits or letters that the user can enter. The value must be between 1 and 32 characters; otherwise the macro execution fails. The user needs to press the Enter soft key to complete data entry.

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 the action that the phone performs after 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, see Use the Local Contact Directory.

## Special Characters

The following special characters are used to implement the enhanced feature key functionality. Macro names and macro labels cannot contain these characters. If they do, you may experience unpredictable behavior.

- ! 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).

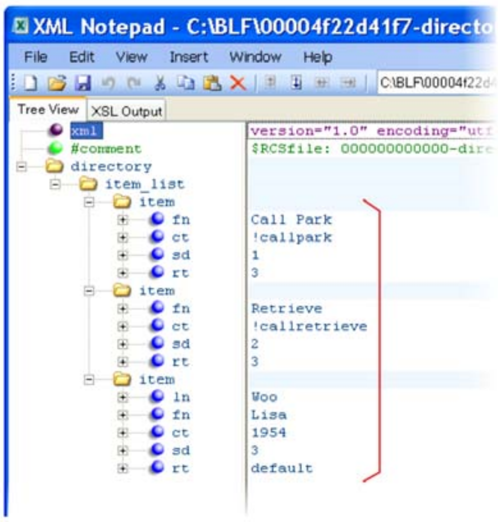
## Example Macros

The action string `$Changup$*444*$P1N4$$Tinvite$$Cwaitconnect$$P2N3$$Cpause2$$Tdtmf$$Changup$`is executed in order 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. After connection, there is a 2 second pause, and then the string *567* is sent using DTMF dialing on the active call.

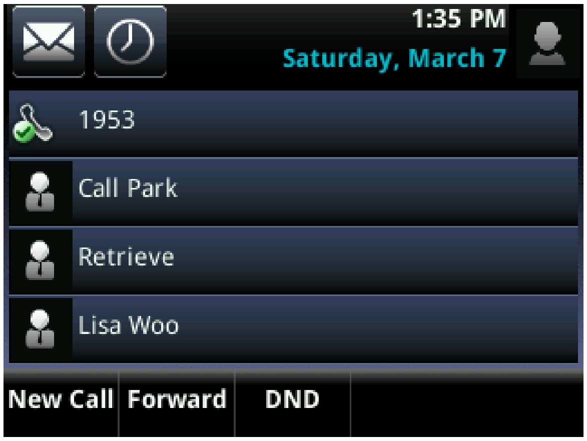
6 The active call is disconnected.

Because line keys and their functions are linked to fields in the directory file, a macro name you enter in `efk.list.x.mname` must match the name you enter to the `contact (ct)` field in the directory file. The macro name you enter in the `(ct)` field of the directory file must begin with the '!' prefix. The following example directory file shows a line key configured with call park, call retrieve, and a speed dial contact Lisa Woo.



```
xml                 version="1.0" encoding="ut
#comment            $RCSfile: 000000000000-dire
directory
  item_list
    item
      fn            Call Park
      ct            !callpark
      sd            1
      rt            3
    item
      fn            Retrieve
      ct            !callretrieve
      sd            2
      rt            3
    item
      ln            Woo
      fn            Lisa
      ct            1954
      sd            3
      rt            default
```

The following illustrates the call park and call retrieve line keys and a speed dial contact Lisa Woo.



For an explanation of all fields in the directory file, see the table Understanding the Local Contact Directory.

## Flexible Line Key Assignment

You can give your phone users the ability to assign a line key function to a line key anywhere on the phone's screen. Normally, functions are assigned line keys in succession, the order in which the line key displays on the phone. This feature enables you to break that ordering and assign a line key function to a line key that displays anywhere on the phone's screen. This feature is available on the VVX 300 series, 400 series, 500 series, 600 series, and VVX Expansion Modules. Refer to the table Flexible Line Key Assignment for the parameters you need to configure to set up this feature.

You can apply this feature to any line key function including line appearance, speed dial, busy lamp field (BLF), and presence. Line keys that you configure using this feature override the default line key assignments as well as any custom line key configurations you may have made. To use this feature, you need to specify the function of each line key on the phone. You do this by assigning a category and an index to each line key, both of which are explained in the example configuration.

Specific conditions apply when you assign busy lamp field (BLF) or presence to line keys. If you are assigning BLF or presence to a line key, assign that line key to index=0 to indicate automatic ordering. BLF and presence line keys are self-ordering, meaning that if you have these features assigned to multiple line keys, you can specify the location of the BLF or presence line key but not the order in which they display. For example, you can assign a BLF line key to index 1, 3, and 5 but you cannot specify how the contacts are ordered, which BLF contacts display on line keys 1, 3, and 5. In addition, to assign BLF and presence to a line key, you need to assign a corresponding registration line. You can configure multiple line keys per registration if each line key has a corresponding reg.x.lineKeys parameter.

**Flexible Line Key Assignment**

| **Parameter Function** | **template** > parameter |
|---|---|
| To enable flexible line key assignment. | **reg-advanced.cfg** > lineKey.reassignment.enabled |
| Specify the line key category. | **reg-advanced.cfg** > lineKey.x.category |
| Specify the line key number (dependent on category). | **reg-advanced.cfg** > lineKey.x.index |

**Note: Line keys are numbered sequentially**

Line keys on VVX phones and expansion modules are numbered sequentially, and the line keys on your expansion module depends on how many lines your phone supports. For example, a VVX 600/601 phone supports 16 lines, numbered 1-16. The first line on an expansion module connected to a VVX 600/601 phone is line 17.

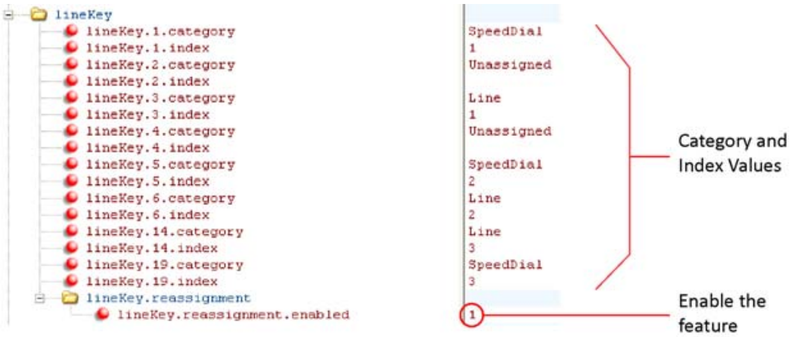
## Example Flexible Line Key Assignment Configuration

To enable flexible line key assignment, in the features.cfg template, set the lineKey.reassignment.enabled parameter to 1. Then assign each line key a category and an index. The category specifies the function of the line key and can include: Unassigned, Line, BLF, Speed Dial, and Presence. Note that the category Unassigned leaves that line key blank. The index specifies the order in which the line keys display on the phone screen. Use the following table to help you assign a category and an index to the line keys on your phone.

**Assigning Flexible Line Keys**

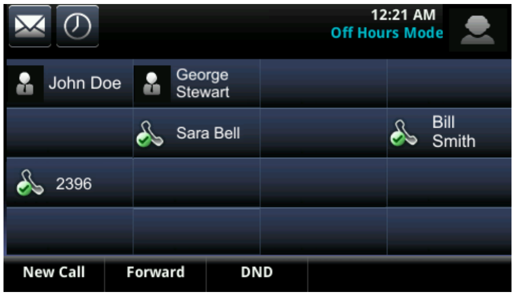
| *Assigning a Category and an Index to Line Keys* | | | | | |
|---|---|---|---|---|---|
| **Category** | unassigned | line | BLF | Speed Dial | Presence |
| **Index** | Null | The line index number | 0 | The speed dial index number | 0 |

The following illustration shows you an example flexible line key assignment configuration in the features.cfg template file:



```
lineKey
    lineKey.1.category              SpeedDial
    lineKey.1.index                 1
    lineKey.2.category              Unassigned
    lineKey.2.index
    lineKey.3.category              Line
    lineKey.3.index                 1
    lineKey.4.category              Unassigned
    lineKey.4.index
    lineKey.5.category              SpeedDial
    lineKey.5.index                 2
    lineKey.6.category              Line
    lineKey.6.index                 2
    lineKey.14.category             Line
    lineKey.14.index                3
    lineKey.19.category             SpeedDial
    lineKey.19.index                3
    lineKey.reassignment
        lineKey.reassignment.enabled    1
```

This configuration displays on a VVX 600/601 phone screen as the following:



## Configure the Phone Keypad

You can customize many of the default key functions on the phone’s keypad interface. The table Configure Phone Keys lists the parameters you can configure to change the layout of your phone’s keypad. Polycom recommends that you configure only those phone keys with removable key caps, including Directories, Applications, Conference, Transfer, Redial, Menu, Messages, Do Not Disturb, and Call Lists.

**Caution: Choosing keys to remap**

Polycom recommends that you remap only those keys with removable key caps. If you remap other keys, your phone may not work properly. You should not remap the following keys: the dial pad, volume control, handsfree, mute, headset, hold, and the navigation arrow keys.

You can configure phone keys in the following ways:

- You can assign function or features to a key.
- You can turn a phone key into a speed dial.
- You can assign enhanced feature key (EFK) operations to a phone key. For example, you can reach a phone menu path to a single key press using a macro code. To find out how to configure EFK functions, see the section Configure Enhanced Feature Keys.
- You can delete all functions and features from a phone key.

### Configure Phone Keys

| Parameter Function | **template** > parameter |
|---|---|
| Set the primary key function for key y on phone model x. | **features.cfg** > key.x.function.prim |
| Set the secondary key function for key y on phone model x. | **features.cfg** > key.x.subPoint.prim |

# Configure Phone Logs and Directory Files

This section provides information on configuring phone logs and phone directory files.

## Polycom Experimental Features

You can enable and evaluate experimental features for the RealPresence Trio solution in a non-production environment. If you decide to try out these features, be aware that they are neither tested nor supported. These features may, or may not, become official features in a future release.

You can enable experimental features in the Polycom Web Configuration Utility.

### To enable the experimental features:

1. In the Web Configuration Utility, click Settings > Polycom Labs.
2. Select and configure the desired Polycom Lab features.

## Polycom Experience Cloud (PEC)

Using the Polycom Experience Cloud (PEC) service, you can view basic diagnostic and phone usage data, including start and stop events, call quality information, packet statistics, call duration, and call logs.

Administrators can configure the PEC service using the Web Configuration Utility or configuration parameters.

# Configure the Call Logs

The phone records and maintains phone events to a call log, also known as a call list. These call logs contain call information such as remote party identification, time and date of the call, and call duration. The log is stored as a file in XML format named **<*MACaddress*>-calls.xml** to your provisioning server. If you want to route the call logs to another server, use the CALL_LISTS_DIRECTORY field in the master configuration file. You can use the call logs to redial previous outgoing calls, return incoming calls, and save contact information from call log entries to the contact directory. All call logs are enabled by default. See the table Configure the Call Logs for instructions on how to enable or disable the call logs.

The phones automatically maintain the call logs in three separate call lists: Missed Calls, Received Calls, and Placed Calls. Each of these call lists can be cleared manually by individual phone users. You can delete individual records or all records in a group (for example, all missed calls). You can also sort the records or filter them by line registration.

As of Polycom UC Software 4.0.1, the VVX 500/501, and 1500 phones remember the previous call history after a restart or reboot.

**Tip: Merged call lists**

On some phones, missed and received calls display in one call list. In these combined lists, you can identify call types by the icons:

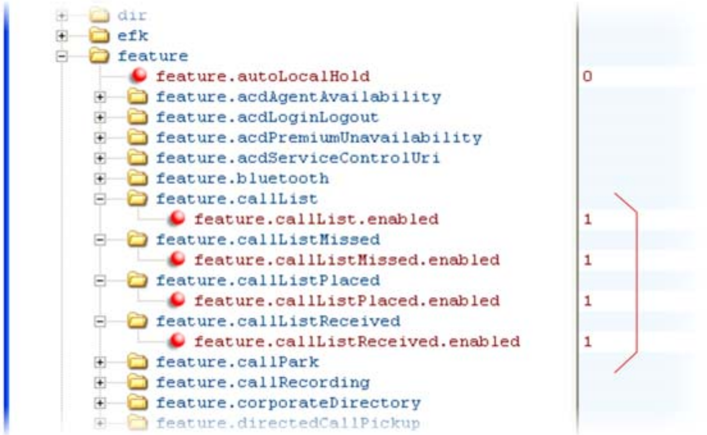
- Missed call icon ✕
- Received call icon ↙

### Configure the Call Logs

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the missed call list. | **features.cfg** > feature.callListMissed.enabled |
| Enable or disable the placed call list. | **features.cfg** > feature.callListPlaced.enabled |
| Enable or disable the received call list. | **features.cfg** > feature.callListReceived.enabled |

## Example Call Log Configuration

The following illustration shows you each of the call log parameters you can enable or disable in the features.cfg template file.



```
dir
efk
feature
    feature.autoLocalHold                     0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
        feature.callList.enabled              1
    feature.callListMissed
        feature.callListMissed.enabled        1
    feature.callListPlaced
        feature.callListPlaced.enabled        1
    feature.callListReceived
        feature.callListReceived.enabled      1
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
```

The following table describes each element and attribute that displays in the call log. Polycom recommends using an XML editor such as XML Notepad 2007 to view and edit the call log. Note that you can place the elements and attributes in any order in your configuration file.

### Call Log Elements and Attributes

| Element | Permitted Values |
|---|---|
| **direction** | **In, Out** |
| Call direction with respect to the user. | |
| **disposition** | **Busy, Forwarded, Normal, Partial, Preempted, Rejected, RemotelyHandled, Transferred** |
| Indicates what happened to the call. When a call entry is first created, the disposition is set to Partial. | |
| **line** | **Positive integer** |
| The line (or registration) index. | |
| **protocol** | **SIP or H323** |
| The line protocol. | |

**Call Log Elements and Attributes**

| | | |
|---|---|---|
| **startTime** | **String** | The start time of the call. For example: 2010-01-05T12:38:05 in local time. |
| **duration** | **String** | The duration of the call, beginning when it is connected and ending when the call is terminated.<br>For example: `PT1H10M59S`. |
| **count** | **Positive Integer** | The number of consecutive missed and abandoned calls from a call destination. |
| **destination** | **Address** | The original destination of the call.<br>For outgoing calls, this parameter designates the outgoing call destination; the name is initially supplied by the local phone (from the name field of a local contact entry) but may later be updated via call signaling. This field should be used for basic redial scenarios.<br>For incoming calls, the called destination identifies the requested party, which may be different than any of the parties that are eventually connected (the destination may indicate a SIP URI which is different from any SIP URI assigned to any lines on the phone). |
| **source** | **Address** | The source of the call (caller ID from the call recipient's perspective). |
| **Connection** | **Address** | An array of connected parties in chronological order.<br>As a call progresses, the connected party at the far end may change, for example, if the far end transfers the call to someone else. The connected element allows the progression of connected parties, when known, to be saved for later use. All calls that contain a connected state must have at least one connection element created. |
| **finalDestination** | **Address** | The final connected party of a call that has been forwarded or transferred to a third party. |

## Use the Local Contact Directory

Polycom phones feature a contact directory file you can use to store frequently used contacts as a template contact directory file named **000000000000-directory~.xml** included with the UC Software download. This template file is loaded to the provisioning server the first time you boot up a phone with UC Software or when you reset the phone to factory default settings.

When you first boot the phone out of the box or when you reset the phone to factory default settings, the phone looks for contact directories in the following order:

- An internally stored local directory
- A personal **<MACaddress>-directory.xml** file
- A global **000000000000-directory.xml** file when the phone substitutes <000000000000> for its own MAC address.

To create a per-phone, personal directory file, replace ***<000000000000>*** in the global file name with the phone's MAC address: ***<MACaddress>*-directory.xml**. Any changes users make to the contact directory from the phone are stored on the phone drive and uploaded to the provisioning server in the personal directory (**<*MACaddress*>-directory.xml**) file, which enables you to preserve a contact directory during reboots.

To create a global directory file that you can use to maintain the directory for all phones from the provisioning server, remove the tilde (~) from the template file name **000000000000-directory.xml**. When you update the global directory file on the provisioning server, the updates are downloaded onto the phone and combined with the phone specific directory.

Using the parameter `voIpProt.SIP.specialEvent.checkSync.downloadDirectory`, you can configure the phones to download the updated directory files upon receipt of a `checksync NOTIFY` message. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates.

Any changes to either the global or personal directory files are reflected in the directory on the phone after a restart or a `checksync NOTIFY` message. When merging the two files, the personal directory always takes precedence over the changes in the global directory. Thus, if a user modifies a contact from the global directory, the contact is saved in the personal directory file, and the contact from the global directory is ignored when the files are next uploaded.

If you created a per-phone **<*MACaddress*>-directory.xml** for a phone and you want that phone to use a global contact directory **000000000000-directory.xml**, delete the **<*MACaddress*>directory.xml** and reset the phone to factory defaults as shown in the section Reset the Phone to Defaults.

**Tip: Ensuring users do not delete definitions in the contact directory**
To avoid users accidentally deleting the definitions in the contact directory, make the contact directory file read only.

The Contact Directory is the central database for several phone features including speed dial (see Configure the Speed Dial Feature), distinctive incoming call treatment (see Apply Distinctive Incoming Call Treatment), presence (see Use Presence), and instant messaging (see Enable Instant Messaging). The following table lists the directory parameters you can configure. The following table lists the maximum number of contacts and maximum file size of the local Contact Directory for each phone. If you want to conserve phone memory, use the parameter `dir.local.contacts.maxNum` to configure the phones to support a lower maximum number of contacts. If you want to conserve phone memory, use the parameter `dir.local.contacts.maxNum` to configure the phones to support a lower maximum number of contacts.

**Maximum File Size and Number of Contacts**

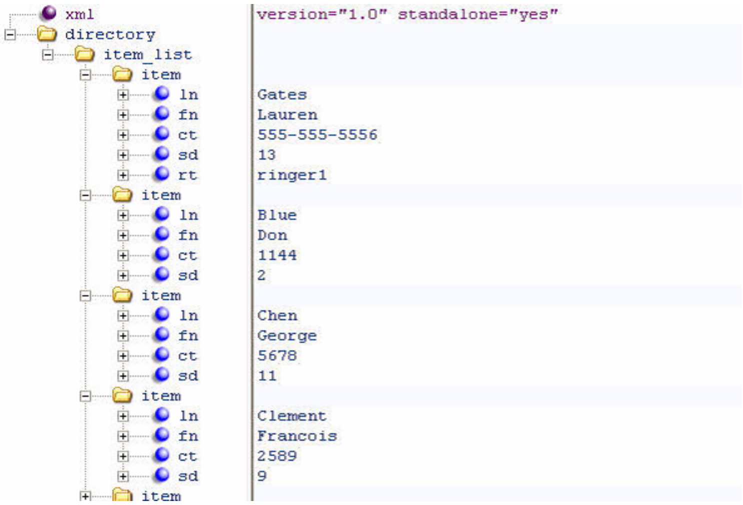
| Phone | Maximum File Size | Maximum Number of Contacts in File |
|---|---|---|
| VVX 300/301/310/311 | 4MB | 500 |
| VVX 400/401/410/411 | 4MB | 500 |
| VVX 500/501 and 600/601 | 4MB | 500 |
| VVX 1500 | 102400 bytes<br>Non-volatile: 100KB | 9999 |
| SoundStructure VoIP Interface | Not applicable | Not applicable |

### Use the Local Contact Directory

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the local contact directory. | **features.cfg** >feature.directory.enabled |
| Specify if the local contact directory is read-only. | **features.cfg** > dir.local.readonly |
| Specify the maximum number of contact entries for each phone. | **features.cfg**> dir.local.contacts.maxNum |
| Specify whether to search the directory by first name or last name. | **features.cfg** > dir.search.field |
| Specify when the phone downloads the updated global directory file. | **feature.cfg >** voIpProt.SIP.specialEvent.checkSync.download Directory |
| The template contact directory file. | 000000000000-directory~.xml |

## Example Configuration

The following illustration shows four contacts configured in a directory file.



```
xml                 version="1.0" standalone="yes"
directory
    item_list
        item
            ln      Gates
            fn      Lauren
            ct      555-555-5556
            sd      13
            rt      ringer1
        item
            ln      Blue
            fn      Don
            ct      1144
            sd      2
        item
            ln      Chen
            fn      George
            ct      5678
            sd      11
        item
            ln      Clement
            fn      Francois
            ct      2589
            sd      9
        item
```

The following table describes each of the parameter elements and permitted values that you can use in the local contact directory.

**Note: GENBAND contact duplication**

You can duplicate contacts in the Contact Directory registered with the GENBAND server.

### Understanding the Local Contact Directory

| Element | Definition | Permitted Values |
|---|---|---|
| **fn** | **First Name** | **UTF-8 encoded string of up to 40 bytes**[1] |

**Understanding the Local Contact Directory**

| | | |
|---|---|---|
| The contact's first name. | | |
| **ln** | **Last Name** | **UTF-8 encoded string of up to 40 bytes[1]** |
| The contact's last name. | | |
| **ct** | **Contact** | **UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL** |
| 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. The maximum field length is 128 characters.<br>Note: This field cannot be null or duplicated. | | |
| **sd** | **Speed Dial Index** | **Null, 1 to 9999** |
| Associates a particular entry with a speed dial key for one-touch dialing or dialing from the speed dial menu. | | |
| **lb** | **Label** | **UTF-8 encoded string of up to 40 bytes[1]** |
| The label for the contact. 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 first and last names form the label. A space is added between first and last names.<br>Note: For GENBAND, the **Label** element is shown as **Nick Name**, and is a mandatory, non-duplicate field. | | |
| **pt** | **Protocol** | **SIP, H323, or Unspecified** |
| The protocol to use when placing a call to this contact. | | |
| **rt** | **Ring Tone** | **Null, 1 to 21** |
| When incoming calls match a directory entry, this field specifies the ringtone to be used. | | |
| **dc** | **Divert Contact** | **UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL** |
| The address to forward calls to if the Auto Divert feature is enabled. | | |
| **ad** | **Auto Divert** | **0 or 1** |
| If set to 1, callers that match the directory entry are diverted to the address specified for the divert contact element.<br>Note: If auto-divert is enabled, it has precedence over auto-reject. | | |

**Understanding the Local Contact Directory**

| | | |
|---|---|---|
| **ar** | **Auto Reject** | **0 or 1** |
| If set to 1, callers that match the directory entry specified for the auto-reject element are rejected.<br>Note: If auto divert is also enabled, it has precedence over auto reject. | | |
| **bw** | **Buddy Watching** | **0 or 1** |
| If set to 1, this contact is added to the list of watched phones. | | |
| **bb** | **Buddy Block** | **0 or 1** |
| If set to 1, this contact is blocked from watching this phone. | | |

[1] In some cases, this will be less than 40 characters due to UTF-8’s variable bit length encoding.

## Configure the Local Digit Map

The phone has a local digit map feature that, when configured, automatically calls a dialed number, eliminating the need to press the **Dial** or **Send** soft key to place outgoing calls. Note that digit maps do not apply to on-hook dialing.

Digit maps are defined by a single string or a list of strings. If a number you dial matches any string of a digit map, the call is automatically placed. If a number you dial matches no string—an impossible match—you can specify the phone’s behavior. If a number ends with #, you can specify the phone’s behavior, called trailing # behavior. You can also specify the digit map timeout, the period of time after you dial a number that the call is placed. The parameter for each of these options is outlined in the following table. The configuration syntax of the digit map is based on recommendations in section 2.1.5 of RFC 3435.

**Web Info: Changing the local digit map on Polycom phones**

For instructions on how to modify the local digit map, see *Changes to Local Digit Maps on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones: Technical Bulletin 11572* at Polycom Engineering Advisories and Technical Notifications.

**Configure the Local Digit Map**

| **Parameter Function** | **template** > parameter |
|---|---|
| Apply a dial plan to dialing scenarios. | **site.cfg** > dialplan.applyTo* |
| Specify the digit map to use for the dial plan. | **site.cfg** > dialplan.digitmap |
| Specify the timeout for each segment of the digit map. | **site.cfg** > dialplan.digitmap.timeOut |
| Specify the behavior if an impossible dial plan match occurs. | **site.cfg** > dialplan.impossibleMatchHandling |
| Specify if trailing # digits should be removed from digits sent out. | **site.cfg** > dialplan.removeEndOfDial |

**Configure the Local Digit Map**

| | |
|---|---|
| Specify the details for emergency dial plan routing. | **site.cfg** > dialplan.routing.emergency.x.* |
| Specify the server that to used for routing calls. | **site.cfg** > dialplan.routing.server.x.* |
| Configure the same parameters as above for a specific registration (overrides the global parameters). | **site.cfg** > dialplan.x.* |
| Specifies the time in seconds that the phone waits before dialing a number when you dial on-hook. | **site.cfg** > dialplanuserDialtimeOut |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

Polycom support for digit map rules varies for open SIP servers and Microsoft Skype for Business Server.

## Use Open SIP Digit Map

The following is a list of digit map string rules for open SIP environments. If you are using a list of strings, each string in the list can be specified as a set of digits or timers, or as an expression which the gateway to use to find the shortest possible match.

Digit map extension letter R indicates that certain matched strings are replaced. Using a *RRR* syntax, you can replace the digits between the first two *R*s with the digits between the last two *R*s. For example, ***R****555****R****604****R*** would replace 555 with 604. 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 illustrate the semantics of the syntax:

- `R9R604Rxxxxxxx`—Replaces *9* with *604*
- `xxR601R600Rxx`—When applied to *1160122* gives *1160022*
- `R9RRxxxxxxx`—Remove *9* at the beginning of the dialed number (replace 9 with nothing)
  - 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 (replace nothing with *604*)
  - For example, if a customer dials *4539400*, *604* is added to the front of the number, so a call to 6044539400 is placed.
- `xR60xR600Rxxxxxxx`—Replace any 60x 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 (.) that 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 VVX 500/501, 600/601, and 1500 phones match the HH**

Only VVX 500/510, 600/611, and 1500 phones match the H. On all other phones, the H is ignored and users 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 VVX 1500 user dials 3302 on an H.323 or dual protocol line, the call is placed after the user dials the last digit.

Take note of the following guidelines:

- The following letters are case sensitive: *x*, *T*, *R*, *S*, and *H*.
- You must use only *, #, +, or 0–9 between the second and third *R*.
- If a digit map does not comply, it is not included in the digit plan as a valid map. That is, no match is made.
- There is no limit to the number of R triplet sets in a digit map. However, a digit map that contains less than a full number of triplet sets (for example, a total of 2 Rs or 5 Rs) is considered an invalid digit map.
- If you use T in the left part of *RRR's* syntax, the digit map will not work. For example, *R0TR322R* will not work.

### Generate Secondary Dial Tone with Digit Maps

You can regenerate dial tone by adding a comma "**,**" to the digit map. In the following example, you can dial seven-digit numbers after dialing "8" as shown next in the rule **8,[2-9]xxxxxxT:**

[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|8,**[2-9]xxxxxxT**|[2-9]xx.T

By adding digit "8", dial tone plays again and users can complete the remaining seven-digit number. In this example, if you also have a 4-digit extension that begins with "8" then you will hear dial tone after the first "8" was dialed because "8" matches the "8" in the digit map.

If you want to generate dial tone without the need to send the "8", replace one string with another using the special character "R" as shown next in the rule **R8RR**. In the following example, replace "8" with an empty string to dial the seven-digit number:

[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|R8RR,[2-9]xxxxxxT|[2-9]xx.T

## Configure the Speed Dial Feature

You can link entries in your local contact directory to speed dial contacts on the phone. The speed dial feature enables you to place calls quickly using dedicated line keys or from a speed dial menu. To set up speed dial through the phone's contact directory, see the section Use the Local Contact Directory. To set up speed dial contacts, you need to become familiar with parameters in the following table, which identifies the directory XML file and the parameters you need to set up your speed dial contacts.

You can assign contacts the following speed dial index ranges.

**Speed Dial Index Ranges**

| Phone Model | Range |
|---|---|
| VVX 101, 201 | 1 – 99 |
| VVX 300 series, 400 series, 500 series, and 600 series | 1 – 500 |
| VVX 1500 | 1 – 9999<br>The maximum number may be limited by the phone’s available memory. |
| SoundStructure VoIP Interface | Not applicable. |

On some call servers, enabling presence for an active speed dial contact displays that contact’s status on the speed dial’s line key label. For information on how to enable presence for contacts, see the section Use Presence.

**Configure the Speed Dial Feature**

| Parameter Function | **template** > parameter |
|---|---|
| Enter a speed dial index number in the <sd>x</sd> element in the <MAC address>-directory.xml file to display a contact directory entry as a speed dial key on the phone. Speed dial contacts are assigned to unused line keys and to entries in the phone’s speed dial list in numerical order. | |
| The template contact directory file. | **000000000000-directory~.xml** |

**Tip: Quick access to the speed dial list**
To access the speed dial list quickly, press the phone’s Up arrow key from the idle display.

## Example Speed Dial Configuration

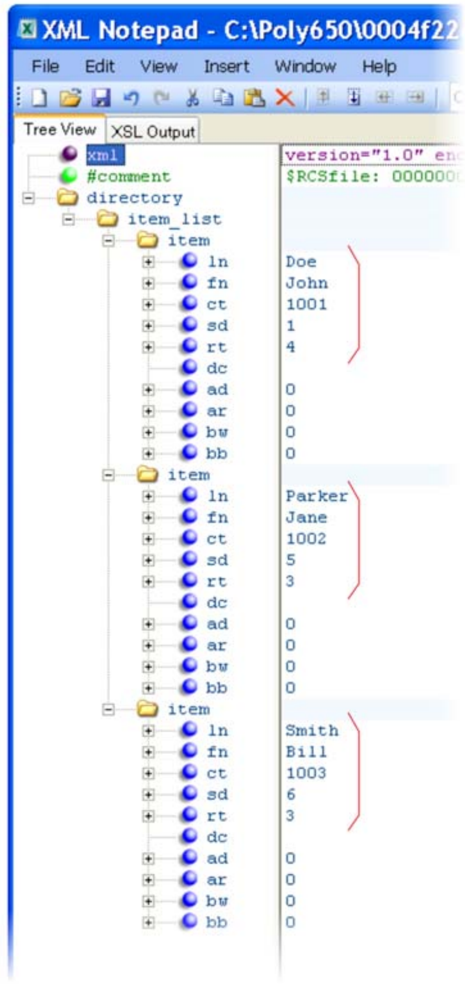
The first time you boot up a phone with UC Software or when you reset the phone to factory default settings, a template contact directory file named **00000000000-directory~.xml** is loaded to the provisioning server. You can edit and use this template file as a global contact directory for a group of phones or you can create your own per-phone directory file. To create a global directory, locate the **00000000000-directory~.xml** template in your UC Software files and remove the tilde (~) from the file name. When you restart, reboot, or reset to factory defaults, the phone substitutes the global file with its own **<MACaddress>-directory.xml** which is uploaded to the server. If you want to create a per-phone directory, replace **<000000000000>** in the global file name with the phone's MAC address, for example, **<MACaddress>-directory.xml**.

When you reset the phone to factory defaults, the phone looks first for its own **<MACaddress>-directory.xml** and then for the global directory. Contact directories stored locally on the phone may or may not override the **<MACaddress>-directory.xml** on the server depending on your server configuration.

For more information on how to use the template directory file 000000000000 directory~.xml, see Use the Local Contact Directory.

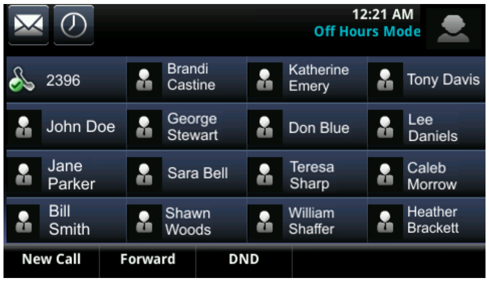
After you have renamed the directory file as a per-phone directory, enter a number in the speed dial `<sd>` field to display a contact directory entry as a speed dial contact on the phone. Speed dial entries automatically display on unused line keys on the phone and are assigned in numerical order.

The example local contact directory file shown next is saved with the phone's MAC address and shows the contact *John Doe* with extension number *1001* as speed dial entry 1 on the phone.



```
xml                 version="1.0" enc
#comment            $RCSfile: 000000
directory
  item_list
    item
      ln            Doe
      fn            John
      ct            1001
      sd            1
      rt            4
      dc
      ad            0
      ar            0
      bw            0
      bb            0
    item
      ln            Parker
      fn            Jane
      ct            1002
      sd            5
      rt            3
      dc
      ad            0
      ar            0
      bw            0
      bb            0
    item
      ln            Smith
      fn            Bill
      ct            1003
      sd            6
      rt            3
      dc
      ad            0
      ar            0
      bw            0
      bb            0
```

This configuration results in the following speed dial keys on the phone.



# Use the Corporate Directory

You can connect your phone to a corporate directory server that supports the Lightweight Directory Access Protocol (LDAP) version 3. The corporate directory is a flexible feature and table Use the Corporate Directory links you to the parameters you can configure. After set up on the phones, the corporate directory can be browsed or searched. You can call numbers and save entries you retrieve from the LDAP server to the local contact directory on the phone.

Polycom phones currently support the following LDAP servers:

- Microsoft Active Directory 2003 SP2
- Sun ONE Directory Server 5.2 p6
- Open LDAP Directory Server 2.4.12
- Microsoft Active Directory Application Mode (ADAM) 1.0 SP1

Polycom phones support corporate directories that support server-side sorting and those that do not. For phones that do not support server-side sorting, sorting is performed on the phone.

**Tip: Better performance with server-side sorting**

Polycom recommends using corporate directories that have server-side sorting for better performance. Consult your LDAP administrator when making any configuration changes for the corporate directory. For more information on LDAP attributes, see *RFC 4510 - Lightweight Directory Access Protocol (LDAP): Technical Specification Road Map*.

**Web Info: Supported LDAP directories**

Configuration of a corporate directory depends on the LDAP server you use. For detailed explanations and examples of all currently supported LDAP directories, see *Best Practices When Using Corporate Directory on Polycom Phones*: *Technical Bulletin 41137* at Polycom Engineering Advisories and Technical Notifications.

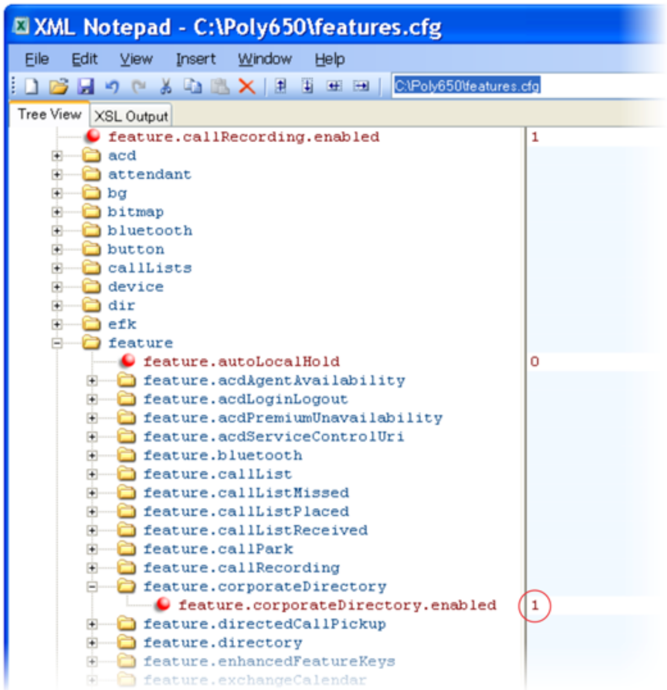
**Use the Corporate Directory**

| Parameter Function | **template** > parameter |
|---|---|
| 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 settings. | **features.cfg** > dir.corp.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

## Example Corporate Directory Configuration

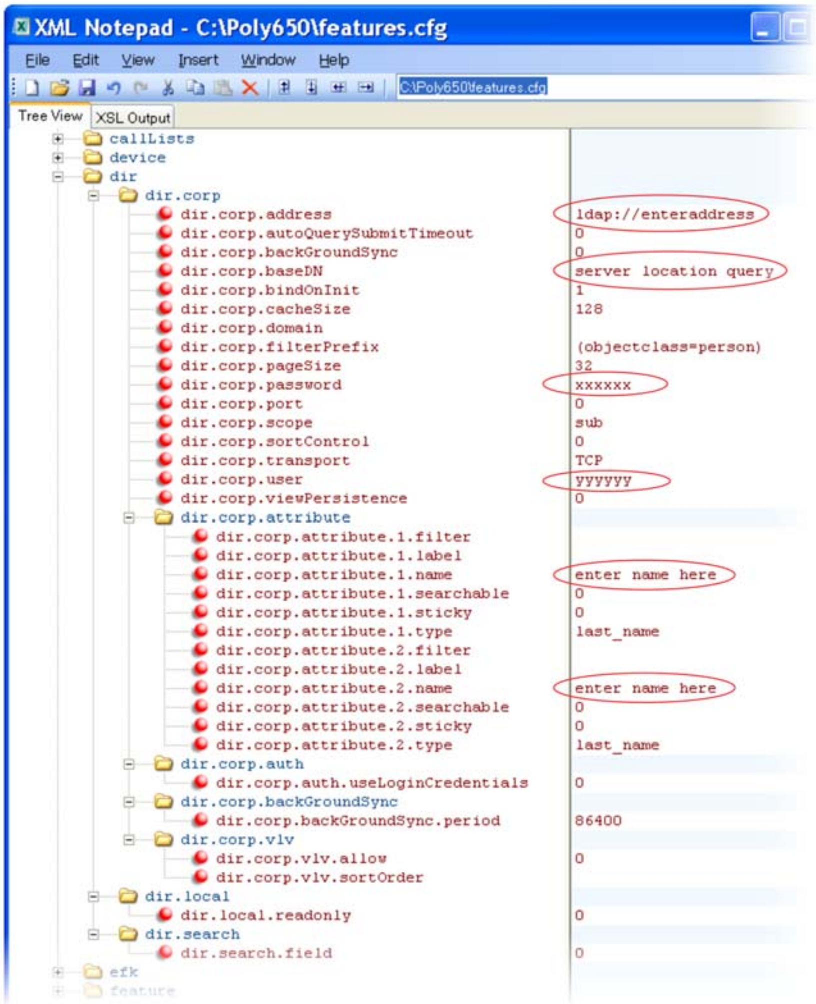
The following example is a representation of the minimum parameters you must set to begin using the corporate directory. The exact parameters and values you need to configure vary with the corporate directory you are using.

First, enable the corporate directory feature in the features.cfg template, as shown next.


XML Notepad - C:\Poly650\features.cfg
File Edit View Insert Window Help
C:\Poly650\features.cfg
Tree View
XSL Output

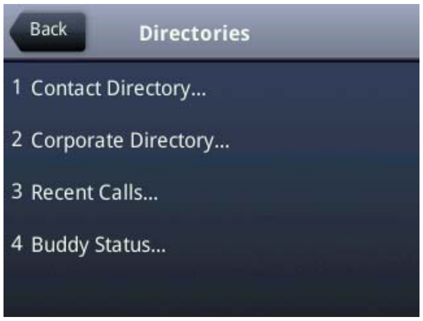
```
feature.callRecording.enabled                      1
acd
attendant
bg
bitmap
bluetooth
button
callLists
device
dir
efk
feature
   feature.autoLocalHold                           0
   feature.acdAgentAvailability
   feature.acdLoginLogout
   feature.acdPremiumUnavailability
   feature.acdServiceControlUri
   feature.bluetooth
   feature.callList
   feature.callListMissed
   feature.callListPlaced
   feature.callListReceived
   feature.callPark
   feature.callRecording
   feature.corporateDirectory
      feature.corporateDirectory.enabled           1
   feature.directedCallPickup
   feature.directory
   feature.enhancedFeatureKeys
   feature.exchangeCalendar
```

The following illustration points you to the minimum parameters you need to set. You need to enter a corporate directory address in dir.corp.address and specify where on the corporate directory server you want to make queries in dir.corp.baseDN. In addition, you require a user name and password. The dir.corp.attribute.x.name must match the attributes in the server.



```
callLists
device
dir
    dir.corp
        dir.corp.address                              ldap://enteraddress
        dir.corp.autoQuerySubmitTimeout               0
        dir.corp.backGroundSync                       0
        dir.corp.baseDN                               server location query
        dir.corp.bindOnInit                           1
        dir.corp.cacheSize                            128
        dir.corp.domain
        dir.corp.filterPrefix                         (objectclass=person)
        dir.corp.pageSize                             32
        dir.corp.password                             xxxxxx
        dir.corp.port                                 0
        dir.corp.scope                                sub
        dir.corp.sortControl                          0
        dir.corp.transport                            TCP
        dir.corp.user                                 yyyyyy
        dir.corp.viewPersistence                      0
        dir.corp.attribute
            dir.corp.attribute.1.filter
            dir.corp.attribute.1.label
            dir.corp.attribute.1.name                 enter name here
            dir.corp.attribute.1.searchable           0
            dir.corp.attribute.1.sticky               0
            dir.corp.attribute.1.type                 last_name
            dir.corp.attribute.2.filter
            dir.corp.attribute.2.label
            dir.corp.attribute.2.name                 enter name here
            dir.corp.attribute.2.searchable           0
            dir.corp.attribute.2.sticky               0
            dir.corp.attribute.2.type                 last_name
        dir.corp.auth
            dir.corp.auth.useLoginCredentials         0
        dir.corp.backGroundSync
            dir.corp.backGroundSync.period            86400
        dir.corp.vlv
            dir.corp.vlv.allow                        0
            dir.corp.vlv.sortOrder
    dir.local
        dir.local.readonly                            0
    dir.search
        dir.search.field                              0
efk
feature
```

To search the corporate directory, press Directories on the phone and select Corporate Directory, as shown next.



# Use Third-Party Servers

This section provides information on configuring phones and features with third-party servers.

## Configure Polycom Phones with Alcatel-Lucent

This section shows you how to configure Polycom phones with Alcatel-Lucent (ALU) CTS.

### Initiate and Manage Alcatel-Lucent Advanced Conferences

When you are signed into the ALU CTS on your VVX phone, you can initiate ad-hoc conference calls with two or more contacts from your phone. You can also create a participant list and manage conference participants. Note you you can have only one active conference call in progress at a time on your phone. This feature is not supported on VVX 101 and 201 phones.

You can configure the number of participants in a conference using the parameter `reg.x.advancedConference.maxParticipants`. The number of participants you configure must match the number of participants allowed on the ALU CTS.

Advance Conference includes the following features:

- **Roster** Provides a list of participants in the conference
- **Conference Controller** The person who creates the conference and can add or drop participants, and mute and unmute participants.
- **Push-to-Conference** Includes participants list when creating a conference call.
- Join two calls into a conference call
- Join a call to an active call to make a conference call

**Configure ALU Advanced Conferences**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable advanced conferences and conference controls. | **features.cfg** > feature.advancedConference.enabled |
| Enable or disable push-to-conference functionality. | **features.cfg** > reg.x.advancedConference.pushToConference |
| Specify the maximum number of participants allowed in a push to conference. | **features.cfg** > reg.x.advancedConference.maxParticipants |
| Enable or disable conference participants to receive notifications for conference events. | **features.cfg** > reg.x.advancedConference.subscribeForConfEvents |
| Enable or disable the conference host to receive notifications for conference events. | **features.cfg** > reg.x.advancedConference.subscribeForConfEventsOnCCPE |

## Barge In and Bridge In for ALU CTS

The Barge In feature is for BLF shared lines, and Bridge In is for Shared Call Appearance lines. This feature enables multiple users in a Shared Call Appearance group to view and bridge into active calls on a shared line. This feature is not supported on VVX 1500 business media phones.

By default, group members can bridge into active calls only. Users cannot bridge into held or incoming calls. Multiple people can bridge into one active call. This feature is disabled by default. You can enable this feature using the parameter reg.x.bridgeInEnabled.

#### Configure Barge In and Bridge In

| Parameter Function | **template** > parameter |
|---|---|
| Enables or disables the Bridge In feature. | **features.cfg** > reg.x.bridgeInEnabled |

## Barge-In for Busy Lamp Field Lines

This feature enables users to barge in on active and held calls on Busy Lamp Field (BLF) lines and supports three barge-in modes: Normal, Whisper and Silent. This feature is not supported on VVX 1500 business media phones. The Barge In feature for BLF lines is disabled by default. You can enable the barge-in feature, the default barge-in mode, and whether or not a tone plays when a contact barges in on a call.

#### Configure Barge In for BLF

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable barge-in and choose the default barge-in mode. | **features.cfg** > attendant.resourceList.x.bargeInMode |
| Enable or disable a tone that plays when a contact barges in on a call. | **features.cfg** > attendant.resourceList.x.requestSilentBargeIn |

## Dual Tone Multi Frequency (DTMF) Relay

This feature enables users to press DTMF commands during active SIP audio calls and conference calls to perform actions. This feature is not supported for H.323 calls.

#### Configure DTMF Relay

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable DTMF relays for active SIP calls. | voIpProt.SIP.dtmfViaSignaling.rfc2976 |
| Controls the behavior of the Star and Pound keys used for DTMF relays for active SIP calls. | voIpProt.SIP.dtmfViaSignaling.rfc2976.nonLegacyEncoding |

## Shared Call Appearance

The Shared Call Appearance feature enables users who share a line to monitor and bridge into calls on the shared line. Each line supports up to 21 call appearances. This feature is disabled by default. You can enable the feature and configure the hold request for the line. For ALU CTS, this feature is supported on VVX 300 series, VVX 400 series, VVX 500 series, and VVX 600 series phones.

Note the following when using shared call appearance with ALU CTS:

- Members of the SCA group cannot resume remotely held calls.
- 21 Shared Call Appearances per line.
- The maximum number of calls associated with a shared call appearance group is the same as the number of calls provisioned for that shared line.
- An incoming call to a shared call appearance group can be presented to the group only as long as there is one available idle call appearance.
- All shared call appearances are able to receive and originate calls, regardless of the call activity on the other shared call appearances.
- Users can bridge into an active SCA call that is in shared mode.

**Configure Shared Call Appearance**

| **Parameter Function** | **template** > parameter |
|---|---|
| Controls the default behavior of a Shared Call Appearance call. By default, an outgoing call from the call group is private. After the call is answered, the user needs to press the **Share** soft key to make the call public so that other people on the line can bridge in to the call. | **features.cfg** ><br>feature.scap.defCallTypeExclusive |
| Specify the Hold request for Shared Call Appearance calls to the ALU server. This value must match what is configured on the ALU server for the Shared Call Appearance hold request. | **features.cfg** ><br>feature.scap.HoldRequestUriUserPart |

## Visitor Desk Phone (VDP)

Visitor desk phone enables users registered with the Alcatel-Lucent CTS to access personal settings on a shared phone by logging in. Administrators configure a common setting for all phones and any user can make calls, including emergency calls, from a phone without having to log in. After the user logs in to the shared phone, personal settings are available as a user profile in <user> phones.cfg and any changes the user makes to phone settings are stored to this file. After the user logs out, another user can log into the phone to access their personal settings. Note that rebooting a phone logs out the user - the phone reboots with default settings and the user must log in. VDP is available on VVX 101, 201, 300 series, 400 series, 500 series, and 600 series business media phones.

On the server, you can configure the duration of a login period after which the user must re-enter credentials to the phone. When the time is nearing expiration, the server calls the phone and plays a message indicating the remaining time and prompts the user to re-enter credentials to extend the session.

If a user logs into a second phone when already logged into a first phone, the user is automatically logged out of the first phone. When logged in or out, users can dial an access code to play a message indicating if that user is logged in to a phone and the remaining time in a session.

The file <user>-directory.xml contains the user's contact list; the phone displays directory updates to the user at each login. Calls a user makes when logged into a phone are stored in call logs <user>-calls.xml. Calls a user makes when not logged in are not stored.

Note that any user phone services you enable, such as message-waiting indicator (MWI), busy lamp field (BLF), or shared call appearance (SCAP), are available to the user only after logging into the phone and the user profile is downloaded to the phone.

**Configure VDP**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable VDP and the Visitor login soft key. | **features.cfg** > feature.VDP.enabled |
| Specify the VDP login service access code. | **features.cfg** > prov.vdp.accessCode.login |
| Specify the VDP logout service access code. | **features.cfg** > prov.vdp.accessCode.logout |

## Configure Polycom Phones with GENBAND Server

GENBAND's application server, also called EXPERiUS™ A2, provides full-featured, IP-based multimedia communications applications for business and consumers. You can deploy EXPERiUS A2 as a standalone server or in combination with a GENBAND CONTiNUUM™ C20 server; features vary depending on your deployment. Polycom has performed interoperability tests with GENBAND C20 with Polycom VVX 300 series, 400 series, 500 series, 600 series, and 1500 phones.

UC Software 5.2 enhances Polycom's interoperability with GENBAND by offering several features on VVX business media phones while improving feature performance. Polycom UC Software 5.2.0 provides phones registered with GENBAND server the following new features or enhancements:

- MADN-SCA—Provides support for conference barge in, privacy, and remote call appearance. MADN-SCA requires you to deploy EXPERiUS A2 and CONTiNUUM C20 server.
- Global Address Book—The global address book (GAB) feature is a corporate directory application managed by the GENBAND server.
- Personal Address Book—The personal address book (PAB) feature is managed by the GENBAND server and allows multiple clients (phones, computer software) to read and modify a user's personal directory of contacts. When one client changes a contact all other clients are immediately notified of the change by the GENBAND server.
- E.911—Enhanced 911 services specific to GENBAND server implementation. You can use E.911 with a C20 server.

### Configure Multiple Appearance Directory Number – Single Call Appearance (MADN-SCA)

Multiple appearance directory number—single call appearance (MADN-SCA) enables a group of users to share a single directory number that displays as a single line to each member of the group. When this feature is enabled, users can initiate or receive calls on this shared line. MADN-SCA requires you to deploy EXPERiUS A2 and CONTiNUUM C20 server.

When you set the line to shared, an incoming call alerts all the members of the group simultaneously, and the call can be answered by any group member. On the server, you can configure a privacy setting that determines whether or not, after the call is answered, other members of the group can barge in to the same call using the Barge In soft key and whether or not a call on hold can be picked up by other members of the group. Only one call can be active on the line at a time on the MADN-SCA shared line. When a call is in progress, any incoming calls to the line receive a busy tone.

Optionally, you can configure star codes on the server that you can dial on the phone to toggle the privacy setting during a single active call. Note the following call behavior. If the line is configured for privacy by default, you can use a star code to toggle privacy on and off during an active call. When the call ends, the line resets to privacy settings. If the line is configured on the server with privacy off, you can use a star code to toggle to privacy on during an active call but you cannot toggle back to privacy off during the call. When the call ends, the line resets to privacy off.

In the UC Software download, Polycom provides the following two sample enhanced feature key (EFK) macros that you can configure to display on the phone to change privacy states: `privacyReleaseRestoreESK.cfg` and `privacyEnableESK.cfg`.

The following table lists all parameters available for MADN-SCA; see Example MDN-SCA Configuration for the minimum parameters you need to configure.

**MADN-SCA Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the user of the registration SIP URI. | **reg-basic.cfg** > reg.x.address |
| Per-registration parameter enabling the GENBAND MADN-SCA feature. | **reg-advanced.cfg** > reg.x.server.y.specialInterop |
| Global parameter enabling the GENBAND MADN-SCA feature. | **sip-interop.cfg** > voIpProt.server.x.specialInterop |
| Specify the line type as private or shared. | **reg-advanced.cfg** > reg.x.type |
| Enable or disable line barge-in. | **reg-advanced.cfg** > reg.x.bargeInEnabled |
| Set the maximum number of concurrent calls for a single registration. | **reg-advanced.cfg** > reg.x.callsPerLineKey |
| User ID to be used for authentication challenges for this registration. | **reg-basic.cfg** > reg.x.auth.userId |
| The password to be used for authentication challenges for this registration. | **reg-basic.cfg** > reg.x.auth.password |
| The IP address or hostname of the SIP server. | **reg-basic.cfg** > reg.x.outboundProxy.address |
| The domain of the authorization server. | **reg-advanced.cfg** > reg.x.auth.domain |
| This field must match the value of the registration which makes up the part of the shared line appearance. | **reg-advanced.cfg** > reg.x.thirdPartyName |

**Note: Setting up the MADN-SCA feature with GENBAND**

If you configure the line-specific parameter **reg.x.server.y.address**, you must also configure values in the line-specific parameter r**eg.x.server.y.specialInterop**.

If you configure the global parameter **voIpProt.server.x.address**, you must also configure values in the global parameter **voIpProt.server.x.specialInterop**.

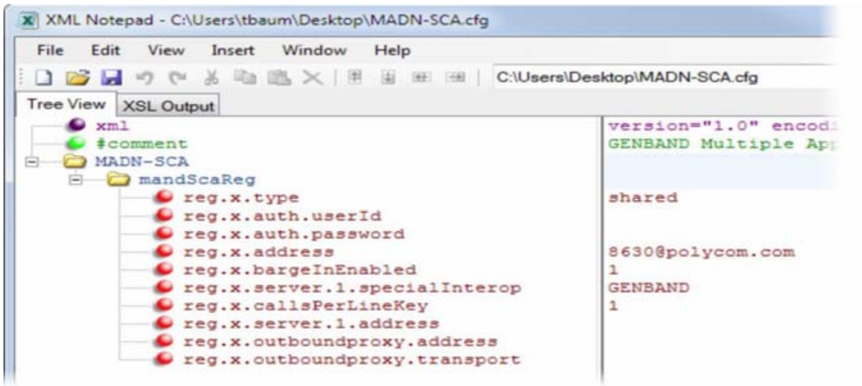
For all deployments, including GENBAND, line-specific configuration parameters override global configuration parameters. If you set values in both line-specific and global parameters, line-specific parameters are applied and global parameters are not applied.

### Example MDN-SCA Configuration

The following example configuration shows the minimum configuration you need to enable MDN-SCA on the phone. You can use the parameters in the template configuration files or create your own configuration file from the parameters.

### To configure MADN-SCA:

1 Enter values for the following parameters in a configuration file and save. The value `8630@polycom.com` is an example registration address.


XML Notepad - C:\Users\tbaum\Desktop\MADN-SCA.cfg
File
Edit
View
Insert
Window
Help
C:\Users\Desktop\MADN-SCA.cfg
Tree View
XSL Output

```
xml                                   version="1.0" encod
#comment                              GENBAND Multiple Ap
MADN-SCA
  mandScaReg
    reg.x.type                        shared
    reg.x.auth.userId
    reg.x.auth.password
    reg.x.address                     8630@polycom.com
    reg.x.bargeInEnabled              1
    reg.x.server.1.specialInterop     GENBAND
    reg.x.callsPerLineKey             1
    reg.x.server.1.address
    reg.x.outboundproxy.address
    reg.x.outboundproxy.transport
```

2 Enter the name of the configuration file to the CONFIG_FILES field of the master configuration file and save.

## Configure the Global Address Book (GAB)

GENBAND's global address book (GAB) is a read-only global contact directory set up by an administrator and can co-exist with other corporate directories on the phone. Users can access the GAB on the phone in the Directories menu and Features menu.

#### GAB Parameters

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the global address book. | **features.cfg** > feature.corporateDirectory.alt.enabled |
| Enter the URL address of the GAB service provided by the server. | **features.cfg** > dir.corp.alt.address |
| The port that connects to the server if a full URL is not provided. | **features.cfg** > dir.corp.alt.port |
| The user name used to authenticate to the GENBAND server. | **features.cfg** > dir.corp.alt.user |
| Displays the results from your last address directory search. | **features.cfg** > dir.corp.alt.viewPersistence |
| Use a filter to set a predefined search string through configuration files. | **features.cfg** > dir.corp.alt.attribute.x.filter |

**GAB Parameters**

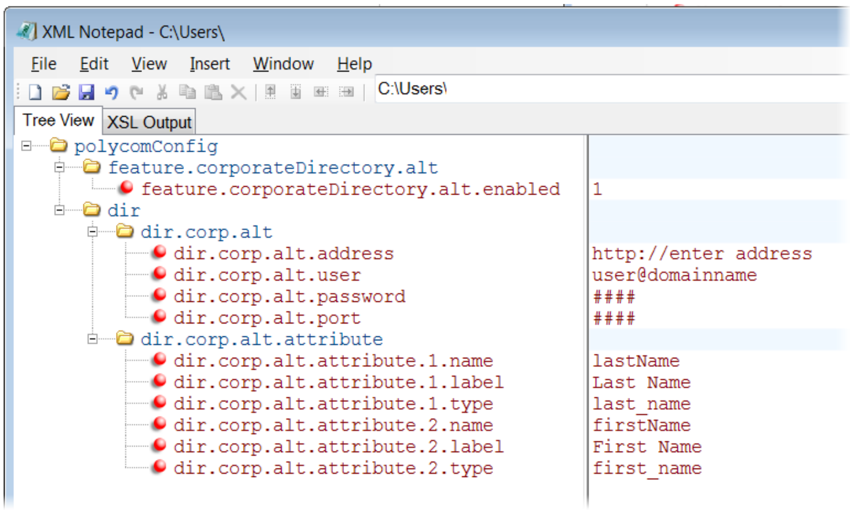
| | |
|---|---|
| Enable or disable a filter string. | **features.cfg** > dir.corp.alt.attribute.x.sticky |
| Provide a label to identify a user. | **features.cfg** > dir.corp.alt.attribute.x.label |
| The name of the parameter to match on the server. | **features.cfg** > dir.corp.alt.attribute.x.name |
| Define how parameter x is interpreted by the phone. | **features.cfg** > dir.corp.alt.attribute.x.type |
| Specify a method for synchronizing the directory and server. | **sip-interop.cfg**, **site.cfg**, > dir.local.serverFeatureControl.method |

## Example GAB Configuration

The following example shows the minimum parameters you need to configure to enable GAB on the phone.

### To configure GAB:

1 Enable GAB by configuring the values in `feature.corporateDirectory.alt` and `dir.corp.alt`. The following illustration includes an example GAB address book parameters in `dir.corp.alt.attribute`.



```
polycomConfig
  feature.corporateDirectory.alt
    feature.corporateDirectory.alt.enabled    1
  dir
    dir.corp.alt
      dir.corp.alt.address                    http://enter address
      dir.corp.alt.user                       user@domainname
      dir.corp.alt.password                   ####
      dir.corp.alt.port                       ####
    dir.corp.alt.attribute
      dir.corp.alt.attribute.1.name           lastName
      dir.corp.alt.attribute.1.label          Last Name
      dir.corp.alt.attribute.1.type           last_name
      dir.corp.alt.attribute.2.name           firstName
      dir.corp.alt.attribute.2.label          First Name
      dir.corp.alt.attribute.2.type           first_name
```

2 Save the configuration file.

3 Enter the name of the configuration file to the CONFIG_FILES field of the master configuration file and save.

## Configure the Personal Address Book (PAB)

The personal address book (PAB) enables users to read and modify a personal directory of contacts on their phone. Users can access the PAB on the phone in the Features or Directory menu.

When you modify contact information using any soft client, desk phone, or mobile client registered to the same line, the change is made on all other clients, and you are notified immediately of the change by the GENBAND server. When you enable server control, five telephone number fields per contact are available.

**PAB Parameters**

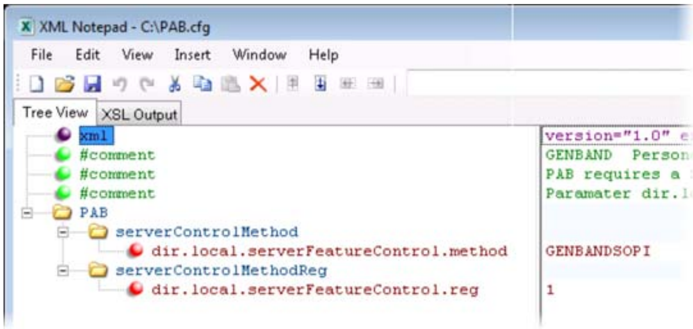
| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the global address book. | **features.cfg** > feature.corporateDirectory.alt.enabled |
| Enter the URL address of the GAB service provided by the server. | **features.cfg** > dir.corp.alt.address |
| The port that connects to the server if a full URL is not provided. | **features.cfg** > dir.corp.alt.port |
| The user name used to authenticate to the GENBAND server. | **features.cfg** > dir.corp.alt.user |
| Displays the results from your last address directory search. | **features.cfg** > dir.corp.alt.viewPersistence |
| Use a filter to set a predefined search string through configuration files. | **features.cfg** > dir.corp.alt.attribute.x.filter |
| Enable or disable a filter string. | **features.cfg** > dir.corp.alt.attribute.x.sticky |
| Provide a label to identify a user. | **features.cfg** > dir.corp.alt.attribute.x.label |
| The name of the parameter to match on the server. | **features.cfg** > dir.corp.alt.attribute.x.name |
| Define how parameter x is interpreted by the phone. | **features.cfg** > dir.corp.alt.attribute.x.type |
| Specify a method for synchronizing the directory and server. | **sip-interop.cfg**, **site.cfg**, > dir.local.serverFeatureControl.method |
| Specify the phone line to enable personnel address book feature on. | **site.cfg** > dir.local.serverFeatureControl.reg |
| Specify the maximum number of contacts available in the GENBAND personnel address book contact directory. | **site.cfg** > dir.genband.local.contacts.maxSize |

## Example PAB Configuration

The following example shows an example PAB configuration.

### To configure PAB:

1 Enter the values shown for the following parameters and save the configuration file.



```
xml                                          version="1.0" e
#comment                                     GENBAND  Person
#comment                                     PAB requires a
#comment                                     Paramater dir.l
PAB
  serverControlMethod
    dir.local.serverFeatureControl.method    GENBANDSOPI
  serverControlMethodReg
    dir.local.serverFeatureControl.reg       1
```

2 Enter the configuration file to the CONFIG_FILES field of the master configuration file and save.

## Configure GENBAND Enhanced 911 (E.911) Location

When using E.911 with GENBAND, you must correctly configure a location tree on the server that the phone downloads on start up. When the phone starts up, the phone prompts you to choose a location, which is stored on the phone to identify the phone location to 911 operators dispatching emergency services. This feature is available for all VVX phones and is disabled by default as it is available only in a GENBAND environment.

Callers can make a 911 call when the phone is locked, regardless of the call state, or when other features are in use. When a 911 call is in progress, the interface soft key options to end, hold, or transfer a call do not display, you cannot use hard keys on the VVX 300/301, 310/311, 400/401, 410/411, and 1500 to end, hold, or transfer a call, and DND or call forwarding enabled do not work.

### To set the location information on the phone:

1. Register the phone.

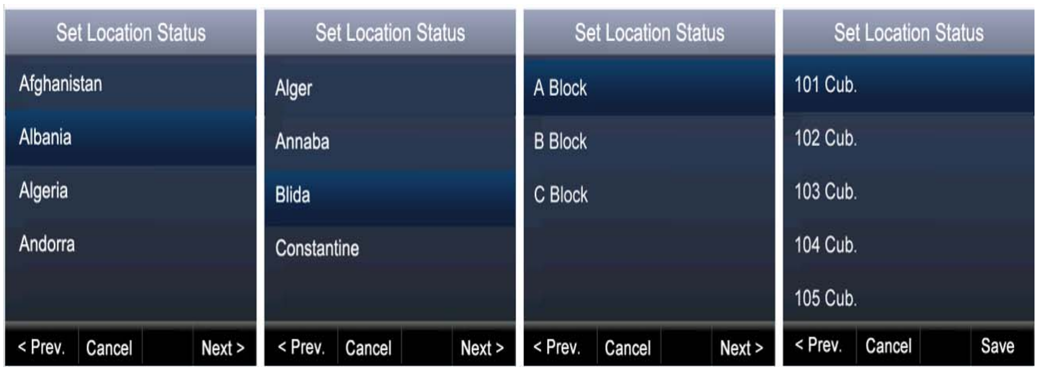
   The phone prompts you with a warning message ‘Set your location’ for 10 seconds.
2. Press the warning message to enter a location. If the warning message disappears, on the phone, go to **Settings > Status > Diagnostics > Warnings**.

   The warning message ‘Set your Location’ displays until you press Clear.
3. Press the **Details** soft key to enter a location to the location tree navigation menu. Example navigation menus are shown in the following figures.
4. Choose a location and press **Save**.
5. On the phone, go to **Status > Location Information**.

   The location information displays in the **Status** menu.

The following figure shows the menu structure for setting the Location Status on VVX phones.

**Location menu structure for VVX 500/501, 600/601, 1500**



**E911 Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the GENBAND E.911 feature. | **features.cfg** > feature.genband.E911.enabled |
| Enter a description of the location of the phone. | **site.cfg** > genband.E911.location.description |

**E911 Parameters**

| | |
|---|---|
| Enter the location ID corresponding to the location description you entered in `genband.E911.location.description.` | **site.cfg** > genband.E911.location.locationID |
| Select the registration line to use to retrieve E.911 location information | **reg-basic.cfg** > genband.E911.registration.line |

## Configure Polycom Phones with BroadSoft

This section shows you how to configure Polycom devices with BroadSoft Server options. You can use the features available on the BroadWorks R18 server or the BroadWorks R20 or later server with the following phones: VVX 300 series, 400 series, 500 series, 600 series, and 1500 phones. Note that you cannot register lines with the BroadWorks R18 server and the R20 and later server on the same phone. All lines on the phone must be registered to the same BroadWorks server.

Some features require you to authenticate the phone with the BroadWorks XSP service interface as described above in the section Configure Authentication for BroadSoft BroadWorks Xtended Service Platform (XSP) Service Interface.

### Configure Authentication for BroadWorks Xtended Service Platform (XSP) Service Interface

You can configure Polycom VVX business media phones to use advanced features available with BroadSoft BroadWorks server. The VVX business media phones support the following he advanced Broadsoft features:

- Broadsoft Enhanced Call Park
- Broadsoft UC-One directory, favorites, and presence
- Broadsoft UC-One personal call control features

To use these features on Polycom devices with a BroadWorks server, you must authenticate the phone with the BroadSoft XSP service interface. The authentication method to use depends on which version of BroadWorks you are running. If your server is running BroadWorks R19 or earlier, enable the following parameters to authenticate on the BroadWorks server using separate XSP credentials:

- `dir.broadsoft.xsp.address`
- `reg.x.broadsoft.userId`
- `reg.x.broadsoft.xsp.password`
- `reg.x.broadsoft.useXspCredentials`

If your server is running BroadWorks R19 Service Pack 1 or later, enable the following parameters to authenticate on the BroadWorks server using the same SIP credentials you used to register the phone lines: dir.broadsoft.xsp.address

- `reg.x.auth.userId`
- `reg.x.auth.password`
- `reg.x.broadsoft.userId`

**Configure BroadWorks XSP Service Interface Authentication**

| Parameter Function | **template** > parameter |
|---|---|
| Enter the password associated with the BroadSoft XSP user account for the line. Required only when `reg.x.broadsoft.useXspCredentials=1`. | **features.cfg** > reg.x.broadsoft.xsp.password |
| The BroadSoft user ID to authenticate with the BroadSoft XSP service interface, If this parameter value is empty, the line might be registered with BroadWorks server, and not have access to advanced features. | **features.cfg** > reg.x.broadsoft.userId |
| Determine the XSP authentication method for the BroadWorks version you are using. | **features.cfg** > reg.x.broadsoft.useXspCredentials |
| Set the BroadSoft Directory XSP address. | **features.cfg** > dir.broadsoft.xsp.address |
| User ID to be used for authentication challenges for this registration when `reg.x.broadsoft.useXspCredentials=0`. | **reg-basic.cfg** > reg.x.auth.userId |
| The password to be used for authentication challenges for this registration when `reg.x.broadsoft.useXspCredentials=0`. | **reg-basic.cfg** > reg.x.auth.password |

## BroadWorks Enhanced Call Park

You can configure BroadWorks Enhanced Call Park per registered line. The following features are available for Enhanced Call Park:

- You can configure Enhanced Call Park only using configuration files; you cannot configure the feature on the Web Configuration Utility or from the local phone interface.
- You can configure Enhanced Call Park for private lines and shared lines. No configuration is necessary to enable the call park notification for monitored BLF lines.
- The default star codes set for the `call.parkedCallRetrieveString` is *88.

The parameter `call.parkedCallRetrieveString` is updated for this feature to configure the access code to retrieve calls.

**Configure Enhanced Call Park**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the BroadWorks Enhanced Call Park feature. | **reg-advanced.cfg** > reg.x.enhancedCallPark.enabled |
| The line extension for a shared line. If there is no extension provided for this parameter, the call park notification is ignored for the shared line. | **reg-advanced.cfg** > reg.x.lineAddress |

**Configure Enhanced Call Park**

| | |
|---|---|
| Enable or disable the audio notifications for parked calls on private and shared lines. | **features.cfg** > feature.enhancedCallPark.allowAudioNotification |
| The access code used to retrieve a parked call. | **features.cfg** > call.parkedCallRetrieveString |

## Configure Polycom BroadSoft UC-One Application

The Polycom BroadSoft UC-One application integrates with BroadSoft Enterprise Directory and BroadCloud services—a set of hosted services by BroadSoft—to provide the following three features:

- **BroadSoft Directory** Displays information for all users in the enterprise, for example, work and mobile phone numbers.
- **BroadCloud Presence** Enables users to share presence information with the BroadTouch Business Communicator (BTBC) client application.
- **BroadCloud Favorites** Enables users to mark contacts as favorites with the BroadTouch Business Communicator (BTBC) client application.

These features are available on Polycom VVX 500 series and VVX 600 series business media phones running Polycom UC Software 4.1.3G or later, and VVX 300 series and VVX 400 series business media phones running Polycom UC Software 5.0.0 or later. These features require support from the BroadSoft BroadWorks R18 SP1 platform with patches and BroadSoft BroadCloud services. For details on how to set up and use these features, see the latest *Polycom VVX Business Media Phones - User Guide* at Latest Polycom UC Software Release.

Polycom's BroadSoft UC-One application enables you to:

- Access the BroadSoft Directory
- Search for contacts in BroadSoft Directory
- View BroadSoft UC-One contacts and groups
- View the presence status of BroadSoft UC-One contacts
- View and filter BroadSoft UC-One contacts
- Activate and control BroadSoft UC-One personal call control features.

The table Configure the BroadSoft UC-One Application lists all parameters available that configure features in the BroadSoft UC-One application.

Administrators can configure the UC-One Call Settings menu and menu options in the Web Configuration Utility.

### To enable UC-One Call Settings in the Web Configuration Utility:

1. Navigate to **Settings > UC-One**.
2. Under **General**, click **Enable** for **BroadSoft UC-One**.

   This enables the UC-One Call Settings menu to display on the phone.

### To enable UC-One Call Settings menu options:

1. In the Web Configuration Utility, navigate to **Settings > UC-One**.
2. Under **Call Settings Features**, enable each feature menu you want available on the phone.

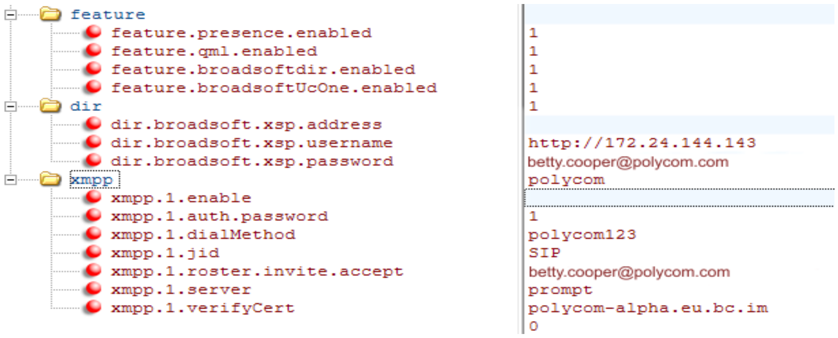
The following table lists all parameters available to configure features in the BroadSoft UC-One application.

**Configure the BroadSoft UC-One Application**

| Parameter Function | **template** > parameter |
|---|---|
| To turn QML application on or off and enable or disable display of the user interface for BroadSoft UC-One directory. | **features.cfg** > feature.qml.enabled |
| To turn BroadSoft Directory on or off. | **features.cfg** > feature.broadsoftdir.enabled |
| To turn BroadSoft UC-One on or off. | **features.cfg** > feature.broadsoftUcOne.enabled |
| To turn Presence on or off. | **features.cfg** > feature.presence.enabled |
| Enable or disable the UC-One Settings icon to display on the Home screen. | **features.cfg** > homeScreen.UCOne.enable |
| To set the BroadSoft Directory XSP home address. | **applications.cfg** > dir.broadsoft.xsp.address |
| To set the BroadSoft Directory XSP user name. | **applications.cfg** > dir.broadsoft.xsp.username |
| To set the BroadSoft Directory XSP password. | **applications.cfg** > dir.broadsoft.xsp.password |
| To set the BroadSoft XMPP password. | **features.cfg** > xmpp.1.auth.password |
| To set the BroadSoft XMPP dial method. | **features.cfg** > xmpp.1.dialMethod |
| To turn BroadSoft XMPP presence on or off. | **features.cfg** > xmpp.1.enable |
| To set the BroadSoft XMPP Jabber Identity used to register with presence server. | **features.cfg** > xmpp.1.jid |
| To turn the BroadSoft XMPP inviter's subscription for presence. | **features.cfg** > xmpp.1.roster.invite.accept |
| To set the BroadSoft XMPP presence server IP or FQDN. | **features.cfg** > xmpp.1.server |
| To turn the verification of the TLS certificate provided by the BroadSoft XMPP presence server on or off. | **features.cfg** > xmpp.1.verifyCert |

## Example BroadSoft UC-One Configuration

The illustration shown next provides an example configuration for the BroadSoft UC-One features.



```
feature
    feature.presence.enabled          1
    feature.qml.enabled               1
    feature.broadsoftdir.enabled      1
    feature.broadsoftUcOne.enabled    1
dir                                   1
    dir.broadsoft.xsp.address
    dir.broadsoft.xsp.username        http://172.24.144.143
    dir.broadsoft.xsp.password        betty.cooper@polycom.com
xmpp                                  polycom
    xmpp.1.enable
    xmpp.1.auth.password              1
    xmpp.1.dialMethod                 polycom123
    xmpp.1.jid                        SIP
    xmpp.1.roster.invite.accept       betty.cooper@polycom.com
    xmpp.1.server                     prompt
    xmpp.1.verifyCert                 polycom-alpha.eu.bc.im
                                      0
```

## Anonymous Call Rejection

Anonymous Call Rejection enables users to automatically reject incoming calls from anonymous parties who have restricted their caller identification. You can enable the Anonymous Call Rejection feature using configuration files or the Web Configuration Utility. After you enable the feature for users, users can turn call rejection on or off from the phone. When a user turns Anonymous Call Rejection on, the phone gives no indication that an anonymous call was received.

You can configure this option in the Web Configuration Utility.

### To enable Anonymous Call Rejection in the Web Configuration Utility:

1. Navigate to **Settings > UC-One**.
2. Under the **Call Setting Features**, click **Enable** for **Anonymous Call Rejection**.

**Configure Anonymous Call Rejection**

| Parameter Function | **template** > parameter |
|---|---|
| Displays the Anonymous Call Rejection menu on the phone. | **features.cfg** > feature.broadsoft.xsi.AnonymousCalReject.enabled |
| Enable or disable all BroadSoft UC-One features. | **features.cfg** > feature.broadsoftUcOne.enabled |
| Enter the BroadSoft user ID to confirm that the line is a registered BroadSoft line. | **features.cfg** > reg.x.broadsoft.userId |

## Simultaneous Ring Personal

The Simultaneous Ring feature enables users to add phone numbers to a list of contacts whose phones ring simultaneously when the user receives an incoming call. When you enable the Simultaneous Ring menu option on the phones for users, users can turn the feature on or off from the phone and define which numbers should be included in the Simultaneous Ring group.

You can enable or disable the Simultaneous Ring feature for users using configuration files or the Web Configuration Utility.

**Configure Simultaneous Ring**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the Simultaneous Ring Personal feature. | **features.cfg** > feature.broadsoft.xsi.SimultaneousRing.enabled |
| Enable or disable all BroadSoft UC-One features. | **features.cfg** > feature.broadsoftUcOne.enabled |

## Line ID Blocking

You can enable or disable the Line ID Blocking menu option on the phone. When you enable the menu for users, users can choose to hide their phone number before making a call. You can configure this feature using configuration parameters or the Web Configuration Utility.

**Configure Line ID Blocking**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the Line ID Blocking feature. | **features.cfg** > feature.broadsoft.xsi.LineIdblock.enabled |
| Enable or disable all BroadSoft UC-One features. If disabled, none of the UC-One features are enabled even if the individual feature parameters are enabled. | **features.cfg** > feature.broadsoftUcOne.enabled |

## BroadWorks Anywhere

BroadWorks Anywhere enables users to use one phone number to receive calls to and dial out from their desk phone, mobile phone, or home office phone. When you enable this feature, users can move calls between phones and perform phone functions from any phone. When enabled, the BroadWorks Anywhere settings menu displays on the phone and users can turn the feature on or off and add BroadWorks Anywhere locations on the phone. You can configure BroadWorks Anywhere using configuration files or the Web Configuration Utility.

**Configure BroadWorks Anywhere**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the BroadWorks Anywhere feature. If set to 0, the feature menu is disabled does not display. | **features.cfg** > feature.broadsoft.xsi.BroadWorksAnywhere.enabled |
| Enable or disable all BroadSoft UC-One features. If disabled, none of the UC-One features are enabled even if the individual feature parameters are enabled. | **features.cfg** > feature.broadsoftUcOne.enabled |

## Configure Remote Office

Remote Office enables users to set up a phone number on their office phone to forward incoming calls to a mobile device or home office number. When enabled, this feature enables users to answer incoming calls to the office phone on the phone, and any calls placed from that phone show the office phone number. You can configure Remote Office using configuration files or the Web Configuration Utility.

**Configure Remote Office**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the Remote Office feature. | **reg-advanced.cfg** > feature.broadsoft.xsi.RemoteOffice.enabled |
| Enter the BroadSoft user ID to confirm that the line is a registered BroadSoft line. If empty, the line is not considered as a BroadSoft line. | **features.cfg** > reg.x.broadsoft.userId |

**Configure Remote Office**

| | |
|---|---|
| Enable or disable all BroadSoft UC-One features. If disabled, none of the UC-One features are enabled even if the individual feature parameters are enabled. | **features.cfg** > feature.broadsoftUcOne.enabled |
| Set the BroadSoft Directory XSP password. | **applications.cfg** > dir.broadsoft.xsp.password |

## BroadSoft UC-One Credentials

Enabling this feature allows users to enter their BroadWorks UC-One credentials on the phone instead of in the configuration files. The parameters `reg.x.broadsoft.useXspCredentials`, and `feature.broadsoftUcOne.enabled` must be enabled to display the UC-One Credentials menu option on the phone.

**Configure XSP User Name an Password**

| **Parameter Function** | **template** > parameter |
|---|---|
| Set the IP address or hostname of the BroadSoft directory XSP home address. | **features.cfg** > dir.broadsoft.xsp.address |
| Enter the BroadSoft user ID to confirm that the line is a registered BroadSoft line. If empty, the line is not considered as a BroadSoft line. | **features.cfg** > reg.x.broadsoft.userId |
| Enable or disable all BroadSoft UC-One features. If disabled, none of the UC-One features are enabled even if the individual feature parameters are enabled. | **features.cfg** > feature.broadsoftUcOne.enabled |
| Set the BroadSoft Directory XSP user name. Note that this value will be overridden by what the user enters through the phone UI | **applications.cfg** > dir.broadsoft.xsp.username |
| Set the BroadSoft Directory XSP password. Note that this value will be overridden by what the user enters through the phone UI | **applications.cfg** > dir.broadsoft.xsp.password |
| Turn BroadSoft Directory on or off. | **features.cfg** > feature.broadsoftdir.enabled |

## Set Up Polycom Phones for Use with Microsoft Skype for Business Server

Lync Server provides a unified communications (UC) solution that enables customers, colleagues, and business partners to communicate instantly by voice, video, or messaging through a single interface, regardless of their location or network.Note that the features available when you are registered with Skype for Business Server vary with the Polycom phone model and Polycom UC Software version you are using. Note that UC Software 5.3.0 does not support the use of VVX 1500 business media phone with Skype for Business Server. Note that the concurrent failover/fallback feature is not compatible in a Microsoft environment.

You can deploy RealPresence Trio 8800 with Lync Server 2013. As of UC Software 5.4.0A, you can register Polycom phones using Microsoft Skype for Business Online.

As of UC Software 5.1.2, Polycom offers devices with an Open SIP or a Skype for Business base profile (a Lync SKU). Polycom devices shipped with a Lync base profile include Skype for Business-qualified UC Software with a feature license included and enable you to start up the phone and register with Skype for Business Server with default settings.

If you are using UC Software 5.3.0 with Skype for Business Server and want to change default settings or customize your deployment, you must set up a provisioning server. For a list of available features and full instructions on deploying RealPresence Trio 8800 with Lync Server, see the latest *Polycom UC Software in a Microsoft Environment – Deployment Guide* on Polycom UC Software for Microsoft Lync Deployments. For full details on the user features available on Polycom phones registered with Microsoft see the latest *Polycom VVX Business Media Phones User Guide* at Latest Polycom UC Software Release.

Polycom UC Software 5.4.1 supports the following devices with Lync Server and Skype for Business:

- VVX 201, 300 series, 400 series, 500 series, and 600 series business media phones
- SoundStructure VoIP Interface

Polycom UC Software 5.4.0A supports the following devices with Lync Server and Skype for Business:

- VVX 201, 300, 310, 400, 410, 500, and 600 business media phones
- SoundStructure VoIP Interface

You can register the following phones with Lync Server 2010 and Lync Server 2013 using UC Software 4.1.x and later:

- VVX 300, 310, 400, 410, 500, and 600
- SoundStructure VoIP Interface

Note the following points when using Polycom phones with Skype for Business Server:

- Polycom UC Software enables you to register a single phone line with Skype for Business Server; you cannot register multiple lines with Skype for Business Server. When you register a line on a Polycom phone using Skype for Business Server you cannot register lines with another server.

Polycom phones shipped with a Lync Base Profile (Lync SKU) include a license. If you are not using Polycom phones shipped with a Lync Base Profile, you must purchase a Skype for Business Feature License from a Polycom reseller or Polycom sales representative to use Polycom VVX products in a Microsoft Skype for Business environment. You can use Polycom phones in a Lync Server environment for trial purposes, without purchasing a license, to a maximum of 30 days.

- When you are running UC Software 4.1.x or 5.0 for use with Lync Server 2010, you have access to two separate contact lists: the default local contact directory on your Polycom phone and a Lync contact list. If you want to disable the local contact directory on your Polycom phone or make it read-only, see the section Use the Local Contact Directory.

**Tip: Workaround for phones using G.722 and retrieving Microsoft Skype for Business voicemail**

If your Polycom phones are configured with G.722 and users find that they do not hear audio when retrieving voicemail from the Microsoft Skype for Business Server, you need to make the following changes to parameters in the site.cfg template file:

- Change `voice.codecPref.G7221.24kbps` from 0 to 5.
- Change `voice.codecPref.G7221.32kbps` from 5 to 0.
- Add `voice.audioProfile.G7221.24kbps.payloadType` and set it to 112.

## Enable Microsoft Exchange Calendar Integration

As of UC Software 4.0.1, VVX phones can display the Microsoft Exchange 2007 and 2010 calendar. The calendar gives you quick access to meeting information and you can dial in to conference calls. To integrate the Microsoft Exchange Calendar features with your phone, configure the parameters in the table Enable Microsoft Exchange Calendar Integration.

You can launch the feature from a calendar widget that displays in the status bar on the VVX phone.

You need a valid Microsoft Windows credentials to access the Microsoft Exchange Calendar information on the phone. You can manage these credentials through the Login Credentials, which are available through **Menu > Settings > Basic > Login Credentials**.

You can view the calendar information in day or month format. On VVX phones, the meeting details displays beside the calendar view.

All possible phone numbers that you can dial to place a call to the meeting displays in the meeting details. You can automatically place a call by pressing a soft key.

A reminder pop-up is displayed 15 minutes before a scheduled meeting. You can dismiss the reminder, select snooze to have the reminder pop up again, open the meeting details view. A tone plays along with the reminder pop-up.

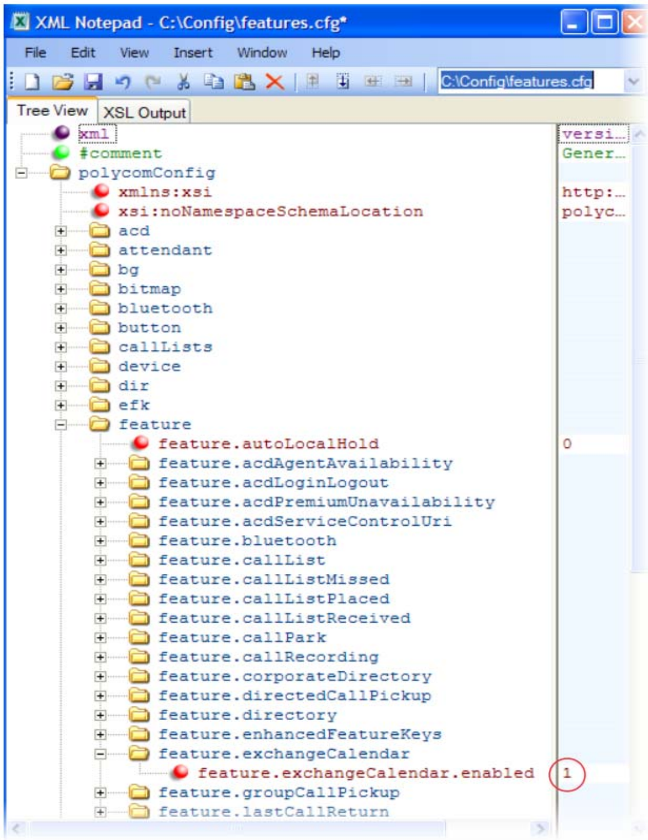
**Web Info: Using Microsoft Exchange Calendar integration**
For user instructions on how to use calendar integration, refer to the user guide for your phone at Polycom Voice Support.

**Enable Microsoft Exchange Calendar Integration**

| Parameter Function | **template** > parameter |
|---|---|
| Turn Microsoft Exchange Calendar Integration on or off. | **features.cfg** > feature.exchangeCalendar.enabled |
| Specify the Microsoft Exchange Server address. | **applications.cfg** > exchange.server.url |
| Specify the pattern to use to identify phone numbers in meeting descriptions. | **applications.cfg** > exchange.meeting.phonePattern |
| Turn the meeting reminder on or off. | **applications.cfg** > exchange.meeting.reminderEnabled |

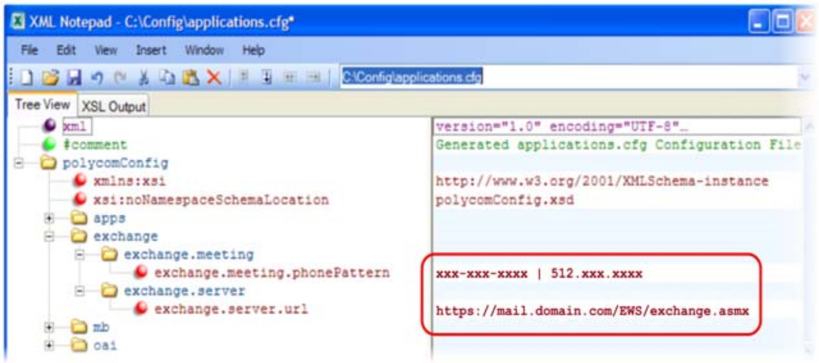
### Example Exchange Calendar Configuration

The following example shows the Calendar feature enabled in features.cfg.



```
xml                                      versi…
#comment                                 Gener…
polycomConfig
  xmlns:xsi                              http:…
  xsi:noNamespaceSchemaLocation          polyc…
  acd
  attendant
  bg
  bitmap
  bluetooth
  button
  callLists
  device
  dir
  efk
  feature
    feature.autoLocalHold                0
    feature.acdAgentAvailability
    feature.acdLoginLogout
    feature.acdPremiumUnavailability
    feature.acdServiceControlUri
    feature.bluetooth
    feature.callList
    feature.callListMissed
    feature.callListPlaced
    feature.callListReceived
    feature.callPark
    feature.callRecording
    feature.corporateDirectory
    feature.directedCallPickup
    feature.directory
    feature.enhancedFeatureKeys
    feature.exchangeCalendar
      feature.exchangeCalendar.enabled   1
    feature.groupCallPickup
    feature.lastCallReturn
```

After you enable the feature, specify the Microsoft Exchange Server address in applications.cfg, as shown next. In this example, a pattern has been specified for meeting numbers. When you specify a pattern, any number in your meeting invitation that matches the pattern displays on a meeting participants' phones as a soft key. Then, participants can press the soft key to dial in to the meeting. You can specify multiple patterns, separated by a bar. In the following example, two patterns are specified.


XML Notepad - C:\Config\applications.cfg*
File
Edit
View
Insert
Window
Help
C:\Config\applications.cfg
Tree View
XSL Output

```
xml                                        version="1.0" encoding="UTF-8"…
#comment                                   Generated applications.cfg Configuration File
polycomConfig
  xmlns:xsi                                http://www.w3.org/2001/XMLSchema-instance
  xsi:noNamespaceSchemaLocation            polycomConfig.xsd
  apps
  exchange
    exchange.meeting
      exchange.meeting.phonePattern        xxx-xxx-xxxx | 512.xxx.xxxx
    exchange.server
      exchange.server.url                  https://mail.domain.com/EWS/exchange.asmx
  mb
  oai
```

# Configure Phone Hardware

This section provides information on configuring the phone hardware.

## Connect to an Ethernet Switch

VVX business media phones have two Ethernet ports—labeled LAN and PC— and an embedded Ethernet switch that runs at full line rate. The SoundStructure VoIP Interface has one Ethernet port, labeled LAN. The Ethernet switch enables you to connect a personal computer and other Ethernet devices to the office LAN by daisy-chaining through the phone, eliminating the need for a stand-alone hub.

Each phone can be powered through an AC adapter or through a Power over Ethernet (PoE) cable connected to the phone's LAN port. To disable the PC Ethernet port, see the section Disable the PC Ethernet Port.

If you are 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 so that audio packets from the phone have priority over packets from the PC port. For more information, see <qos/>.

## USB Port Lockdown

The USB Port Lock down feature enables you to choose which phone headset ports to enable or disable and choose which of the phone's USB ports to power on or off.

The port lockdown feature is available on the VVX 500/501, 600/601, and 1500 phones. The VVX 1500 has a single USB port, and the VVX 500/501 and 600/601 support two USB ports, one port on the top and rear of the phones. The phones support various USB devices such as USB mass storage devices and a USB headset. The top USB port on the VVX 500/501 and 600/601 support the VVX Camera.

**Caution: Remove and replace feature.usbTop.power.enabled**

Two parameters `feature.usbTop.power.enabled` and `feature.usbRear.power.enabled` replace `feature.usb.power.enabled`. You must replace feature.usb.power.enabled with these two new parameters and set both to 0 to disable USB ports.

Note how the parameters affect ports on the phones:

- The parameters `feature.usbTop.power.enabled` and `feature.usbRear.power.enabled` enable and disable the USB ports on VVX 401/411, 500/501, 600/601, and 1500 phones.
- Only the parameter `feature.usbTop.power.enabled` applies to VVX 401/411 and 1500 phones. The parameter `feature.usbRear.power.enabled` does not apply to VVX 401/411 and 1500 phones because the VVX 401/411 and 1500 only have one USB port.
- You can control the VVX 500/501 and 600/601 top and rear USB ports independently using `feature.usbTop.power.enabled` to control the top USB port and `feature.usbRear.power.enabled` to control the rear USB port.

**Note: Disabling the top USB port on VVX 500/501 and 600/601 phones**

If you set the parameter `feature.usbTop.power.enabled` to 0 to disable the top USB port on VVX 500/501 and 600/601 phones, you must set the parameter `video.enable` to 0 as well.

The top and rear USB ports are enabled by default. Disabling the USB port makes the following features unavailable:

- Call recording
- Picture frame
- USB headset

- USB camera for video calls on the VVX 500/501 and 600/601 - no video calls
- USB charging device on the rear port of the VVX 500/501 and 600/601

**Note: A power adapter can cause issues on VVX 500/501 phones**

When you connect a power adapter to a VVX 500/501, the USB ports are powered on even if the parameters feature.usbTop.power.enabled and feature.usbRear.power.enabled are disabled. This can cause issues during phone reboots when USB devices are connected to the phone.

**Configure USB Port Lockdown**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable power to USB ports. | **template** > feature.usbTop.power.enabled |
| Enable or disable power to USB port 2. | **template** > feature.usbRear.power.enabled |
| Enable or disable power to the top USB port on VVX 500/501 and 600/601 phones. | **video.cfg** > video.enable |

## Connect Polycom VVX Expansion Modules

The Polycom VVX Expansion Modules are consoles you can connect to Polycom VVX business media phones to add additional lines. VVX Expansion Modules enable you to handle large call volumes on a daily basis and expand the functions of your phone. You can accept, screen, dispatch, and monitor calls with VVX Expansion Module and reduce the number of lost customer calls, shorten transaction times, and increase the accuracy of call routing.

You can capture the current screen of expansion modules, see the section

Polycom VVX Expansion Modules are available for the following Polycom VVX business media phones running UC Software 4.1.6 or later:

- VVX 300 series and 400 series business media phones
- VVX 500 series and 600 series business media phones

**Web Info: Polycom Support documentation for expansion modules**

For all documents that help you set up and use the Polycom VVX expansion modules with your VVX phones see Polycom VVX Expansion Modules Support page.

The following features are available on the VVX LCD Color Expansion Modules and VVX Expansion Modules with a paper display:

- **VVX Expansion Modules – LCD Color Display** VVX Color Expansion Modules feature an easy-to-navigate 272x480 LCD display. Each color expansion module provides you with 28 line keys and 3 display pages, supporting a total of 84 lines that you can set up as registrations, favorites, busy lamp field contacts, or Microsoft Skype for Business presence contacts. You can connect up to three color expansion modules to your phone to support an additional 252 line keys per phone. If you are registering Polycom phones with Skype for Business Server, you can use only the LCD color display expansion modules; you cannot use the paper display expansion modules for phones registered with Skype for Business Server.

- **VVX Expansion Modules – Paper Display** VVX paper display expansion modules provide you with 40 line keys that you can set up as registrations, favorites, or busy lamp field contacts. You can connect up to three expansion modules to your phone to support an additional 120 line keys per phone.

**Note: Line keys are numbered sequentially**

The number of line keys the phone supports varies by phone model. For a list of see the column in the table Flexible Call Appearances. Line keys on VVX phones and expansion modules are numbered sequentially, and the line key numbering on your expansion module depends on how many lines your phone supports. For example, a VVX 600/601 phone supports 16 lines, numbered 1-16. The first line on an expansion module connected to a VVX 600/601 phone is line 17.

The following figure illustrates the LCD color and paper expansion modules.

Polycom VVX Expansion Modules

**Expansion Module LCD color display and paper display**

## VVX Expansion Module Power Values

Polycom VVX phones use more power when you connect expansion module. The following table outlines the power each phone uses when you connect an expansion module, as well as the power value sent in LLDP-MED. For a list of power values for all Polycom phones without an expansion module attached, see Power Values.

**VVX Expansion Module Power Values**

| Model | Power Usage (Watts) | Power Value Sent in LLDP-MED Extended Power Via MDI TLV |
|---|---|---|
| VVX 300/301 | 5.0 | 5000mW |
| VVX 310/311 | 5.0 | 5000mW |
| VVX 400/401 | 5.0 | 5000mW |
| VVX 410/411 | 5.0 | 5000mW |

**VVX Expansion Module Power Values**

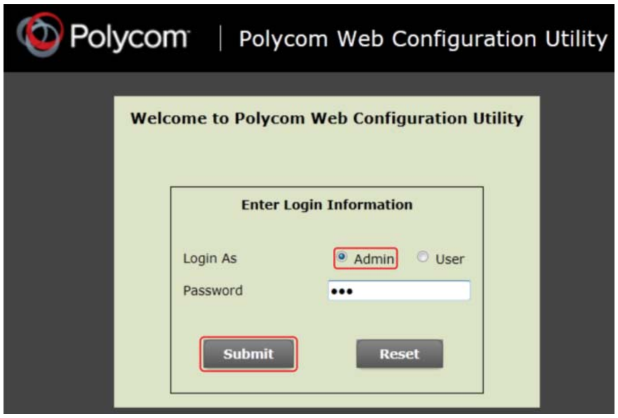
| | | |
|---|---|---|
| VVX 500/501 | 8.0 | 8000mW |
| VVX 600/601 | 8.0 | 8000mW |

## Generate Configured Line Key Information

Using the Web Configuration Utility, you can generate and download a PDF file with the line key configuration for each paper display expansion module connected to your VVX phone. The generated PDF enables you to print line key information for line keys on your expansion modules and insert the PDF as a directory card on your modules.

### To generate and download the line key information PDF using the Web Configuration Utility:

1. In your Internet browser, enter your phone's IP address into your browser's address bar.
2. Log in as an **Admin**, enter the default password, and select **Submit**, as shown next.



Polycom | Polycom Web Configuration Utility

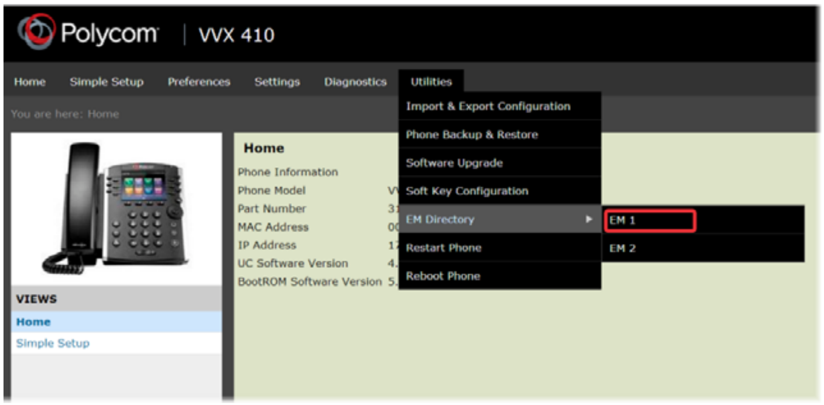
Welcome to Polycom Web Configuration Utility

Enter Login Information

Login As ● Admin ○ User

Password •••

3. Select **Utilities > EM Directory**.
4. Select the expansion module you want to generate a PDF for. For example, EM1 is chosen in the following figure.



5. In the confirmation dialog, select **Yes** to download the PDF for the configured lines for your expansion module.

6 Select **Save > Open**.

The PDF with the configured line key information for your expansion module displays.

After you download the PDF with configured line key information for your expansion module, you can print the PDF and insert the PDF as the directory card for the expansion module.

# Configure Smart Paging

The smart paging feature arranges line key assignments and distributes pages on the VVX Color Expansion Modules based on the number of expansion modules connected to a VVX phone. Smart paging is automatically enabled for color expansion modules connected to VVX phones with UC Software 5.1.0 or later, and is not available on the VVX Expansion Modules with a paper display. Note that when the flexible line key feature is enabled, the expansion module ignores the smart paging configuration and line key assignments display on the designated line key.

**Note: Line keys are numbered sequentially**

Line keys on VVX phones and expansion modules are numbered sequentially, and the line key numbering on your expansion module depends on how many lines your phone supports. For example, a VVX 600/601 phone supports 16 lines, numbered 1-16. The first line on an expansion module connected to a VVX 600/601 phone is line 17.

The following table lists the configuration parameter you need to enable and disable the smart paging feature.

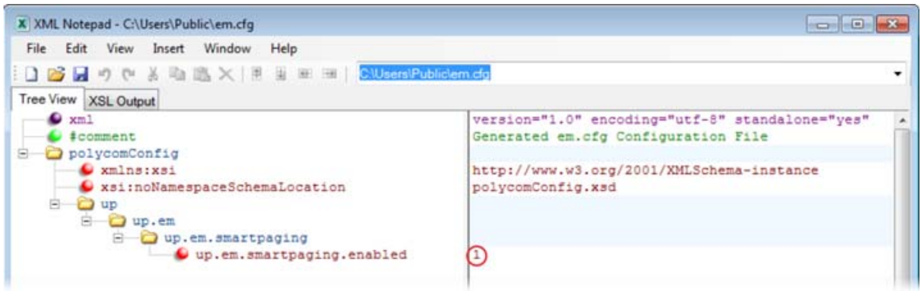
**Configuring Smart Paging**

| Parameter Function | **template** > parameter |
|---|---|
| Enable and disable smart paging. | **em.cfg** > up.em.smartpaging.enabled |

## Example Smart Paging Configuration

In the following example, smart paging is enabled in the `up.em.smartpaging.enabled` parameter in the em.cfg template. By default, the `up.em.smartpaging.enabled` parameter is set to 1, as shown in the following figure.

**Example of Smart Paging configuration**



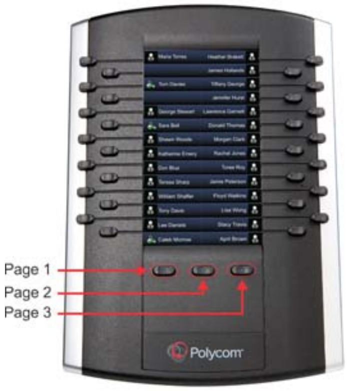
```
xml                                          version="1.0" encoding="utf-8" standalone="yes"
#comment                                     Generated em.cfg Configuration File
polycomConfig
    xmlns:xsi                                http://www.w3.org/2001/XMLSchema-instance
    xsi:noNamespaceSchemaLocation            polycomConfig.xsd
    up
        up.em
            up.em.smartpaging
                up.em.smartpaging.enabled    1
```

When you enable smart paging, the pages on the color expansion module are distributed across the connected expansion modules, as described in the following scenarios.

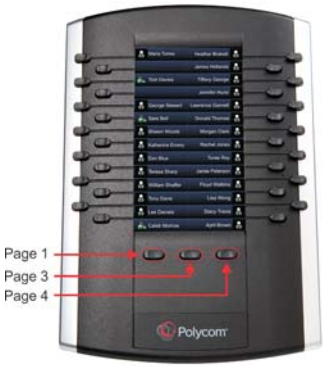
**Note: Smart paging available with multiple expansion modules**

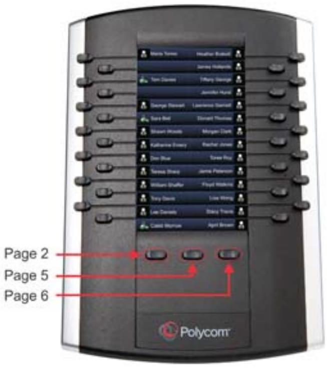
Smart paging applies only when you connect more than one expansion module to your VVX phone. If you connect one expansion module, the order of pages is sequential even if smart paging is disabled.

- If only one expansion module is connected to the VVX phone, the pages are ordered sequentially on the module: pages 1, 2, and 3.

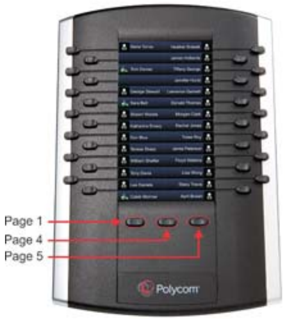


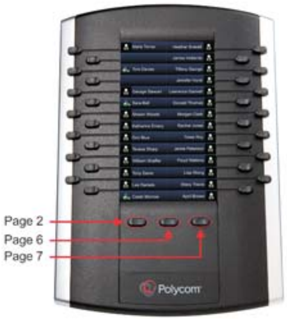
- If two expansion modules are connected, the pages are ordered non-sequentially across both expansion modules where pages 1, 3, and 4 are on the first expansion module, and pages 2, 5, and 6 are on the second expansion module.

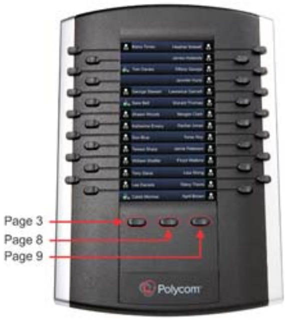




- If you are using three connected expansion modules, the pages are distributed across all modules where pages 1, 4, and 5 are on the first expansion module, pages 2, 6, and 7 are on the second expansion module, and pages 3, 8, and 9 are on the third expansion module.







## Enable the Power-Saving Feature

The VVX 500, 600, and 1500 phones support a power-saving feature that is enabled by default. This feature has a number of options you can configure:

- Turn on the phone’s power-saving feature during non-working hours and working hours. When you enable power-saving mode on VVX 500 and 600, the phone display screen does not automatically turn back on after going idle.
- If you want to turn on power-saving during non-working hours, you can configure the power-saving feature around your work schedule.
- On the VVX 1500 only, use the `powerSaving.userDetectionSensitivity.*` parameters to configure the sensitivity of the built-in motion detection system and an idle time after which the phone enters the power-saving mode.

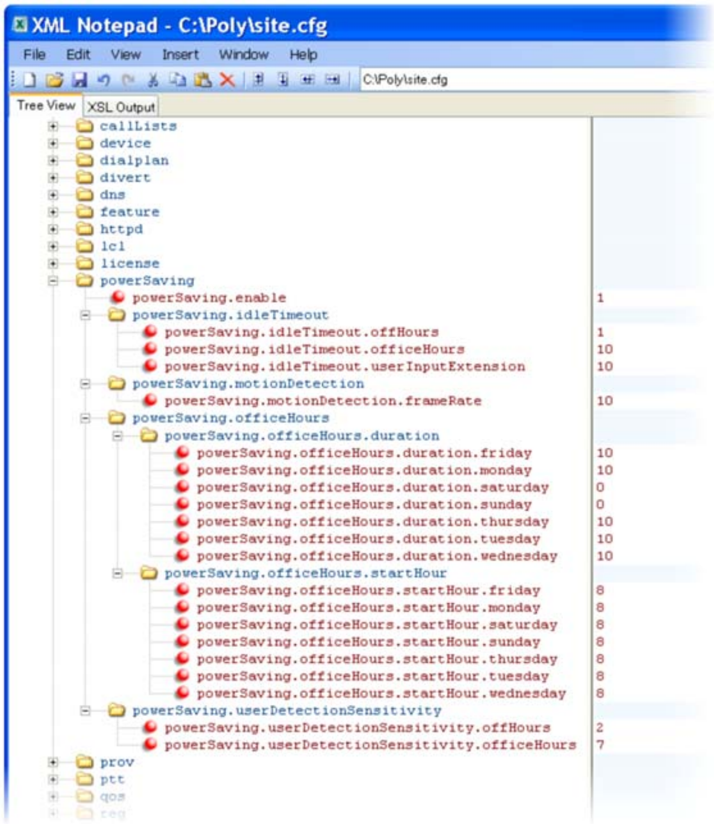
When you enable power-saving mode and the phone is in low power state, the red LED indicator flashes at three second intervals to show that the phone still has power.

**Power-Saving Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Turn the power-saving feature on or off. | **site.cfg** > powerSaving.enable |
| Specify the amount of time before the phone screen goes idle. | **site.cfg** > powerSaving.idleTimeout.* |
| Set the office hour start time and duration for each day of the week. | **site.cfg** > powerSaving.officeHours.* |
| Set the phone’s motion detection sensitivity. | **site.cfg** > powerSaving.userDetectionSensitivity.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

## Example Power-Saving Configuration

The following illustration shows the power-saving default settings, which reflect the hours of a typical work week.



```
callLists
device
dialplan
divert
dns
feature
httpd
lcl
license
powerSaving
    powerSaving.enable                                    1
    powerSaving.idleTimeout
        powerSaving.idleTimeout.offHours                  1
        powerSaving.idleTimeout.officeHours               10
        powerSaving.idleTimeout.userInputExtension        10
    powerSaving.motionDetection
        powerSaving.motionDetection.frameRate             10
    powerSaving.officeHours
        powerSaving.officeHours.duration
            powerSaving.officeHours.duration.friday       10
            powerSaving.officeHours.duration.monday       10
            powerSaving.officeHours.duration.saturday     0
            powerSaving.officeHours.duration.sunday       0
            powerSaving.officeHours.duration.thursday     10
            powerSaving.officeHours.duration.tuesday      10
            powerSaving.officeHours.duration.wednesday    10
        powerSaving.officeHours.startHour
            powerSaving.officeHours.startHour.friday      8
            powerSaving.officeHours.startHour.monday      8
            powerSaving.officeHours.startHour.saturday    8
            powerSaving.officeHours.startHour.sunday      8
            powerSaving.officeHours.startHour.thursday    8
            powerSaving.officeHours.startHour.tuesday     8
            powerSaving.officeHours.startHour.wednesday   8
    powerSaving.userDetectionSensitivity
        powerSaving.userDetectionSensitivity.offHours     2
        powerSaving.userDetectionSensitivity.officeHours  7
prov
ptt
qos
reg
```

## Enable Polycom Desktop Connector Integration

With the Polycom® Desktop Connector™ application installed on a computer, you can use your mouse and keyboard to enter information and navigate screens on your VVX phone running Polycom UC Software 4.0.1 or later. This feature enables users to enter phone numbers or to select screen objects without having to use the phone’s keypad or touch screen. To use this feature, the phone and computer must be on the same network or directly connected through the phone’s PC port.

You need to download and install the Polycom Desktop Connector application. The Polycom Desktop Connector is compatible with computers running Microsoft® Windows XP®, Windows Vista®, and Windows® 7.

Once Polycom Desktop Connector is installed, you need to pair the VVX phone and the computer (to configure the connection). If they are directly connected, there is no need to enter the VVX phone’s IP address; just press the Reconnect soft key. If they are connected through a switch or hub, enter the computer’s IP address using the phone’s user interface, and press the Reconnect soft key. You can also change the configuration by editing the phone’s configuration parameters shown in the following table or by using the Web Configuration Utility.

**Web Info: Installing and enabling the Polycom Desktop Connector application**

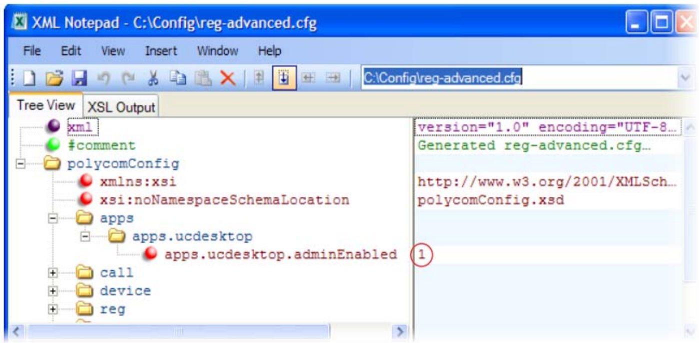
For details on how to install Polycom Desktop Connector application and enable it for use on VVX phones, see the latest *Polycom VVX Business Media Phones User Guide* at Latest Polycom UC Software Release.

**Enable Polycom Desktop Connector Integration**

| **Parameter Function** | **template** > parameter |
|---|---|
| Turn the Desktop Connector on or off for administrators. | **applications.cfg** > apps.ucdesktop.adminEnabled |
| Specify the user name of the user's computer. | **applications.cfg** > apps.ucdesktop.desktopUserName |
| Turn the Desktop Connector on or off for users. | **applications.cfg** > apps.ucdesktop.enabled |
| Specify if the phone is positioned to the left or right of your computer. | **applications.cfg** > apps.ucdesktop.orientation |
| Specify the server address of the user's computer. | **applications.cfg** > apps.ucdesktop.ServerAddress |
| Specify the server port number for the connection. | **applications.cfg** > apps.ucdesktop.ServerPort |

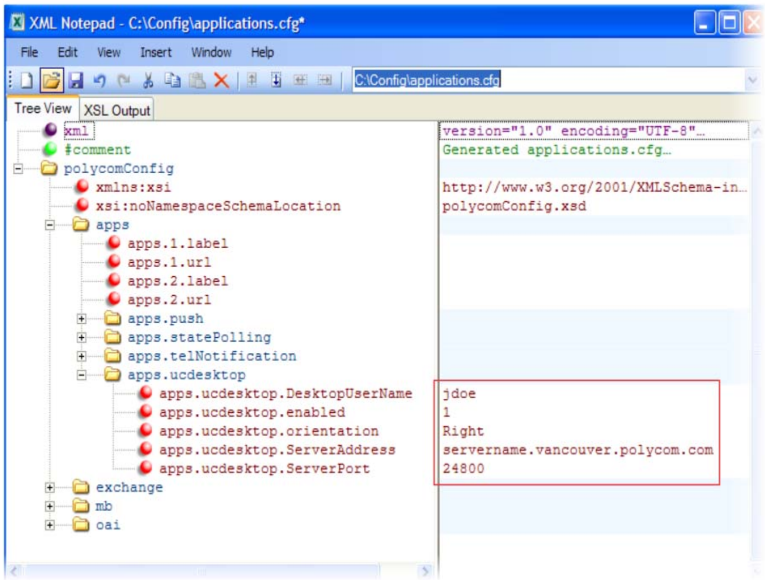
## Example Desktop Connector Configuration

To use the Polycom Desktop Connector, ensure that the `apps.ucdesktop.adminEnabled` in the applications.cfg template parameter is enabled, as shown next. By default, the parameter is enabled.


XML Notepad - C:\Config\reg-advanced.cfg
File
Edit
View
Insert
Window
Help
C:\Config\reg-advanced.cfg
Tree View
XSL Output

```
xml                                  version="1.0" encoding="UTF-8…
#comment                             Generated reg-advanced.cfg…
polycomConfig
  xmlns:xsi                          http://www.w3.org/2001/XMLSch…
  xsi:noNamespaceSchemaLocation      polycomConfig.xsd
  apps
    apps.ucdesktop
      apps.ucdesktop.adminEnabled    1
  call
  device
  reg
```

The following illustration shows the parameters in applications.cfg that you need to configure to use the Polycom Desktop Connector on your phones. You'll have to enable the feature, as well as specify a user name, server address and port, and specify the phone's position relative to your computer.


XML Notepad - C:\Config\applications.cfg*
File
Edit
View
Insert
Window
Help
C:\Config\applications.cfg
Tree View
XSL Output

| | |
|---|---|
| xml | version="1.0" encoding="UTF-8"… |
| #comment | Generated applications.cfg… |
| polycomConfig | |
| xmlns:xsi | http://www.w3.org/2001/XMLSchema-in… |
| xsi:noNamespaceSchemaLocation | polycomConfig.xsd |
| apps | |
| apps.1.label | |
| apps.1.url | |
| apps.2.label | |
| apps.2.url | |
| apps.push | |
| apps.statePolling | |
| apps.telNotification | |
| apps.ucdesktop | |
| apps.ucdesktop.DesktopUserName | jdoe |
| apps.ucdesktop.enabled | 1 |
| apps.ucdesktop.orientation | Right |
| apps.ucdesktop.ServerAddress | servername.vancouver.polycom.com |
| apps.ucdesktop.ServerPort | 24800 |
| exchange | |
| mb | |
| oai | |

# Customize Phone Audio

After you set up your Polycom phones on the network, phone users can send and receive calls using the default configuration. However, you might consider modifications that optimize the audio quality of your network. This section provides information on configuring phone audio.

Frequency bandwidth is one of the most critical elements affecting the intelligibility of speech in telephony. The frequency range that the human ear is most sensitive to is far beyond the capabilities of the plain old telephony system (POTS). In fact 80 percent of the frequencies in which speech occurs are not even used by public telephone networks because they only operate from 300Hz to 3.5 kHz. Complicating the intelligibility of telephony speech in today's world is background noise, variations in environmental reverberation, and communication among persons speaking a variety of native languages. While VoIP technology can broaden the frequency bandwidth and improve sound quality and intelligibility, it can also increase the network load and create a demand for lower raw bit rates. As Audio Codec Specifications shows, Polycom offers phones with a range of codecs, including codecs with high frequency bandwidth and low raw bit rates.

This section describes the audio sound quality features and options you can configure for your Polycom phones. Use these features and options to optimize the conditions of your organization's phone network system.

## Polycom Acoustic Fence™

Available on all VVX business media phones, the Polycom Acoustic Fence feature enables you to enhance background noise suppression when using the phone handset or a headset. This feature is particularly useful in call center environments where background noise can impact far-end audio quality.

**Configure Noise Suppression Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable noise suppression for headsets. | **features.cfg** > voice.ns.hd.enable |
| Enable or disable Acoustic Fence noise suppression for headsets. | **features.cfg** > voice.ns.hd.enhanced |
| Increase or decrease the noise suppression threshold on headsets. | **features.cfg** > voice.ns.hd.nonStationaryThresh |
| Enable or disable noise suppression for handsets. | **features.cfg** > voice.ns.hs.enable |
| Enable or disable Acoustic Fence noise suppression for handsets. | **features.cfg** > voice.ns.hs.enhanced |
| Increase or decrease the noise suppression threshold for handsets. | **features.cfg** > voice.ns.hs.nonStationaryThresh |

## Customize Audio Sound Effects

You can customize the audio sound effects that are used for incoming calls and other alerts using synthesized tones or sampled audio files. You can replace the default sampled audio files with your own custom .wav audio file format. The phone supports the following .wav audio file formats:

- 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 L16/44100 (16-bit dynamic range, 44.1 kHz sample rate)
- mono L16/48000 (16-bit dynamic range, 48-kHz sample rate)

Your custom sampled audio files must be available at the path or URL specified by saf.x in the following table so the phone can download them. Include the name of the file and the .wav extension in the path.

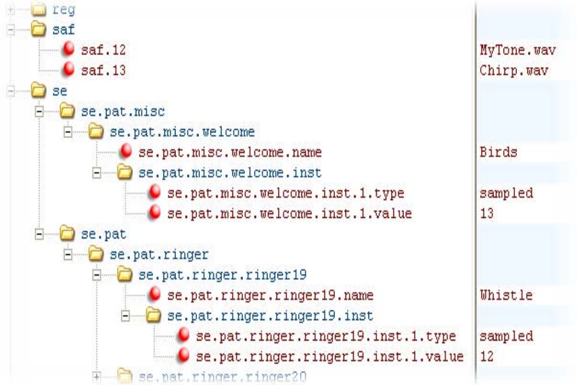
**Customize Audio Sound Effects**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify a path or URL for the phone to download a custom audio file. | **site.cfg** > saf.x |
| Specify the name, type, and value for a custom sound effect. | **region.cfg** > se.pat.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

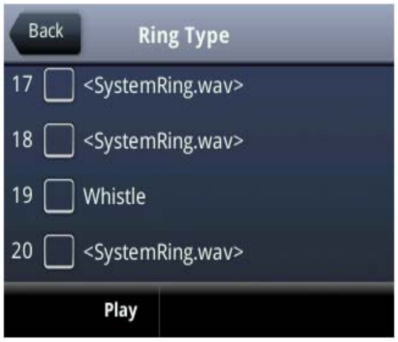
## Example Configuration

The following example configuration illustrates how to add a custom sound effect from a sampled audio file. In the example, the custom audio files MyTone.wav and Chirp.wav have been added as sound effects 12 and 13. The welcome sound has been customized to use the sampled audio file 13 (Chirp.wav) with the label Birds. Ringtone 19 is named Whistle and is configured to use sampled audio file 12 (MyTone.wav).



```
reg
saf
    saf.12                                          MyTone.wav
    saf.13                                          Chirp.wav
se
    se.pat.misc
        se.pat.misc.welcome
            se.pat.misc.welcome.name                Birds
            se.pat.misc.welcome.inst
                se.pat.misc.welcome.inst.1.type     sampled
                se.pat.misc.welcome.inst.1.value    13
    se.pat
        se.pat.ringer
            se.pat.ringer.ringer19
                se.pat.ringer.ringer19.name         Whistle
                se.pat.ringer.ringer19.inst
                    se.pat.ringer.ringer19.inst.1.type    sampled
                    se.pat.ringer.ringer19.inst.1.value   12
            se.pat.ringer.ringer20
```

The following illustration shows the custom ring tone Whistle as it displays on the phone menu:



## Adjust Context Sensitive Volume Control

The parameters shown in the following table enable you to adjust the volume of phone sound effects—such as the ringer and the volume of receiving call audio—separately for the speakerphone, handset, and headset. While transmit levels are fixed according to the TIA/EIA-810-A standard, you can adjust the receive volume. The receiving volume of the handset and headset resets after each call to comply with regulatory requirements. The hands free speakerphone volume level remains at the same level as the previous call.

**Context Sensitive Volume Control**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify if a Bluetooth headset should be used for every call (VVX 600/601 only). | **site.cfg** > voice.volume.persist.bluetooth.headset |
| Specify if the volume level of the handset, headset, and speakerphone should reset after each call. | **site.cfg** > voice.volume.persist.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

## Use Voice Activity Detection

The purpose of voice activity detection (VAD) is to detect periods of silence in the transmit data path so the phone doesn't have to transmit unnecessary data packets for outgoing audio. This process conserves network bandwidth. The VAD parameters in the following table help you set up this feature. For compression algorithms without an inherent VAD function, such as G.711, the phone uses the codec-independent comfort noise transmission processing specified in RFC 3389. The RFC 3389 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, to efficiently regenerate a facsimile of the background noise at the remote end.

**Voice Activity Detection Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify if G.729 Annex B should be signaled. | **site.cfg** > voice.vad.signalAnnexB |
| Enable or disable voice activity detection. | **site.cfg** > voice.vadEnable |
| Specify the threshold between active voices and background voices. | **site.cfg** > voice.vadThresh |

## Control Comfort Noise Payload Packets

When enabled, the Comfort Noise payload type is negotiated in SDP with the default of 13 for 8 KHz codecs and a configurable value between 96 and 127 for 16 KHz codecs.

The following table lists the parameters you can use to enable Comfort Noise Control.

**Comfort Noise Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Specify if the support for Comfort Noise in the SDP body of the INVITE message is published with the supported comfort noise payloads included in the media line for audio. | **site.cfg** > voice.CNControl |
| Specify the dynamic payload type used for Comfort Noise RTP packets. | **site.cfg** > voice.CN16KPayload |

## Generate Dual-Tone Multi-Frequency (DTMF) Tones

The phone generates dual-tone multi-frequency (DTMF) tones in response to user dialing on the dial pad. Use the parameters in the following table to set up this feature. These tones, commonly referred to as *touch 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 endpoint.

**Dual-Tone Multi-Frequency (DTMF) Tone Generation**

| Parameter Function | **template** > parameter |
|---|---|
| Specify if DTMF tones should be played through the speakerphone. | **sip-interop.cfg** > tone.dtmf.chassis.masking |
| Specify the frequency level of DTMF digits. | **sip-interop.cfg** > tone.dtmf.level |
| Specify how long the phone should wait between DTMF digits. | **sip-interop.cfg** > tone.dtmf.onTime |
| Specify how long the phone should play each DTMF tone for. | **sip-interop.cfg** > tone.dtmf.onTime |
| Enable or disable DTMF encoding in an RTP stream. | **sip-interop.cfg** > tone.dtmf.viaRtp |

## Set 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—or otherwise use—DTMF events received from the remote end of the call. Use the parameters in the following table to set up this feature.

**DTMF Event RTP Payload**

| Parameter Function | **template** > parameter |
|---|---|
| Specify if the phone use RFC 2833 to encode DTMF. | **sip-interop.cfg** > tone.dtmf.rfc2833Control |
| Specify the phone-event payload encoding in the dynamic range to be used in SDP offers. | **sip-interop.cfg** > tone.dtmf.rfc2833Payload |

## Acoustic Echo Cancellation

Your Polycom phone uses advanced acoustic echo cancellation (AEC) for handsfree operation using the speakerphone. See the table Audio Codec Priority for a list of audio codecs available for each phone and their priority. The phone also supports headset echo cancellation. The phones use both linear and non-linear techniques to aggressively reduce echo while permitting natural, full-duplex communication patterns.

**Caution: Contact Polycom Support before modifying acoustic echo cancellation parameters**
Consult Polycom Support before you make changes to any acoustic echo cancellation parameters.

## Supported Audio Codecs

The following table details the audio codec support and priority for Polycom phones.

**Settings: Video disables G.722.1C codec**
On the VVX 500/501 and 600/601, when you enable video, the G.722.1C codec is disabled.

**Audio Codec Priority**

| Phone | Supported Audio Codecs | Priority |
|---|---|---|
| VVX 101, 201 | G.711µ-law | 6 |
| | G.711a-law | 7 |
| | G.722 | 4 |
| | G.722.1 (32kbps) | 5 |
| | G.729AB | 8 |
| | iLBC (13.33kbps, 15.2kbps) | 0, 0 |
| VVX 300/301, 310/311, 400/401, 410/411 | G.711µ-law | 6 |
| | G.711a-law | 7 |
| | G.722 | 4 |
| | G.722.1 (32kbps) | 5 |
| | G.729AB | 8 |
| | iLBC (13.33kbps, 15.2kbps) | 0, 0 |

**Audio Codec Priority (continued)**

| | | |
|---|---|---|
| VVX 500/501, 600/601 | G.711 µ -law | 6 |
| | G.711a-law | 7 |
| | G.722 | 4 |
| | G.722.1 (32kbps) | 5 |
| | G.722.1C (48kbps) | 2 |
| | G.729AB | 8 |
| | Opus | 0 |
| | iLBC (13.33kbps, 15.2kbps) | 0, 0 |
| VVX 1500 | G.711 µ -law | 6 |
| | G.711a-law | 7 |
| | G.719 (64kbps) | 0 |
| | G.722 | 4 |
| | G.722.1 (32kbps) | 5 |
| | G.722.1C (48kbps) | 2 |
| | G.729AB | 8 |
| | Siren14 (48kbps) | 3 |
| | iLBC (13.33kbps, 15.2kbps) | 0, 0 |
| SoundStructure VoIP Interface | G.711 µ -law | 6 |
| | G.711a-law | 7 |
| | G.722 | 4 |
| | G.722.1 (32kbps) | 5 |
| | G.722.1C (48kbps) | 2 |
| | G.729AB | 8 |
| | iLBC (13.33kbps, 15.2kbps) | 0, 0 |

The table following summarizes the audio codecs supported on Polycom phones.

**Audio Codec Specifications**

| Algorithm | Reference | Raw Bit Rate | IP Bit Rate | Sample Rate | Default Payload Size | Effective Audio Bandwidth |
|---|---|---|---|---|---|---|
| G.711 µ -law | RFC 1890 | 64 Kbps | 80 Kbps | 8 Ksps | 20 ms | 3.5 KHz |
| G.711 a-law | RFC 1890 | 64 Kbps | 80 Kbps | 8 Ksps | 20 ms | 3.5 KHz |
| G.719 | RFC 5404 | 32 Kbps<br>48 Kbps<br>64 Kbps | 48 Kbps<br>64 Kbps<br>80 Kbps | 48 Ksps | 20 ms | 20 KHz |
| G.711 | RFC 1890 | 64 Kbps | 80 Kbps | 16 Ksps | 20 ms | 7 KHz |
| G.722[1] | RFC 3551 | 64 Kbps | 80 Kbps | 16 Ksps | 20 ms | 7 KHz |
| G.722.1 | RFC 3047 | 16 Kbps<br>24 Kbps<br>32 Kbps | 32 Kbps<br>40 Kbps<br>48 Kbps | 16 Ksps | 20 ms | 7 KHz |
| G.722.1C | G7221C | 224 Kbps<br>32 Kbps<br>48 Kbps | 40 Kbps<br>48 Kbps<br>64 Kbps | 32 Ksps | 20 ms | 14 KHz |
| G.729AB | RFC 1890 | 8 Kbps | 24 Kbps | 8 Ksps | 20 ms | 3.5 KHz |
| Opus | RFC 6716 | 8 - 24 Kbps | 24 - 40 Kbps | 8 Ksps<br>16 Ksps | 20 ms | 3.5 KHz<br>7 KHz |
| Lin16 | RFC 1890 | 128 Kbps<br>256 Kbps<br>512 Kbps<br>705.6 Kbps<br>768 Kbps | 132 Kbps<br>260 Kbps<br>516 Kbps<br>709.6 Kbps<br>772 Kbps | 8 Ksps<br>16 Ksps<br>32 Ksps<br>44.1 Ksps<br>48 Ksps | 10 ms | 3.5 KHz<br>7 KHz<br>14 KHz<br>20 KHz<br>22 KHz |
| Siren14 | SIREN14 | 24 Kbps<br>32 Kbps<br>48 Kbps | 40 Kbps<br>48 Kbps<br>64 Kbps | 32 Ksps | 20 ms | 14 KHz |
| Siren22 | SIREN22 | 32 Kbps<br>48 Kbps<br>64 Kbps | 48 Kbps<br>64 Kbps<br>80 Kbps | 48 Ksps | 20 ms | 22 KHz |
| iLBC | RFC 3951 | 13.33 Kbps<br>15.2 Kbps | 31.2 Kbps<br>24 Kbps | 8 Ksps | 30 ms<br>20 ms | 3.5 KHz |

[1] Per RFC 3551. Even though the actual sampling rate for G.722 audio is 16,000 Hz (16 ksps), the RTP clock rate advertised for the G.722 payload format is 8,000 Hz because that value was erroneously assigned in RFC 1890 and must remain unchanged for backward compatibility.

**Note: Network bandwidth requirements for encoded voice**
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 48 kbps for both the receive and transmit signals consumes about 100 kbps of network bandwidth (two-way audio).

Use parameters in the following table to specify the priority for audio codecs on your Polycom phones.

**Audio Codec Priorities**

| **Parameter Function** | **template** > parameter |
|---|---|
| To specify the priority for a codec. | **site.cfg** > voice.codecPref.<nameOfCodec> |

## Set 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. See the following table for available parameters. 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.

**IP Type-of-Service (ToS)**

| **Parameter Function** | **template** > parameter |
|---|---|
| Set the IP header bits for call control. | **site.cfg** > qos.ip.callControl.* |
| Set the IP header bits for RTP. | **site.cfg** > qos.ip.rtp.* |
| Set the IP header bits for RTP video. | **site.cfg** > qos.ip.rtp.video.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

## Set IEEE 802.1p/Q

The phone tags all Ethernet packets it transmits with an 802.1Q VLAN header when:

- A valid VLAN ID specified in the phone’s network configuration.
- The phone is instructed to tag packets through Cisco Discovery Protocol (CDP) running on a connected Ethernet switch.
- A VLAN ID is obtained from DHCP or LLDP (see Modify DHCP Settings)

Use the following table to set values. 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.

**IEEE 802.1p/Q**

| Parameter Function | **template** > parameter |
|---|---|
| Set the user priority for packets without a per-packet protocol setting (including 802.1p/Q). | **site.cfg** > qos.ethernet.other.user_priority |

## Configure Voice Quality Monitoring (VQMon)

You can configure the phones to generate various quality metrics you can use to monitor sound and listening quality. These metrics can be sent between the phones in RTCP XR packets, which are compliant with RFC 3611—RTP Control Extended Reports (RTCP XR). The packets are sent to a report collector as specified in draft RFC Session initiation Protocol Package for Voice Quality Reporting Event. The metrics can also be sent as SIP PUBLISH messages to a central voice quality report collector.

As of UC Software 5.3.0, you can use Real Time Control Protocol Extended Report (RTCP XR) to report voice quality metrics to remote endpoints. This feature is available on VVX phones and supports RFC6035 compliance as well as draft implementation for voice quality reporting.

Voice quality monitoring metrics are supported on the VVX phones. You require a license key to activate the VQMon feature on the VVX 300/301, 310/311, 400/401, and 410/411. This feature is available for open SIP and is not available with Skype for Business Server. For more information on VQMon, contact your Certified Polycom Reseller.

You can enable three types of voice quality reports:

- **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.

You can generate a wide range of performance metrics, the parameters for which are shown in the following table. 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.

**Voice Quality Monitoring (VQM)**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the warning threshold for alerts. | **features.cfg** > voice.qualityMonitoring.collector.alert.* |
| Enable the generation of quality reports. | **features.cfg** > voice.qualityMonitoring.collector.enable.* |
| Specify the server address and port. | **features.cfg** > voice.qualityMonitoring.collector.server.x.* |
| Enable the generation of RTCP-XR packets. | **features.cfg** > voice.qualityMonitoring.rtcpxr.enable |
| Specify the standards compliance. | **features.cfg** > voice.qualityMonitoring.rfc6035.enable |
| Enable or disables re-registration on failover. | **features.cfg** > voice.qualityMonitoring.failover.enable |

**Voice Quality Monitoring (VQM)**

| | |
|---|---|
| Specify the device location with a valid location string. | **features.cfg** > voice.qualityMonitoring.location |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Configure Audible Ringer Location

You can choose where all audio alerts, including incoming call alerts, are played out on devices running UC Software 3.3.0 or later. Use the following table to specify where you hear audio. You can specify the audio to play from the handsfree speakerphone (default), the handset, the headset, or the active location. If you choose the active location, audio alerts play out through the handset or headset if they are off hook. Otherwise, alerts play through the speakerphone.

**Audible Ringer Location**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify where audio alerts play out from. | **reg-advanced.cfg** > se.destination |

## Use Bluetooth Headset Support

You can use Bluetooth v2.1 headsets with VVX 600/601 business media phones. To use a Bluetooth headset, you need to enable the Bluetooth headset feature and turn on the Bluetooth radio, as shown in the following table.

**Troubleshooting: Using a Bluetooth headset affects my phone’s voice quality**
You may not experience the highest voice quality if you use a Bluetooth headset while the 2.4 GHz band is enabled or while you are in an environment with many other Bluetooth devices. This possible loss in voice quality is due to inherent limitations with Bluetooth technology.

**Bluetooth Headset Support**

| **Parameter Function** | **template** > parameter |
|---|---|
| To enable or disable the Bluetooth headset feature. | **features.cfg** > feature.bluetooth.enabled |
| To turn the Bluetooth radio (transmitter/receiver) on or off. | **features.cfg** > bluetooth.radioOn |

# Set Up Phone Video

After you set up Polycom phones on your network with the default configuration, users can place and answer video calls. Polycom's Open SIP UC Software enables you to make custom configurations to optimize video calling. This section provides information on configuring phone video.

The Polycom VVX 500/501, 600/601 and 1500 phones using the Polycom VVX Camera support transmission and reception of high quality video images. Polycom Open SIP 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.

## Configure Video Transmission

By default, at the start of a video call, the VVX 1500 and VVX phones using the VVX Camera transmit 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 Stop or Start soft key.

You can configure:

- Video Transmission Parameters
- Video and Camera View Parameters
- Video Camera Parameters

Use the parameters in the following table to configure video transmission on your VVX phones.

**Video Transmission Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify if video calls should use a full screen layout. | **video.cfg** > video.autoFullScreen |
| Specify when video transmission should start in a call. | **video.cfg** > video.autoStartVideoTx |
| Set the call rate for a video call (can be changed on the phone). | **video.cfg** > video.callRate |
| Specify whether the phone is forced to send RTCP feedback messages to request fast update I-frames for video calls. | **video.cfg** > video.forceRtcpVideoCodecControl |
| Set the maximum call rate for a video call (the maximum rate set from the phone cannot exceed this). | **video.cfg** > video.maxCallRate |
| Specify the quality of video to be shown in a call or conference. | **video.cfg** > video.quality |

You can use the parameters in the following table to set the video and local camera view settings on your VVX phones.

**Video and Camera View Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the view of the video window in normal viewing mode. | **video.cfg** > video.screenMode |

**Video and Camera View Parameters (continued)**

| | |
|---|---|
| Specify the view of the video window in full screen viewing mode. | **video.cfg** > video.screenModeFS |
| Specify if the local camera view is shown in the full screen layout. | **video.cfg** > video.localCameraView.fullscreen.enabled |
| Determine how the local camera view is shown. | **video.cfg** > video.localCameraView.fullscreen.mode |

You can use the parameters in the following table to configure the video camera on your VVX phones.

**Video Camera Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Set the brightness level. | **video.cfg** > video.camera.brightness |
| Set the contrast level. | **video.cfg** > video.camera.contrast |
| Specify if flicker avoidance is automatic, suited for Europe/Asia, or North America. | **video.cfg** > video.camera.flickerAvoidance |
| Set the frame rate. | **video.cfg** > video.camera.frameRate |
| Set the saturation level. | **video.cfg** > video.camera.saturation |
| Set the sharpness level. | **video.cfg** > video.camera.sharpness |

## Supported Video Codecs

See the following table for a summary of the VVX phone’s video codec support.

**Video Codec Specifications**

| Algorithm | MIME Type | Frame Size | Bit Rate (kbps) | Frame Rate (fps) |
|---|---|---|---|---|
| H.261 | H261/90000 | Tx Frame size: CIF, QCIF, SQCIF<br>RX Frame size: CIF, QCIF | 64 to 768 | 5 to 30 |
| H.263 | H263/90000, H263-1998/90000 | Tx Frame size:CIF, QCIF<br>Rx Frame size:CIF, QCIF, SQCIF, QVGA, SVGA, SIF | 64 to 768 kbps | 5 to 30 |
| H.264 | H264/90000 | Tx Frame size:CIF, QCIF<br>Rx Frame size:CIF, QCIF, SQCIF, QVGA, SVGA, SIF | 64 to 768 | 5 to 30 |

You can configure the parameters in the following table to prioritize and adjust the video codecs that your VVX phones use.

**Video Codec Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Prioritize the video codecs from 1 to 4. | **video.cfg** > video.codecPref.* |
| Adjust the parameters for the H261, H263, H2631998, and H264 codec profiles. | **video.cfg** > video.profile.<codec>.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Configure H.323 Protocol

As of SIP 3.2.2, the VVX 1500 phone and VVX camera-enabled VVX 500/501 and 600/601 phones support telephony signaling via the H.323 protocols. This protocol enables direct communication with H.323 endpoints, gatekeepers, call servers, media servers, and signaling gateways.

**Note: Activating H.323 video**
You need a license key to activate H.323 video on your VVX 1500 phone; the license is installed on the VVX 1500D. For more information, contact your Certified Polycom Channel Partner.

The VVX 500/501, 600/601, and 1500 can support SIP and H.323 signaling simultaneously, and you can bridge both types of calls during multi-party conference calls. The phone can automatically detect the correct or optimal signaling protocol when dialing a call from the contact directory or the corporate directory. While SIP supports server redundancy and several transport options, only a single configured H.323 gatekeeper address per phone is supported. The phone does not require H.323 gatekeepers, but use then if available. If an H.323 gatekeeper is not configured or is unavailable, you can still enable your phone to make H.323 calls.

Support of the SIP protocol for telephony signaling can be disabled on the VVX 500/501, 600/601, and 1500 such that all calls route via the H.323 protocol.

This section provides detailed information on:

- Supported H.323 Video Standards
- Supported Polycom Interoperability
- Use the H.323 Protocol
- FQDN Support for H.323 Gatekeeper Failover

For a list of all H.323 parameters, see the following table.

**H.323 Protocol Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify if the user is presented with protocol routing choices. | **reg-advanced.cfg** and **site.cfg**> up.manualProtocolRouting |
| Set soft keys for protocol routing. | **reg-advanced.cfg** and **site.cfg** > up.manualProtocolRouting.softKeys |

**H.323 Protocol Parameters (continued)**

| | |
|---|---|
| Enable or disable auto-answer for all H.323 calls. | **reg-advanced.cfg** and **h323.cfg** > call.autoAnswer.H323 |
| Specify if the phone can make calls using H.323 even if an H.323 gatekeeper is not configured or is unavailable. | **sip-interop.cfg** > call.enableOnNotRegistered |
| Specify if video should begin immediately after a call is auto-answered. | **reg-advanced.cfg** > call.autoAnswer.videoMute |
| Specify whether SIP or H.323 is the preferred call protocol. | **video.cfg** > call.autoRouting.preferredProtocol |
| Specify if calls should be routed by line or by protocol. | **sip-interop.cfg** > call.autoRouting.preference |
| Enable or disable H.323 signaling for the line registration. | **sip-interop.cfg** > reg.x.protocol.H323 |
| Specify the H.323 server settings for a specific registration. | **site.cfg** > reg.x.server.H323.* |
| Specify the H.323 protocol settings. | **h323.cfg** > voIpProt.H323.* |
| Specify the H.323 server settings. | **h323.cfg** > voIpProt.server.H323.* |
| Configure the H.323 media encryption parameters. | **site.cfg** > sec.H235.mediaEncryption.* |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Supported H.323 Video Standards

The following table lists the standards the H.323 feature supports.

**Supported Video Standards**

| 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 |
| 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 |

## Supported Polycom Interoperability

Polycom endpoints/bridges/call servers (or gatekeepers)/media servers support the video calls listed in the following table.

**Supported Polycom Product Interoperability**

| Polycom Product | Protocol | Software Version |
|---|---|---|
| Polycom HDX® 9000 series | SIP/ISDN/H.323 | SW 2.6.0 |
| Polycom HDX 8000 series | SIP/ISDN/H.323 | SW 2.6.0 |
| Polycom HDX7000 series | SIP/ISDN/H.323 | SW 2.6.0 |
| Polycom HDX 6000 | SIP/ISDN/H.323 | SW 2.6.0 |
| Polycom HDX 4000 series | SIP/ISDN/H.323 | SW 2.6.0 |
| Polycom RMX® 2000 | H.323 | SW 4.0.2.7 |
| Polycom Quality Definition Experience™ (QDX™) | H.323 | SW 4.0, 4.0.1 |
| Polycom RMX 1000 | H.323 | SW 1.1.1.8787 |
| Polycom RMX 2000 | H.323 | SW 5.0.1.24, 6.0 |
| Polycom RSS™ | H.323 | SW 6.0 |
| Polycom VBP™ 6400-ST series | H.323 | SW 9.1.5.1 |
| Polycom VBP 5300-ST series | H.323 | SW 9.1.5.1 |
| Polycom VBP 5300-E series | H.323 | SW 9.1.5.1 |
| Polycom VBP 4350 series | H.323 | SW 9.1.5.1 |
| Polycom VBP 200 | H.323 | SW 9.5.2 |
| Polycom VSX® 8000 | SIP/ISDN/H.323 | SW 9.0.6 |
| Polycom VSX 7000s and VSX 7000e | SIP/ISDN/H.323 | SW 9.0.6 |
| 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 |
| RealPresence® Group Series 300 | H.239/H.261/H.263\H.264/SIP/TIP | SW 4.1.1 |
| RealPresence Group Series 500 | H.239/H.261/H.263\H.264/SIP/TIP | SW 4.1.1 |
| RealPresence Group Series 700 | H.239/H.261/H.263\H.264/SIP/TIP | SW 4.1.1 |

**Web Info: Viewing an updated list of Polycom video support with third-party products**

See the *UC Software Release Notes* on the Latest Polycom UC Software Release page 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.

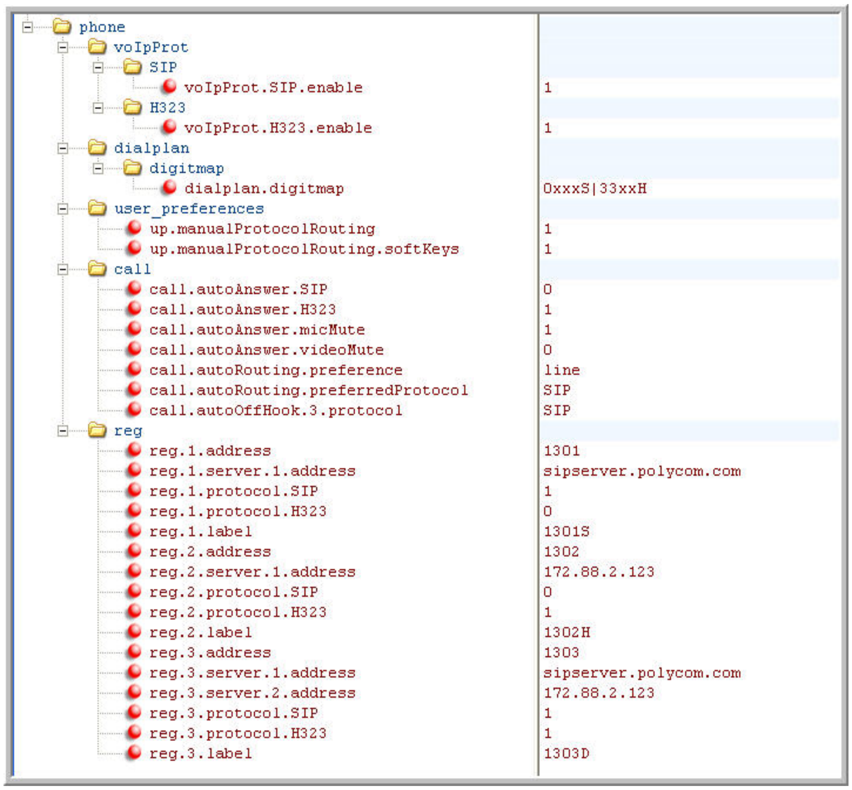
## Use the H.323 Protocol

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 continues to work as a SIP-only phone; however, the phone is not registered (you are able to send and receive SIP URL calls).
- The phone stores the protocol used to place a call in the placed call list.
- 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, when more than one protocol is available, each protocol displays as a soft key and the user can choose which protocol to use.
- Calls made using H.323 cannot be forwarded or transferred.
  - The Transfer and Forward soft keys do not display during an H.323 call on a VVX 500/501, 600/601, or 1500 phone. The Forward soft key does not display on the idle screen on a VVX 500/501, 600/601, or 1500 phone if the primary line is an H.323 line.
  - If a VVX 500/501, 600/601, or 1500 user presses the Transfer soft key during an H.323 call, 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 call and one of the parties is connected by H.323, that party is not transferred to the other party that was part of the conference call.

### Example H.323 Configuration

The following illustrates an example of a sip-h323.cfg file and the parameters you need to configure.



```
phone
    voIpProt
        SIP
            voIpProt.SIP.enable                      1
        H323
            voIpProt.H323.enable                     1
    dialplan
        digitmap
            dialplan.digitmap                        0xxxS|33xxH
    user_preferences
        up.manualProtocolRouting                     1
        up.manualProtocolRouting.softKeys            1
    call
        call.autoAnswer.SIP                          0
        call.autoAnswer.H323                         1
        call.autoAnswer.micMute                      1
        call.autoAnswer.videoMute                    0
        call.autoRouting.preference                  line
        call.autoRouting.preferredProtocol           SIP
        call.autoOffHook.3.protocol                  SIP
    reg
        reg.1.address                                1301
        reg.1.server.1.address                       sipserver.polycom.com
        reg.1.protocol.SIP                           1
        reg.1.protocol.H323                          0
        reg.1.label                                  1301S
        reg.2.address                                1302
        reg.2.server.1.address                       172.88.2.123
        reg.2.protocol.SIP                           0
        reg.2.protocol.H323                          1
        reg.2.label                                  1302H
        reg.3.address                                1303
        reg.3.server.1.address                       sipserver.polycom.com
        reg.3.server.2.address                       172.88.2.123
        reg.3.protocol.SIP                           1
        reg.3.protocol.H323                          1
        reg.3.label                                  1303D
```

Use these parameters:

- To configure 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 display, and you 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.
- 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.

## FQDN Support for H.323 Gatekeeper Failover

This enhancement, available only for registration failover scenarios, enables fully qualified domain name (FQDN) configuration for H.323 Gatekeeper. Gatekeeper IP addresses are resolved from a DNS server when the Gatekeeper sends a DNS A query or through the local static cache. This enhancement supports a maximum of two IP addresses based on the DNS response irrespective of the number of records received. Note that this enhancement does not apply if you are using the parameter `voIpProt.H323.autoGateKeeperDiscovery` for autodiscovery.

## Toggle Between Audio-only or Audio-Video Calls

When this feature is enabled on the VVX 1500, and VVX camera-enabled VVX 500/501 and 600/601 business media phones, a soft key displays to toggle calls between audio-only or audio-video. Note that this feature applies only to outbound calls from your phone; incoming video calls to your phone are answered using video even when you set the feature to use audio-only. When you enable this feature using f`eature.audioVideoToggle.enabled`, calls are audio-only by default, and you must toggle the call to use audio-video before the call begins. After a video call has ended, the phone returns to audio-only. If you want a call mode setting to persist until users manually change the call mode, also enable `audioVideoToggle.callMode.persistent`. Use the following table to locate available parameters.

**Voice and Video Toggle Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the audio/video toggle feature. | **features.cfg** > feature.audioVideoToggle.enabled |
| Enable or disable the last call mode set by the user. | **video.cfg** > audioVideoToggle.callMode.persistent |
| Allow the user to select the call mode to use when using SIP protocol only. | **video.cfg** > video.callMode.default |

## Switch Between Voice and Video During Calls

You can enable VVX 1500, VVX camera-enabled VVX 500/501 and 600/601 phones to switch between voice and video during calls. Use the following table to locate the available parameters. If this feature is enabled, users can switch between audio-only calls, and calls with audio and video. Users can make audio calls by default, and select a Voice/Video if they want to add video to the call. After a video call has ended, the phone switches back to audio-only.

**Voice and Video Toggle Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the audio/video toggle feature. | **features.cfg** > feature.audioVideoToggle.enabled |
| Allow the user to select the call mode to use when using SIP protocol only. | **video.cfg** > video.callMode.default |

With a monitor connected to a RealPresence Trio Visual+ paired with the RealPresence Trio 8800, you can show content during in-person meetings, video conference calls, and point-to-point video calls.

To show content you can use Polycom People + Content™ IP, Polycom RealPresence Desktop, or Polycom RealPresence Mobile applications.

## Configure Phone Security

After you set up Polycom devices on your network with the default configuration, users can place and answer calls. Polycom's Open SIP UC Software enables you to make custom configurations to optimize security settings. This section provides information on configuring phone security.

## SIP Instance Support

In environments where multiple phones are registered using the same address of record (AOR), the phones are identified by their IP address. However, firewalls set up in these environments can change the IP addresses regularly for security purposes. This feature provides support for SIP instance to identify individual phones instead of using IP addresses. This feature complies with RFC 3840.

Support for SIP instance is available on VVX 300 series, 400 series, 500 series, 600 series, and 1500 business media phones. The parameter reg.x.gruu provide a unique identification (a contact address) to a specific user agent (UA) instance. This helps to route the request to the UA instance and is required in cases in which the REFER request must be routed to the correct UA instance, for example, a call transfer.

**Configure SIP Instance Support**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify if the phone sends sip.instance in the REGISTER request. | reg.x.gruu |

## Set Local User and Administrator Passwords

The phone prompts you for a user or administrator password before you can access certain menu options. To set the administrator password, change the default administrator password on the phone user interface in the Settings menu. The default user password is 123 and the default administrative password is 456. Note that you can use an administrator password where a user password is required, but a user cannot access administrator settings with a user password. If you do not change the default administrative password, the phone displays a reminder message each time the phone reboots.

To remove settings made using the phone’s user interface, go to **Reset Local Configuration** menu on the phone.

If the phone requires the administrator password, you may be able to use the user password, but you are presented with limited menu options. If the phone prompts you for the user password, you may use the administrator password (you will see the same menus as the user). If you are registering Polycom phones with Microsoft Skype for Business Server, a message displays on the phone screen prompting you to change the default password.

The Web Configuration Utility is protected by the user and administrator password and displays different features and options depending on which password you use. The default user password is 123 and the default administrator password is 456. You should change the administrator password from the default value. You may want to change the user password for security reasons, see the following table for all parameters.

**Local User and Administrator Password Settings**

| **Parameter Function** | **template** > parameter |
|---|---|
| Set the minimum length for the administrator password. | **site.cfg** > sec.pwd.length.admin |
| Set the minimum length for the user password. | **site.cfg** > sec.pwd.length.user |
| Enable or disable masking of password characters as you type. | **site.cfg** > up.echoPasswordDigits |

**Local User and Administrator Password Settings (continued)**

| | |
|---|---|
| Set the phone’s local administrator password. | **device.cfg** > device.auth.localAdminPassword |
| Set the phone’s local user password. | **device.cfg** > device.auth.localUserPassword |

## Disable External Ports and Features

You can disable unused external phone ports and phone features to increase the security of devices in your deployment. You can disable the following ports and features:

- **Web Configuration Utility** A web browser-based interface that enables you to configure settings on the phone.
- **PC port** Acts as a pass-through switch for externally attached devices such as a personal computer or laptop. The PC port informs the main switch of any secondary (PC) port link status changes.
- **Aux port** This port is used to connect Polycom Expansion Modules to Polycom VVX phones. This port uses a synchronous peripheral cable interface (SPI) cable.
- **USB Port** Use this port to plug in devices for local call recording, the picture frame feature, or for the Polycom VVX Camera. VVX 500/501 and 600/601 business media phones have two USB ports. To enable or disable the phone USB port, disable the following three parameters:
  - ➢ `feature.callRecording.enabled=0`
  - ➢ `feature.pictureFrame.enabled=0`
  - ➢ `diags.pcap.enabled=0`

  To disable the second USB port on the VVX 500/501 and 600/601, use the parameter:
  - ➢ `video.enable=0`
- **Speakerphone** Disable the phone’s audio speakerphone.
- **Headset** Disable the phone’s headset jack.
- **Handset** Disable the phone’s handset.
- **Call forwarding** Disable the call forwarding feature.
- **Do Not Disturb** Disable the Do Not Disturb feature and soft key.
- **Push-to-Talk (PTT)** Disable the push-to-talk feature.
- **Bluetooth** Bluetooth capability is disabled by default. To enable or disable Bluetooth headset capability, use the configuration parameter `bluetooth.radioOn`. Note that if `up.headsetmodeenabled` is set to 0 and `bluetooth.radioOn` is set to 1, you can pair a Bluetooth device but no audio is available.
- **Autoanswer menu** Disable the Autoanswer menu on the phone interface.
- **Applications icon** Disable the Applications icon on the phone's Home screen.

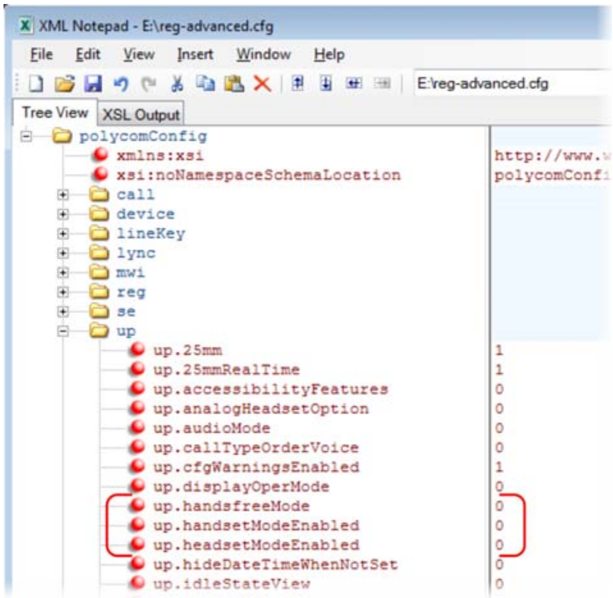
**Note: One port must be enabled to send and receive calls**
At least one audio port must be enabled to send and receive calls.

**Disable Unused Ports and Features**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable the PC port mode that sets the network speed over Ethernet. | **device.cfg** > device.net.etherModePC |
| Use or do not use all enabled device.xxx fields to set parameters. | **device.cfg** > device.set |
| Enable or disable the phone auxiliary port. | **device.cfg** > device.auxPort.enable |
| Enable or disable the complete httpd web client. | **site.cfg** > httpd.enabled |
| Enable or disable the Web Configuration Utility. | **site.cfg** > httpd.cfg.enabled |
| Enable or disable push-to talk mode. | **site.cfg** > ptt.pttMode.enable |
| Enable or disable the phone USB port for local call recording. | **features.cfg** > feature.callRecording.enabled |
| Enable or disable handsfree mode. | **reg-advanced.cfg** > up.handsfreeMode |
| Enable or disable the headset port. | **reg-advanced.cfg** > up.headsetModeEnabled |
| Enable or disable the handset port. | **reg-advanced.cfg** > up.handsetModeEnabled |
| Enable or disable call forwarding. | **features.cfg** > feature.forward.enable |
| Turn on or off display of the call forward icon on the phone Home screen. | **features.cfg** > homeScreen.forward.enable |
| Enable or disable Do Not Disturb (DND). | **features.cfg** > feature.doNotDisturb.enable |
| Enable or disable display of the DND icon on the phone’s Home screen. | **features.cfg** > homeScreen.doNotDisturb.enable |
| Enable or disable the DND soft key on the phone. | **features.cfg** > softkey.feature.doNotDisturb |
| Enable or disable the phone's Autoanswer menu. | **features.cfg** > call.autoAnswerMenu.enable |
| Enable or disable the Applicatios icon on the phone's Home screen. | **features.cfg** > homeScreen.application.enable |

## Example Configuration

This section shows you how to disable external ports and features. You must modify settings on BroadSoft server and for the phones in your deployment. The following example shows the headset, handset, and speakerphone ports disabled.



```
polycomConfig
    xmlns:xsi                          http://www.w
    xsi:noNamespaceSchemaLocation      polycomConfi
    call
    device
    lineKey
    lync
    mwi
    reg
    se
    up
        up.25mm                        1
        up.25mmRealTime                1
        up.accessibilityFeatures       0
        up.analogHeadsetOption         0
        up.audioMode                   0
        up.callTypeOrderVoice          0
        up.cfgWarningsEnabled          1
        up.displayOperMode             0
        up.handsfreeMode               0
        up.handsetModeEnabled          0
        up.headsetModeEnabled          0
        up.hideDateTimeWhenNotSet      0
        up.idleStateView               0
```

## Set Visual Security Classification

This feature enables all phones to display the security classification of an active call. The security classification of a call is determined by the lowest security classification among all participants connected to the call. For example, a Top Secret classification displays when all participants in a call have a Top Secret classification level. If User A is classified as Top Secret and User B has a lower classification level of Restricted, User A and B are connected to the call as Restricted.

**Caution: Security classifications**

Calls classification is determined by the lowest classification among all participants in the call. You can safely exchange information classified no higher than the call's security classification.

This feature is supported on Polycom VVX business media phones. To enable this feature, you must configure settings on the BroadSoft BroadWorks server v20 or higher and on the phones. If a phone has multiple registered lines, administrators can assign a different security classification to each line. The following table shows the parameters you can configure on your provisioning server.

An administrator can configure security classifications as names or strings and set the priority of each on the server in addition to the default security classification level Unclassified. The default security classification Unclassified displays until you set classifications on the server. When you establish a call to a phone not connected to this feature, your phone displays as Unclassified.

Note that phone users can modify their assigned security classification level to a value lower than their assigned level during a call, as shown in Modify Security Classification Level. When the call is over, the server resets your classification level to its original state. You cannot change security classification when the phone is in the idle state. You can view the security classification for your phone line by navigating to **Status > Lines.** If your phone has multiple registered lines, you can view the security classification level assigned to each line on the phone in idle state by going to **Home > Settings > Status > Lines > Line x**.

**Configure Visual Security Classification**

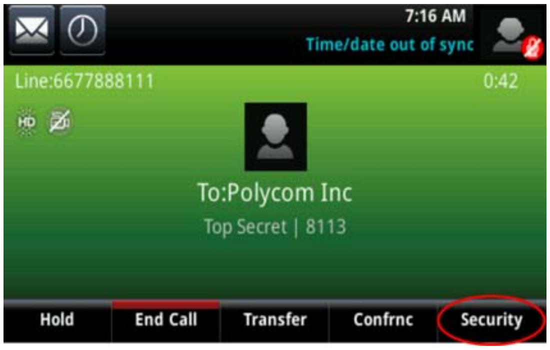
| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable Visual Security Classification for all lines on a phone. | **sip-interop.cfg** ><br>voIpProt.SIP.serverFeatureControl.securityClassification |
| Enable or disable Visual Security Classification for a specific phone line. | **reg-advanced.cfg** ><br>reg.x.serverFeatureControl.securityClassification |

## Modify Security Classification Level

Phone users have the option to modify their classification to a lower classification level at any point during an active call. For example, a user classified as Top Secret can override the classification level to a level lower than Top Secret and revert back to the Top Secret classification at any point during an active call.

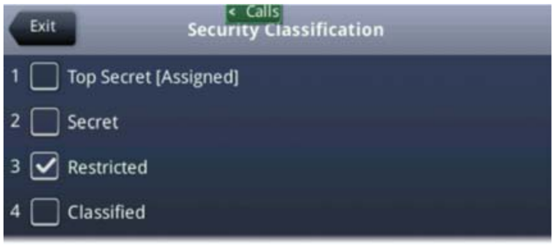
### To modify a security classification level:

1 Press the **Security** soft key that displays only during an active call.



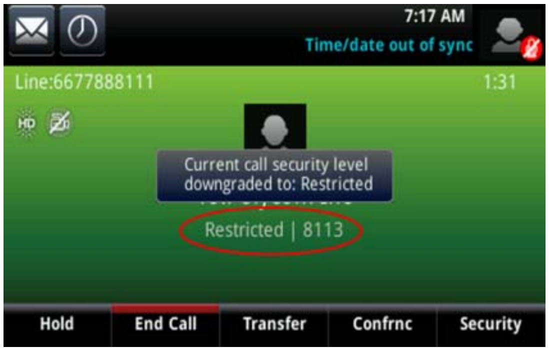
2 In the **Security Classification** screen, choose an available security level and press **Exit**.

In this example, the user changes classification level from **Top Secret** to **Restricted**.



1 □ Top Secret [Assigned]
2 □ Secret
3 ☑ Restricted
4 □ Classified

When a participant changes classification level and the change results in a lower classification level for the active call, a message displays to all participants indicating a change in the call's security classification, as illustrated in the following figure.



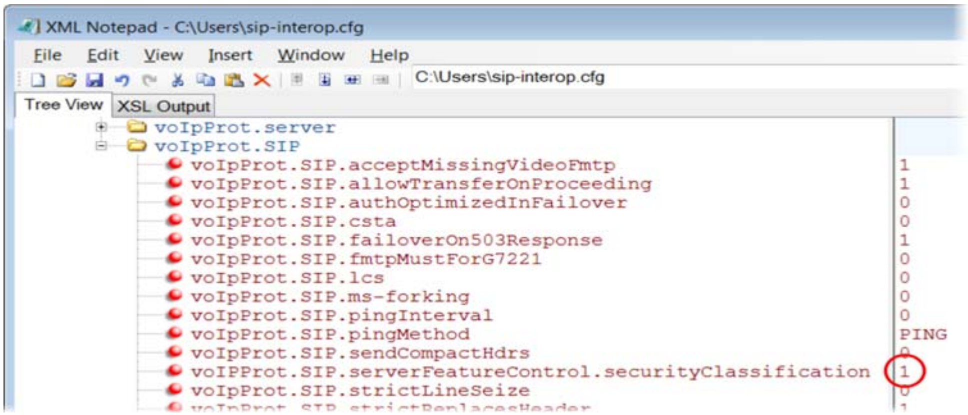
Current call security level downgraded to: Restricted

Restricted | 8113

## Example Configuration

You must make changes on the BroadSoft r20 server. On your provisioning server, enable the parameter voIpProt.SIP.serverFeatureControl.securityClassification as shown in the following figure.

**Enable security classification**



```
voIpProt.server
voIpProt.SIP
    voIpProt.SIP.acceptMissingVideoFmtp                        1
    voIpProt.SIP.allowTransferOnProceeding                     1
    voIpProt.SIP.authOptimizedInFailover                       0
    voIpProt.SIP.csta                                          0
    voIpProt.SIP.failoverOn503Response                         1
    voIpProt.SIP.fmtpMustForG7221                              0
    voIpProt.SIP.lcs                                           0
    voIpProt.SIP.ms-forking                                    0
    voIpProt.SIP.pingInterval                                  0
    voIpProt.SIP.pingMethod                                    PING
    voIpProt.SIP.sendCompactHdrs                               0
    voIPProt.SIP.serverFeatureControl.securityClassification  1
    voIpProt.SIP.strictLineSeize                               0
    voIpProt.SIP.strictReplacesHeader                          1
```

# Choose Incoming Signaling Validation

You can choose from three optional levels of security for validating incoming network signaling:

- Source IP address validation
- Digest authentication
- Source IP address validation and digest authentication

See the following table for the parameters that specify the validation type, method, and the events you want to validate.

**Incoming Signal Validation Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Specify what type of validation to perform. | **sip-interop.cfg** ><br>voIp.SIP.requestValidation.x.method |

**Incoming Signal Validation Parameters**

| | |
|---|---|
| Set the name of the method for which validation is applied. | **sip-interop.cfg** ><br>voIp.SIP.requestValidation.x.request |
| Determine which events within the Event header should be validated. | **sip-interop.cfg** ><br>voIp.SIP.requestValidation.x.request.y.event |

## Encrypt Configuration Files

You can encrypt configuration files, contact directories, and configuration override files can all be encrypted. Note that you cannot encrypt the master configuration file.

You can determine whether encrypted files are the same as unencrypted files and use the SDK to facilitate key generation. Use the following table to configure the parameters used to encrypt files.

**Configuration File Encryption Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify if configuration files uploaded from the phone to the provisioning server should be encrypted. | **site.cfg** > sec.encryption.upload.config |
| Specify if the contact directory is encrypted when it is uploaded from the phone to the provisioning server. | **site.cfg** > sec.encryption.upload.dir |
| Specify if the configuration overrides file should be encrypted when it is uploaded from the phone to the server. | **site.cfg** > sec.encryption.upload.overrides |
| Specify an encryption key so the phone can download encrypted files from the provisioning server. | **device.cfg** > device.sec.configEncryption.key |

## Use Digital Certificates

Polycom phones are installed with a Polycom-authenticated RSA certificate. You can use this certificate to create a secure connection between phone and server when initiating TLS communications over protocols such as HTTPS and SIP. You can download the Polycom Root CA from http://pki.polycom.com/pki. The certificate is set to expire on March 9, 2044.

**Web Info: Digital certificates on Polycom phones**

For details on installing digital credentials on VVX phones, see *Device Certificates on Polycom SoundPoint IP, SoundStation IP, and VVX Phones: Technical Bulletin 37148* at Polycom Engineering ADvisories and Technical Notifications.

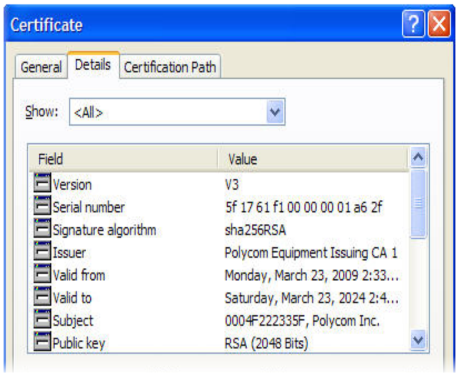
Polycom uses the X.509 standard, which defines what information can go into a certificate. An X.509 digital certificate is a digitally signed statement. All X.509 certificates have the following fields, in addition to the signature:

- **Version** This identifies which version of the X.509 standard applies to this certificate, which in turn affects what information can be specified in the certificate.
- **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 and indicates that you trust the entity that signed this certificate.
- **Validity Period** Each certificate is valid 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 that specifies to which public key cryptographic system this key belongs and any associated key parameters.

Polycom supports the use of Subject Alternative Names (SAN) with TLS security certificates. Polycom does not support the use of the asterisk (*) or wildcard characters in the Common Name field of a Certificate Authority's public certificate. If you want to enter multiple hostnames or IP addresses on the same certificate, use the SAN field.

The following is an example of a Polycom device certificate when viewed in a browser.



Show: <All>

| Field | Value |
|---|---|
| Version | V3 |
| Serial number | 5f 17 61 f1 00 00 00 01 a6 2f |
| Signature algorithm | sha256RSA |
| Issuer | Polycom Equipment Issuing CA 1 |
| Valid from | Monday, March 23, 2009 2:33... |
| Valid to | Saturday, March 23, 2024 2:4... |
| Subject | 0004F222335F, Polycom Inc. |
| Public key | RSA (2048 Bits) |

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, see Public Key Infrastructure (X.509) and RFC 2459: Internet X.509 Public Key Infrastructure.

**Web Info: Using custom device certificates with Polycom phones**
As of UC Software 4.0.0, you can install custom device certificates on your Polycom phones. These certificates are installed in the same way custom CA certificates are installed. See *Technical Bulletin 17877: Using Custom Certificates With Polycom Phones*.

### To determine if there is a device certificate on a Polycom phone:

1. Navigate to **Settings > Advanced > Admin Settings > TLS Security > Custom Device Certificates**.

   You can view the Polycom device certificate on the phone at **Settings > Status > Platform > Phone**.

2. Press the **Info** soft key to view the certificate.

   One of the following messages display:

- Device Certificate: Installed or Device Certificate: Factory Installed is displayed if the certificate is available in flash memory, all the certificate fields are valid (listed above), and the 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.

**Note: Device certificate shown as "self-signed"**

If your phone reports the device certificate as 'self-signed" rather than "Factory installed," you should return the equipment to receive a replacement.

## Generate a Certificate Signing Request

You may need a certificate to perform a number of tasks, for example, multiple TLS authentication. By default, the phone requests a 512-bit certificate. If you require a stronger certificate, use OpenSSL or another certificate signing request utility.

### To obtain a certificate you need to:

1. Request a certificate from a Certificate Authority (CA) by creating a certificate signing request (CSR).
2. Forward the CSR to a CA to create a certificate. If your organization doesn't have its own CA, you need to forward the CSR to a company like Symantec.

   If successful, the CA sends back a certificate that has been digitally signed with their private key.

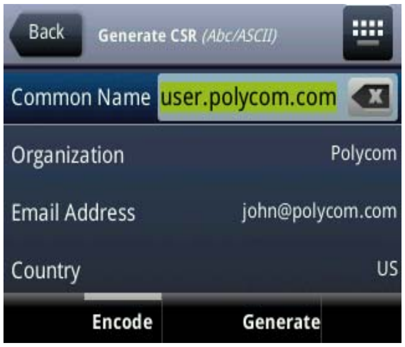
After you receive the certificate, you can download it to the phone in the following ways:

- Using a configuration file
- Through the phone's user interface
- Through the Web Configurable Utility

### To generate a certificate signing request on a Polycom phone:

1. Navigate to **Settings > Advanced > Admin Settings > Generate CSR**.
2. When prompted, enter the administrative password and press the **Enter** soft key. The default administrative password is **456**.
3. From the **Generate CSR Screen**, fill in the **Common Name** field - the Organization, Email Address, Country, and State fields are optional.

The following figure shows the Generate CSR screen on a VVX 500/501 phone.



Back Generate CSR *(Abc/ASCII)*

Common Name: user.polycom.com

Organization: Polycom

Email Address: john@polycom.com

Country: US

4 Press **Generate**.

A message “CSR generation completed” displays on the phone’s screen. The MAC.csr (certificate request) and MAC-private.pem (private key) are uploaded to the phone’s provisioning server.

# Configure TLS Profiles

The Transport Layer Security (TLS) profiles describe a collection of custom CA and device certificates installed on the Polycom phones and the features where these certificates are used for authentication.

Your phone can trust certificates issued by widely recognized certificate authorities when trying to establish a connection to a provisioning server for application provisioning. There are a number of parameters you can use to configure TLS profiles listed in the tableTLS Platform Profile and TLS Application Profile Parameters. For the complete list of trusted Certificate Authorities, see the section Trusted Certificate Authority List.

Custom CA and device certificates can be added to the phone and set up to be used by different features. For example, the phone's factory-installed or custom device certificate could be used for authentication when phone provisioning is performed by an HTTPS server. A custom CA certificate could also be used when accessing content through the microbrowser or browser.

After you install certificates on the phone, you can to determine which TLS platform profiles or TLS application profiles use these certificates. By default, TLS Platform Profile 1 uses every CA certificate and the default device certificate. Also, each TLS application uses TLS Platform Profile 1 as the default profile. You can quickly apply a CA certificate to all TLS applications by installing it on the phone and keeping the default TLS profile and default TLS application values.

Lastly, you must choose which TLS platform profile or application profile to use for each TLS application. The profiles can be used for phone provisioning, with the applications running on the microbrowser and browser, and for 802.1X, LDAP, and SIP authentication. Some applications, such as Syslog, can only use a TLS platform profile, not a TLS application profile. See <TLS/> for the list of applications.

For more information on device (or digital) certificates installed on the phones at the factory, see the section Use Digital Certificates.

**Web Info: Using custom certificates**

For more information on using custom certificates, see *Technical Bulletin 17877: Using Custom Certificates With Polycom Phones*.

The following table shows parameters for TLS Platform Profile 1. To configure TLS Platform Profile 2, use a 2 at the end of the parameter instead of a 1. For example, set `device.sec.TLS.profile.caCertList2` instead of `.caCertList1`.

**TLS Platform Profile and TLS Application Profile Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| **TLS Platform Profile Parameters** (use 2 at the end of each parameter (instead of 1) to set up platform profile 2) | |
| Specify which CA certificates to use. | **device.cfg** > device.sec.TLS.profile.caCertList1 |
| Specify the cipher suite. | **device.cfg** > device.sec.TLS.profile.cipherSuite1 |
| Select the default cipher suite or a custom cipher suite. | **device.cfg** > device.sec.TLS.profile.cipherSuiteDefault1 |
| Specify a custom certificate. | **device.cfg** > device.sec.TLS.customCaCert1 |
| Specify which device certificates to use. | **device.cfg** > device.sec.TLS.profile.deviceCert1 |

**TLS Platform Profile and TLS Application Profile Parameters (continued)**

| **TLS Application Profile Parameters** | |
|---|---|
| Specify which CA certificates to use. | **site.cfg** > sec.TLS.profile.x.caCert.* |
| Specify the cipher suite. | **site.cfg** > sec.TLS.profile.x.cipherSuite |
| Select the default cipher suite or a custom cipher suite. | **site.cfg** > sec.TLS.profile.x.cipherSuiteDefault |
| Specify a custom certificate. | **site.cfg** > sec.TLS.customCaCert.x |
| Specify which device certificates to use. | **site.cfg** > sec.TLS.profile.x.deviceCert |
| Specify the custom device key. | **site.cfg** > sec.TLS.customDeviceKey.x |

.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

This section includes the following topics:

- Download Certificates to a Polycom Phone
- Set TLS Profiles

## Download Certificates to a Polycom Phone

You can download certificates to a Polycom phone by specifying a URL where the certificate is currently stored. You can install up to eight CA certificates and eight device certificates on the phone. You can refresh certificates when they expire or are revoked. You can delete any CA certificate or device certificate that you install.

**Note: Maximum size for certificates**

For VVX 1500 phones, the maximum certificate size on Platform CA1 is 1536KB and 4KB for Platform CA2.

### To download a certificate to a Polycom phone:

1 Navigate to **Menu > Settings > Advanced > Administrative Settings > TLS Security** and select **Custom CA Certificates** or **Custom Device Certificates**.

2 When prompted, enter the administrative password and press the **Enter** soft key. The default administrative password is **456**.

3 Select the **Install** soft key.

4 Enter the URL where the certificate is stored.

For example, http://bootserver1.vancouver.polycom.com/ca.crt

5 Select the **Enter** soft key.

The certificate is downloaded. The certificate’s MD5 fingerprint displays to verify that the correct certificate is to be installed.

6 Select the **Accept** soft key.

The certificate is installed successfully.

The appropriate certificate menu displays the certificate’s common name.

## Set TLS Profiles

By default, all Polycom-installed profiles are associated with the default cipher suite and use trusted and widely recognized CA certificates for authentication. Use the following table to refer to parameters. You can change the cipher suite, CA certificates, and device certificates for the two platform profiles and the six application profiles. You can then map profiles directly to the features that use certificates.

**TLS Profile Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Specify the TLS profile to use for each application (802.1X and Provisioning). | **device.cfg** > device.sec.TLS.profileSelection.* |
| Specify the TLS profile to use for each application (other applications). | **device.cfg** > sec.TLS.profileSelection.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

## Support Mutual TLS Authentication

Mutual Transport Layer Security (MTLS) 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 or a custom device certificate installed on it. For more information, refer to the section Use Digital Certificates.

Prior to SIP 3.2, and in cases where the phones do not have device certificates, the phone authenticates to the server as part of the TLS authentication, but the server cannot cryptographically authenticate the phone. This is sometimes 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 requests 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 device 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
- 802.1X authentication, if the server is configured for mutual authentication (optional for EAP-TLS)

**Note: You cannot modify the factory-installed certificate or private key**

Users cannot modify or update the digital certificate or the associated private key installed on the phone during manufacturing. Users can install a custom device certificate to be used instead of, or in addition to, the factory-installed certificate.

You can download the Polycom Root CA from http://pki.polycom.com/pki. 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 or your organization's CA must be downloaded onto the HTTPS server.

The following operating system/web server combinations have been tested and verified:

- Microsoft Internet Information Services 6.0 on Microsoft Windows Server 2003
- Apache v1.3 on Microsoft Windows XP

**Web Info: Provisioning using Microsoft Internet Information Services**

For more information on using Mutual TLS with Microsoft Internet Information Services (IIS) 6.0, see *Mutual Transport Layer Security Provisioning Using Microsoft Internet Information Services 6.0: Technical Bulletin 52609* at Polycom Engineering Advisories and Technical Notifications.

## Configurable TLS Cipher Suites

The phone administrator can control which cipher suites to offer/accept during TLS session negotiation. The phone supports the cipher suites listed in the following table and you can use the parameters listed in the table Configurable TLS Cipher Suites to configure TLS Cipher Suites. The 'Null Cipher' listed in the following table is a special case option which does not encrypt the signaling traffic, and is useful for troubleshooting purposes.

**TLS Cipher Suites**

| Cipher | Cipher Suite |
|---|---|
| ADH | ADH-RC4-MD5, ADH-DES-CBC-SHA, ADH-DES-CBC3-SHA, ADH-AES128-SHA, ADH-AES256-SHA |
| AES128 | AES128-SHA |
| AES256 | AES256-SHA |
| DES | DES-CBC-SHA, DES-CBC3-SHA |
| DHE | DHE-DSS-AES128-SHA, DHE-DSS-AES256-SHA, DHE-RSA-AES128-SHA, DHE-RSA-AES256-SHA |
| EXP | EXP-RC4-MD5, EXP-DES-CBC-SH, EXP-EDH-DSS-DES-CBC-SHA, EXP-DES-CBC-SHA, EXP-ADH-RC4-MD5, EXP-ADH-DES-CBC-SHA, EXP-EDH-RSA-DES-CBC-SHA |
| EDH | EDH-RSA-DES-CBC-SHA, EDH-DSS-DES-CBC3-SHA, EDH-DSS-CBC-SHA |
| NULL | NULL-MD5, NULL-SHA |
| RC4 | RC4-MD5, RC4-SHA |

**Tip: Changes to the default TLS cipher suites in UC Software 4.0.0**
Changes have been made to the default TLS cipher suites in UC Software 4.0.0. If you created customized TLS cipher suites in a previous release of the UC Software, your changes are lost unless you backup the configuration files.

**Configurable TLS Cipher Suites**

| Parameter Function | **template** > parameter |
|---|---|
| Specify the global cipher list. | **site.cfg** > sec.TLS.cipherList |
| Specify the cipher list for a specific TLS Platform Profile or TLS Application Profile. | **site.cfg** > sec.TLS.<application>.cipherList |

## Secure Real-Time Transport Protocol

Secure Real-Time Transport Protocol (SRTP) provides a way of encrypting audio stream(s) to avoid interception and eavesdropping on phone calls. As described in RFC 3711, both RTP and RTCP signaling may be encrypted using an advanced encryption standard algorithm. The parameters used to configure SRTP are shown in the following table . When this feature is enabled, phones negotiate with the other end-point the type of encryption and authentication to use for the session. This negotiation process is compliant with RFC4568—Session Description Protocol (SDP) Security Descriptions for Media Streams.

**Web Info: SRTP RFC resources**
For more information on SRTP, see RFC 3711. For the procedure describing how two phones set up SRTP for a call, see RFC 4568.

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 sounds like noise and cannot be understood. Only the receiver knows the key to restore the data.

A number of session parameters have been added to enable you to turn off authentication and encryption for RTP and RTCP streams. This is done mainly to reduce the phone’s processor usage.

If the call is completely secure (RTP authentication and encryption and RTCP authentication and RTCP encryption are enabled), a padlock symbol displays in the last frame of the connected context animation (two arrows moving towards each other).

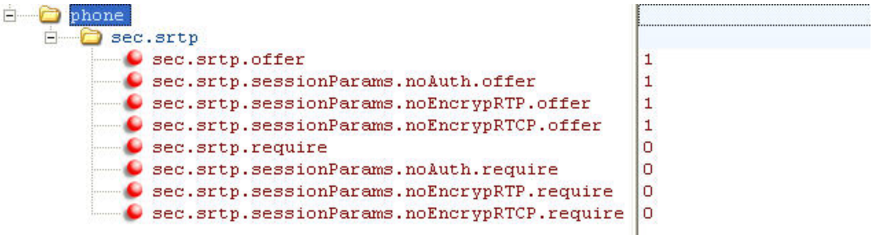
**Secure Real Time Transport Protocol Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable SRTP. | **sip-interop.cfg** > sec.srtp.enable |
| Include secure media in SDP of SIP INVITE. | **sip-interop.cfg** > sec.srtp.offer |
| Include crypto in offered SDP. | **sip-interop.cfg** > sec.srtp.offer.* |
| Secure media stream required in all SIP INVITEs. | **sip-interop.cfg** > sec.srtp.require |

**Secure Real Time Transport Protocol Parameters (continued)**

| | |
|---|---|
| Check tag in crypto parameter in SDP. | **sip-interop.cfg** > sec.srtp.requireMatchingTag |
| Specify if the phone offers and/or requires: RTP encryption, RTP authentication, and RTCP encryption. | **sip-interop.cfg** > sec.srtp.sessionParams.* |
| .* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information. | |

In example 1, the **srtp_1.cfg** configuration file is shown below:



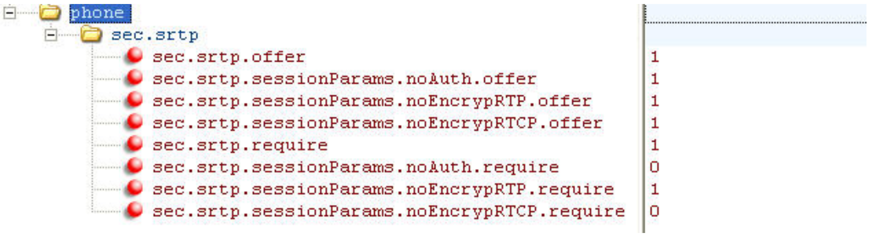
```
phone
  sec.srtp
    sec.srtp.offer                               1
    sec.srtp.sessionParams.noAuth.offer          1
    sec.srtp.sessionParams.noEncrypRTP.offer     1
    sec.srtp.sessionParams.noEncrypRTCP.offer    1
    sec.srtp.require                             0
    sec.srtp.sessionParams.noAuth.require        0
    sec.srtp.sessionParams.noEncrypRTP.require   0
    sec.srtp.sessionParams.noEncrypRTCP.require  0
```

This results in an offer (SIP INVITE with SDP) with 8 crypto attributes with the following session parameters:

```
<no session parameters> UNENCRYPTED_SRTCP UNENCRYPTED_SRTP UNAUTHENTICATED_SRTP
UNAUTHENTICATED_SRTP,UNENCRYPTED_SRTCP UNENCRYPTED_SRTP,UNENCRYPTED_SRTCP
UNAUTHENTICATED_SRTP,UNENCRYPTED_SRTP
UNAUTHENTICATED_SRTP,UNENCRYPTED_SRTP,UNENCRYPTED_SRTCP
```

In the above example, the crypto attributes are ordered ‘most secure’ to ‘least secure’ (more security turned off). The phone receiving this call should chose the most secure crypto it can support based on the SRTP require settings in sip.cfg and reply with it in the SDP of a 200 OK SIP message.

In this example 2, the **srtp_2.cfg** configuration file is shown below:



```
phone
  sec.srtp
    sec.srtp.offer                               1
    sec.srtp.sessionParams.noAuth.offer          1
    sec.srtp.sessionParams.noEncrypRTP.offer     1
    sec.srtp.sessionParams.noEncrypRTCP.offer    1
    sec.srtp.require                             1
    sec.srtp.sessionParams.noAuth.require        0
    sec.srtp.sessionParams.noEncrypRTP.require   1
    sec.srtp.sessionParams.noEncrypRTCP.require  0
```

This results in an offer (SIP INVITE with SDP) with 4 crypto attributes with the following session parameters:

```
UNENCRYPTED_SRTP UNENCRYPTED_SRTP,UNENCRYPTED_SRTCP
UNAUTHENTICATED_SRTP,UNENCRYPTED_SRTP
UNAUTHENTICATED_SRTP,UNENCRYPTED_SRTP,UNENCRYPTED_SRTCP
```

In the above example, every crypto includes the UNENCRYPTED_SRTP session parameter because it is required.

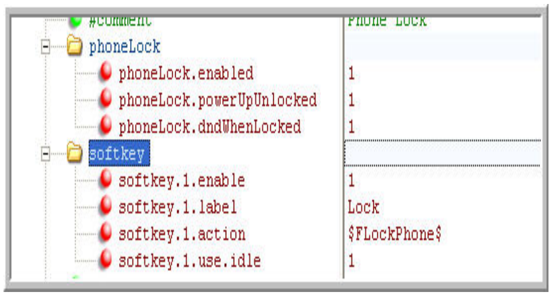
If nothing compatible is offered based on the receiving phone’s STRP ‘require’ settings, then the call is rejected or dropped.

# Lock the Phone

As of Polycom UC Software 3.3.0, users can lock their phones, and prevent access to the menu or key presses, by pressing the Lock soft key or through the phone menu.

**Note: Displaying the lock soft key on your phone**
You need to enable the enhanced feature key (EFK) feature if you want your phone to display a Lock soft key. See feature.enhancedFeatureKeys.enabled.

The following configuration file snippet shows how to display the Lock soft key.



```
#comment                      Phone Lock
phoneLock
    phoneLock.enabled          1
    phoneLock.powerUpUnlocked  1
    phoneLock.dndWhenLocked    1
softkey
    softkey.1.enable           1
    softkey.1.label            Lock
    softkey.1.action           $FLockPhone$
    softkey.1.use.idle         1
```

After the phone is locked, all user features and access to menus are disabled. The messages “The phone is locked.” and “Authorized calls only.” display on the screen. Incoming calls to the phone may receive a Do Not Disturb message. You can specify the authorized numbers to which users can place calls.

Using the New Call soft key, users can place calls using up to five authorized numbers including the emergency number. If the user places a call —using the keypad— to a number that matches an authorized number, the call proceeds. This is to ensure that certain numbers such as emergency numbers can be placed from the phone.

To unlock the phone, the user presses the Unlock soft key and enters their password; if it is entered correctly, the phone returns to its normal idle state.

In case the user forgets their password, the system administrator can unlock their phone either by entering the administrator password or by disabling (and re-enabling) the phone lock feature. The latter method facilitates remote unlocking and avoids disclosing the administrator password to the user. See the following table for the parameters that configure the phone lock feature.

**Note: Shared lines on locked phones**
If a locked phone has a registered shared line, calls to the shared line displays on the locked phone and the phone’s user can answer the call.

**Phone Lock Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable enhanced feature keys. | **features.cfg** > feature.enhancedFeatureKeys.enabled |
| Enable or disable phone lock. | **features.cfg** > phoneLock.enabled |
| Specify an authorized contact (description and value) who can be called while the phone is locked. | **features.cfg** > phoneLock.authorized.* |

**Phone Lock Parameters (continued)**

| | |
|---|---|
| Specify the scenarios when phone lock should be enabled. | **features.cfg** > phoneLock.* |

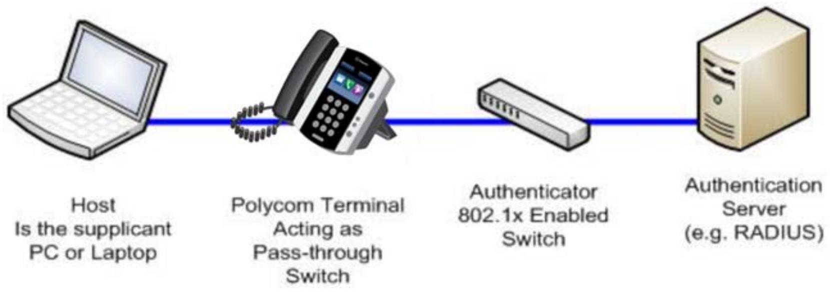
.* indicates grouped parameters. See the section Example Two: Configuring Grouped Parameters for more information.

## Secondary Port Link Status Report

Devices equipped with a secondary PC port can act as a pass-through switch for externally attached devices. The phone informs the network switch (authenticator) of any secondary PC port link status changes.

As of Polycom UC Software 3.3.0, Polycom phones include this feature.

**A Polycom terminal acting as a pass–through switch**



If you want to configure this feature, see the table Secondary Port Link Status Report Parameters for the parameters you need to set. Polycom devices detect an externally connected host connection/disconnection, informing the authenticator switch to initiate the authentication process or drop an existing authentication. This feature ensures that the port authenticated by the externally attached device switches to unauthenticated upon device disconnection so that other unauthorized devices cannot use it. It also ensures that the externally attached device can move to another port in the network and start a new authentication process. This feature extends Cisco Discovery Protocol (CDP) to include a Second Port Status Type, Length, Value (TLV) that informs an authenticator switch of the status of devices connected to a device's secondary PC port.

To reduce the frequency of CDP packets, the phone does not send link up status CDP packets before a certain time period. The phone immediately sends all link-down indication to ensure that the port security is not compromised. You can configure the required elapse time—sleep time—between two CDP UPs dispatching (see sec.hostmovedetect.cdp.sleepTime1).

If the externally attached device (the host) supports 802.1X authentication, then the device can send an EAPOL-Logoff on behalf of the device after it is disconnected from the secondary PC port. This informs the authenticator switch to drop the authentication on the port corresponding with the previously attached device.

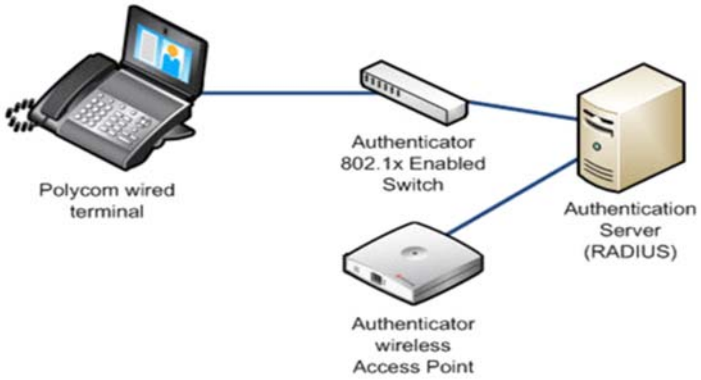
**Secondary Port Link Status Report Parameters**

| Parameter Function | **template** > parameter |
|---|---|
| Enable or disable EAPOL logoff. | **site.cfg** > sec.dot1x.eapollogoff.enabled |
| Specify if the LAN port link should be reset or not. | **site.cfg** > sec.dot1x.eapollogoff.lanlinkreset |
| Specify the phone should indicate to a host that it has been connected or disconnected to the host's secondary (PC) port. | **site.cfg** > sec.hostmovedetect.cdp.enabled |
| Set the time interval between link-up and link-down reporting. | **site.cfg** > sec.hostmovedetect.cdp.sleepTime |

## Support 802.1X Authentication

IEEE 802.1X is a Port-Based Network Access Control (PNAC). It provides an authentication mechanism to devices trying to attach to a local area network (LAN). IEEE 802.1X is based on the Extensible Authentication Protocol (EAP). As of Polycom UC Software 3.3.0, Polycom phones support standard IEEE 802.1X authentication. The following figure shows a typical 802.1X network configuration with wired Polycom phones.

**A typical 802.1X network configuration**


Polycom wired terminal
Authenticator 802.1x Enabled Switch
Authentication Server (RADIUS)
Authenticator wireless Access Point

Polycom phones support the following EAP authentication methods:

- EAP-TLS (requires Device and CA certificates)
- EAP-PEAPv0/MSCHAPv2 (requires CA certificates)
- EAP-PEAPv0/GTC (requires CA certificates)
- EAP-TTLS/MSCHAPv2 (requires CA certificates)
- EAP-TTLS/GTC (requires CA certificates)
- EAP-FAST (optional Protected Access Credential (PAC) file, if not using in-band provisioning)
- EAP-MD5

To set up an EAP method that requires a device or CA certificate, you need to configure TLS Platform Profile 1 or TLS Platform Profile 2 to use with 802.1X. You can use the parameters in the following table S to configure 802.1X Authentication. For more information see Configure TLS Profiles.

**Web Info: EAP authentication protocol**
For more information, see RFC 3748: Extensible Authentication Protocol.

**Set 802.1X Authentication Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the 802.1X feature. | **device.cfg** > device.net.dot1x.enabled |
| Specify the identity (username) for authentication. | **device.cfg** > device.net.dot1x.identity |
| Specify the 802.1X EAP method. | **device.cfg** > device.net.dot1x.method |
| Specify the password for authentication. | **device.cfg** > device.net.dot1x.password |
| To enable EAP In-Band Provisioning for EAP-FAST. | **device.cfg** > device.net.dot1x.eapFastInBandProv |
| Specify a PAC file for EAP-FAST (optional). | **device.cfg** > device.pacfile.data |
| Specify the optional password for the EAP-FAST PAC file. | **device.cfg** > device.pacfile.password |

## Set User Profiles

Parameters listed in the table User Profile Parameters enable users to access their personal phone settings from any phone in the organization. This means that users can access their contact directory and speed dials, as well as other phone settings, even if they temporarily change work areas. This feature is particularly useful for remote and mobile workers who do not have a dedicated work space and conduct their business in more than one location. The user profile feature is also useful if an office has a common conference phone from which multiple users need to access their personal settings.

If you set up the user profile feature, a user can log in to a phone by entering their user ID and password. The default password is **123**. If the user profile feature is set up on your company’s phones, users can:

- Log in to a phone to access their personal phone settings.
- Log out of a phone after they finish using it.
- Place a call to an authorized number from a phone that is in the logged out state.
- Change their user password.

If a user changes any settings while logged in to a phone, the settings save and display the next time the user logs in to a phone. When a user logs out, the user’s personal phone settings are no longer displayed.

**Tip: Calling authorized and emergency numbers from shared devices**
You can configure the phones so that anyone can call authorized and emergency numbers when not logged in to a phone. For more information, see dialplan.routing.emergency.outboundIdentity.

When you set up the user profile feature, you must decide whether to require users always to log in to a phone or not. If you set up the user profile feature as enabled, and users are not required to log in, users have the option to use the phone as is—without access to their personal settings—or they can log in to display their personal settings. You can also specify if a user is logged out of the phone when the phone restarts or reboots, or if they remain logged in.

You can choose to define default credentials for the phone (see the section Create a Phone Configuration File). If you specify a default user ID and password, the phone automatically logs itself in each time an actual user logs out or the phone restarts or reboots. When the phone logs itself in using the default login credentials, a default phone profile displays, and users retain the option to log in and view their personal settings.

To set up the user profile feature, you need to perform the following procedures on the provisioning server:

- Create a phone configuration file, or update an existing file, to enable the feature's settings.
- Create a user configuration file in the format **<user>.cfg** to specify the user's password and registration, and other user-specific settings that you want to define.

**Tip: Resetting a user's password**
You can reset a user's password by removing the password parameter from the override file. This causes the phone to use the default password in the <user>.cfg file.

After you complete these procedures, update the phone's configuration to affect your changes.

**User Profile Parameters**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable the user profile feature. | **site.cfg** > prov.login.enabled |
| Specify the amount of time before a non-default user is logged out. | **site.cfg** > prov.login.automaticLogout |
| Specify the default password for the default user. | **site.cfg** > prov.login.defaultPassword |
| Specify if the phone can have users other than the default user. | **site.cfg** > prov.login.defaultOnly |
| Specify the name of the default user. | **site.cfg** > prov.login.defaultUser |
| Specify the password used to validate the user login. | **site.cfg** > prov.login.localPassword |
| Specify if a user should remain logged in after the handset reboots. | **site.cfg** > prov.login.persistent |
| Specify if a user must log in while the feature is enabled. | **site.cfg** > prov.login.required |

## Create a Phone Configuration File

Create a phone configuration file for the user login feature, and then add and set the attributes for the feature. Or, if you already have a phone configuration file, update the file to include the user login parameters you want to change.

**Tip: Creating a default user password for all users**
Polycom recommends that you create a single default user password for all users.

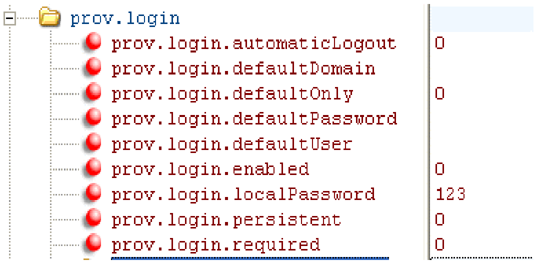
### To define the feature's settings:

1. Create a **site.cfg** file for the phone and place it on the provisioning server.

   You can base this file on the sample configuration template that is in your software package. To find the file, navigate to **<provisioning server location>/Config/site.cfg**.

2 In site.cfg, open the **<prov.login/>** attribute, and then add and set values for the user login attributes.

The following example is an example site.cfg file. Your file contains different values, depending on how you want the feature to work.



```
prov.login
    prov.login.automaticLogout      0
    prov.login.defaultDomain
    prov.login.defaultOnly          0
    prov.login.defaultPassword
    prov.login.defaultUser
    prov.login.enabled              0
    prov.login.localPassword        123
    prov.login.persistent           0
    prov.login.required             0
```

## Create a User Configuration File

Create a configuration file for each user that you want to be able to log in to the phone. The name of the file specifies the user’s login ID. In the file, specify any user-specific settings that you want to define for the user.

**Tip: Converting a phone-based deployment to a user-based deployment**

To convert a phone-based deployment to a user-based deployment, copy the **<MACaddress>-phone.cfg** file to **<user>-phone.cfg** and copy **phoneConfig<MACaddress>.cfg** to **<user>.cfg**.

### To create a user configuration file:

1 On the provisioning server, create a user configuration file for each user to log in to the phone. The name of the file is the user’s ID to log in to the phone. For example, if the user’s login ID is **user100**, the name of the user’s configuration file is **user100.cfg**.

2 In each **<user>.cfg** file, you can add and set values for the user’s login password (optional).

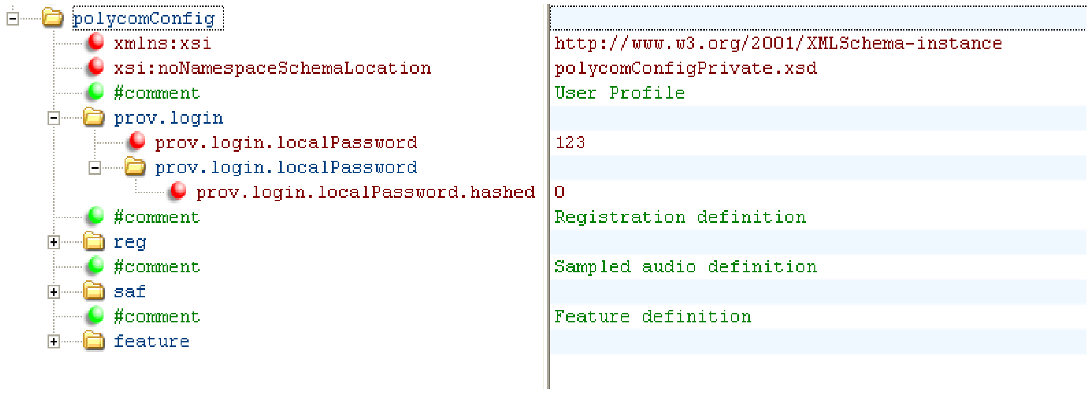
3 Add and set values for any user-specific parameters, such as:

- Registration details (for example, the number of lines the profile displays and line labels).
- Feature settings (for example, microbrowser settings).

**Caution: Adding user-specific parameters**

If you add optional user-specific parameters to <user>.cfg, add only those parameters that will not cause the phone to restart or reboot when the parameter is updated. For information on which parameters cause the phone to restart or reboot, see the reference section Configuration Parameters.

The following is a sample user configuration file.



| | |
|---|---|
| polycomConfig | |
| xmlns:xsi | http://www.w3.org/2001/XMLSchema-instance |
| xsi:noNamespaceSchemaLocation | polycomConfigPrivate.xsd |
| #comment | User Profile |
| prov.login | |
| prov.login.localPassword | 123 |
| prov.login.localPassword | |
| prov.login.localPassword.hashed | 0 |
| #comment | Registration definition |
| reg | |
| #comment | Sampled audio definition |
| saf | |
| #comment | Feature definition |
| feature | |

If a user updates their password or other user-specific settings using the Main Menu on the phone, the updates are stored in **<user>-phone.cfg**, not **<MACaddress>-phone.cfg**.

If a user updates their contact directory while logged in to a phone, the updates are stored in **<user>-directory.xml**. Directory updates display each time the user logs in to a phone. For certain phones (for example, the VVX 1500 phone), an up-to-date call lists history is defined in **<user>-calls.xml**. This list is retained each time the user logs in to their phone. Configuration parameter precedence (from first to last) for a phone that has the user profile feature enabled is:

- <user>-phone.cfg
- Web Configuration Utility (through a browser)
- Configuration files listed in the master configuration file (including <user>.cfg)
- Default values

## Configure I-Frames

When video streams initialize, devices transmit video packets called I-frames (reference frames) that contain information to display a complete picture. Subsequent video packets, known as P-frames, are smaller and not as complete to consume less bandwidth. Due to packet loss, jitter, or corruption, devices occasionally need to make multiple requests for a complete I-frame in order to reset the full frame, after which devices can revert to P-frame updates.

You can set parameters to control an I-frame request. The following table indicates parameter dependencies and messaging behavior when setting an I-frame request method.

**I-Frame Parameter Dependencies**

| video.<br>forceRtcpVideoCodecControl | video.<br>dynamicControlMethod | voIpProt.<br>SDP.offer.rtcpVideoCodecControl | Behavior when requesting video I-frame updates |
|---|---|---|---|
| **0** | **0 (n/a)** | **0** | Only SIP INFO messages are sent. No RTCP-FB is offered in SDP. |
| **0** | **1 (n/a)** | **0** | Only SIP INFO messages are sent. No RTCP-FB is offered in SDP. |
| **0** | **0 (n/a)** | **1** | RTCP-FB is offered in SDP. If SDP responses do not contain the required RTCP-FB attribute, then only SIP INFO requests are used. |
| **0** | **1 (n/a)** | **1** | RTCP-FB is offered in SDP. If SDP responses do not contain the required RTCP-FB attribute, then only SIP INFO requests are used. |
| **1** | **0** | **0** | The SDP attribute a=rtcp-fb is not included in SDP offers. Both RTCP-FB and SIP INFO messages are attempted. |
| **1** | **1** | **0** | The SDP attribute a=rtcp-fb is not included in SDP offers. Both RTCP-FB and SIP INFO messages are attempted. If no RTCP-FB messages are received, only SIP INFO messages are sent. If no response is received for SIP INFO messages then, again, both RTCP-FB and SIP INFO messages are attempted. |

**I-Frame Parameter Dependencies (continued)**

| video.<br>forceRtcpVideoCodecControl | video.<br>dynamicControlMethod | voIpProt.<br>SDP.offer.rtcpVideoCodecControl | Behavior when requesting video I-frame updates |
|---|---|---|---|
| **1** | **0** | **1** | RTCP-FB is offered in SDP. Even if the SDP response does not include an accepted a=rtcp-fb attribute both RTCP-FB and SIP INFO messages are sent. |
| **1** | **1** | **1** | RTCP-FB is offered in SDP. Even if the SDP response does not include an accepted a=rtcp-fb attribute both RTCP-FB and SIP INFO messages are sent initially. If no RTCP-FB response is received, only SIP INFO messages are sent afterwards. |

# Troubleshoot Your Polycom Phones

This section describes tools and techniques for troubleshooting Polycom phones running Polycom UC Software. The phone can provide feedback in the form of on-screen error messages, status indicators, and log files for troubleshooting issues.

This section includes the following troubleshooting topics:

- Understand Error Message Types
- Manage the Phone's Memory Resources
- Test Phone Hardware
- Upload a Phone's Configuration
- Perform Network Diagnostics
- Ports Used on Polycom Phones

This section also addresses phone issues, likely causes, and corrective actions. Issues are grouped as follows:

- Power and Startup Issues
- Screen and System Access Issues
- Calling Issues
- Display Issues
- Audio Issues
- Licensed Feature Issues
- Upgrade Issues

Review the latest UC Software Release Notes on Polycom UC Software Support Center for known problems and possible workarounds. If a problem is not listed in this section or in the latest Release Notes, contact your Certified Polycom Reseller for support.

## Understand Error Message Types

Several types of errors can occur while the phone is booting. If an error occurs, the phone informs you by displaying an error message. Errors can affect how the phone boots up. If the error is fatal, the phone is not able to boot until the error is resolved. If the error is recoverable, the phone continues to boot but the phone's configuration may change.

## Updater Error Messages

Most of the following errors are logged to the phone's boot log. However, if you are having trouble connecting to the provisioning server, the phone is not likely to upload the boot log.

### 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 set to a VLAN that is different from the DHCP server. Check the DHCP configuration.

### Application <file name> is not compatible with this phone!

When the Updater displays the error *The application is not compatible*, an application file was downloaded from the provisioning server but cannot be installed on this phone. You can usually resolve this issue by finding a software image that is compatible with the hardware or the BootROM and installing it on the provisioning server. Be aware that there are various different hardware and software dependencies.

### Could not contact boot server using existing configuration

The phone could not contact the provisioning server, but the causes may be numerous. Possible causes include:

- cabling issues
- DHCP configuration
- a problem with the provisioning server

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!

This message indicates that the phone has no application stored in device settings, that the phone could not download an application, and that the phone cannot boot. To resolve this issue, you must download compatible Polycom UC Software to the phone using one of the supported provisioning protocols. You must resolve the issue of connecting the phone to the provisioning server and provide a compatible software image on the provisioning server. This error is fatal, but recoverable.

## Read Polycom UC Software Error Messages

The warning notification feature, added in UC Software 4.0.0, provides users a visual indication that one or more error conditions exist. When the warning notification displays, users see:

- An informative message when the warning is first detected
- An icon in the status bar on the idle display, as shown next
    - ➢ On VVX 400 series, 500 series, 600 series, and 1500 phones (VVX 300 series grayscale) ⚠

You can view the list of warnings on your phone. Access to the Warnings menu varies by phone model:

- ➢ **VVX 1500** Menu > Status > Diagnostics > Warnings
- ➢ **VVX 300/301, 310/311, 400/401, 410/411, 500/501, 600/601** Settings > Status > Diagnostics > Warnings

## Config file error: Files contain invalid params: <filename1>, <filename2>,...
## Config file error: <filename> contains invalid params.
## The following contain pre-3.3.0 params: <filename>

These messages display if any of the following pre-Polycom UC Software 3.3.0 parameters are found in the configuration files:

- tone.chord.ringer.x.freq.x
- se.pat.callProg.x.name
- ind.anim.IP_500.x.frame.x.duration
- ind.pattern.x.step.x.state
- feature.2.name
- feature.9.name

This message also displays if any configuration file contains:

- More than 100 unknown parameters, or
- More than 100 out-of-range values, or
- More than 100 invalid values.

To update the configuration files to use the correct parameters, see Change Parameter Values for details.

## Line: Unregistered

This message displays if a line fails to register with the call server.

## Login credentials have failed. Please update them if information is incorrect.

This message displays when the user enters incorrect login credentials: **Status** > **Basic** > **Login Credentials**.

## Missing files, config. reverted

This message displays when errors in the configuration and a failure to download the configuration files force the phone to revert to its previous (known) condition with a complete set of configuration files. This also displays if the files listed in the ***<MAC Address>*.cfg** file are not present on the provisioning server.

## Network Authentication Failure

This message displays if 802.1X authentication with the Polycom phone fails. The error codes shown in the following table display on the phone’s screen—if the **Details** soft key is selected—and in the log files:

**Event Codes and Descriptions**

| Event Code | Description | Comments |
|---|---|---|
| 1 | Unknown events | This includes any event listed in this table. |
| 2 | Mismatch in EAP Method type<br>Authenticating server's list of EAP methods does not match with clients’. | |

**Event Codes and Descriptions**

| | | |
|---|---|---|
| 30xxx | TLS Certificate failure<br>TLS certificate-related failures. When using a non-zero value, ‘xxx’ is the standard TLS alert message code. For example, if a bad/invalid certificate (on the basis of its signature and/or content) is presented by the phone, ‘xxx’ is 042. If you don’t know why the certificate is invalid, then the generic certificate error code is xxx=000. | See section 7.2 of RFC 2246<br>for further TLS alert codes and error codes. |
| 31xxx | Server Certificate failure<br>Certificate presented by the server is considered invalid.<br>‘xxx’ can take the following values:<br>•009 - Certificate not yet Valid<br>•010 - Certificate Expired<br>•011 - Certificate Revocation List<br>(CRL) not yet Valid<br>•012 - CRL Expired | |
| 4xxx | Other TLS failures<br>This is due to TLS failure other than certification related errors. The reason code (the TLS alert message code) is represented by "xxx". For example, if the protocol version presented by the server is not supported by the phone, then xxx is 70, and the EAP error code is 4070. | See section 7.2 of RFC 2246<br>for further TLS alert codes and error codes. |

### Network link is down

Because the Polycom phones do not have an LED indicating network LINK status, link failures are indicated with the message *Network link is down*. This message displays on the screen whenever the phone is not in the menu system and persists until the link problem is resolved. Call-related functions, soft keys, and line keys are disabled when the network is down; however the menu works.

## Use the Status Menu

Debugging of s single phone may be possible by examining of the phone’s status menu. Press **Menu**, select **Status**, and press the **Select** soft key to view the Status menu. Scroll to one of the Status menu items and press the **Select** soft key. Each of the menu items is explained next.

**Status Menu**

| Menu Item | Description |
|---|---|
| Platform | Information about the following:<br>• phone's serial number or MAC address<br>• current IP address<br>• Updater version<br>• application version<br>• name of the configuration files in use<br>• address of the provisioning server |
| Network | Information about the following:<br>• TCP/IP Setting<br>• Ethernet port speed<br>• connectivity status of the PC port (if it exists)<br>• statistics on packets sent and received since last boot.<br>You can also find out the last time the phone rebooted.<br>The Call Statistics screen shows packets sent and received on the last call. |
| Lines | Details about the status of each line that has been configured on the phone |
| Diagnostics | Offers a series of hardware tests to verify correct operation of the microphone, speaker, handset, and third party headset, if present.<br>You can also test that each of the keys on the phone is working, and display the function assigned to each of the keys in the configuration.<br>In addition to the hardware tests, the Diagnostics menu has a series of real-time graphs for CPU, network, and memory use that can be helpful for diagnosing performance issues. |

## Use Log Files

Polycom phones log various events to files stored in the flash file system and periodically uploads 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 Updater and one for the UC Software. 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 Updater (boot) 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.

Both log files can be uploaded on demand using a multiple key combination described in Use Multiple Key Combinations. 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 Configuration Utility. You should not modify the default logging levels unless directed to by Polycom Customer 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 Customer Support.

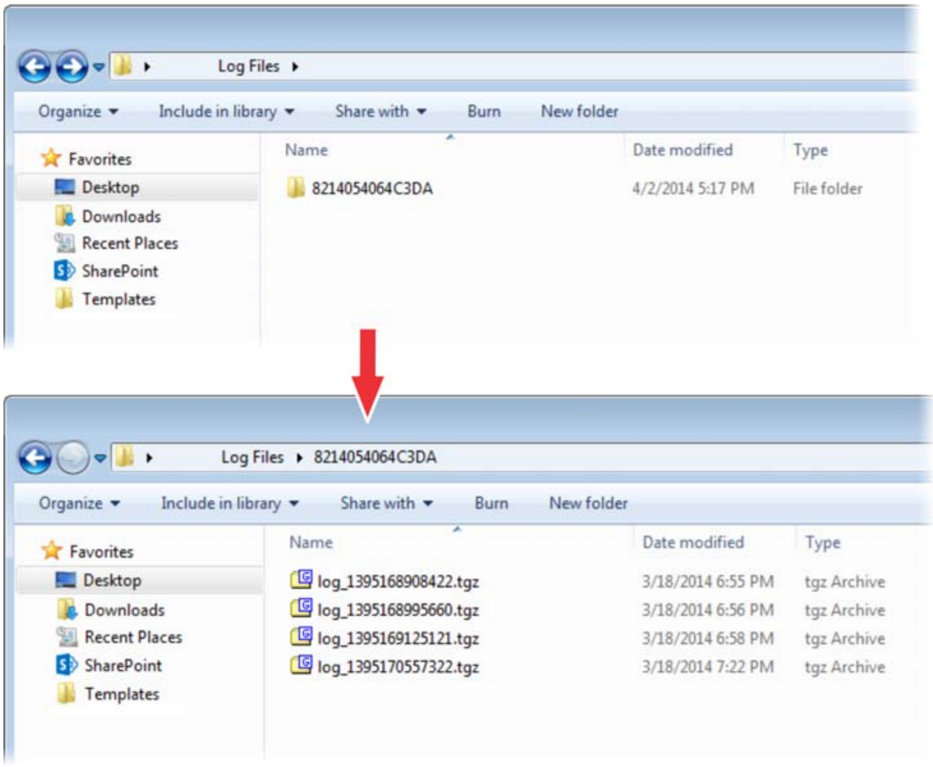
## Logging Options

Each component of the Polycom UC Software is capable of logging events of different severity. This enables you to capture lower severity events in one part of the application, and high severity events for other components.

You can use log level parameter setting for retrieving system log files:

The following figure shows an example folder with log files.

**Device folder and log files**



| Name | Date modified | Type |
|---|---|---|
| 8214054064C3DA | 4/2/2014 5:17 PM | File folder |

| Name | Date modified | Type |
|---|---|---|
| log_1395168908422.tgz | 3/18/2014 6:55 PM | tgz Archive |
| log_1395168995660.tgz | 3/18/2014 6:56 PM | tgz Archive |
| log_1395169125121.tgz | 3/18/2014 6:58 PM | tgz Archive |
| log_1395170557322.tgz | 3/18/2014 7:22 PM | tgz Archive |

The parameters for log level settings are found in the **techsupport.cfg** configuration file. They are `log.level.change.module_name`. Log levels range from 0 to 6 (0 for the most detailed logging, 6 for critical errors only). There are many different log types that can be adjusted to assist with the investigation of different problems. The exact number of log types is dependent on the phone model.

When testing is complete, remove the configuration parameter from the configuration files.

There are other logging parameters, described next, that you can modify. Changing these parameters does not have the same impact as changing the logging levels, but you should still understand how your changes affect the phone and the network.

**Logging Parameters**

| Parameter | Description |
|---|---|
| `log.render.level` | Sets the lowest level that can be logged<br>(default=1) |
| `log.render.file.size` | Maximum size before log file is uploaded<br>(default=32 kb) |
| `log.render.file.upload.period` | Frequency of log uploads<br>(default is 172800 seconds = 48 hours) |
| `log.render.file.upload.append` | Controls whether log files on the provisioning server are overwritten or appended, not supported by all servers<br>(default=1 so files are appended) |
| `log.render.file.upload.append.sizeLimit` | Controls the maximum size of log files on the provisioning server<br>(default=512 kb) |
| `log.render.file.upload.append.limitMode` | Controls whether to stop or delete logging when the server log reaches its maximum size<br>(default=delete) |

## Scheduled Logging

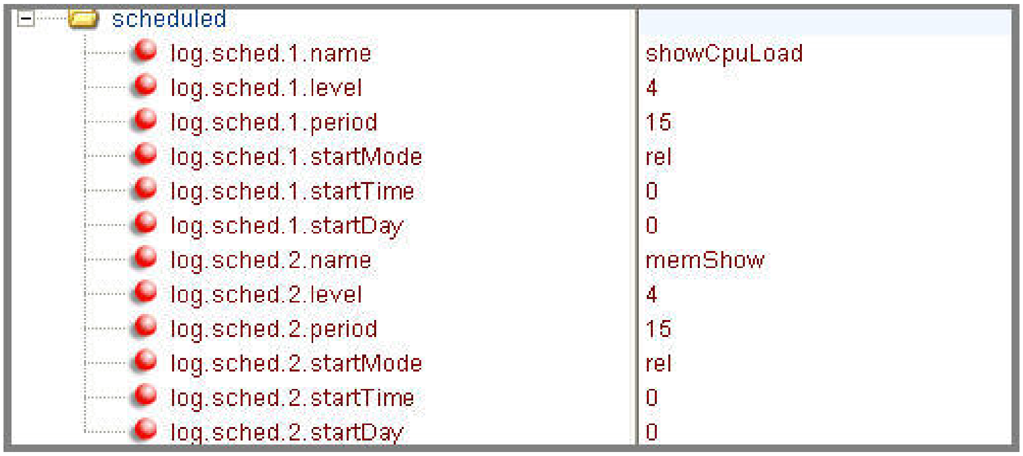
Schedules logging is a powerful tool that can help you troubleshoot issues that occur after the phone has been operating for some time.

The output of these instructions is written to the application log, and can be examined later for trend data.

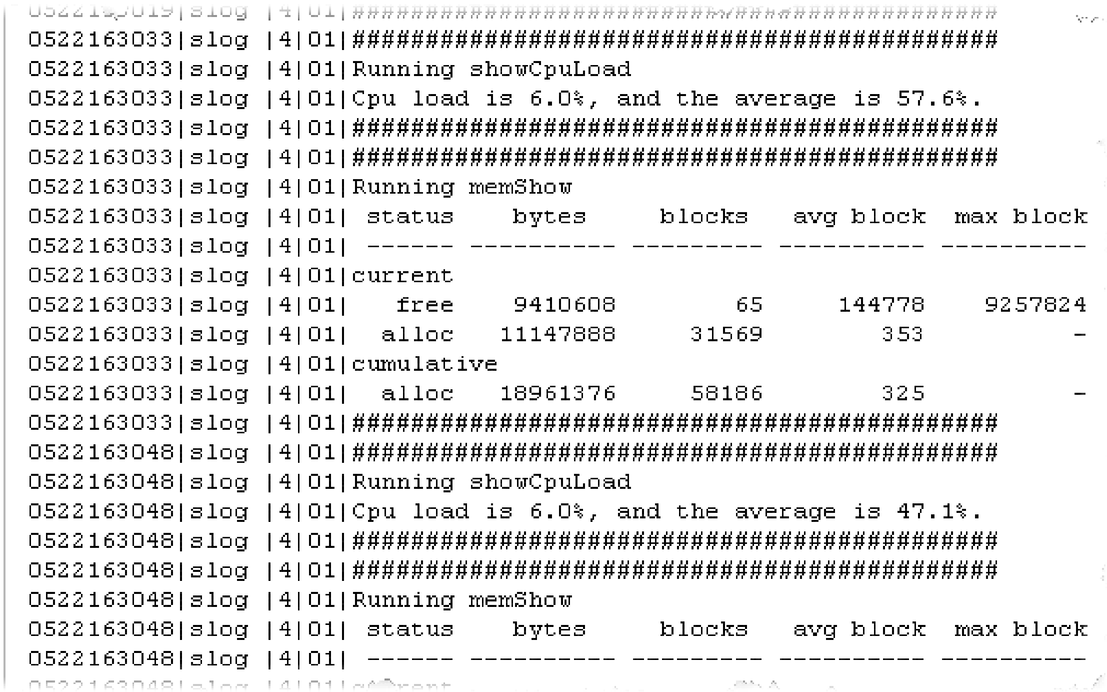
The parameters for scheduled logging are found in the **techsupport.cfg** configuration file. They are `log.sched.`*`module_name`*.

For an example of a configuration file and the resulting log file, see the following figure.

**Scheduled Logging Log File**



```
scheduled
    log.sched.1.name         showCpuLoad
    log.sched.1.level        4
    log.sched.1.period       15
    log.sched.1.startMode    rel
    log.sched.1.startTime    0
    log.sched.1.startDay     0
    log.sched.2.name         memShow
    log.sched.2.level        4
    log.sched.2.period       15
    log.sched.2.startMode    rel
    log.sched.2.startTime    0
    log.sched.2.startDay     0
```



```
0522163019|slog |4|01|############################################
0522163033|slog |4|01|############################################
0522163033|slog |4|01|Running showCpuLoad
0522163033|slog |4|01|Cpu load is 6.0%, and the average is 57.6%.
0522163033|slog |4|01|############################################
0522163033|slog |4|01|############################################
0522163033|slog |4|01|Running memShow
0522163033|slog |4|01| status    bytes     blocks   avg block  max block
0522163033|slog |4|01| ------ ---------- --------- ---------- ----------
0522163033|slog |4|01|current
0522163033|slog |4|01|   free    9410608        65     144778    9257824
0522163033|slog |4|01|  alloc   11147888     31569        353          -
0522163033|slog |4|01|cumulative
0522163033|slog |4|01|  alloc   18961376     58186        325          -
0522163033|slog |4|01|############################################
0522163048|slog |4|01|############################################
0522163048|slog |4|01|Running showCpuLoad
0522163048|slog |4|01|Cpu load is 6.0%, and the average is 47.1%.
0522163048|slog |4|01|############################################
0522163048|slog |4|01|############################################
0522163048|slog |4|01|Running memShow
0522163048|slog |4|01| status    bytes     blocks   avg block  max block
0522163048|slog |4|01| ------ ---------- --------- ---------- ----------
0522163048|slog |4|01|current
```

## Manual Log Upload

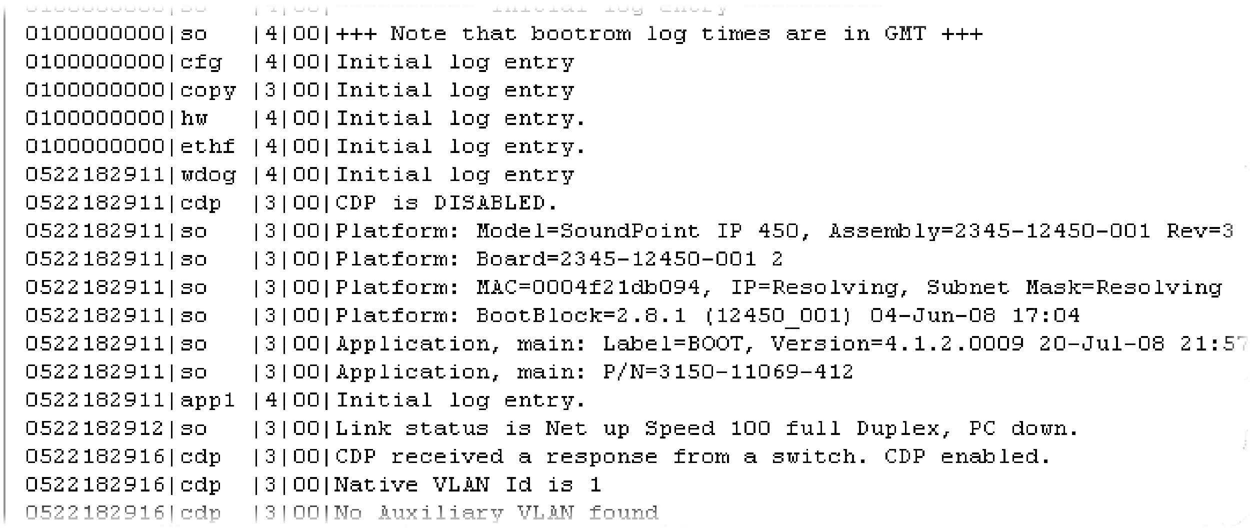
If you want to look at the log files without having to wait for the phone to upload them (which could take as long as 24 hours or more), initiate an upload by pressing the correct multiple key combination on the phone (see Use Multiple Key Combinations).

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 File

See the figure Boot Log File for an example of a boot log file:

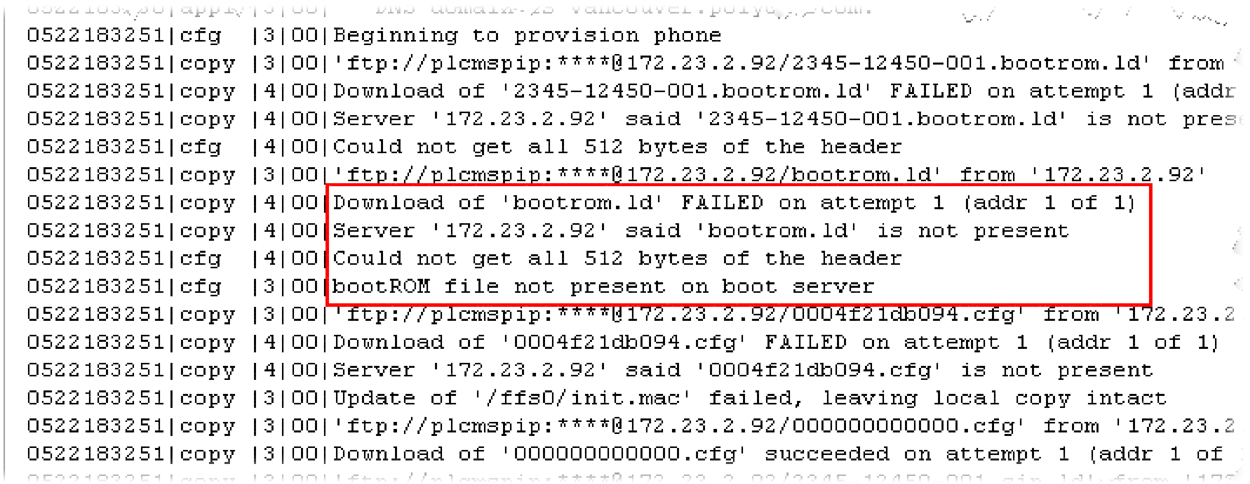
**Boot Log File**



```
0100000000|so   |4|00|+++ Note that bootrom log times are in GMT +++
0100000000|cfg  |4|00|Initial log entry
0100000000|copy |3|00|Initial log entry
0100000000|hw   |4|00|Initial log entry.
0100000000|ethf |4|00|Initial log entry.
0522182911|wdog |4|00|Initial log entry
0522182911|cdp  |3|00|CDP is DISABLED.
0522182911|so   |3|00|Platform: Model=SoundPoint IP 450, Assembly=2345-12450-001 Rev=3
0522182911|so   |3|00|Platform: Board=2345-12450-001 2
0522182911|so   |3|00|Platform: MAC=0004f21db094, IP=Resolving, Subnet Mask=Resolving
0522182911|so   |3|00|Platform: BootBlock=2.8.1 (12450_001) 04-Jun-08 17:04
0522182911|so   |3|00|Application, main: Label=BOOT, Version=4.1.2.0009 20-Jul-08 21:57
0522182911|so   |3|00|Application, main: P/N=3150-11069-412
0522182911|app1 |4|00|Initial log entry.
0522182912|so   |3|00|Link status is Net up Speed 100 full Duplex, PC down.
0522182916|cdp  |3|00|CDP received a response from a switch. CDP enabled.
0522182916|cdp  |3|00|Native VLAN Id is 1
0522182916|cdp  |3|00|No Auxiliary VLAN found
```

The figure Boot Failure Messages shows a number of boot failure messages:

**Boot Failure Messages**

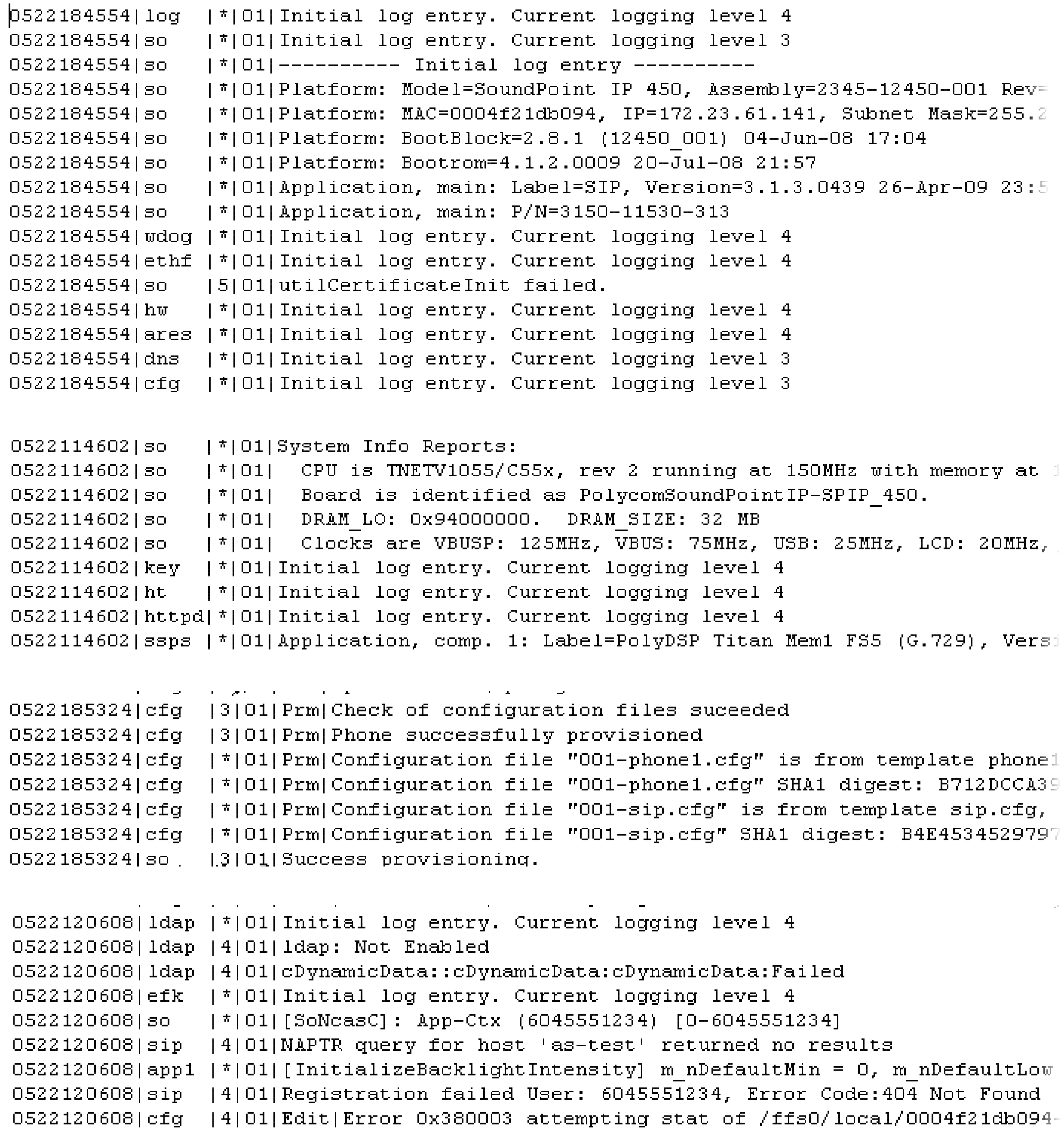


```
0522183251|cfg  |3|00|Beginning to provision phone
0522183251|copy |3|00|'ftp://plcmspip:****@172.23.2.92/2345-12450-001.bootrom.ld' from
0522183251|copy |4|00|Download of '2345-12450-001.bootrom.ld' FAILED on attempt 1 (addr
0522183251|copy |4|00|Server '172.23.2.92' said '2345-12450-001.bootrom.ld' is not pres
0522183251|cfg  |4|00|Could not get all 512 bytes of the header
0522183251|copy |3|00|'ftp://plcmspip:****@172.23.2.92/bootrom.ld' from '172.23.2.92'
0522183251|copy |4|00|Download of 'bootrom.ld' FAILED on attempt 1 (addr 1 of 1)
0522183251|copy |4|00|Server '172.23.2.92' said 'bootrom.ld' is not present
0522183251|cfg  |4|00|Could not get all 512 bytes of the header
0522183251|cfg  |3|00|bootROM file not present on boot server
0522183251|copy |3|00|'ftp://plcmspip:****@172.23.2.92/0004f21db094.cfg' from '172.23.2
0522183251|copy |4|00|Download of '0004f21db094.cfg' FAILED on attempt 1 (addr 1 of 1)
0522183251|copy |4|00|Server '172.23.2.92' said '0004f21db094.cfg' is not present
0522183251|copy |3|00|Update of '/ffs0/init.mac' failed, leaving local copy intact
0522183251|copy |3|00|'ftp://plcmspip:****@172.23.2.92/000000000000.cfg' from '172.23.2
0522183251|copy |3|00|Download of '000000000000.cfg' succeeded on attempt 1 (addr 1 of
```

# Reading an Application Log File

The figure Application Log File shows portions of an application log file:

### Application Log File



```
0522184554|log  |*|01|Initial log entry. Current logging level 4
0522184554|so   |*|01|Initial log entry. Current logging level 3
0522184554|so   |*|01|---------- Initial log entry ----------
0522184554|so   |*|01|Platform: Model=SoundPoint IP 450, Assembly=2345-12450-001 Rev=
0522184554|so   |*|01|Platform: MAC=0004f21db094, IP=172.23.61.141, Subnet Mask=255.2
0522184554|so   |*|01|Platform: BootBlock=2.8.1 (12450_001) 04-Jun-08 17:04
0522184554|so   |*|01|Platform: Bootrom=4.1.2.0009 20-Jul-08 21:57
0522184554|so   |*|01|Application, main: Label=SIP, Version=3.1.3.0439 26-Apr-09 23:5
0522184554|so   |*|01|Application, main: P/N=3150-11530-313
0522184554|wdog |*|01|Initial log entry. Current logging level 4
0522184554|ethf |*|01|Initial log entry. Current logging level 4
0522184554|so   |5|01|utilCertificateInit failed.
0522184554|hw   |*|01|Initial log entry. Current logging level 4
0522184554|ares |*|01|Initial log entry. Current logging level 4
0522184554|dns  |*|01|Initial log entry. Current logging level 3
0522184554|cfg  |*|01|Initial log entry. Current logging level 3


0522114602|so   |*|01|System Info Reports:
0522114602|so   |*|01|  CPU is TNETV1055/C55x, rev 2 running at 150MHz with memory at 
0522114602|so   |*|01|  Board is identified as PolycomSoundPointIP-SPIP_450.
0522114602|so   |*|01|  DRAM_LO: 0x94000000.  DRAM_SIZE: 32 MB
0522114602|so   |*|01|  Clocks are VBUSP: 125MHz, VBUS: 75MHz, USB: 25MHz, LCD: 20MHz,
0522114602|key  |*|01|Initial log entry. Current logging level 4
0522114602|ht   |*|01|Initial log entry. Current logging level 4
0522114602|httpd|*|01|Initial log entry. Current logging level 4
0522114602|ssps |*|01|Application, comp. 1: Label=PolyDSP Titan Mem1 FS5 (G.729), Vers


0522185324|cfg  |3|01|Prm|Check of configuration files suceeded
0522185324|cfg  |3|01|Prm|Phone successfully provisioned
0522185324|cfg  |*|01|Prm|Configuration file "001-phone1.cfg" is from template phone1
0522185324|cfg  |*|01|Prm|Configuration file "001-phone1.cfg" SHA1 digest: B712DCCA39
0522185324|cfg  |*|01|Prm|Configuration file "001-sip.cfg" is from template sip.cfg,
0522185324|cfg  |*|01|Prm|Configuration file "001-sip.cfg" SHA1 digest: B4E4534529797
0522185324|so   |3|01|Success provisioning.


0522120608|ldap |*|01|Initial log entry. Current logging level 4
0522120608|ldap |4|01|ldap: Not Enabled
0522120608|ldap |4|01|cDynamicData::cDynamicData:cDynamicData:Failed
0522120608|efk  |*|01|Initial log entry. Current logging level 4
0522120608|so   |*|01|[SoNcasC]: App-Ctx (6045551234) [0-6045551234]
0522120608|sip  |4|01|NAPTR query for host 'as-test' returned no results
0522120608|app1 |*|01|[InitializeBacklightIntensity] m_nDefaultMin = 0, m_nDefaultLow
0522120608|sip  |4|01|Registration failed User: 6045551234, Error Code:404 Not Found
0522120608|cfg  |4|01|Edit|Error 0x380003 attempting stat of /ffs0/local/0004f21db094-
```

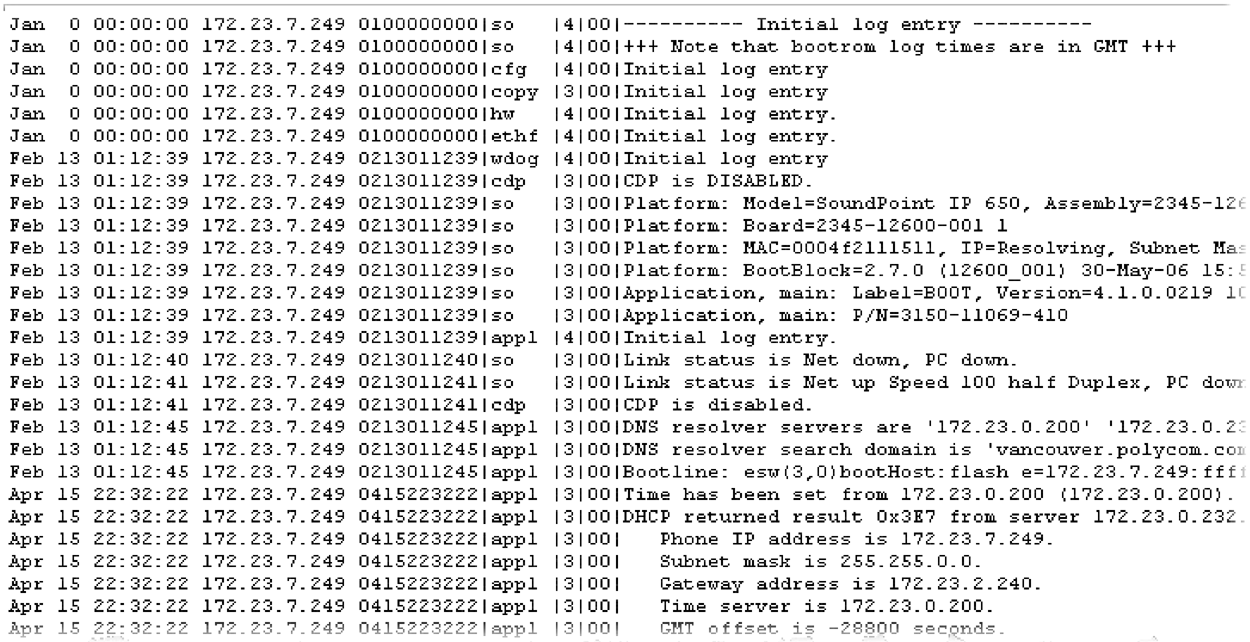
**Caution: Passwords display in configuration log file**

Passwords display in a level 1 cfg log file.

## Reading a Syslog File

The figure Syslog File shows a portion of a syslog log file. Note that the messages look identical to the normal log except for the addition of a timestamp and IP address:

**Syslog File**



```
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=2345-12
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 Ma
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Platform: BootBlock=2.7.0 (12600_001) 30-May-06 15:
Feb 13 01:12:39 172.23.7.249 0213011239|so   |3|00|Application, main: Label=BOOT, Version=4.1.0.0219 1
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 dow
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' '172.23.0.2
Feb 13 01:12:45 172.23.7.249 0213011245|appl |3|00|DNS resolver search domain is 'vancouver.polycom.co
Feb 13 01:12:45 172.23.7.249 0213011245|appl |3|00|Bootline: esw(3,0)bootHost:flash e=172.23.7.249:fff
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.
```

**Web Info: Using Syslog on Polycom phones**

For more information about syslog, see *Technical Bulletin 17124: Using Syslog on Polycom Phones*.

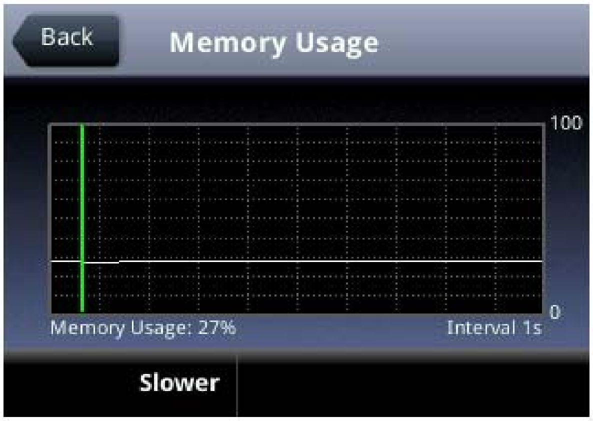
# Manage the Phone’s Memory Resources

Polycom phones are designed to operate optimally in a variety of deployments and real-world environments. Each new software release adds new features and capabilities that require varying degrees of the phone’s memory resources. To ensure your phones and their configured features operate smoothly, verify whether the phones have adequate available memory resources. If you are using a range of phone features—especially customized or advanced features—you may need to manage phone memory resources. To help you optimize your phone features and memory resources, Polycom provides several tools and troubleshooting tips.

## Identify Symptoms

When the phone memory resources start to run low, you may notice one or more of the following symptoms:

- The phones reboot or freeze up.
- The phones do not download all ringtones, directory entries, backgrounds, or XML dictionary files.



ɪrowser stop or do not run at all.

ıe's available memory and manage the phone features

## ory

you need to manage your phone's memory. Before you ł files you want to make available on the phone.

ouch pad interface, choose **Status > Diagnostics >**

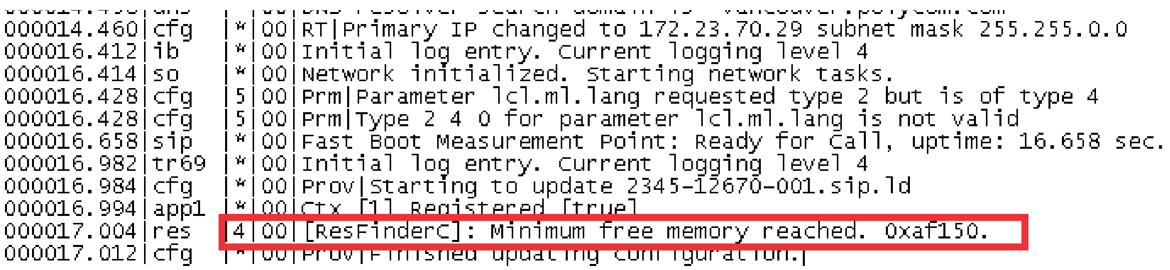
Use the *Memory Usage* chart to check what the current Memory Usage amount is. Typically, you want to ensure that the phone is running at less than 95 percent of its available memory.

If the phone is using more than 95 percent of its available memory, you may need to take steps to reduce this amount. For information and tips on freeing memory on the phone, see Manage the Phone Features.

The second method you can use to confirm whether you need to manage your phone's memory is to check the app log files. The app log file is enabled by default and is saved to your provisioning server directory with the MAC address of the phone prepended to the app log file. For example, if the MAC address of your phone is **0004f2203b0**, the app log file name is **0004f2203b0-app.log**.

Open the app log. If you see the message shown next in the figure Application Log Error Message, you may need to manage your phone's memory resources.

**Application Log Error Message**



```
000014.498|dns  |*|00|DNS resolver search domain is  vancouver.polycom.com
000014.460|cfg  |*|00|RT|Primary IP changed to 172.23.70.29 subnet mask 255.255.0.0
000016.412|ib   |*|00|Initial log entry. Current logging level 4
000016.414|so   |*|00|Network initialized. Starting network tasks.
000016.428|cfg  |5|00|Prm|Parameter lcl.ml.lang requested type 2 but is of type 4
000016.428|cfg  |5|00|Prm|Type 2 4 0 for parameter lcl.ml.lang is not valid
000016.658|sip  |*|00|Fast Boot Measurement Point: Ready for Call, uptime: 16.658 sec.
000016.982|tr69 |*|00|Initial log entry. Current logging level 4
000016.984|cfg  |*|00|Prov|Starting to update 2345-12670-001.sip.ld
000016.994|app1 |*|00|Ctx [1] Registered [true]
000017.004|res  |4|00|[ResFinderC]: Minimum free memory reached. 0xaf150.
000017.012|cfg  |*|00|Prov|Finished updating configuration.|
```

**Web Info: Reading the app log files**

For more information on reading the log files see Use Log Files.

## Manage the Phone Features

This section provides tips for managing the phone features to conserve phone memory resources. This section is especially useful if you are customizing features or using several advanced features.

If you are using a mixed deployment, such as a combination of phone models, consider configuring each phone model separately with their own features instead of applying all phone features to different phone models.

All phone features are designed to operate optimally on Polycom phones. The features listed in the following table are customizable, advanced features that can take up significant memory. Use this table as a reference guide to the amount of memory a feature can use and for tips on balancing features so that you can optimize the phone features you want for your deployment.

**Managing the Phone Features**

| Feature | Typical Memory Size |
|---|---|
| **Idle Browser** | **Variable. Optimized to display three or four elements.** |
| The idle browser is optimized to display three or four application elements. If you display complex pages that include large table or images, try to display a simplified page. If the page cannot be simplified, try reducing the number of available ringtones or display backgrounds, or disable the main browser. | |
| **Custom Idle Display Image** | **15KB** |
| The average size of Polycom display images is 15KB. If you are using custom images, Polycom recommends limiting the file size to 15KB for images on the idle display. If your phone does not display your custom image and the file size is less than 15KB, try reducing the number of available ringtones or idle display and image backgrounds. | |
| **Main Browser** | **Variable. Optimized to display three or four elements.** |
| The main browser is optimized to display three or four application elements. As with the idle browser, try simplifying the content to conserve memory resources. If the content cannot be simplified, try reducing the number of available ringtones or image backgrounds, or disable the idle browser. | |
| **Local Contact Directory** | **170 bytes per entry** |
| Polycom phones are optimized to display four contact attributes to a maximum of 250 contact entries. Each entry averages about 170 bytes of memory.<br>If you need more space for the contact directory, try disabling the idle browser, reducing the number of available ringtones or image backgrounds. | |
| **Corporate Directory** | **Varies by server** |
| The Corporate Directory feature is optimized to display five contact attributes up to a maximum of eight on Polycom phones. Because the corporate directory entries are saved to a server, the size of each entry and the corporate directory as a whole vary with the server you are using. If the phone has difficulty displaying directory search results with more than five attributes, try reducing the number of available ringtones or image backgrounds, or disable the idle browser or main browser. | |
| **Ringtones** | **16KB** |
| Polycom provides a number of audio files for ringtones that are designed to work correctly with the phones. Polycom ringtones can range in size from 30KB to 125KB. If you want to use custom ringtones, Polycom recommends limiting the file size to 16KB. If you want to make more room for custom ringtones, try disabling the idle browser or main browser, or reduce the number of custom or image backgrounds. If you want to make room for other features, try reducing the number of available ringtones. | |
| **Background Images** | **8 – 32KB** |
| Polycom phones are optimized to display background images of about 50KB. If you want to display background images having a file size of more than 50KB or make room for more images, try disabling the idle browser or main web browser, or reduce the number of available ringtones. If you want to make room for other features, try reducing the number and size of available background images. | |

**Managing the Phone Features**

| Feature | Typical Memory Size |
|---|---|
| **Phone Interface Language** | **90KB** |
| The average size of the VVXLocalization XML dictionary files for languages that display on the phone's interface is about 90KB. Some of these language files use an expanded character set that can increase the file size to 115KB. To conserve memory resources, Polycom recommends using only those XML language files for the languages you need. | |
| **Web Configuration Utility Interface** | **250KB** |
| The average size of the languages XML dictionary files for languages that display on the Web Configuration Utility interface is about 250KB. Some of these language files use an expanded character set that can increase the file size to 370KB. To conserve memory resources, Polycom recommends using only those XML language files for the languages you need. | |

If you are still having difficulty freeing up sufficient space on your phones, contact Polycom Voice Product Support.

# Test Phone Hardware

You can view diagnostic information from the **Diagnostics** menu on your phone (**Menu** > **Settings** > **Status** > **Diagnostics**).

If you select **Diagnostics** > **Test Hardware**, you can select one of the following menu items to perform a hardware diagnostic test:

- **Audio Diagnostics** Test the speaker, microphone, handset, and a third party headset**.**
- **Keypad Diagnostics** Verify the function assigned to each keypad key.
- **Display Diagnostics** Test the LCD for faulty pixels.
- **LED Diagnostics** Test the LED lights on your phone.
- **Touch Screen Diagnostics** Test the touch screen response.

# Upload a Phone's Configuration

As of Polycom UC Software 3.3.0, you can upload the files representing a phone's current configuration. A number of files can be uploaded to the provisioning server, one for every active source as well as the current non-default configuration set.

As of Polycom UC Software 4.0.0, you can upload the phone's configuration through the Web Configuration Utility.

This is primarily a diagnostics tool to help find configuration errors.

### To upload the phone's current configuration:

1 Navigate to the Upload Configuration menu on the phone (**Menu** > **Settings** > **Advanced** > **Admin Settings** > **Upload Configuration**).

2 Choose to upload the configuration from one of **All Sources**, **Configuration Files**, or **Web**.

You can select **Device Settings** if you perform this task using the Web Configuration Utility.

3 Press the **Upload** soft key.

The phone uploads the configuration file to the location that you specify in `prov.configUploadPath`. For example, if you select **All Sources**, a file ***<MACaddress>*-update-all.cfg** is uploaded.

# Perform Network Diagnostics

In Polycom UC Software 4.0.0, ping and traceroute are added to the phone’s diagnostics tools. These diagnostics can be used for troubleshooting network connectivity problems. Both tools are accessible by pressing the **Menu** key and selecting **Status** > **Diagnostics** > **Network**.

Enter a URL address, for example, http://www.google.com, or any IP address, for example, the system IP address or any other phone’s IP address, and then press the **Enter** soft key.

# Ports Used on Polycom Phones

The following table lists the ports currently used by the Polycom UC Software.

**Ports Used by Polycom Phones**

| Port Number | Protocol | Outgoing | Incoming | UDP or TCP |
|---|---|---|---|---|
| 21 | FTP | Provisioning, Logs | | TCP |
| 22 | SSH | Admin | Admin | TCP |
| 23 | Telnet[1] | Admin | | TCP |
| 53 | DNS | | | UDP |
| 67 | DHCP | Server | | UDP |
| 68 | DHCP | Client | | UDP |
| 69 | TFTP | Provisioning, Logs | | UDP |
| 80 | HTTP | Provisioning, Logs, Pull<br>Web interface, Poll | | TCP |
| 123 | NTP | Time Server | | UDP |
| 389 | LDAP | | | |
| 443 | HTTPS | Provisioning, Logs | HTTP Pull Web interface, HTTP Push | TCP |
| 514 | Syslog | Logs | | |
| 636 | LDAP | | | |
| 1719 | H.323[2] | RAS Signaling | RAS Signaling | |

**Ports Used by Polycom Phones**

| Port Number | Protocol | Outgoing | Incoming | UDP or TCP |
|---|---|---|---|---|
| 1720 | H.323[2] | Signaling | Signaling | |
| 2222 | RTP[3] | Media Packets | Media Packets | |
| 2223 | RTCP[3] | Media Packet Statistics | Media Packet Statistics | |
| 5060 | SIP | SIP signaling | SIP signaling | |
| 5061 | SIP over TLS | Secure signaling | Secure signaling | |
| 24800 | PDC | PDC Client messages | PDC Server messages | TCP |

[1] Telnet is disabled by default on VVX phones.

[2] H.323 is available only on the VVX 500/501, 600/601, and 1500.

[3] RTP and RTCP can use any even port between 2222 and 2269 (2317 on VVX 500/501, 600/601, or 1500), but you can configure ports by setting `tcpIpApp.port.rtp.mediaPortRangeStart`.

# Power and Startup Issues

The following table describes possible solutions to power and startup issues.

**Troubleshooting Power and Startup Issues**

| The phone has power issues or the phone has no power. |
|---|
| Determine whether the problem is caused by the phone, the AC outlet, or the PoE switch. Do one of the following:<br>• Verify that no lights appear on the unit when it is powered up.<br>• Check if the phone is properly plugged into a functional AC outlet.<br>• Make sure that the phone isn't plugged into an outlet controlled by a light switch that is off.<br>• If plugged into a power strip, try plugging directly into a wall outlet instead. |
| **The phone does not boot.** |
| If your phone does not boot, there may be a corrupt or invalid firmware image or configuration on the phone:.<br>• Ensure that the provisioning server is accessible on the network and a valid software load and valid configuration files are available.<br>• Ensure that the phone is pointing to the provisioning server on the network.<br>• Reboot the phone. |

# Dial Pad Issues

The following table describes possible solutions to issues with the dial pad.

**Troubleshooting Dial Pad Issues**

| **The dial pad does not work.** |
|---|
| If the dial pad on your phone does not respond, do one of the following:<br>• Check for a response from other feature keys or from the dial pad.<br>• Place a call to the phone from a known working telephone. Check for display updates.<br>• Navigate to **Menu > System Status > Server Status** to check if the telephone is correctly registered to the server.<br>• Navigate to **Menu > System Status > Network Statistics**. Scroll down to see whether LAN port shows Active or Inactive.<br>• 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. |

# Screen and System Access Issues

The following table describes possible solutions to screen and system access issues.

**Troubleshooting Screen and System Access Issues**

| **There is no response from feature key presses.** |
|---|
| If your phone is not in the active state, do one of the following:<br>• Press the keys more slowly.<br>• Check to see whether or not the key has been mapped to a different function or disabled.<br>• Make a call to the phone to check for inbound call display and ringing. If successful, try to press feature keys while a call is active to access a directory or buddy status.<br>• Navigate to **Menu > Status > Lines** to confirm the line is actively registered to the call server.<br>• Reboot the phone to attempt re-registration to the call server. Refer to Reboot the Phone. |
| **The display shows the message *Network Link is Down.*** |
| If you see this message, the LAN cable is not properly connected. Do one of the following:<br>• Check termination at the switch or hub (furthest end of the cable from the phone).<br>• Check that the switch or hub is operational (flashing link/status lights).<br>• Press Menu followed by **Status > Network**. Scroll down to verify that the LAN is active.<br>• Ping the phone from another machine.<br>• Reboot the phone to attempt re-registration to the call server. Navigate **to Menu > Settings > Advanced > Reboot Phone**). |

# Calling Issues

The following table provides possible solutions to generic calling issues.

**Troubleshooting Calling Issues**

**There is no dial tone.**

If there is no dial tone, power may not be correctly supplied to the phone. Try 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 you are using in-line powering, have your system administrator check that the switch is supplying power to the phone.

**The dial tone is not present on one of the audio paths.**

If dial tone is not present on one of the audio paths, do one of the following:

- Switch between handset, headset (if present), or handsfree speakerphone to see whether the dial tone is present on another path.
- If the dial tone exists on another path, connect a different handset or headset to isolate the problem.
- Check configuration for gain levels.

**The phone does not ring.**

If there is no ring tone but the phone displays a visual indication when it receives an incoming call, do the following:

- Adjust the ring level from the front panel using the volume up/down keys.
- Check the status of handset, headset (if connected), and handsfree speakerphone.

**The line icon shows an unregistered line icon.**

If you see one of the following icons the phone line is unregistered. Register the line and try to place a call.

**Unregistered Line Icons**: ☏ (most phones) ✗.

**Registered Line Icons**: ☎ (most phones) ✓ .

# Display Issues

The following table provides tips for resolving display screen issues.

**Troubleshooting Display Issues**

**There is no display or the display is incorrect.**

If there is no display, power may not be correctly supplied to the phone. Do one of the following:

- Check that the display is illuminated.
- Make sure the power is inserted properly at the rear of the phone.
- If your are using PoE powering, check that the PoE switch is supplying power to the phone.
- Use the screen capture feature to determine if the display on the phone is incorrect. Refer to Capture Your Device's Current Screen.

**The display is too dark or too light.**

**Troubleshooting Display Issues**

| |
|---|
| The phone contrast may be set incorrectly. To adjust the contrast, do one of the following:<br>• Adjust the contrast. Refer the phone’s User Guide.<br>• Reboot the phone to obtain the default level of contrast. Refer to Reboot the Phone.<br>• Use the screen capture feature to verify whether the screen displays properly in the capture. Refer to Capture Your Device's Current Screen. |
| **The display is flickering.** |
| Certain types of older fluorescent lighting cause the display to flicker. If your phone is in an environment lit with fluorescent lighting, do one of the following:<br>• Move the Polycom phone away from the lights.<br>• Replace the lights. |
| **The time and date are flashing.** |
| If the time and date are flashing, you have disconnected the phone from the LAN or there is no SNTP time server configured. Do one of the following; refer to Set the Time and Date Display.<br>• Reconnect the phone to the LAN.<br>• Configure an SNTP server.<br>• Disable the time and date if you do not want to connect your phone to a LAN or SNTP server. |

# Audio Issues

The following table describes possible solutions to audio issues.

**Troubleshooting Audio Issues**

| There is no audio on the headset |
|---|
| If there is no audio on your headset, the connections may not be correct. Do one of the following:<br>• Ensure the headset is plugged into the jack marked Headset at the rear of the phone.<br>• Ensure the headset amplifier (if present) is turned on and adjust the volume. |

# Licensed Feature Issues

The following table describes issues for features that require a license.

**Troubleshoot Feature License Issues**

| **Voice Quality Monitoring or H.323 is not available on the phone.** |
|---|
| If you cannot access features, check your licenses on the phone by navigating to **Menu > Status > Licenses**.<br>• You need a license to use voice quality monitoring on the VVX 300/301, 310/311, 400/401, and 410/411. You do not need a license to use voice quality monitoring on the VVX 500/501, 600/601, and 1500.<br>• You need a license to use H.323 on VVX 1500. You do not need a license to use H.323 on the VVX 500/501, 600/601. Note that H.323 is not supported on VVX 300/301, 310/311, 400/401, 410/411, and SoundStructure VOIP Interface.<br>• If your phone is not installed with UC Software version 4.0.0 or later, you also require a license for conference management, corporate directory, and call recording. |

# Upgrade Issues

The following table describes possible solutions to issues that may occur during or after a software upgrade.

**Troubleshooting Software Upgrade Issues**

| **Some settings or features are not working as expected on the phone.** |
|---|
| The phone’s configuration may be incorrect or incompatible.<br>Check for errors on the phone by navigating to **Menu > Status > Platform > Configuration**. If there are *Errors Found*, *Unknown Params,* or *Invalid values*, correct your configuration files and restart the phone. |
| **The phone displays a *Config file error* message for five seconds after it boots up.** |
| Pre-UC Software 3.3.0 configuration files are being used with UC Software 3.3.0. Specifically, the following parameters are in the configuration files:<br>• one.chord.ringer.x.freq.1<br>• se.pat.callProg.x.name<br>• ind.anim.IP_500.x.frame.x. duration<br>• ind.pattern.1.step.x.state<br>• feature.2.name<br>• feature.9.name<br>Also the configuration files contain:<br>• more than 100 “unknown” parameters<br>• more than 100 “out-of-range” parameters<br>• more than 100 “invalid” parameters<br>Correct the configuration files, remove the invalid parameters, and restart the phone. |

### Troubleshooting Software Upgrade Issues

| **When you are upgrading phone software by using the Web Configuration Utility, the phone is unable to connect to the Polycom Hosted Server.** |
|---|
| Occasionally, the phone is unable to connect to the Polycom hosted server because of the following:<br>• The Polycom hosted server is temporarily unavailable.<br>• There is no software upgrade information for the phone to receive.<br>• The network configuration is preventing the phone from connecting to the Polycom hosted server.<br>*Note: UC Software 4.0.0 does not support internet access for software upgrades through a web proxy.*<br>To troubleshoot the issue:<br>Try upgrading your phone later.<br>Verify that new software is available for your phone using the *Polycom UC Software Release Matrix for VVX Phones*.<br>Verify that your network's configuration allows the phone to connect to http://downloads.polycom.com.<br>If the issue persists, try manually upgrading your phone's software. To upgrade phone software using this method, refer to Set Up the Provisioning Server. |

# Update and Maintain Polycom Devices and UC Software

This section provides information on updating and maintaining your devices and the UC Software.

You can upgrade the software that is running on the Polycom phones in your organization. The upgrade process varies with the version of Polycom UC Software that is currently running on your phones and with the version that you want to upgrade to.

- As of UC Software 5.3.0, you can update software with the user-controlled software update feature explained in the section User-Controlled Software Update.
- If you are updating software from UC Software 4.0.x, see the section Update Phones from UC Software 4.0.x.

The Updater, UC Software executable, and configuration files can all be updated using centralized provisioning.

## Update Polycom UC Software

You can upgrade the software that is running on the Polycom phones in your organization. The upgrade process varies with the version of Polycom UC Software that is currently running on your phones and with the version that you want to upgrade to. The Updater, UC Software executable, and configuration files can all be updated using centralized provisioning.

**Administrative Tip: Updating UC Software on a single phone**

You can use the software upgrade tool in the Web Configuration Utility to update the UC Software version running on a single phone. Note that configuration changes made to individual phones using the Web Configuration Utility override configuration settings made using central provisioning. For instructions on how to update UC Software, see *Use the Software Upgrade Tool in the Web Configuration Utility: Feature Profile 67993* at Polycom Engineering Advisories and Technical Notifications.

To continue setting up a provisioning server, administrators can use the instructions shown in Update Phones from UC Software 4.0.x.

**Web Info: Downgrading from UC Software 4.0.0 or later**

After you have deployed the phones using UC Software 4.0.0 or later, you can downgrade to a previous software release by following the instructions in *Upgrading Polycom Phones to and Downgrading Phones from Polycom UC Software 4.0.0*: *Technical Bulletin 64731 at* Polycom Engineering Advisories and Technical Notifications.

# User-Controlled Software Update

This feature, available as of UC Software 5.3.0, enables phone users to choose when to update their phone with the latest software update provided by an administrator. Administrators can use this feature to send a software version that is later or earlier than the current version on the phone. This feature is available on VVX phones. You can use the Web Configuration Utility to update phone software if this feature is enabled or disabled.

Any configuration changes you make on the server are included in the software update. When users postpone software updates, configuration changes are postponed along with new software updates. When the phone updates the software, configuration changes are made as well. This applies to configuration changes imported using the Web Configuration Utility.

Depending on the polling policy you set, the phone displays a notification to update the phone software when the polling time expires. Users can choose to postpone the software after the polling expires. For example, if the polling policy is set to poll for every four hours, after four hours, the phone polls for new software on the server and displays a notification letting the user know that a software update is available. Users can choose to update the software or postpone it for up to six hours.

The polling policy is disabled after the phone displays the software update notification. After the software postponement ends, the phone displays the software update notification again. Users can postpone the software update up to three times. After the last postponement, the phone automatically updates the software.

This feature does not work if you have enabled ZTP or Skype for Business Device Update, and is not available with Skype for Business Server.

**Configure User-Controlled Software Update**

| **Parameter Function** | **template** > parameter |
|---|---|
| Enable or disable display of the software update notification. | **site.cfg** > prov.usercontrol.enabled |
| Configure a time interval for software update notifications. | **site.cfg** > prov.usercontrol.postponeTime |

# Update Phones from UC Software 4.0.x

If your Polycom phones are running a version of UC Software 4.0.x or later, you can upgrade to a more recent UC Software version using the instructions in this section. If your phones are running a software release earlier than UC Software 4.0.x, you can upgrade to UC Software 4.0.x by following the instructions in Technical Bulletin 64731: Upgrading Polycom Phones to and Downgrading Phones From Polycom UC Software 4.0.0.

### To update phones to Polycom UC Software 4.0.1:

1. Back up your existing application and configuration files.
2. Create your new configuration using UC Software 4.1.0.

   Configuration file changes and enhancements are explained in the Release Notes that accompany the software.

**Caution: Mandatory changes to configuration files**

To ensure predictable phone behavior, the configuration files listed in CONFIG_FILES attribute of the master configuration file must be updated when the software is updated. You must add new configuration files to the CONFIG_FILES attribute in the appropriate order.

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

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

- ➢ You should reboot your phone using the multiple-key combination as a backup option only if another reboot method fails. For details on using a multiple key combination to reboot your phone, see the section Use Multiple Key Combinations.
- ➢ You can boot the phones remotely through the SIP signaling protocol. See <voIP.SIP.specialEvent.*/>.

You can configure the phones to periodically poll the provisioning server for changed configuration files or application executables. If a change is detected, the phone may reboot to download the change.

# Trusted Certificate Authority List

Polycom maintains and publishes a list of trusted certificate authorities (CAs) supported by each major Polycom UC Software release. To find the list of supported CAs for your UC Software version, see *Certificate Updates for Polycom UC Software – Technical Update* on Polycom Engineering Advisories and Technical Notifications. Polycom publishes the following details for each trusted CA:

- Certificate Common Name (CN)
- RSA public key size
- Signature algorithm
- Start and end date of certificate validity

**Troubleshoot why your certificate authority is not listed**

Polycom endeavors to maintain a built-in list of the most commonly used Certificate Authority (CA) certificates. Due to memory constraints, we cannot ensure a complete set of certificates.

If you are using a certificate from a commercial CA not in the list above, you can submit a feature request for Polycom to add your CA to the trusted list. At this point, you can use the custom certificate method to load your particular CA certificate into the phone. See *Using Custom Certificates on Polycom Phones (Technical Bulletin 17877)*.

# OpenSSL Versions List

To view release notes for all Open SSL versions, see OpenSSL Release Notes.

**OpenSSL Versions**

| *UC Software Version* | *OpenSSL Version* |
|---|---|
| UC Software 5.3.0 | OpenSSL 1.0.1j 15 Oct 2014 |
| UC Software 5.2.2 | OpenSSL 1.0.1j 15 Oct 2014 |

**OpenSSL Versions**

| *UC Software Version* | *OpenSSL Version* |
|---|---|
| UC Software 5.2.0 | OpenSSL 1.0.1h 5 Jun 2014 |
| UC Software 5.1.3 | OpenSSL 1.0.1h 5 Jun 2014 |
| UC Software 5.1.2 | OpenSSL 1.0.1h 5 Jun 2014 |
| UC Software 5.0.2 | OpenSSL 1.0.1c 10 May 2012 |
| UC Software 5.0.1 | OpenSSL 1.0.1c 10 May 2012 |
| UC Software 5.0.0 | OpenSSL 1.0.1c 10 May 2012 |

# Encrypt Configuration Files

Polycom phones can download encrypted files from the provisioning server and encrypt files before uploading them to the provisioning server. You can encrypt all configuration files except the master configuration file, contact directories, and configuration override files from the Web Configuration Utility and local device interface.

To encrypt files, you must provide the phone an encryption key. You can generate your own 32 hex-digit, 128 bit key or use the Polycom Software Development Kit (SDK) to generate a key and to encrypt and decrypt configuration files on a UNIX or Linux server. The SDK is distributed as source code that runs under the UNIX operating system.

Note that the SDK generates a random key and applies Advanced Encryption Standard (AES) 128 in Cipher Block Chaining (CBC) mode. For example, a key can look like this:

```
Crypt=1;KeyDesc=companyNameKey1;Key=06a9214036b8a15b512e03d53412006;
```

**Web Info: Using the SDK to encrypt files**

To request the SDK and quickly install the generated key, see *When Encrypting Polycom UC Software Configuration Files*: *Quick Tip 67442 at* Polycom Engineering Advisories and Technical Notifications.

You can use the `device.set, device.sec.configEncryption.key, and device.sec.configEncryption.key.set` configuration parameters to set the key on the phone.

If the phone doesn't have a key, you must download the key to the phone in plain text, which is a potential security concern 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. To manage your provisioning server efficiently, Polycom recommends that you give each key a unique descriptive string in order to identify which key was used to encrypt a file.

After encrypting a configuration file, it is useful to rename the file to avoid confusing it with the original version, for example, rename **site.cfg** to **site.enc**. However, the directory and override filenames cannot be changed this way.

**Troubleshooting: My phone keeps displaying an error message for my encrypted file**

If a phone downloads an encrypted file that it cannot decrypt, the action is logged, and an error message displays. The phone continues to do this until the provisioning server provides an encrypted file that can be read, an unencrypted file, or until the file is removed from the list in the master configuration file.

For security purposes, you can change the key on the phones and the server from time to time.

### To change a key on the phone:

1. Put all encrypted configuration files on the provisioning server to use the new key.

   The phone may reboot multiple times.

   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.
2. Put the new key into a configuration file that is in the list of files downloaded by the phone, specified in **000000000000.cfg** or ***<MACaddress>*.cfg**.
3. Use the `device.sec.configEncryption.key` parameter to specify the new key.
4. Provision the phone again so that it downloads the new key. The phone automatically reboots a second time to use 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 replaces them when it successfully boots.

### To check whether an encrypted file is the same as an unencrypted file:

1. Run the *configFileEncrypt* utility, available from Polycom Support, on the unencrypted file with the "-d" option. This shows the "digest" field.
2. Look at the encrypted file using a text editor, 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.

# Use Multiple Key Combinations

You can use multiple key combinations on your Polycom phones to reboot the phone, to restore the phone to factory default values, or to upload log files from the phone to your provisioning server.

**Web Info: Resetting and rebooting your phone**

For other methods for resetting and rebooting your Polycom phones, refer to *Updating, Troubleshooting, and Resetting SoundPoint IP, SoundStation IP, and VVX 1500 Phones*: *Quick Tip 18298 at* Polycom Engineering Advisories and Technical Notifications.

## Reboot the Phone

Rebooting the phone downloads new software and new configuration files if they exist on the provisioning server.

**Tip: Download new configuration files without rebooting your phone**

As of UC Software 3.3.0, not all configuration parameter changes require the phone to restart or reboot. You can update your phone’s configuration by navigating to **Menu > Settings > Basic** and selecting **Update Configuration**.

If there is new software on the provisioning server, the phone restarts or reboots to download the software. If there are configuration file changes, your phone only restarts if it is necessary. Otherwise, the phone downloads the new configuration files without restarting.

You can use a multiple key combination to reboot your phone. Depending on your phone model, press and hold the following keys simultaneously until you hear a confirmation tone (for about three seconds):

- VVX 300 and 400 series: dial pad keys 0, 1, and 3 while the phone is off hook
- VVX 500 and 600 series: dial pad keys 0, 1, and 3
- VVX 1500: Delete, Volume-, Volume+, and Select

**Tip: Quickly restart your phone**

As of SIP 3.2.0, users can restart their phones by pressing the **Menu** key and selecting **Settings > Basic > Restart Phone**. If new Updater or Polycom UC Software is available on the provisioning server, the phone downloads the software when it restarts.

# Reset the Phone to Defaults

You can reset a phone to default settings. This is useful when you use more than one method to configure phones and phone features. Resetting the phone to factory defaults clears the flash parameters, removes log files, user data, and cached data, and resets the administrator password to 456.

The following table describes options for resetting the phone. To access these settings, go to **Menu > Settings > Advanced > Administration Settings > Reset to Defaults**.

**Phone Reset Options**

| Setting | Description |
|---|---|
| Reset Local Configuration | Clears the override file generated by changes using the phone user interface. |
| Reset Web Configuration | Clears the override file generated by changes using the Web Configuration Utility. |
| Reset Device Settings | Resets the phone’s flash file system settings that are not stored in an override file.<br>These are your network and provisioning server settings and include custom certificates and encryption keys. Local, web, and other configuration files remain intact. |
| Format File System | Formats the phone’s flash file system and deletes the UC Software application, log files, configuration, and override files.<br>Note that if the override file is stored on the provisioning server, the phone re-downloads the override file when you provision the phone again. Formatting the phone’s file system does not delete those device settings affecting network and provisioning, and any certificates and encryption keys remain on the phone. |
| Reset to Factory | Removes the Web and local override files, any stored configuration files in the flash file system, as well as any custom certificates and encryption keys.<br>All network and provisioning settings are reset but the UC Software application and updater remain intact. |

You can also use a multiple key combination to reset your phone to the factory defaults. Depending on your phone model, press and hold the following keys simultaneously during the updater/BootROM countdown process until the password prompt displays:

- VVX 1500: 4, 6, 8, and * dial pad keys

- VVX 300 and 400 series: dial pad keys 1, 3, and 5
- VVX 500 and 600 series: dial pad keys 1, 3, and 5

**Tip: Old reset behavior**

Before UC Software 4.0.0, this multiple key combination performed a device reset only, clearing the flash parameters and deleting all log files. Within the updater, this is still true.

Enter the administrator password to initiate the reset. Resetting to factory defaults also resets the administrator password (factory default password is 456). Polycom recommends that you change the administrative password from the default value.

**Settings: Resetting a VVX 1500 D to default values disables the H.323 protocol**

After you reset to factory defaults on a Polycom VVX 1500 D phone, you must re-enable the H.323 protocol through a configuration file change or by using the Web Configuration Utility. Refer to the section Configure H.323 Protocol.

## Update Log Files

Uploading the log files copies the log files from the phone to the provisioning server. The files called ***<MACaddress>-now-xxx.log*** are created.

You can use a multiple key press to upload log files to your provisioning server. Depending on your phone model, press and hold one the following keys simultaneously until you hear a confirmation tone for about three seconds:

- VVX 1500: Up, Down, Left, and Right arrow keys
- VVX 300 and 400 series: 1, 5, and 9
- VVX 500 and 600 series: dial pad keys 1, 5, and 9

## Set the Base Profile

Setting the base profile allows for quick setup of Polycom phones with Microsoft Lync Server 2010 and Skype for Business Server.

You can use a multiple key combination to set the base profile on a single Polycom phone. Depending on your phone model, press and hold the following keys simultaneously for about three seconds until you hear a confirmation tone:

- VVX 300 and 400 series: 1, 4, and 9 dial pad keys
- VVX 500 and 600 series: 1, 4, and 9 dial pad keys
- VVX 1500: 1, 4, and 9 dial pad keys

A login screen displays. Enter the administrator password (default 456) to initiate the setup. Polycom recommends that you change the administrative password from the default value.

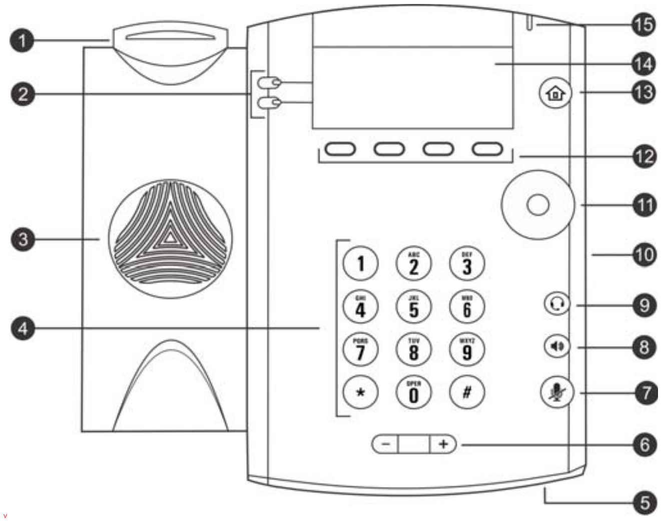
# Define the Phone Key Layout

You can redefine certain hard key functions using parameters in the configuration files. The following figures and tables show the default key layouts for the following phones:

- VVX 101 and 201 Default Phone Key Functions
- VVX 300 Series Default Phone Key Functions
- VVX 400 Series Default Phone Key Functions
- VVX 500 Series Default Phone Key Functions
- VVX 600 Series Default Phone Key Functions
- VVX 1500 Default Phone Key Functions

## VVX 101 and 201

The following figure and table show the available phone key functions.

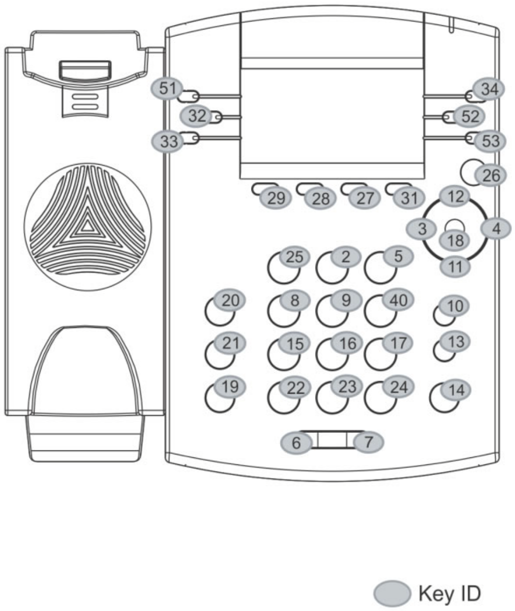


**VVX 101 and 201 Default Phone Key Functions**

| KEY ID | Function | KEY ID | Function |
|---|---|---|---|
| 1 | Hookswitch | 9 | Headset key |
| 2 | Line keys | 10 | Security slot (on side) |
| 3 | Speaker | 11 | Navigation keys / Select key |
| 4 | Dial pad keys | 12 | Soft keys |
| 5 | Microphone | 13 | Home key |
| 6 | Volume keys | 14 | Screen |
| 7 | Mute key | 15 | Message Waiting Indicator |
| 8 | Speakerphone key | | |

## VVX 300 Series

The following figure and table show the available phone key functions. IDs that have no function are described as n/a.



**VVX 300 Series Default Phone Key Functions**

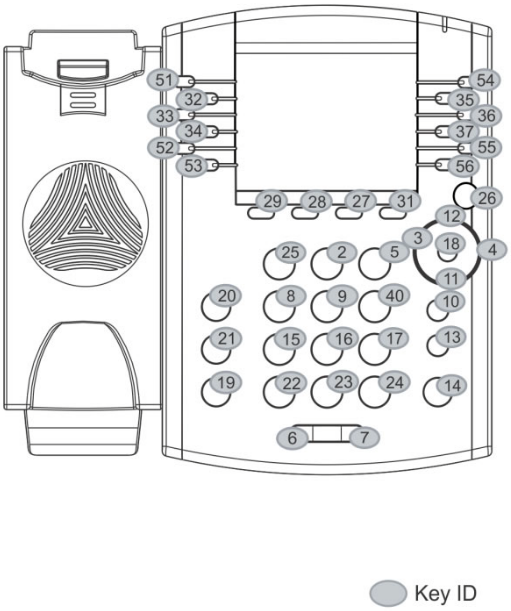
| KEY ID | Function | KEY ID | Function | KEY ID | Function | KEY ID | Function |
|---|---|---|---|---|---|---|---|
| 1 | n/a | 15 | Dialpad7 | 29 | SoftKey1 | 43 | n/a |
| 2 | Dialpad2 | 16 | Dialpad8 | 30 | n/a | 44 | n/a |
| 3 | ArrowLeft | 17 | Dialpad9 | 31 | SoftKey4 | 45 | n/a |
| 4 | ArrowRight | 18 | Select | 32 | Line2 | 46 | n/a |
| 5 | Dialpad3 | 19 | Hold | 33 | Line3 | 47 | n/a |
| 6 | VolDown | 20 | Transfer | 34 | Line4 | 48 | n/a |
| 7 | VolUp | 21 | Messages | 35 | n/a | 49 | n/a |
| 8 | Dialpad4 | 22 | DialpadStar | 36 | n/a | 50 | n/a |
| 9 | Dialpad5 | 23 | Dialpad0 | 37 | n/a | 51 | Line1 |
| 10 | Headset | 24 | DialpadPound | 38 | n/a | 52 | Line5 |
| 11 | ArrowDown | 25 | Dialpad1 | 39 | n/a | 53 | Line6 |
| 12 | ArrowUp | 26 | Home | 40 | Dialpad6 | | |

**VVX 300 Series Default Phone Key Functions**

| KEY ID | Function | KEY ID | Function | KEY ID | Function | KEY ID | Function |
|---|---|---|---|---|---|---|---|
| 13 | Handsfree | 27 | SoftKey3 | 41 | n/a | | |
| 14 | MicMute | 28 | SoftKey2 | 42 | n/a | | |

## VVX 400 Series

The following figure and table show the available phone key functions. IDs that have no function are described as n/a.



**VVX 400 Series Default Phone Key Functions**

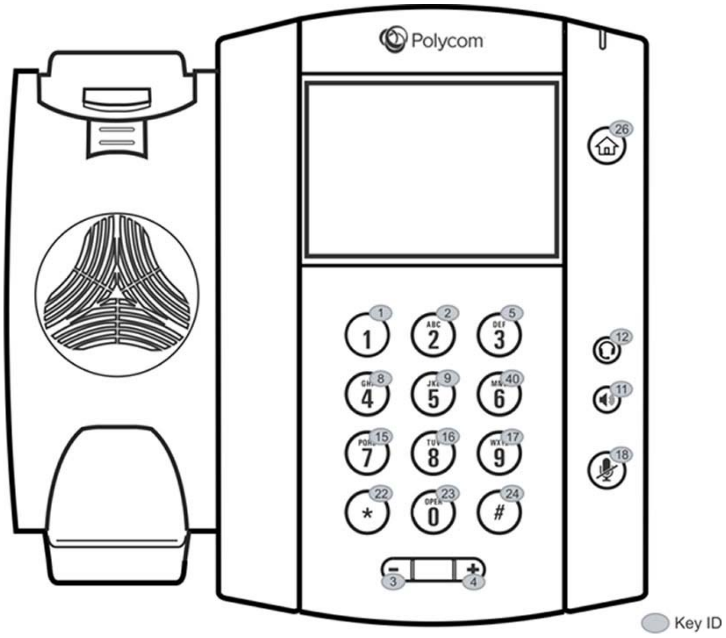
| KEY ID | Function | KEY ID | Function | KEY ID | Function | KEY ID | Function |
|---|---|---|---|---|---|---|---|
| 1 | n/a | 15 | Dialpad7 | 29 | SoftKey1 | 43 | n/a |
| 2 | Dialpad2 | 16 | Dialpad8 | 30 | n/a | 44 | n/a |
| 3 | ArrowLeft | 17 | Dialpad9 | 31 | SoftKey4 | 45 | n/a |
| 4 | ArrowRight | 18 | Select | 32 | Line2 | 46 | n/a |
| 5 | Dialpad3 | 19 | Hold | 33 | Line3 | 47 | n/a |
| 6 | VolDown | 20 | Transfer | 34 | Line4 | 48 | n/a |
| 7 | VolUp | 21 | Messages | 35 | Line8 | 49 | n/a |
| 8 | Dialpad4 | 22 | DialpadStar | 36 | Line9 | 50 | n/a |
| 9 | Dialpad5 | 23 | Dialpad0 | 37 | Line10 | 51 | Line1 |
| 10 | Headset | 24 | DialpadPound | 38 | n/a | 52 | Line5 |
| 11 | ArrowDown | 25 | Dialpad1 | 39 | n/a | 53 | Line6 |
| 12 | ArrowUp | 26 | Home | 40 | Dialpad6 | 54 | Line7 |

**VVX 400 Series Default Phone Key Functions**

| KEY<br>ID | Function | KEY<br>ID | Function | KEY<br>ID | Function | KEY<br>ID | Function |
|---|---|---|---|---|---|---|---|
| 13 | Handsfree | 27 | SoftKey3 | 41 | n/a | 55 | Line11 |
| 14 | MicMute | 28 | SoftKey2 | 42 | n/a | 56 | Line12 |

## VVX 500 Series

The following figure and table show the available phone key functions. IDs that have no function are described as n/a.

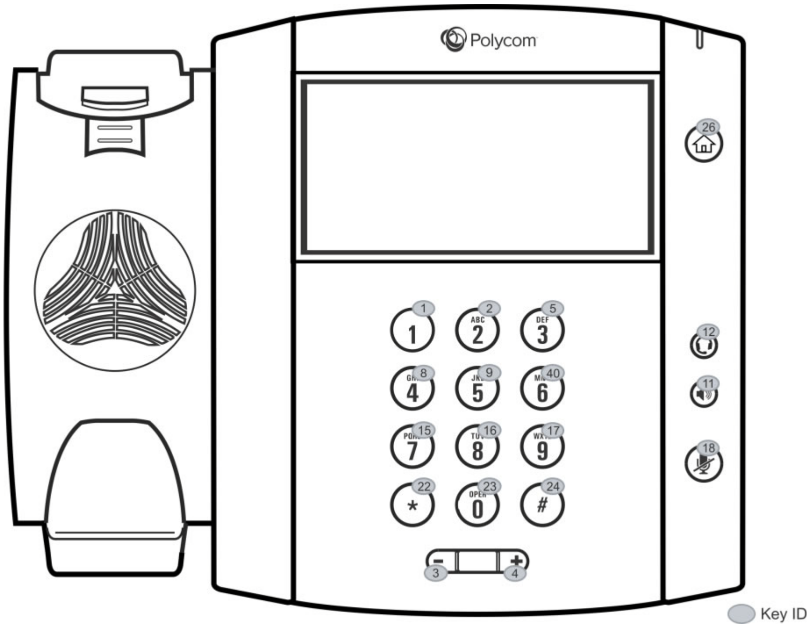


**VVX 500 Series Default Phone Key Functions**

| KEY ID | Function | KEY ID | Function | KEY ID | Function | KEY ID | Function |
|---|---|---|---|---|---|---|---|
| 1 | Dialpad1 | 12 | Headset | 23 | Dialpad0 | 34 | n/a |
| 2 | Dialpad2 | 13 | n/a | 24 | DialpadPound | 35 | n/a |
| 3 | VolDown | 14 | n/a | 25 | n/a | 36 | n/a |
| 4 | VolUp | 15 | Dialpad7 | 26 | Home | 37 | n/a |
| 5 | Dialpad3 | 16 | Dialpad8 | 27 | n/a | 38 | n/a |
| 6 | n/a | 17 | Dialpad9 | 28 | n/a | 39 | n/a |
| 7 | n/a | 18 | MicMute | 29 | n/a | 40 | Dialpad6 |
| 8 | Dialpad4 | 19 | n/a | 30 | n/a | 41 | n/a |
| 9 | Dialpad5 | 20 | n/a | 31 | n/a | 42 | n/a |
| 10 | n/a | 21 | n/a | 32 | n/a | | |
| 11 | Handsfree | 22 | DialpadStar | 33 | n/a | | |

## VVX 600 Series

The following figure and table show the available phone key functions. IDs that have no function are described as n/a.

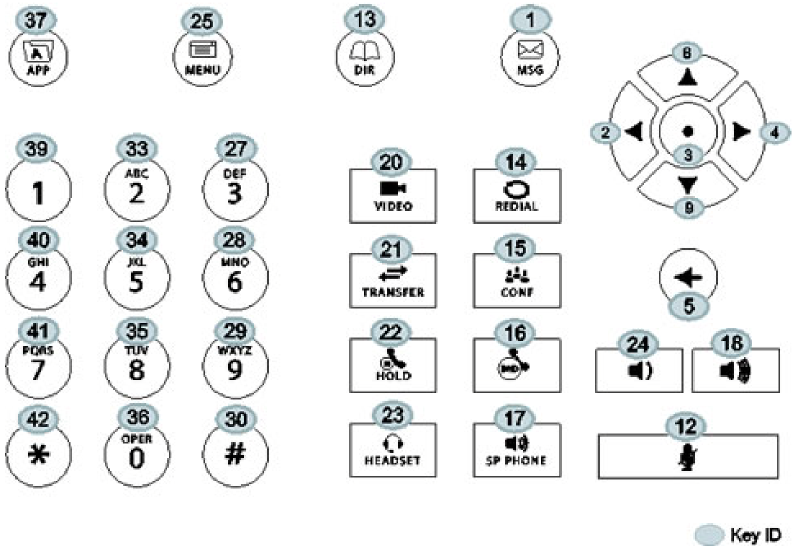


**VVX 600 Series Default Phone Key Functions**

| KEY ID | Function | KEY ID | Function | KEY ID | Function | KEY ID | Function |
|---|---|---|---|---|---|---|---|
| 1 | Dialpad1 | 12 | Headset | 23 | Dialpad0 | 34 | n/a |
| 2 | Dialpad2 | 13 | n/a | 24 | DialpadPound | 35 | n/a |
| 3 | VolDown | 14 | n/a | 25 | n/a | 36 | n/a |
| 4 | VolUp | 15 | Dialpad7 | 26 | Home | 37 | n/a |
| 5 | Dialpad3 | 16 | Dialpad8 | 27 | n/a | 38 | n/a |
| 6 | n/a | 17 | Dialpad9 | 28 | n/a | 39 | n/a |
| 7 | n/a | 18 | MicMute | 29 | n/a | 40 | Dialpad6 |
| 8 | Dialpad4 | 19 | n/a | 30 | n/a | 41 | n/a |
| 9 | Dialpad5 | 20 | n/a | 31 | n/a | 42 | n/a |
| 10 | n/a | 21 | n/a | 32 | n/a | | |
| 11 | Handsfree | 22 | DialpadStar | 33 | n/a | | |

## VVX 1500

The following figure and table show the available phone key functions. IDs that have no function are described as n/a.



**VVX 1500 Default Phone Key Functions**

| 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 | VolDown | 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 | | |

# Map Internal Key Functions

A complete list of internal key functions for enhanced feature keys and hard key mappings is shown in the table Key Labels and Internal Functions.

Note the following guidelines:

- The ***Function*** value is case sensitive.
- Some functions are dependent on call state. Generally, if the soft key displays on a call screen, the soft key function is executable.
- Some functions depend on the feature being enabled. For example, the BuddyStatus and MyStatus soft keys require the presence feature to be enabled.
- Hard key remappings do not require the enhanced feature key feature to be enabled. This includes the speed dial function on older platforms. On newer platforms, use line key functions.

The table below shows only line1 to line 6 functions.

**Key Labels and Internal Functions**

| Function | Description | Notes |
|---|---|---|
| ACDAvailable | ACD available from idle | |
| ACDLogin | Login to ACD | |
| ACDLogout | Log out of ACD | |
| ACDUnavailable | ACD unavailable from idle | |
| Answer | Answer | Call screen only |
| Applications | Main Browser | |
| ArrowDown | Move arrow down | |
| ArrowLeft | Move arrow left | |
| ArrowRight | Move arrow right | |
| ArrowUp | Move arrow up | |
| BargeIn | Barge In to show appearances, Barge In | Call screen only |
| BuddyStatus | Buddy Status | |
| Callers | Callers | |
| CallList | Call Lists | |
| CallPark | ParkEntry | Call screen only |
| CallPickup | Pickup a call | Call screen only |
| Conference | Begin a conference call | Call screen only |
| Delete | Delete | |
| Dialpad0 | Dialpad 0 | |
| Dialpad1 | Dialpad 1 | |
| Dialpad2 | Dialpad 2 | |

**Key Labels and Internal Functions**

| Function | Description | Notes |
|---|---|---|
| Dialpad3 | Dialpad 3 | |
| Dialpad4 | Dialpad 4 | |
| Dialpad5 | Dialpad 5 | |
| Dialpad6 | Dialpad 6 | |
| Dialpad7 | Dialpad 7 | |
| Dialpad8 | Dialpad 8 | |
| Dialpad9 | Dialpad 9 | |
| DialpadPound | Dialpad pound sign | |
| DialpadStar | Dialpad star sign | |
| DialpadURL | Dial name | Call screen only |
| DirectedPickup | Directed pickup | Call screen only |
| Directories | Directories | |
| Divert | Forward | |
| DoNotDisturb | Do Not Disturb menu | |
| EnterRecord | Enter a call record | Call screen only |
| Exit | Exit existing menu | Menu only |
| GroupPickup | Group pickup | |
| Handsfree | Use handsfree | |
| Headset | Use headset | Desktop phones only |
| Hold | Toggle hold | |
| Join | Join | Call screen only |
| LCR | Last call return | |
| 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 | |

**Key Labels and Internal Functions**

| Function | Description | Notes |
|---|---|---|
| LockPhone | Lock the phone | |
| Menu | Menu | |
| Messages | Messages menu | |
| MicMute | Mute the microphone | |
| MyStatus | View my status | |
| NewCall | New call | Call screen only |
| Null | Do nothing | |
| Offline | Offline for presence | |
| Page | Group Paging | |
| ParkedPickup | Parked pickup | Call screen only |
| QuickSetup | Quick Setup feature | Call screen only |
| Redial | Redial | Call screen only |
| Select | Select | |
| ServerACDAgentAvailable | serverACDAgentAvailable | |
| ServerACDAgentUnavailable | serverACDAgentUnavailable | |
| ServerACDSignIn | serverACDSignIn | |
| ServerACDSignOut | serverACDSignOut | |
| Setup | Settings menu | |
| Silence | RingerSilence | Call screen only |
| SoftKey1 | SoftKey 1 | |
| SoftKey2 | SoftKey 2 | |
| SoftKey3 | SoftKey 3 | |
| SoftKey4 | SoftKey 4 | |
| Softkey5 | Softkey 5 | |
| SpeedDial | SpeedDial | |
| Split | Split | Call screen only |
| Talk | Push-to-Talk | |
| Transfer | Transfer | Call screen only |
| Video | Video | Polycom VVX 500/501, 600/601, and 1500 |

**Key Labels and Internal Functions**

| Function | Description | Notes |
|---|---|---|
| VolDown | Set volume down | |
| VolUp | Set volume up | |

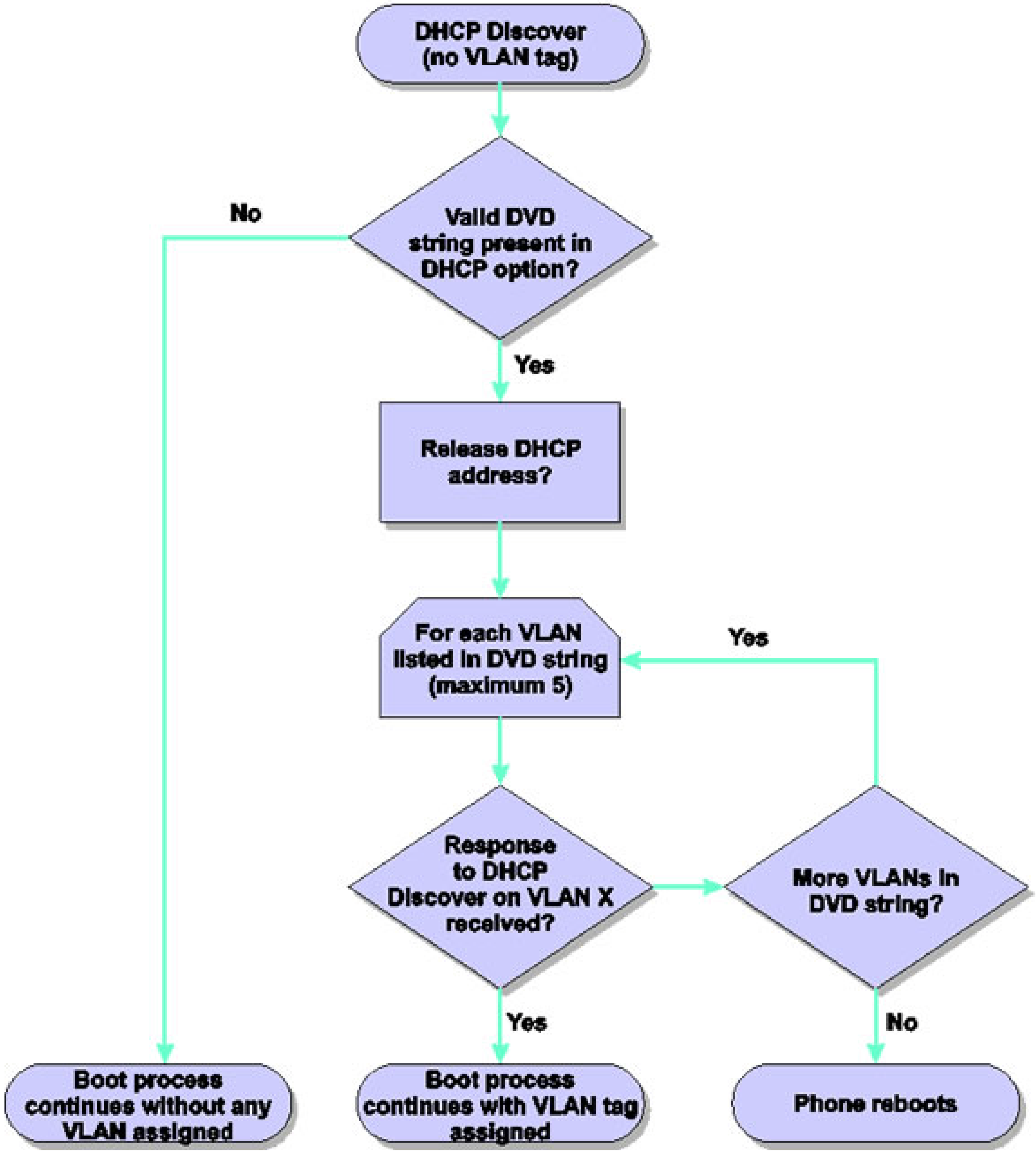
# Assign a VLAN ID Using DHCP

In deployments where is not possible or desirable to assign a virtual local area network (VLAN) statically in the phone’s network configuration menu or use Cisco Discovery Protocol (CDP) or Link-Layer Discovery Protocol (LLDP) to assign a VLAN ID, it is possible to assign a VLAN ID to the phone by distributing the VLAN ID via DHCP.

When using this method to assign the phone’s VLAN ID, the phone first boots on the default VLAN (or statically configured VLAN, if first configured in the phone’s network configuration menu), obtains its intended VLAN ID from the DHCP offer, then continues booting (including a subsequent DHCP sequence) on the newly obtained VLAN.

See the figure VLAN Using DHCP Phone Boot Up Sequence to understand the phone boot-up sequence when assigning a VLAN ID via DHCP.

**VLAN using DHCP phone boot-up sequence**


DHCP Discover (no VLAN tag)
Valid DVD string present in DHCP option?
No
Yes
Release DHCP address?
For each VLAN listed in DVD string (maximum 5)
Yes
Response to DHCP Discover on VLAN X received?
More VLANs in DVD string?
Yes
No
Boot process continues without any VLAN assigned
Boot process continues with VLAN tag assigned
Phone reboots

### 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 examines DHCP options 128,144, 157 and 191 in that order for a valid DVD string.

When set to Custom, a value set in the VLAN ID Option are 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 semi colon “;”
- All characters after the semi colon “;” are ignored
- There must be no white space before the semi colon “;”
- VLAN IDs may be decimal, hex, or octal

The following DVD strings result in the phone using VLAN 10:

```
VLAN-A=10;
VLAN-A=0x0a;
VLAN-A=012;
```

**Note: VLAN tags assigned by CDP or LLDP**

If a VLAN tag is assigned by CDP or LLDP, DHCP VLAN tags are ignored.

# Parse 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. RFC 2132 does not specify the format of this option's data, and can be defined by each vendor.

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).

The following example is a sample decode of a packet (DHCP Option 60) from a VVX 500/501:

```
3c 7a
```

- Option 60, length of Option data (part of the DHCP specification)

```
00 00 36 3d
```

- Polycom signature (always 4 octets)

```
75
```

- Length of Polycom data

```
01 07 50 6f 6c 79 63 6f 6d
```

- sub-option 1 (company), length, "Polycom"

```
02 0b 56 56 58 2d 56 56 58 5f 34 31 30
```

- sub-option 2 (part), length, "VVX-VVX_500/501"

```
03 10 33 31 31 31 2d 34 36 31 36 32 2d 30 30 31 2c 37
```

- sub-option 3 (part number), length, "3111-44500-001,7"

```
04 1e 53 49 50 2f 35 2e 32 2e 30 2e 58 58 58 58 2f 30 36 2d 41 75 67 2d 31 34
20 32 30 3a 35 35
```

- sub-option 4 (Application version), length, "SIP/5.2.0.XXXX/06-Aug-14 20:55"

```
05 1d 55 50 2f 35 2e 34 2e 30 2e 58 58 58 58 2f 30 36 2d 41 75 67 2d 31 34 20
32 31 3a 30 34
```

- sub-option 5 (Updater version), length, "UP/5.4.0.XXXX/06-Aug-14 21:04”

```
06 0c 64 73 6c 66 6f 72 75 6d 2e 6f 72 67
```

- sub-option 6 “dslforum.org”

# Disable the PC Ethernet Port

You can disable the Ethernet port and the PC Ethernet port on all devices from the phone interface.

### To disable Ethernet:

1. Navigate to the phone’s Ethernet Menu (**Menu > Settings > Advanced** (default password 456) > **Administration Settings > Network Configuration > Ethernet Menu**).
2. Scroll down to **PC Port Mode** and press the **Edit** soft key.
3. Select **Disabled** and press the **OK** soft key.
4. Press the **Exit** soft key and select **Save Config**.

   The phone reboots. When the reboot is complete, the PC Ethernet port is disabled.

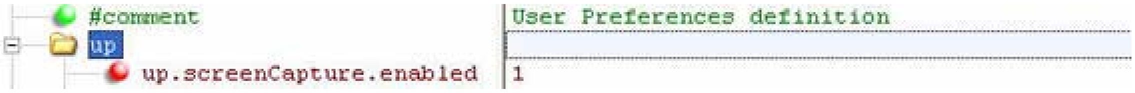
# Capture Your Device's Current Screen

You can capture your phone or expansion module's current screen using a web browser. Note that you must enable the phone's web server using the parameter `httpd.enabled` before you can take a screen capture. To capture the current screen of expansion modules, you must connect the expansion modules to a phone.

### To capture a device's current screen:

1. Modify your configuration file to enable the screen capture feature.

   Open your configuration file in an XML editor and add the following line:



```
#comment                          User Preferences definition
up
   up.screenCapture.enabled       1
```

2. Save the configuration file and update your device's configuration.
3. On the device, turn on the screen capture feature from the **Screen Capture** menu (**Menu > Settings > Basic > Preferences > Screen Capture**).

   Turn the screen capture on again (repeat this step) each time the device restarts or reboots.
4. In a web browser, enter https://<phoneIPaddress>/captureScreen as the browser address.

   To find your phone's IP address, navigate to **Menu > Status > Platform > Phone**.

   The web browser displays an image showing the phone's current screen. 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.

**Web Info: Using the LLDP protocol**
The protocol was formally ratified as IEEE standard 802.1AB in May 2005. Refer to section 10.2.4.4 of the LLDP-MED standard.

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.

**Settings: Enabling VLAN using multiple methods**
There are four ways to obtain VLAN on the phone and they can all be enabled, but the VLAN used is chosen by the priority of each method: 1. LLDP; 2. CDP; 3. Static (the VLAN ID is entered through the phone's user interface); 4. DVD (VLAN Via DHCP).

The following mandatory and optional 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 is recognized as LLDP only if it contains mandatory TLVs. Polycom phones running the UC Software 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.

## LLDP-MED Location Identification

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 phones do not have the capability to determine their physical location automatically or provision to a statically configured location. because of these limitations, the phones do 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.

## Supported TLVs

The basic TLV format is as follows:

- TLV Type (7 bits) [0-6]
- TLV Length (9 bits) [7-15]
- TLV Information (0-511 bytes)

The following table lists the supported TLVs.

**Supported TLVs**

| No | Name | Type (7 bits) [0-6] | Length (9 bits) [7-15] | Type Length | Org. Unique Code (3 bytes) | Sub Type |
|---|---|---|---|---|---|---|
| **1** | **Chassis-Id[1]** | **1** | **6** | **0x0206** | **-** | **5** |
| IP address of phone (4 bytes). Note that 0.0.0.0 is not 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 and Model Names. | | | | | | |
| **6** | **System description** | **6** | **min len > 0, max len <= 255** | **-** | **-** | **-** |
| Manufacturer's name - "Polycom"; 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.<br>Enabled Capabilities: Telephone and Bridge if phone has PC port support, it is not disabled and PC port is connected to PC. | | | | | | |
| **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. | | | | | | |
| **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<br>Note: After support for configuring location Identification information is locally available:<br>Capabilities - 0x37 (LLDP-Med capabilities, Network policy, Location Identification, Extended Power Via MDI-PD, Inventory) Class Type III | | | | | | |

**Supported TLVs**

<table>
<tr><th>No</th><th>Name</th><th>Type<br>(7 bits)<br>[0-6]</th><th>Length (9 bits) [7-15]</th><th>Type<br>Length</th><th>Org. Unique<br>Code (3<br>bytes)</th><th>Sub<br>Type</th></tr>
<tr><td>11</td><td>LLDP-MED network policy[2]</td><td>127</td><td>8</td><td>0xfe08</td><td>0x0012bb</td><td>2</td></tr>
<tr><td colspan="7">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</td></tr>
<tr><td>12</td><td>LLDP-MED network policy[2]</td><td>127</td><td>8</td><td>0xfe08</td><td>0x0012bb</td><td>2</td></tr>
<tr><td colspan="7">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.<br>**Note**: Voice signaling TLV is sent only if it contains configuration parameters that are different from voice parameters.</td></tr>
<tr><td>13</td><td>LLDP-MED network policy[2]</td><td>127</td><td>8</td><td>0xfe08</td><td>0x0012bb</td><td>2</td></tr>
<tr><td colspan="7">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.<br>**Note**: Video conferencing TLV is sent only from video-capable phones: VVX 500/501, 600/601, and 1500 business media phones.</td></tr>
<tr><td>14</td><td>LLDP-MED location identification[3]</td><td>127</td><td>min len > 0, max len <= 511</td><td>-</td><td>0x0012bb</td><td>3</td></tr>
<tr><td colspan="7">ELIN data format: 10 digit emergency number configured on the switch. Civic Address: physical address data such as city, street number, and building information.</td></tr>
<tr><td>15</td><td>Extended power via MDI</td><td>127</td><td>7</td><td>0xfe07</td><td>0x0012bb</td><td>4</td></tr>
<tr><td colspan="7">PowerType -PD device PowerSource-PSE&local Power Priority -Unknown PowerValue - Refer to Power Values.</td></tr>
<tr><td>16</td><td>LLDP-MED inventory hardware revision</td><td>127</td><td>min len > 0, max len <= 32</td><td>-</td><td>0x0012bb</td><td>5</td></tr>
<tr><td colspan="7">Hardware part number and revision</td></tr>
<tr><td>17</td><td>LLDP-MED inventory firmware revision</td><td>127</td><td>min len > 0, max len <= 32</td><td>-</td><td>0x0012bb</td><td>6</td></tr>
<tr><td colspan="7">BootROM revision</td></tr>
<tr><td>18</td><td>LLDP-MED inventory software revision</td><td>127</td><td>min len > 0, max len <= 32</td><td>-</td><td>0x0012bb</td><td>7</td></tr>
<tr><td colspan="7">Application (SIP) revision</td></tr>
</table>

**Supported TLVs**

| No | Name | Type<br>(7 bits)<br>[0-6] | Length (9 bits) [7-15] | Type Length | Org. Unique Code (3 bytes) | Sub Type |
|---|---|---|---|---|---|---|
| **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** |
| **22** | **LLDP-MED inventory asset ID** | **127** | **4** | **0xfe08** | **0x0012bb** | **11** |
| Empty (Zero length string) | | | | | | |
| **23** | **End of LLDP DU** | **0** | **0** | **0x0000** | **-** | **-** |

[1] For other subtypes, refer to IEEE 802.1AB, March 2005.
[2] For other application types, refer to TIA Standards 1057, April 2006.
[3] At this time, this TLV is not sent by the phone.

## System and Model Names

The following table outlines Polycom phone models, and their system and model names.

**Phone System and Model Names**

| Model | System Name | Model Name |
|---|---|---|
| VVX 101 | Polycom VVX 101 | VVX-VVX_101 |
| VVX 201 | Polycom VVX 201 | VVX-VVX_201 |
| VVX 300 | Polycom VVX 300 | VVX-VVX_300 |
| VVX 301 | Polycom VVX 301 | VVX-VVX_301 |
| VVX 310 | Polycom VVX 310 | VVX-VVX_310 |
| VVX 311 | Polycom VVX 311 | VVX-VVX_311 |
| VVX 400 | Polycom VVX 400 | VVX-VVX_400 |
| VVX 401 | Polycom VVX 401 | VVX-VVX_401 |
| VVX 410 | Polycom VVX 410 | VVX-VVX_410 |
| VVX 411 | Polycom VVX 411 | VVX-VVX_411 |
| VVX 500 | Polycom VVX 500 | VVX-VVX_500 |

**Phone System and Model Names**

| Model | System Name | Model Name |
|---|---|---|
| VVX 501 | Polycom VVX 501 | VVX-VVX_501 |
| VVX 600 | Polycom VVX 600 | VVX-VVX_600 |
| VVX 601 | Polycom VVX 601 | VVX-VVX_601 |
| VVX 1500 | Polycom VVX 1500 | VVX-VVX_1500 |
| SoundStructure | SoundStructure VoIP Interface | SoundStructure VoIP Interface |

## PMD Advertise and Operational MAU

The following table lists values for the PMD advertise and operational MAU.

**PMD Advertise and Operational MAU Type**

| 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: Default PMD advertise capability values**

By default, all phones have the PMD advertise capability set for 10HD, 10FD, 100HD, and 100FD bits. The VVX 310/311, 410/411, 500/501, 600/601, and 1500 have Gigabit Ethernet and support the PMD advertise capability set for 1000FD bit.

## Power Values

The following table outlines the power usage for each phone, as well as the power value sent in LLDP-MED.

### Power Consumption – Network Standby

In accordance with section 7 of the EU regulation 801/2013, Polycom provides the power consumption figures for its VoIP telephones when in their network standby state. To view this information, see Polycom Environment Compliance.

**Phone Power Values**

| Model | Power Usage (Watts) | Power Value Sent in LLDP-MED Extended Power Via MDI TLV |
|---|---|---|
| VVX 101 | 5.0 | 50 |
| VVX 201 | 5.0 | 50 |
| VVX 300 | 5.0 | 50 |
| VVX 301 | 5.0 | 50 |
| VVX 310 | 5.0 | 50 |
| VVX 311 | 5.0 | 50 |
| VVX 400 | 5.0 | 50 |
| VVX 401 | 5.0 | 50 |
| VVX 410 | 5.0 | 50 |
| VVX 411 | 5.0 | 50 |
| VVX 500 | 8.0 | 80 |
| VVX 501 | 8.0 | 80 |
| VVX 600 | 8.0 | 80 |
| VVX 601 | 8.0 | 80 |
| VVX 1500 | 11.8 | 118 |
| SoundStructure | 4.78 | na |

# Configuration Parameters

This section is a reference guide to the UC Software configuration parameters you use to configure devices and call controls. This section provides a description of each configuration parameter, and permitted and default values.

## <acd/>

The following table shows <acd/> parameters for the SIP-B automatic call distribution (ACD) and feature synchronized ACD features.

**Automatic Call Distribution Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **acd.reg**[1] | **1 to 34** | **1** |
| The index of the registration (line) used to support BroadSoft server-based ACD. | | |
| **acd.stateAtSignIn** | **0 or 1** | **1** |
| The state of the user when signing in. If 1, the user is available. If 0, the user is unavailable. | | |
| **acd.x.unavailreason.active**<br>If 1, the reason code is active. If 0, the code is inactive. | **0 or 1** | **0** |
| acd.x.unavailreason.codeValue[1]<br>The code value. For example, 1000100000 | String | Null |
| acd.x.unavailreason.codeName[1]<br>The code name. For example, Out to Lunch | string | Null |
| These three parameters configure the unavailable reason codes used for premium feature-synchronized ACD features, where x is the index of up to 100 codes. | | |

[1] Change causes phone to restart or reboot.

## <apps/>

The following table lists <apps/> parameters you can use to control telephone notification events, state polling events, and push server controls.

**Application Parameters**

| **Parameter** | **Permitted Values** | **Default** |
|---|---|---|
| **apps.push.alertSound** | **0 or 1** | **0** |
| If 0, there is no sound when an alert is pushed. If 1, there is sound. | | |
| **apps.push.messageType** | **0 to 5** | **0** |

**Application Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| Choose a priority level for push messages from the application server to the phone.<br>**1** (None) Discard push messages<br>**2** (Normal) Allows only normal push messages<br>**3** (Important) Allows only important push messages<br>**4** (High) Allows only priority push messages<br>**5** (Critical) Allows only critical push<br>**6** (All) Allows all push messages | | |
| **apps.push.password** | **string** | **null** |
| The password to access the push server URL. | | |
| **apps.push.secureTunnelEnabled** | **0 or 1** | **1** |
| If 0, the web server is not connected through a secure tunnel. If 1, the web server is connected through a secure tunnel. | | |
| **apps.push.secureTunnelPort** | **1 to 65535** | **443** |
| The port that the phone should use to communicate to the web server when the secure tunnel is used. | | |
| **apps.push.secureTunnelRequired** | **0 or 1** | **1** |
| If 0, communications to the web server do not require a secure tunnel. If 1, communications require a secure tunnel. | | |
| **apps.push.serverRootURL** | **URL** | **null** |
| The URL of the application server you enter here is combined with the phone address and sent to the phone's browser. 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 sent to the microbrowser is `http://172.24.128.85:8080/sampleapps/examples/sample.html`. You can use HTTP or HTTPS. | | |
| **apps.push.username** | **string** | **null** |
| The user name to access the push server URL. Note: To enable the push functionality, you must set values for the parameters `apps.push.username` and `apps.push.password` (not null). | | |
| **apps.statePolling.password** | **string** | **null** |
| Enter the password that the phone requires to authenticate phone state polling. | | |
| **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 parameters `apps.statePolling.URL`, `apps.statePolling.username`, and `apps.statePolling.password` must be set to non-null values. | | |
| **apps.statePoling.responseMode** | **0 or 1** | **1** |
| The mode of sending requested polled data. If 1, requested polled data is sent to a configured URL. If 0, the data is sent in the HTTP response. | | |
| **apps.statePolling.username** | **string** | **null** |
| Enter the user name that the phone requires to authenticate phone state polling. | | |

**Application Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **apps.telNotification.callStateChangeEvent** | **0 or 1** | **0** |
| If 0, call state change notification is disabled. If 1, notification is enabled. | | |
| **apps.telNotification.incomingEvent** | **0 or 1** | **0** |
| If 0, incoming call notification is disabled. If 1, notification is enabled. | | |
| **apps.telNotification.lineRegistrationEvent** | **0 or 1** | **0** |
| If 0, line registration notification is disabled. If 1, notification is enabled. | | |
| **apps.telNotification.offhookEvent** | **0 or 1** | **0** |
| If 0, off-hook notification is disabled. If 1, notification is enabled. | | |
| **apps.telNotification.onhookEvent** | **0 or 1** | **0** |
| If 0, on-hook notification is disabled. If 1, notification is enabled. | | |
| **apps.telNotification.outgoingEvent** | **0 or 1** | **0** |
| If 0, outgoing call notification is disabled. If 1, notification is enabled. | | |
| **apps.telNotification.URL** | **URL** | **null** |
| The URL to which the phone sends notifications of specified events. You can use HTTP or HTTPS. | | |
| **apps.telNotification.x.URL** | **URL** | **null** |
| The URL to which the phone sends notifications of specified events, where x 1 to 9. You can use HTTP or HTTPS. | | |
| **apps.telNotification.userLogInOutEvent** | **0 or 1** | **0** |
| If 0, user login/logout notification is disabled. If 1, notification is enabled. | | |
| **apps.ucdesktop.adminEnabled**[1] | **0 or 1** | **1** |
| If 0, the Polycom Desktop Connector is disabled on the administrative level. If 1, it is enabled on the administrative level. | | |
| **apps.ucdesktop.desktopUserName** | **string** | **null** |
| The user's name, supplied from the user's computer, for example, `bsmith`. | | |
| **apps.ucdesktop.enabled** | **0 or 1** | **0** |
| If 0, the Polycom Desktop Connector is disabled for users. If 1, it is enabled for users. | | |
| **apps.ucdesktop.orientation** | **Unspecified, Left, Right** | **Unspecified** |
| The location of the VVX 500/501 and 1500 with respect to the user's computer. For example, to the `Left` of the computer. | | |
| **apps.ucdesktop.ServerAddress** | **string** | **null** |
| The user's computer as a fully qualified domain name (FQDN), for example, computer@yourcompany.com. | | |

**Application Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **apps.ucdesktop.ServerPort** | **1 to 65535** | **24800** |
| The port number. Note: This value should be the same as the one that is used on the user's computer, otherwise the connection is not established. | | |

[1] Change causes phone to restart or reboot.

# <attendant/>

The busy lamp field (BLF)/attendant console feature enhances support for phone-based monitoring. The parameters listed in the following table are supported on the VVX 300 series, 400 series, 500 series, and 600 series phones. The maximum number of BLF entries for these phones is 50.

In the following table, x in a parameter is the number of the BLF entry in the list. If you are using static BLF, you need to configure the number of each entry.

**Attendant/Busy Lamp Field Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **attendant.reg**[1] | **positive integer** | **1** |
| The index of the registration to use to send a SUBSCRIBE to the list SIP URI specified in `attendant.uri`. For example, `attendant.reg = 2` means the second registration is used. | | |
| **attendant.ringType** | **default, ringer1 to ringer24** | **ringer1** |
| The ringtone to play when a BLF dialog is in the offering state. | | |
| **attendant.uri**[1] | **string** | **Null** |
| The list SIP URI on the server. If this is just a user part, the URI is constructed with the server hostname/IP.<br>Note: If this parameter is set, then the individually addressed users configured by `attendant.resourceList` and `attendant.behaviors` are ignored. | | |
| **attendant.behaviors.display.spontaneousCallAppearances.normal**[1]<br>**Normal** | **0 or 1** | **1** |
| **attendant.behaviors.display.spontaneousCallAppearances.automata**[1]<br>**Automatic** | **0 or 1** | **0** |
| If 1, the normal or automatic call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played). If 0, the call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter. Note that the values of these call appearance parameters depend on the values applied to `attendant.resourceList.x.type`. | | |
| **attendant.behaviors.display.remoteCallerID.normal**[1]<br>**Normal** | **0 or 1** | **1** |
| **attendant.behaviors.display.remoteCallerID.automata**[1]<br>**Automatic** | | |

**Attendant/Busy Lamp Field Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| These parameters depend on the value set for the parameter `attendant.resourceList.x.type`. If the parameter `attendant.resourceList.x.type` is set to normal, use the parameter `attendant.behaviors.display.remoteCallerID.normal`. If the parameter `attendant.resourceList.x.type` is set to automata, use the parameter `attendant.behaviors.display.remoteCallerID.automata`.<br>If 1, normal and automatic remote party caller ID information is presented to the attendant. If 0, the string `unknown` is substituted for both name and number information. | | |
| **attendant.resourceList.x.address**[1] | **string that constitutes a valid SIP URI (sip:6416@pol ycom.com) or contains the user part of a SIP URI (6416)** | **Null** |
| The user referenced by `attendant.reg=""` subscribes to this URI for dialog. If a user part is present, the phone subscribes to a sip URI constructed from the user part and domain of the user referenced by `attendant.reg`. | | |
| **attendant.resourceList.x.bargeInMode** | **All, Normal, Listen, Whisper** | |
| Enable or disable barge -in and choose the default barge-in mode.<br>Note: The default value for this parameter is empty. If no value is entered, the Barge In feature is disabled. | | |
| **attendant.resourceList.x.callAddress**[1] | **string** | **Null** |
| If the BLF call server is not at the same address as the BLF presence server, calls are sent to this address instead of the address specified by `attendant.resourceList.x.address`. | | |
| **attendant.resourceList.x.label** | **UTF-8 encoded string** | **Null** |
| The text label displays adjacent to the associated line key. If set to Null, the label is derived from the user part of `attendant.resourceList.x.address`. | | |
| **attendant.resourceList.x.proceedingIsRecipient** [1] | **0 or 1** | **0** |
| A flag to determine if pressing the associated line key for the monitored user picks up the call. | | |
| **attendant.resourceList.x.requestSilentBargeIn** | **0 or 1** | **0** |
| Enable or disable a tone that plays when a contact barges in on a call. If 0, a tone plays when a contact barges in on a call. If 1, no tone is played when a contact barges in on a call. | | |

**Attendant/Busy Lamp Field Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **attendant.resourceList.x.type** | **normal or automata** | **normal** |
| The type of resource being monitored and the default action to perform when pressing the line key adjacent to monitored user x.<br>If `normal`, the default action 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. Note that the value `normal` applies the call appearance setting `attendant.behaviors.display.*.normal`.<br>If `automata`, the default action 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. Note that the value `automata` applies the call appearance setting `attendant.behaviors.display.*.automata=0`. | | |

[1] Change causes phone to restart or reboot.

# <audioVideo/>

Use this parameter to toggle between audio-only or audio-video calls.

**Audio Video Persist Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **audioVideoToggle.callMode.persistent** | **0 or 1** | **0** |
| If 0, the phone returns to audio-only for outgoing calls. If 1, the last call mode the user selected persists until you change modes using the soft key toggle. Note that you must enable `feature.audioVideoToggle.enabled="1"` to apply this parameter. | | |

# <bg/>

The parameters listed in the following table control how you display background images.

**Background Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **bg.background.enabled** | **0 or 1** | **1** |
| Enable or disable the background image change feature. When set to 1, the user can set the background image on the phone screen. When set to 0, the user cannot set the background image of the phone screen and:<br>The background image option is not available on the phone menu or in the Web Configuration Utility when logged in as a user.<br>The icon to set the displayed image as a background in the picture frame menu does not display.<br>Administrators can change the background image using a configuration file or by logging into the Web Configuration Utility as an administrator. | | |
| **bg.color.selection** | **w,x** | **1,1** |

**Background Parameters  (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| Set the background. Specify which type of background (w) and index (x) for that type is selected on reboot. The default selection is 2,1 the first solid background.<br>Use w=1 and x=1 (1,1) to select the built-in image.<br>Use w=2 and x= 1 to 4 to select one of the four `solid` backgrounds.<br>Use w=3 and x= 1 to 6 to select one of the six background `bm` images<br>You can set backgrounds for specific phone models by adding the model name, for example:<br>`bg.color.VVX500.selection, bg.color.VVX1500.selection`<br>Note that although the VVX 300 series phones use a grayscale background, use this parameter to set the background. | | |
| **bg.color.bm.x.name**<br>**Phone screen background image file** | URL or file path of a BMP or JPEG image | |
| **bg.color.bm.x.em.name**<br>**Expansion module (EM) background image file** | URL or file path of a BMP or JPEG image | |
| The name of the image file (including extension).<br>Note: If the file is missing or unavailable, the built-in default solid pattern is displayed. | | |

# <bluetooth/>

The following table specifies the Bluetooth parameter for the VVX 600/601 phone.

**Bluetooth Radio Transmitter Parameter**

| Parameter | Permitted Values | Default |
|---|---|---|
| **bluetooth.pairedDeviceMemorySize** | **0 – 10** | **10** |

# <call/>

The phone supports an optional per-registration feature that enables automatic call placement when the phone is off-hook.

The phone supports a per-registration configuration that determines which events cause the missed-calls counter to increment.

You can enable/disable missed call tracking on a per-line basis.

In the following table, x is the registration number. To view the list of maximum registrations for each phone model see the table Flexible Call Appearances.

**Call Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **call.advancedMissedCalls.addToReceivedList** | **0 or 1** | **0** |
| Applies to calls on that are answered remotely. If 0, calls answered from the remote phone are not added to the local receive call list. If 1, calls answered from the remote phone are added to the local receive call list. | | |
| **call.advancedMissedCalls.enabled** | **0 or 1** | **1** |
| If 1, improved missed call handling for shared lines is enabled (shared lines can correctly count missed calls). If 0, the old missed call handling is used for shared lines (shared lines may not correctly count missed calls). | | |
| **call.advancedMissedCalls.reasonCodes** | **comma-separated list of indexes** | **200** |
| A comma separated list of reason code indexes that are interpreted to mean that a call should not be considered as a missed call. | | |
| **call.autoAnswer.H323** | **0 or 1** | **0** |
| You can use this parameter for the VVX 500/501, 600/601, and 1500. If 0, auto-answer is disabled for H.323 calls. If 1, auto-answer is enabled for all H.323 calls. | | |
| **call.autoAnswer.micMute** | **0 or 1** | **1** |
| If 0, the microphone is active immediately after a call is auto-answered. If 1, the microphone is initially muted after a call is auto-answered. | | |
| **call.autoAnswer.ringClass** | **see the list of ring classes in <rt/>.** | **ringAutoAnswer** |
| The ring class 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 are overridden such that a ringtone of `visual` (no ringer) applies. | | |
| **call.autoAnswer.SIP** | **0 or 1** | **0** |
| You can use this parameter on the VVX 300 series, 400 series, 500 series, 600 series, and 1500. If 0, auto-answer is disabled for SIP calls. If 1, auto-answer is enabled for all SIP calls. | | |
| **call.autoAnswer.videoMute** | **0 or 1** | **0** |
| You can use this parameter for the VVX 500/501, 600/601, and 1500. If 0, video begins transmitting (video Tx) immediately after a call is auto-answered. If 1, video transmission (video Tx) is initially disabled after a call is auto-answered. | | |
| **call.autoAnswer.enable** | **0 or 1** | **1** |
| If 1, the autoanswer menu displays and is available to the user to configure. If 0, the autoanswer menu is disabled and is not available to the user to configure. | | |

**Call Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **call.autoOffHook.x.enabled**[1]<br>**Enable or disable the feature** | **0 or 1** | **0** |
| **call.autoOffHook.x.contact**[1]<br>**The contact address to where the call is placed** | **a SIP URL** | **Null** |
| **call.autoOffHook.x.protocol**[1]<br>**The calling protocol to use** | **SIP or H323** | **Null** |
| If `enabled` is set to 0, no call is placed automatically when the phone goes off hook, and the other parameters are ignored. If enabled is set to 1, a call is automatically placed to the `contact` using the calling `protocol`, when the phone goes off hook.<br>Only the VVX 500/501, 600/601, and 1500 phones use the `protocol` parameter. If no protocol is specified, the phone uses the protocol specified by `call.autoRouting.preferredProtocol`. If a line is configured for a single protocol, the configured protocol is used.<br>The `contact` must be an ASCII-encoded string containing digits, either the user part of a SIP URL (for example, 6416), or a full SIP URL (for example, 6416@polycom.com). | | |
| **call.autoRouting.preferredProtocol** | **SIP or H323** | **SIP** |
| You can use this parameter for the VVX 500/501, 600/601, and 1500. 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. | | |
| **call.autoRouting.preference** | **line or protocol** | **line** |
| You can use this parameter for the VVX 500/501, 600/601, and 1500. 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 is used (`call.autoRouting.preferredProtocol`). If set to **protocol**, the first available line with the preferred protocol activated is used, if available. If not available, the first available line is used. Note: Auto-routing is used when manual routing selection features (`up.manualProtocolRouting`) are disabled. | | |
| **call.BlindTransferSpecialInterop** | **0 or 1** | **0** |
| Set the value to 1 to wait for an acknowledgment from the transferee before ending the call. | | |
| **call.callsPerLineKey** | **Varies by phone model** | **Varies by phone model** |
| Set the maximum number of concurrent calls per line key. This parameter applies to all registered lines. The maximum number of concurrent calls per line key varies by phone model and is listed for each phone in the column *Calls Per Line Key* in the table Flexible Call Appearances. For more information on all types of call appearances see the section Call Forward on Shared Lines. Note that this parameter may be overridden by the per-registration parameter of `reg.x.callsPerLineKey`. | | |
| **call.callWaiting.enable** | **0 or 1** | **1** |
| If 1, the phone alerts you to an incoming call while you are in an active call. If 0, you are not alerted to incoming calls while in an active call and the incoming call is treated as if you did not answer it. If 1, and you end the active call during a second incoming call, you are alerted to the second incoming call. | | |
| **call.callWaiting.ring**[1] | **beep, ring, silent** | **beep** |
| Specifies the ringtone of incoming calls when another call is active. If set to Null, the default value is beep. | | |

**Call Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **call.defaultTransferType** | **Consultative or Blind** | **Generic SIP = Consultative**<br>**Lync = Blind** |
| Set the transfer type the phone uses when transferring a call. If Blind, pressing the Transfer soft key immediately transfers the call to another party. If Consultative, pressing the Transfer soft key puts the call on hold while placing a new call to the other party.<br>The user can press and hold the Transfer soft key to change the transfer type temporarily. The user can also set the default transfer type by going to **Settings > Basic > Preferences > Default Transfer Type**. | | |
| **call.dialtoneTimeOut**[1] | **positive integer** | **60** |
| The time is seconds that a dial tone plays before a call is dropped. If set to 0, the call is not dropped. | | |
| **call.directedCallPickupMethod**[1] | **native or legacy** | **legacy** |
| Specifies how the phone performs a directed call pick-up from a BLF contact.<br>• **native** Indicates the phone uses a native protocol method (in this case SIP INVITE with the Replaces header).<br>• **legacy** Indicates the phone uses the method specified in `call.directedCallPickupString`. | | |
| **call.directedCallPickupString**[1] | **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.donotdisturb.perReg**[1] | **0 or 1** | **0** |
| This parameter determines if the do-not-disturb feature applies to all registrations on the phone (globally), or apply on a per-registration basis. If 0, DND applies to all registrations on the phone when it is active. If 1, the user can activate DND on a per-registration basis. Note: If `voIpProt.SIP.serverFeatureControl.dnd` is set to 1 (enabled), this parameter is ignored. | | |
| **call.enableOnNotRegistered**[1] | **0 or 1** | **1** |
| If 1, users can make calls when the phone is not registered. If 0, calls are not permitted without registration. Setting this parameter to 1 allows Polycom VVX 500/501, 600/601, and 1500 phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured. | | |
| **call.hold.localReminder.enabled**[1] | **0 or 1** | **0** |
| If 1, users are reminded of calls that have been on hold for an extended period of time. If 0, there is no hold reminder. | | |
| **call.hold.localReminder.period**[1] | **non-negative integer** | **60** |
| Specify the time in seconds between subsequent hold reminders. | | |
| **call.hold.localReminder.startDelay**[1] | **non-negative integer** | **90** |
| Specify a time in seconds to wait before the initial hold reminder. | | |
| **call.hold.remoteNotification.enabled** | **0 or 1** | **0** |

**Call Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **call.hold.remoteNotification.period** | **1 - 3600** | **60** |
| **call.hold.remoteNotification.startDelay** | **1 - 3600** | **60** |
| **call.internationalDialing.enabled** | **0 or 1** | **1** |
| Use this parameter to enable or disable the key tap timer that converts a double tap of the asterisk “*” symbol to the “+” symbol used to indicate an international call. By default, this parameter is enabled so that a quick double tap of “*” converts immediately to “+”. To enter a double asterisk “**”, tap “*” once and wait for the key tap timer to expire to enter a second “*”.<br>When you disable this parameter, you cannot dial”+” and you must enter the international exit code of the country you are calling from to make international calls.<br>Changes you make to this parameter cause a restart or reboot.<br>Note that this parameter applies to all numeric dial pads on the phone, including for example, the contact directory. | | |
| **call.lastCallReturnString**[1] | **string of maximum length 32** | ***69** |
| The string sent to the server when the user selects the last call return action. The string is usually a star code. | | |
| **call.localConferenceEnabled**[1] | **0 or 1** | **1** |
| If set to 1, on the VVX 300 series, 400 series, 500 series, and 600 series, the Conference and Join soft keys display during an active call and you can establish conferences on the phone.<br>If set to 0, on the VVX 300 series, 400 series, 500 series, and 600 series, the Conference and Join soft keys do not display during an active call.<br>If set to 0, and you press the Conference hard key on the VVX 1500, an ‘Unavailable’ message displays. | | |
| **call.missedCallTracking.x.enabled**[1] | **0 or 1** | **1** |
| If set to 1, missed call tracking is enabled.<br>If `call.missedCallTracking.x.enabled` is set to 0, then missed call 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.<br>If `call.missedCallTracking.x.enabled` is set to 1 and call.serverMissedCalls.x.enabled is set to 0, then the number of missed calls is incremented regardless of how the server is configured.<br>If `call.missedCallTracking.x.enabled` is set to 1 and `call.serverMissedCalls.x.enabled` is set to 1, then the handling of missed calls depends on how the server is configured. | | |
| **call.offeringTimeOut**[1] | **positive integer** | **60** |
| Specify a time in seconds that an incoming call rings before the call is dropped, 0=infinite.<br>Note**:** The call diversion, no answer feature takes precedence over this feature if enabled. | | |
| **call.parkedCallRetrieveMethod**[1] | **native or legacy** | **Null** |
| The method the phone uses to retrieve a BLF resource's call which has dialog state confirmed. **native** indicates the phone uses a native protocol method (in this case SIP INVITE with the Replaces header). **legacy** indicates the phone uses the method specified in `call.parkedCallRetrieveString`. | | |

**Call Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **call.parkedCallRetrieveString**[1] | **star code** | ***88** |
| The star code used to initiate retrieval of a parked call. | | |
| **call.rejectBusyOnDnd**[1] | **0 or 1** | **1** |
| If 1, and DND is turned on, the phone rejects incoming calls with a busy signal. If set to 0, and DND is turned on, the phone gives a visual alert of incoming calls and no audio ringtone alert.<br>Note: This parameter does not apply to shared lines since not all users may want DND enabled. | | |
| **call.ringBackTimeOut**[1] | **positive integer** | **60** |
| Specify a time in seconds to allow an outgoing call to remain in the ringback state before dropping the call, 0=infinite. | | |
| **call.serverMissedCall.x.enabled**[1] | **0 or 1** | **0** |
| If 0, all missed-call events 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 BroadSoft Synergy call server only (previously known as Sylantro). | | |
| **call.shared.disableDivert**[1] | **0 or 1** | **1** |
| If set to 1, the diversion feature for shared lines is disabled. Note: This feature is disabled on most call servers. | | |
| **call.shared.exposeAutoHolds**[1] | **0 or 1** | **0** |
| If 1, a re-INVITE is sent to the server when setting up a conference on a shared line. If 0, no re-INVITE is sent to the server. | | |
| **call.shared.oneTouchResume**[1] | **0 or 1** | **0** |
| If set to 1, all users on a shared line can resume held calls by pressing the shared line key. If more than one call is on hold, the first held call is selected and resumed.<br>If set to 0, selecting the shared line opens all current calls that the user can choose from.<br>A quick press and release of the line key resumes a call whereas pressing and holding down the line key shows a list of calls on that line. | | |
| **call.shared.remoteActiveHoldAsActive** | **0 or 1** | **1** |
| If 1, shared remote active/hold calls are treated as a active call on the phone. If 0, shared remote active/hold calls are not treated as a active call on the phone. | | |
| **call.shared.seizeFailReorder**[1] | **0 or 1** | **1** |
| If set to 1, play re-order tone locally on shared line seize failure. | | |
| **call.singleKeyPressConference** | **0 or 1** | **0** |
| If set to 1, a conference is initiated when 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.<br>If set to 0, sound effects are heard only by the conference initiator. | | |
| **call.stickyAutoLineSeize**[1] | **0 or 1** | **0** |

**Call Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| If set to 1, the phone uses sticky line seize behavior. This helps with features that need a second call object to work with. The phone attempts 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 uses the line index for the previous call. Dialing through the call list when there is an active call uses the current active call line index. Dialing through the contact directory uses the current active call line index.<br>If set to 0, the feature is disabled (this was the behavior in SIP 1.6.6). Dialing through the call list uses the line index for the previous call. Dialing through the contact directory uses a random line index.<br>Note: This may fail due to glare issues in which case the phone may select a different available line for the call. | | |
| **call.stickyAutoLineSeize.onHookDialing**[1] | **0 or 1** | **0** |
| If `call.stickyAutoLineSeize` is set to 1, this parameter has no effect. The regular stickyAutoLineSeize behavior is followed.<br>If `call.stickyAutoLineSeize` is set to 0 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.)<br>If `call.stickyAutoLineSeize` is set to 0 and this parameter is set to 0, there is no difference between hot dial and new call scenarios.<br>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.transferOnConferenceEnd**[1] | **0 or 1** | **1** |
| The behavior when the conference host exits a conference. If 0, all parties are disconnected when the conference host exits the conference. If 1, the other parties are left connected when the host exits the conference (the host performs an attended transfer to the other parties). | | |
| **call.urlModeDialing**[1] | **0 or 1** | **0** |
| If 0, URL dialing is disabled. If 1, URL dialing is enabled. | | |

[1] Change causes phone to restart or reboot.

# <callLists/>

The call lists (or call log) parameters listed in the following table are supported on VVX 300 series, 400 series, 500 series, 600 series, and 1500 phones.

**Call List (Call Log) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **callLists.collapseDuplicates** | **0 or 1** | **1** |
| If 0, all calls are archived and presented in the call lists. If 1, consecutive incomplete between the same party in the same direction (outgoing/incoming) are collapsed into one record with the most recent call displaying. | | |
| **callLists.logConsulationCalls** | **0 or 1** | **0** |
| If 1, all consultation calls are logged. (Calls made to a third party—while the original party is on hold—when settings up a conference call are called consultation calls.)<br>If 0, consultation calls are not logged. | | |

**Call List (Call Log) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **callLists.size** | **10 to 99** | **99** |
| The maximum number of retained records of each type (incoming, outgoing, and missed). When the maximum number is reached, new records overwrite existing records. You can clear the list using the phone's menu system. If you want to prevent the records from uploading to the provisioning server, enter a false URL in the CALL_LISTS_DIRECTORY field in the master configuration file. | | |
| **callLists.writeDelay.journal** | **1 to 600** | **5** |
| The delay (in seconds) before changes due to an in-progress call are flushed to the file system as a journal. | | |
| **callLists.writeDelay.terminated** | **10 to 600** | **60** |
| The minimum period between writing out the complete XML file to the local file system and, optionally, to the provisioning server. | | |

# <device/>

The <device/> parameters—also known as device settings—contain default values that you can use to configure basic settings for multiple phones.

**Web Info: Default device parameter values**

The default values for the <device/> parameters are set at the factory when the phones are shipped. For a list of the default values, see the latest Product Shipping Configuration Change Notice at Polycom Engineering Advisories and Technical Notifications.

Polycom provides a global device.set parameter that you must enable to install software and change device parameters. In addition, each <device/> parameter has a corresponding .set parameter that enables or disables the value for that device parameter. You need to enable the corresponding .set parameter for each parameter you want to apply.

After you complete the software installation or configuration changes to device parameters, remove device.set to prevent the phones from rebooting and triggering a reset of device parameters that phone users might have changed after the initial installation.

If you configure any parameter values using the <device/> parameters, any subsequent configuration changes you make from the Web Configuration Utility or phone local interface do not take effect after a phone reboot or restart.

The <device/> parameters are designed to be stored in flash memory, and are therefore not added to the <MAC>-web.cfg or <MAC>-phone.cfg override files whether the changes are made through the web interface or the phone interface. This design protects the ability to manage and access the phones using the standard set of parameters on a provisioning server after the initial installation.

## .set Parameter Exception

Each <device/> parameter has a corresponding .set parameter that enables or disables the parameter. There is one exception to this rule: the device.sec.TLS.customDeviceCertX.set parameter applies to device.sec.TLS.customDeviceCertX.publicCert and to device.sec.TLS.customDeviceCertX.privateKey.

**Settings: Each <device/> parameter has a corresponding .set parameter with one exception**

Note that each `<device/>` parameter has a corresponding `.set` parameter that enables or disables the parameter. There is one exception to this rule: the `device.sec.TLS.customDeviceCertX.set` parameter applies to `device.sec.TLS.customDeviceCertX.publicCert` and to `device.sec.TLS.customDeviceCertX.privateKey`.

## Use Caution When Changing Device Parameters

Use caution when changing `<device/>` parameters as incorrect settings may apply the same IP address to multiple phones.

Note that some parameters may be ignored. For example, if DHCP is enabled it will still override the value set with `device.net.ipAddress`.

Though individual parameters are checked to see whether they are in range, the interaction between parameters is not checked. If a parameter is out of range, an error message displays in the log file and parameter will not be used.

Incorrect configuration can put the phones into a reboot loop. For example, server A has a configuration file that specifies that server B should be used, and server B has a configuration file that specifies that server A should be used.

To detect errors, including IP address conflicts, Polycom recommends that you test the new configuration files on two phones before initializing all phones.

## Types of Device Parameters

The following table outlines the three types of <device/> parameters, their permitted values, and the default value.

### Device Parameter Types

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.set**[1] | **0 or 1** | **0** |
| If set to 0, do not use any `device.xxx` fields to set any parameters. Set this to 0 after the initial software installation.<br>If set to 1, use the `device.xxx` fields that have `device.xxx.set=1`. Set this to 1 only for the initial software installation. | | |
| **device.xxx**[1] | **string** | |
| Configuration parameter. | | |
| **device.xxx.set**[1] | **0 or 1** | **0** |
| If set to 0, do not use the `device.xxx` value. If set to 1, use the `device.xxx` value.<br>For example, if `device.net.ipAddress.set=1`, then use the value set for `device.net.ipAddress`. | | |

[1] Change causes phone to restart or reboot

The following table lists each of the <device/> parameters that you can configure.

**Device Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.auth.localAdminPassword** | **string (32 character max)** | |
| The phone's local administrative password. The minimum length is defined by `sec.pwd.length.admin<XREF>`. | | |
| **device.auth.localUserPassword** | **string (32 character max)** | |
| The phone user's local password. The minimum length is defined by `sec.pwd.length.user<XREF>`. | | |
| **device.auxPort.enable**[1] | **0 or 1** | **1** |
| Enable or disable the phone auxiliary port. | | |
| **device.baseProfile** | **Generic, Lync** | |
| Choose the Base Profile that the phone operates with. | | |
| **device.dhcp.bootSrvOpt**[1] | **Null, 128 to 254** | |
| When the boot server is set to Custom or Custom+Option66, specify the numeric DHCP option that the phone looks for. | | |
| **device.dhcp.bootSrvOptType**[1] | **IP address or string** | |
| The type of DHCP option the phone looks for its provisioning server (if `device.dhcp.bootSrvUseOpt` is set to `Custom`). If IP, the IP address provided must specify the format of the provisioning server. If String, the string provided must match one of the formats specified by `device.prov.serverName`. | | |
| **device.dhcp.bootSrvUseOpt**[1] | **Default, Custom, Static, CustomAndDefault** | |
| **Default** The phone looks 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 `device.prov.serverName`.<br>**Custom** The phone looks for the option number specified by `device.dhcp.bootSrvOpt`, and the type specified by `device.dhcp.bootSrvOptType` in the response received from the DHCP server.<br>**Static** The phone uses the boot server configured through the provisioning server `device.prov.*` parameters.<br>**Custom** and **Default** The phone uses the custom option first or use Option 66 if the custom option is not present. | | |
| **device.dhcp.enabled**[1] | **0 or 1** | |
| If 0, DHCP is disabled. If 1, DHCP is enabled. | | |
| **device.dhcp.option60Type**[1] | **Binary, ASCII** | |
| The DHCP option 60 type. `Binary:` vendor-identifying information is in the format defined in RFC 3925. `ASCII`: vendor-identifying information is in ASCII format. | | |
| **device.dhcp.dhcpVlanDiscUseOpt**[1] | **Disabled, Fixed, Custom** | |

**Device Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| VLAN Discovery. Disabled, no VLAN discovery through DHCP. Fixed, use predefined DHCP vendor-specific option values of 128, 144, 157 and 191 (device.dhcp.dhcpVlanDiscOpt is ignored). Custom, use the number specified by device.dhcp.dhcpVlanDiscOpt. | | |
| **device.dhcp.dhcpVlanDiscOpt[1]** | **128 to 254** | |
| The DHCP private option to use when device.dhcp.dhcpVlanDiscUseOpt is set to Custom. | | |
| **device.dns.altSrvAddress[1]** | **server address** | |
| The secondary server to which the phone directs domain name system (DNS) queries. | | |
| **device.dns.domain[1]** | **string** | |
| The phone's DNS domain. | | |
| **device.dns.serverAddress[1]** | **string** | |
| The primary server to which the phone directs DNS queries. | | |
| **device.host.hostname[1]** | **string** | |
| This parameter enables you to specify a hostname for the phone when using DHCP by adding a hostname string to the phone's configuration. If device.host.hostname.set=1, and device.host.hostname=Null, the DHCP client uses Option 12 to send a predefined hostname to the DHCP registration server using Polycom_<MACaddress>. Note that the maximum length of the hostname string is <=255 bytes. The valid character set is defined in RFC1035. | | |
| **device.lync.timeZone** | **0 or 1** | **Lync = 1** |
| If 1, the sign-in wizard asks to set the time zone. | | |
| **device.net.cdpEnabled[1]** | **0 or 1** | |
| If set to 1, the phone attempts to determine its VLAN ID and negotiate power through CDP. | | |
| **device.net.dot1x.anonid[1]** | **string** | |
| EAP-TTLS and EAP-FAST only. The anonymous identity (user name) for 802.1X authentication. | | |
| **device.net.dot1x.enabled[1]** | **0 or 1** | |
| If 0, 802.1X authentication is disabled. If 1, 802.1X authentication is enabled. | | |
| **device.net.dot1x.identity[1]** | **string** | |
| The identity (user name) for 802.1X authentication. | | |
| **device.net.dot1x.method** | **EAP-None, EAP-TLS, EAP-PEAPv0-MSCHAPv2, EAP-PEAPv0-GTC, EAP-TTLS-MSCHAPv2, EAP-TTLS-GTC, EAP-FAST, EAP-MD5** | |
| Specify the 802.1X authentication method, where EAP-NONE means no authentication. | | |

**Device Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.net.dot1x.password**[1] | **string** | |
| The password for 802.1X authentication. This parameter is required for all methods except EAP-TLS. | | |
| **device.net.etherModeLAN**[1] | **Auto, 10HD, 10FD, 100HD, 100FD, 1000FD** | |
| The LAN port mode that sets the network speed over Ethernet. HD means half-duplex and FD means full duplex. Note: Polycom recommends that you do not change this setting. | | |
| **device.net.etherModePC**[1] | **Disabled, Auto, 10HD, 10FD, 100HD, 100FD, 1000FD** | **Auto** |
| The PC port mode that sets the network speed over Ethernet. If set to `Disabled`, the PC port is disabled. HD means half duplex and FD means full duplex. | | |
| **device.net.etherStormFilter**[1] | **0 or 1** | |
| If 1, DoS storm prevention is enabled and received Ethernet packets are filtered to prevent TCP/IP stack overflow caused by bad data or too much data. If 0, DoS storm prevention is disabled. | | |
| **device.net.etherVlanFilter**[1] | **0 or 1** | |
| VLAN filtering for VVX phones is done by the Linux operating system and it cannot be disabled. | | |
| **device.net.ipAddress**[1] | **string** | |
| The phone's IP address. Note: This parameter is disabled when DHCP is enabled (`device.dhcp.enabled` is set to 1. | | |
| **device.net.IPgateway**[1] | **IP address** | |
| The phone's default router. | | |
| **device.net.lldpEnabled**[1] | **0 or 1** | |
| If set to 1, the phone attempts to determine its VLAN ID and negotiate power through LLDP. | | |
| **device.net.subnetMask**[1] | **subnet mask** | |
| The phone's subnet mask. Note: This parameter is disabled when DHCP is enabled (`device.dhcp.enabled` is set to 1). | | |
| **device.net.vlanId**[1] | **Null, 0-4094** | |
| The phone's 802.1Q VLAN identifier. If Null, no VLAN tagging. | | |
| **device.prov.maxRedunServers**[1] | **1 to 8** | |
| The maximum number of IP addresses to use from the DNS. | | |
| **device.prov.password**[1] | **string** | |
| The password for the phone to log in to the provisioning server. Note that a password may not be required. Note: If you modify this parameter, the phone re-provisions. The phone may also reboot if the configuration on the provisioning server has changed. | | |

**Device Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.prov.redunAttemptLimit**[1] | **1 to 10** | |
| The maximum number of attempts to attempt a file transfer before the transfer fails.<br>When multiple IP addresses are provided by DNS, 1 attempt is considered to be a request sent to each server. | | |
| **device.prov.redunInterAttemptDelay**[1] | **0 to 300** | |
| The number of seconds to wait after a file transfer fails before retrying the transfer. When multiple IP addresses are returned by DNS, this delay only occurs after each IP has been tried. | | |
| **device.prov.serverName** | **IP address, domain name string, or URL** | |
| The IP address, domain name, or URL of the provisioning server, followed by an optional directory and optional configuration filename. This parameter is used if DHCP is disabled (`device.dhcp.enabled` is `0`), if the DHCP server does not send a boot server option, or if the boot server option is static (`device.dhcp.bootSrvUseOpt` is `static`). Note: If you modify this parameter, the phone re-provisions. The phone may also reboot if the configuration on the provisioning server has changed. | | |
| **device.prov.serverType**[1] | **FTP, TFTP, HTTP, HTTPS, FTPS** | |
| The protocol the phone uses to connect to the provisioning server. Note: Active FTP is not supported for BootROM version 3.0 or later. Note: Only implicit FTPS is supported. | | |
| **device.prov.upgradeServer** | **string** | |
| A browser-based Software Upgrade button that enables the user to upgrade the phone with a compatible software version available on the Polycom provisioning server. | | |
| **device.prov.tagSerialNo** | **0 or 1** | |
| If 0, the phone's serial number (MAC address) is not included in the User-Agent header of HTTPS/HTTPS transfers and communications to the microbrowser and web browser. If 1, the phone's serial number is included. | | |
| **device.prov.user** | **string** | |
| The user name required for the phone to log in to the provisioning server (if required). Note: If you modify this parameter, the phone re-provisions. The phone may also reboot if the configuration on the provisioning server has changed. | | |
| **device.prov.ztpEnabled** | **0 or 1** | |
| If 0, Disable the ZTP feature. If 1, enable the ZTP feature. For information, see Polycom Zero Touch Provisioning Solution. | | |
| **device.sec.configEncryption.key**[1] | **string** | |
| The configuration encryption key used to encrypt configuration files. For more information, see the section Encrypt Configuration Files. | | |
| **device.sec.coreDumpEncryption.enabled** | **0 or 1** | **1** |
| This parameter enables you to bypass the encryption of the core dump. When set to 1, the core dump is encrypted. When set to 0, encryption of the core dump is bypassed. | | |

**Device Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.sec.TLS.customCaCert1 (TLS Platform Profile 1)**<br>**device.sec.TLS.customCaCert2 (TLS Platform Profile 2)** | **string, PEM format** | |
| The custom certificate to use for TLS Platform Profile 1 and TLS Platform Profile 2 and TLS Application Profile 1 and TLS Application Profile 2 `device.sec.TLS.profile.caCertList` must be configured to use a custom certificate. Custom CA certificate cannot exceed 4096 bytes total size. | | |
| **device.sec.TLS.customDeviceCert1.publicCert**<br>**device.sec.TLS.customDeviceCert2.publicCert** | **Enter the signed custom device certificate in PEM format (X.509)** | |
| **device.sec.TLS.customDeviceCert1.privateKey**<br>**device.sec.TLS.customDeviceCert2.privateKey** | **Enter the corresponding signed private key in PEM format (X.509)** | |
| **device.sec.TLS.customDeviceCert1.set**<br>**device.sec.TLS.customDeviceCert2.set** | **0 or 1** | **0** |
| Note that you use a single `.set` parameter to enable or disable only these two related `<device/>` parameters - `device.sec.TLS.customDeviceCertX.publicCert` and `device.sec.TLS.customDeviceCertX.privateKey.` All other `<device/>` parameters have their own corresponding `.set` parameter that enables or disables that parameter.<br>Size constraints are: 4096 bytes for the private key, 8192 bytes for the device certificate. | | |
| **device.sec.TLS.profile.caCertList1 (TLS Platform Profile 1)**<br>**device.sec.TLS.profile.caCertList2 (TLS Platform Profile 2)** | **Builtin, BuiltinAndPlatform1, BuiltinAndPlatform2, All, Platform1, Platform2, Platform1AndPlatform2** | |
| Choose the CA certificate(s) to use for TLS Platform Profile 1 and TLS Platform Profile 2 authentication:<br>The built-in default certificate<br>The built-in and Custom #1 certificates<br>The built-in and Custom #2 certificates<br>Any certificate (built in, Custom #1 or Custom #2)<br>Only the Custom #1 certificate<br>Only the Custom #2 certificate<br>Either the Custom #1 or Custom #2 certificate | | |
| **device.sec.TLS.profile.cipherSuite1 (TLS Platform Profile 1)**<br>**device.sec.TLS.profile.cipherSuite2 (TLS Platform Profile 2)** | **string** | |
| The cipher suites to use for TLS Platform Profile 1 and TLS Platform Profile 2) | | |
| **device.sec.TLS.profile.cipherSuiteDefault1 (TLS Platform Profile 1)**<br>**device.sec.TLS.profile.cipherSuiteDefault2 (TLS Platform Profile 2)** | **0 or 1** | |
| The cipher suite to use for TLS Platform Profile 1 and TLS Platform profile 2. If set to 0, the custom cipher suite is used. If set to 1, the default cipher suite is used. | | |

**Device Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.sec.TLS.profile.deviceCert1 (TLS Platform Profile 1)**<br>**device.sec.TLS.profile.deviceCert2 (TLS Platform Profile 2)** | **Builtin, Platform1, Platform2** | |
| Choose the device certificate(s) for TLS Platform Profile 1 and TLS Platform Profile 2 to use for authentication. | | |
| **device.sec.TLS.profile.profileSelection.dot1x** | **PlatformProfile1, PlatformProfile2** | |
| Choose the TLS Platform Profile to use for 802.1X, either TLS Platform Profile 1 or TLS Platform Profile 2. | | |
| **device.sec.TLS.profileSelection.provisioning[1]** | **PlatformProfile1, PlatformProfile2** | |
| The TLS Platform Profile to use for provisioning, either TLS Platform Profile 1 or TLS Platform Profile 2. | | |
| **device.sec.TLS.profileSelection.syslog[1]** | **PlatformProfile1, PlatformProfile2** | |
| The TLS Platform Profile to use for syslog, either TLS Platform Profile 1 or TLS Platform Profile 2. | | |
| **device.sec.TLS.prov.strictCertCommonNameValidation** | **0 or 1** | **1** |
| If set to 1, provisioning always verifies the server certificate for commonName/SubjectAltName match with the server hostname that the phone is trying to connect. | | |
| **device.sec.TLS.syslog.strictCertCommonNameValidation** | **0 or 1** | **1** |
| If set to 1, syslog always verifies the server certificate for commonName/SubjectAltName match with the server hostname that the phone is trying to connect. | | |
| **device.sntp.gmtOffset** | **-43200 to 46800** | |
| The GMT offset—in seconds—to use for daylight savings time, corresponding to -12 to +13 hours. | | |
| **device.sntp.serverName** | **IP address or domain name string** | |
| The SNTP server from which the phone obtains the current time. | | |
| **device.syslog.facility** | **0 to 23** | |
| A description of what generated the log message. For more information, see RFC 3164. | | |
| **device.syslog.prependMac[1]** | **0 or 1** | |
| If 1, the phone's MAC address is prepended to the log message sent to the syslog server. | | |
| **device.syslog.renderLevel[1]** | **0 to 6** | |
| Specify the logging level that displays in the syslog. Note that when you choose a log level, you are including all events of an equal or greater severity level and excluding events of a lower severity level. The logging level you choose determines the lowest severity of events to log.<br>**0** or **1**: SeverityDebug(7). **2** or **3**: SeverityInformational(6). **4**: SeverityError(3). **5:** SeverityCritical(2). **6:** SeverityEmergency(0). | | |

**Device Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **device.syslog.serverName** | **IP address or domain name string** | |
| The syslog server IP address or domain name string. | | |
| **device.syslog.transport** | **None, UDP, TCP, TLS** | |
| The transport protocol that the phone uses to write to the syslog server. If set to None, transmission is turned off but the server address is preserved. | | |
| **device.wifi.country** | **Two-letter country code** | **Null** |
| Enter the two-letter code for the country in which you are operating the RealPresence Trio 8800 solution with Wi-Fi enabled. | | |

[1] Change causes phone to restart or reboot.

# <diags/>

Use these parameters to enable and set up the remote packet capture feature.

**Remote Packet Capture Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **diags.dumpcore.enabled**[1] | **0 or 1** | **1** |
| When enabled, the phone generates a core file if it crashes. When disabled, the phone does not generate a core file when it crashes. The default value is 1, enabled. | | |
| **diags.pcap.enabled** | **0 or 1** | **0** |
| Enable or disable all on-board packet capture features. | | |
| **diags.pcap.remote.enabled** | **0 or 1** | **0** |
| Enable or disable the remote packet capture server. | | |
| **diags.pcap.remote.password** | **alphanumeric** | **<MAC Address>** |
| Enter the remote packet capture password. | | |
| **diags.pcap.remote.port** | **Valid TCP Port** | **2002** |
| Specify the TLS profile to use for each application. | | |

[1] Change causes phone to restart or reboot.

# <dialplan/>

The parameters listed in the following table enable you to create a specific routing path for outgoing SIP calls independent of other default configurations.

The dial plan (or digit map) is not applied against placed call list, voicemail, last call return, remote control dialed numbers, or on-hook dialing.

**Dial Plan (Digit Map) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dialplan.applyToCallListDial**[1] | **0 or 1** | **1** |
| If 0, the dial plan does not apply to numbers dialed from the received call list or missed call list. If 1, the dial play is applied to numbers dialed from the received call and missed call lists, including sub-menus. | | |
| **dialplan.applyToDirectoryDial**[1] | **0 or 1** | **0** |
| If 0, the dial plan is not applied to numbers dialed from the directory or speed dial list. If 1, the dial plan is applied to numbers dialed from the directory or speed dial, including auto-call contact numbers. | | |
| **dialplan.applyToForward**[1] | | |
| If 0, the dial plan does not apply to forwarded calls. If 1, the dial plan applies to forwarded calls. | | |
| **dialplan.applyToTelUriDial**[1] | **0 or 1** | **1** |
| If 0, the dial plan does not apply to URI dialing. If 1, the dial plan applies to URI dialing. | | |
| **dialplan.applyToUserDial**[1] | **0 or 1** | **1** |
| If 0, the dial plan does not apply to calls made when the user presses the **Dial** soft key to place a call. If 1, the dial plan applies to calls placed using the **Dial** soft key. | | |
| **dialplan.applyToUserSend**[1] | **0 or 1** | **1** |
| If 0, the dial plan does not apply to calls placed when the user presses the **Send** soft key to place a call. If 1, the dial plan applies to calls placed using the **Send** soft key. | | |
| **dialplan.digitmap**[1] | **string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435** | **[2-9]11\|0T\|**<br>**+011xxx.T\|**<br>**0[2-9]xxxxxxxxx\|**<br>**+1[2-9]xxxxxxxx\|**<br>**[2-9]xxxxxxxxx\|**<br>**[2-9]xxxT** |
| The digit map used for the dial plan. The string is limited to 2560 bytes and 100 segments of 64 bytes; a comma is also allowed; a comma turns dial tone back on;'+' is allowed as a valid digit; extension letter 'R' is used as defined above. This parameter enables the phone to automatically initiate calls to numbers that match a digit map pattern. | | |
| **dialplan.digitmap.timeOut**[1] | **string of positive integers separated by '\|'** | **3 \| 3 \| 3 \| 3 \| 3\| 3** |
| Specify a timeout in seconds for each segment of digit map. After you press a key, the phone waits this many seconds before matching the digits to a dial plan and dialing the call. Note: If there are more digit maps than timeout values, the default value of 3 is used. If there are more timeout values than digit maps, the extra timeout values are ignored. | | |
| **dialplan.filterNonDigitUriUsers**[1] | **0 or 1** | **0** |
| If 0, allow do not filter out (+) in the dial plan. If 1, filter out (+) from the dial plan. | | |

**Dial Plan (Digit Map) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dialplan.impossibleMatchHandling**[1] | **0, 1 or 2** | **0** |
| This parameter applies to digits you enter in dial mode, the dial mode when you pick up the handset, headset, or press the New Call key. The phone is not in dial mode when you are hot dialing, contact dialing, or call list dialing. If set to 0, the digits entered up to and including the point 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.<br>Note that if a call orbit number begins with '#' or '*', you need to set this parameter to 2 to retrieve the call using off-hook dialing. | | |
| **dialplan.removeEndOfDial**[1] | **0 or 1** | **1** |
| If set to 1, strip trailing # digit from digits sent out. | | |
| **dialplan.routing.emergency.outboundIdentity** | **10-25 digits, a SIP, or a TEL URI** | **Null** |
| Choose how your phone is identified when you place an emergency call. You can use one of three formats: a 10-25 digit number, a valid SIP, or a TEL URI. If using a URI, the full URI is included verbatim in the P-A-I header. For example:<br>• dialplan.routing.emergency.outboundIdentity="5551238000"<br>• dialplan.routing.emergency.outboundIdentity=sip:john@emergency.com<br>• dialplan.routing.emergency.outboundIdentity="tel:+16045558000" | | |
| **dialplan.routing.emergency.preferredSource** | **Config or ELIN** | **ELIN** |
| Use this parameter to set the precedence of the source of emergency outbound identities. When set to ELIN, the outbound identity used in the SIP P-Asserted-Identity header is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN).<br>When set to Config, the parameter `dialplan.routing.emergency.outboundIdentity` has priority when enabled; the LLDP-MED ELIN value is used if `dialplan.routing.emergency.outboundIdentity` is null. | | |
| **dialplan.routing.emergency.x.description**[1]<br>**Emergency contact description** | **string** | **x=1:Emergency, Others: Null** |
| **dialplan.routing.emergency.x.server.y**[1]<br>**Emergency server** | **positive integer** | **x=1: 1, others: Null** |
| **dialplan.routing.emergency.x.value**<br>**Emergency URL values** | **SIP URL (single entry)** | **x=1: 911, others: Null** |
| 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.<br>`description:` The label or description for the emergency address<br>`server.y`: The index representing the server to use for emergency routing (`dialplan.routing.server.x.address` where x is the index).<br>`value`: The URLs that should be watched for. When the user dials one of the URLs, the call is directed to the emergency server defined by `address`.<br>Note**:** Blind transfer for 911 (or other emergency calls) may not work if registration and emergency servers are different entities. | | |

**Dial Plan (Digit Map) Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dialplan.routing.server.x.address[1]** | **IP address or hostname** | **Null** |
| The IP address or hostname of a SIP server to use for routing calls. Multiple servers can be listed starting with x=1 to 3 for fault tolerance. Note: Blind transfer for 911 (or other emergency calls) may not work if registration and emergency servers are different entities. | | |
| **dialplan.routing.server.x.port[1]** | **1 to 65535** | **5060** |
| The port of a SIP server to use for routing calls. | | |
| **dialplan.routing.server.x.transport[1]** | **DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly** | **DNSnaptr** |
| The DNS lookup of the first server to be dialed is used if there is a conflict with the others. For example, if `dialplan.routing.server.1.transport="UDPOnly"` and `dialplan.routing.server.2.transport = "TLS"`, then `UDPOnly` is used. | | |
| **dialplan.userDial.timeOut** | **0 – 99 seconds** | **Generic Profile=0** |
| This parameter specifies the time in seconds that the phone waits before dialing a number you enter while the phone is on hook. You can apply `dialplan.userDial.timeOut` only when its value is lower than up.IdleTimeOut. | | |
| **dialplan.x.conflictMatchHandling** | **0 or 1** | **Generic Profile=0** |
| This is the per-registration parameter of `dialplan.conflictMatchHandling`. This parameter takes priority over the general parameter, `dialplan.conflictMatchHandling`. | | |

[1] Change causes phone to restart or reboot.

All of the parameters listed in the following table have a per-registration equivalent that you can configure. All of the per-registration parameters are listed in the following table. Note that the per-registration parameters override the general parameters where x is the registration number, for example, `dialplan.x.applyToTelUriDial` overrides `dialplan.applyToTelUriDial` for registration x.

The maximum number of registrations (x) for each phone model is listed in the table Flexible Call Appearances under the column *Registrations*.

**Per-Registration Dial Plan (Digit Map) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dialplan.conflictMatchHandling** | **0 or 1** | **Generic Profile=0**<br>**Lync Profile=1** |
| **dialplan.x.applyToCallListDial[1]** | **0 or 1** | **1** |
| **dialplan.x.applyToDirectoryDial[1]** | **0 or 1** | **0** |
| **dialplan.x.applyToForward** | **0 or 1** | **0** |

**Per-Registration Dial Plan (Digit Map) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dialplan.x.applyToTelUriDial**[1] | **0 or 1** | **1** |
| **dialplan.x.applyToUserDial**[1] | **0 or 1** | **1** |
| **dialplan.x.applyToUserSend**[1] | **0 or 1** | **1** |
| **dialplan.x.digitmap**[1] | **string - max number of characters 2560** | **Null** |
| **dialplan.x.digitmap.timeOut**[1] | **string - max number of characters 100** | **Null** |
| **dialplan.x.e911dialmask** | **string - max number of characters 256** | **Null** |
| **dialplan.x.e911dialstring** | **string - max number of characters 256** | **Null** |
| **dialplan.x.applyToForward** | **0 or 1** | **0** |
| **dialplan.x.impossibleMatchHandling**[1] | **0 to 2** | **0** |
| **dialplan.x.originaldigitmap** | **string - max number of characters 2560** | **Null** |
| **dialplan.x.removeEndOfDial**[1] | **0 or 1** | **1** |
| **dialplan.x.routing.emergency.y.value**[1] | **string - max number of characters 64** | **Null** |
| **dialplan.x.routing.emergency.y.server.z**[1] | **0 to 3** | **0 For all x, y, and z = 1 to 3** |
| **dialplan.x.routing.server.y.address**[1] | **string - max number of characters 256** | **Null** |
| **dialplan.x.routing.server.y.port**[1] | **1 to 65535** | **5060** |
| **dialplan.x.routing.server.y.transport**[1] | **DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly** | **DNSnaptr** |
| **dialplan.userDial.timeOut** | **0 – 99 seconds** | **Generic Profile=0**<br>**Lync Profile=3** |

[1] Change causes phone to restart or reboot.

# <dir/>

This parameter definition includes:

- <broadsoft/>  Polycom BroadSoft UC-One directory definitions
- <corp/>  The corporate directory definition
- <genband/>  Local contact directory for GENBAND

- <local/> The local directory definition

## <broadsoft/>

Use the parameters listed in the following table with the Polycom BroadSoft UC-One directory.

**Polycom BroadSoft UC-One Feature Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dir.broadsoft.regMap** | **0 - Const_NumLineReg** | **1** |
| Specify the registration line credentials you want to use to retrieve directory information from BroadSoft UC-One directory when `dir.broadsoft.useXspCredentials=0`. This parameter is available with BroadSoft R20 Server or later. | | |
| **dir.broadsoft.useXspCredentials** | **0 or 1** | **1** |
| If 1, the phone uses BroadSoft XSP credentials. If 0, the phone uses SIP credentials from `dir.broadsoft.regMap`. | | |
| **dir.broadsoft.xsp.address** | **dotted-decimal IP address, hostname or FQDN** | **Null** |
| Set the IP address or hostname of the BroadSoft directory XSP home address. For example, `host.domain.com` or `http://xxx.xxx.xxx.xxx`. | | |
| **dir.broadsoft.xsp.username** | **UTF-8 encoding string** | **Null** |
| Set the username used to authenticate to the BroadSoft Directory XSP server. | | |
| **dir.broadsoft.xsp.password** | **UTF-8 encoding string** | **Null** |
| Set the password used to authenticate to the BroadSoft Directory XSP server. | | |

## <corp/>

Use the parameters in the following table to configure a corporate directory. 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.

**Corporate Directory Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dir.corp.address**[1] | **IP address, hostname, or FQDN** | **Null** |
| The IP address or hostname of the LDAP server interface to the corporate directory. For example, host.domain.com. | | |
| **dir.corp.alt.address** | **hostname or FQDN** | **Null** |
| Enter the URL address of the GAB service provided by the server. | | |
| **dir.corp.alt.attribute.x.filter** | **UTF-8 encoded string** | **Null** |
| Use a filter to set a predefined search string through configuration files. | | |
| **dir.corp.alt.attribute.x.label** | **UTF-8 encoded string** | **Null** |

**Corporate Directory Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Provide a label to identify a user. | | |
| **dir.corp.alt.attribute.x.name** | **UTF-8 encoded string** | **Null** |
| The name of the parameter to match on the server. Each name must be unique; however, a global address book entry can have multiple parameters with the same name. You can configure up to eight parameters (x = 1 to 8). | | |
| **dir.corp.alt.attribute.x.sticky** | **0 or 1** | **0** |
| Enable or disable a filter string. If 0, the filter criteria for attribute x is reset after a reboot. If 1, the filter criteria are retained through a reboot. If you set an attribute to be sticky (set this parameter to 1), a '*' displays before the label of the attribute on the phone. | | |
| **dir.corp.alt.attribute.x.type** | **first_name, last_name, phone_number, SIP_address, other** | **last_name** |
| Define how parameter x is interpreted by the phone. Entries can have multiple parameters of the same type. The value `other` is for display purposes only.<br>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 global address book directory. | | |
| **dir.corp.alt.password** | **UTF-8 encoded string** | **Null** |
| The password used to authenticate to the GENBAND server. | | |
| **dir.corp.alt.port** | **0, Null, 1 to 65535** | **0** |
| The port that connects to the server if a full URL is not provided. | | |
| **dir.corp.alt.protocol** | **UTF-8 encoded string** | **sopi** |
| A directory protocol used to communicate to the corporate directory. The default value is sopi. | | |
| **dir.corp.alt.transport** | **TCP or TLS** | **TCP** |
| A transport protocol used to communicate to the corporate directory. The default value is TCP. | | |
| **dir.corp.alt.user** | **UTF-8 encoded string** | **Null** |
| The user name used to authenticate to the GENBAND server. | | |
| **dir.corp.alt.viewPersistence** | **0 or 1** | **0** |
| Displays the results from your last address directory search. | | |
| **dir.corp.attribute.x.addstar**[1] | **0 or 1** | **1** |
| If 1, the wildcard character, asterisk(*), is appended to the LDAP query field. If 0, the wildcard character, asterisk(*), is not appended to the query field. | | |
| **dir.corp.attribute.x.filter**[1] | **UTF-8 encoded string** | **Null** |
| The filter string for this parameter, which is edited when searching. | | |
| **dir.corp.attribute.x.label**[1] | **UTF-8 encoded string** | **Null** |
| The label when data is displayed. | | |

**Corporate Directory Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dir.corp.attribute.x.name**[1] | **UTF-8 encoded string** | **Null** |
| The name of the parameter to match on the server. Each name must be unique; however, an LDAP entry can have multiple parameters with the same name. Up to eight parameters can be configured (x = 1 to 8). | | |
| **dir.corp.attribute.x.searchable**[1] | **0 or 1** | **0** |
| If 0, quick search on parameter x (if x is 2 or more) is disabled. If 1, quick search on x (if x is 2 or more) is enabled. | | |
| **dir.corp.attribute.x.sticky**[1] | **0 or 1** | **0** |
| If 0, the filter criteria for attribute x is reset after a reboot. If 1, the filter criteria are retained through a reboot. If you set an attribute to be sticky (set this parameter to 1), a '*' displays before the label of the attribute on the phone. | | |
| **dir.corp.attribute.x.type**[1] | **first_name, last_name, phone_number SIP_address, H323_address URL, other** | **last_name** |
| Defines how parameter x is interpreted by the phone. Entries can have multiple parameters of the same type. The value other is used for display purposes only.<br>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.autoQuerySubmitTimeout**[1] | **0 to 60 seconds** | **0** |
| The timeout (in seconds) between when the user stops entering characters in the quick search and when the search query is automatically submitted. If 0, there is no timeout (automatic submit is disabled). | | |
| **dir.corp.backGroundSync**[1] | **0 or 1** | **0** |
| If 0, background downloading from the LDAP server is disabled. If 1, background downloading is enabled. | | |
| **dir.corp.backGroundSync.period**[1] | **3600 to 604800** | **86400** |
| The corporate directory cache is refreshed after the corporate directory feature has not been used for this period of time seconds. The default period is 24 hours (86400 seconds). The minimum is 1 hour and the maximum is 7 days. | | |
| **dir.corp.baseDN**[1] | **UTF-8 encoded string** | **Null** |
| The base domain name. This is the starting point for making queries on the LDAP server. | | |
| **dir.corp.bindOnInit**[1] | **0 or 1** | **1** |
| If 0, do not use bind authentication on initialization. If 1, use bind authentication on initialization. | | |
| **dir.corp.cacheSize**[1] | **8 to 256** | **128** |
| The maximum number of entries that can be cached locally on the phone. | | |
| **dir.corp.customError** | **UTF-8 encoded string** | **Null** |
| Configure the error message to display on the phone when the LDAP server finds an error. | | |
| **dir.corp.filterPrefix**[1] | **UTF-8 encoded string** | **(objectclass=person )** |

**Corporate Directory Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Predefined filter string for search queries. | | |
| **dir.corp.pageSize**[1] | **8 to 64** | **32** |
| The maximum number of entries requested from the corporate directory server with each query. | | |
| **dir.corp.password**[1] | **UTF-8 encoded string** | **Null** |
| The password used to authenticate to the LDAP server. | | |
| **dir.corp.port**[1] | **0, Null, 1 to 65535** | **389 (TCP) 636 (TLS)** |
| The port that connects to the server if a full URL is not provided. | | |
| **dir.corp.querySupportedControlOnInit** | **0 or 1** | **1** |
| When enabled, the phone makes an initial query to check the status of the server when booting up. | | |
| **dir.corp.scope**[1] | **one, sub, base** | **sub** |
| The type of search that is performed. If one, a search of one level below the base domain name (DN). If sub, a recursive search of all levels below the base DN. If base, a search at the base DN level. | | |
| **dir.corp.sortControl**[1] | **0 or 1** | **0** |
| Control how a client can make queries and sorts entries locally. If 0, leave sorting as negotiated between the client and server. If 1, force sorting of queries (this causes excessive LDAP queries and should only be used to diagnose LDAP servers with sorting problems). | | |
| **dir.corp.transport**[1] | **TCP, TLS, Null** | **TCP** |
| Specify whether a TCP or TLS connection is made with the server, if a full URL is not provided. | | |
| **dir.corp.user**[1] | **UTF-8 encoded string** | **Null** |
| The user name used to authenticate to the LDAP server. | | |
| **dir.corp.viewPersistence**[1] | **0 or 1** | **0** |
| If 0, the corporate directory search filters and browsing position are reset each time the user accesses the corporate directory. If 1, the search filters and browsing position from the previous session are displayed each time the user accesses the corporate directory. | | |
| **dir.corp.vlv.allow**[1] | **0 or 1** | **0** |
| If 0, virtual view list (VLV) queries are disabled. If 1, VLV queries are enabled and can be made if the LDAP server supports VLV. | | |
| **dir.corp.vlv.sortOrder**[1] | **list of parameters** | **Null** |
| The list of parameters —in exact order—for the LDAP server to use when indexing. For example: `sn`, `givenName`, `telephoneNumber`. | | |

[1] Change causes phone to restart or reboot.

# <genband/>

The parameters listed in this section are for use with the GENBAND personnel address book (PAB) contact directory.

**GENBAND Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dir.genband.local.contacts.maxSize** | **1 - 100** | **100** |
| Specify the maximum number of contacts available in the GENBAND personnel address book contact directory. | | |

# <local/>

The next table lists parameters you can configure for your local contact directory. The maximum local directory size is limited based on the amount of flash memory in the phone and varies by phone model. The maximum number of contacts and maximum file size for phone models is listed in the table Maximum File Size and Number of Contacts. Polycom recommends that you configure a provisioning server that allows uploads to ensure a back-up copy of the directory when the phone reboots or loses power.

Note that on the VVX 1500, the local directory is by default stored in the phone's non-volatile device settings and you have to option to use the phone's volatile RAM and set the maximum file size.

**Local Contact Directory Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **dir.local.contacts.maxNum**[1] | **VVX 300/301/310/311, 400/401/410/411, 500/501, 600/601 = 1-500**<br>**VVX1500 = 1–9999**<br>**SoundStructure VoIP Interface = not applicable** | **500**<br>**9999** |
| Maximum number of contacts allowed in the local contact directory. | | |
| **dir.local.nonVolatile.maxSize** | **1 - 100KB** | **VVX1500=100KB** |
| On the VVX 1500, set the maximum file size of the local contact directory stored on the phone's non-volatile memory. | | |
| **dir.local.readonly**[1] | **0 or 1** | **0** |
| If 0, the local contact directory can be edited. If 1, the local contact directory is read-only.<br>**Notes**:<br>If provisioning polling is enabled and **dir.local.readonly=1** is enabled, the phone will look for a `<mac>-directory.xml` when matching the polling event.<br>If provisioning polling is enabled and **dir.local.readonly=0** is disabled, the phone will not look for a `<mac>-directory.xml` when matching the polling event. | | |
| **dir.local.serverFeatureControl.method** | **None or GENBANDSOPI** | **None** |
| Specify a method for synchronizing the directory and server. When set to GENBANDSOPI, the GENBANDSOPI protocol is enabled on the phone to get the personnel address book service from the GENBAND server. | | |
| **dir.local.serverFeatureControl.reg** | **1 - 34** | **1** |

**Local Contact Directory Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Specify the phone line to enable personel address book feature on. | | |
| **dir.local.volatile** | **0 or 1** | **0** |
| On the VVX 1500, enable or disable the use of volatile memory for the local contact directory. By default the VVX 1500 uses non-volatile memory. | | |
| **dir.local.volatile.maxSize** | **1 - 200KB** | **VVX1500=200KB** |
| On the VVX 1500, set the maximum file size of the local contact directory stored on the phone's volatile memory. | | |
| **dir.search.field** | **0 or 1** | **0** |
| If 0, contact directory searches are sorted by contact's last name. If 1, contact directory searches are sorted by first name. | | |

[1] Change causes phone to restart or reboot.

# <divert/>

The phone has a flexible call forward/diversion feature for each registration. In all cases, a call is diverted only if a non-Null contact has been configured.

The following numbers apply when no expansion module is attached. In the following table, x is the registration number. VVX 300 series: x=1-6; VVX 400 series: x=1-8; VVX 500series: x=1-12; VVX 600 series: x=1-16; VVX 1500: x=1-6. Attaching an expansion module increases the number to 34 (x=1-34).

**Call Diversion (Call Forwarding) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **divert.x.contact**[1] | **contact address: 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<br>)** | **Null** |
| The forward-to contact used for all automatic call diversion features. All automatically forwarded calls are directed to this contact. The contact can be overridden by a busy contact, DND contact, or no-answer contact as specified by the `busy`, `dnd`, and `noAnswer` parameters that follow. | | |
| **divert.x.sharedDisabled**[1] | **0 or 1** | **1** |
| If 0, call diversion features can be used on shared lines. If 1, call diversion features are disabled on shared lines. | | |
| **divert.x.autoOnSpecificCalle**[2] | **0 or 1** | 1 |
| If 0, the auto divert feature of the contact directory is disabled for registration x. If 1, calls on registration x may be diverted using auto divert, you may specify to divert individual calls or divert all calls. | | |
| **divert.busy.x.enabled**[2]<br>**divert.busy.x.contact**[1] | **0 or 1**<br>**contact address** | **1**<br>**Null** |

**Call Diversion (Call Forwarding) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Divert incoming calls that reach a busy signal. If `enabled` is set to 1, calls are diverted when registration x is busy. Calls are sent to the busy contact's address if it is specified; otherwise calls are sent to the default contact specified by `divert.x.contact`. If `enabled` is set to 0, calls are not diverted if the line is busy. | | |
| **divert.dnd.x.enabled**[2]<br>**divert.dnd.x.contact**[1] | **0 or 1**<br>**contact address** | **0**<br>**Null** |
| Divert calls when do not disturb is enabled. If `enabled` is set to 1, calls are diverted when DND is enabled on registration x. Calls are sent to the DND contact's address if it is specified; otherwise calls are sent to the default contact specified by `divert.x.contact`. | | |
| **divert.fwd.x.enabled**[2] | **0 or 1** | **1** |
| If 0, the user cannot enable universal call forwarding (automatic forwarding for all calls on registration x). If 1, a Forward soft key displays on the phone's Home screen that you can use to enable universal call forwarding. | | |
| **divert.noanswer.x.enabled**[2]<br>**divert.noanswer.x.contact**[1]<br>**divert.noanswer.x.timeout**[1] | **0 or 1**<br>**contact address**<br>**positive integer** | **1**<br>**Null**<br>**55** |
| If no-answer call diversion is `enabled`, calls that are not answered after the number of seconds specified by timeout are sent to the no-answer `contact`. If the no-answer `contact` is set to Null, the call is sent to the default contact specified by `divert.x.contact`. If `enabled` is set to 0, calls are diverted if they are not answered. | | |

[1] Change causes phone to restart or reboot.

[2] Change causes phone to restart or reboot. If server-based call forwarding is enabled, this parameter is disabled.

# <dns/>

The <dns/> parameters include:

- DNS-A
- DNS-NAPTR
- DNS-SRV

You can enter a maximum of 12 record entries for DNS-A, DNS-NAPTR, and DNS-SRV. records.

## DNS-A

Add up to 12 DNS-A record entries using the parameters in the following table. Specify the address, name, and cache time interval for DNS-A record *x*, where x is from 1 to 12.

**DNA-A Parameters**

| *Parameter* | *Permitted values* | *Default* |
|---|---|---|
| **dns.cache.A.x.address** | **IP version 4 address** | **Null** |
| IP address. | | |
| **dns.cache.A.x.name** | **valid hostname** | **Null** |

**DNA-A Parameters (continued)**

| *Parameter* | *Permitted values* | *Default* |
|---|---|---|
| Hostname | | |
| **dns.cache.A.x.ttl** | **300 to 536870912 (2^29), seconds** | **300** |
| The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and its reset TTL timer again. | | |

## DNS-NAPTR

Add up to 12 DNS-NAPTR record entries using parameters in the following table. Specify each parameter for DNS-NAPTR record *x*, where x is from 1 to 12.

**DNS-NAPTR Parameters**

| *Parameter* | *Permitted values* | *Default* |
|---|---|---|
| **dns.cache.NAPTR.x.flags** | **A single character from [A-Z, 0-9]** | **Null** |
| The flags to control aspects of the rewriting and interpretation of the fields in the record. Characters are case-sensitive. At this time, only ‘S’, ‘A’, ‘U’, and ‘P’ are defined as flags. See RFC 2915 for details of the permitted flags. | | |
| **dns.cache.NAPTR.x.name** | **domain name string** | **Null** |
| The domain name to which this resource record refers. | | |
| **dns.cache.NAPTR.x.order** | **0 to 65535** | **0** |
| An 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 are processed before high numbers. | | |
| **dns.cache.NAPTR.x.regexp** | **string containing a substitution expression** | **Null** |
| This parameter is currently unused.<br>Applied to the original string held by the client. The substitution expression is applied in order to construct the next domain name to look up. The grammar of the substitution expression is given in RFC 2915. | | |
| **dns.cache.NAPTR.x.replacement** | **domain name string with SRV prefix** | **Null** |
| The next name to query for NAPTR records depending on the value of the flags field. It must be a fully qualified domain-name. | | |
| **dns.cache.NAPTR.x.service** | **string** | **Null** |
| Specifies the service(s) available down this rewrite path. For more information, see RFC 2915. | | |

**DNS-NAPTR Parameters (continued)**

| *Parameter* | *Permitted values* | *Default* |
|---|---|---|
| **dns.cache.NAPTR.x.ttl** | **300 to 536870912 (2^29), seconds** | **300** |
| The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and its reset TTL timer again. | | |

## DNS-SRV

Add up to 12 DNS-SRV record entries using parameters in the following table. Specify each parameter for DNS-SRV record *x*, where x is from 1 to 12.

**DNS-SRV Parameters**

| *Parameter* | *Permitted values* | *Default* |
|---|---|---|
| **dns.cache.SRV.x.name** | **domain name string with SRV prefix** | **Null** |
| The domain name string with SRV prefix. | | |
| **dns.cache.SRV.x.port** | **0 to 65535** | **0** |
| The port on this target host of this service. For more information, see RFC 2782. | | |
| **dns.cache.SRV.x.priority** | **0 to 65535** | **0** |
| The priority of this target host. For more information, see RFC 2782. | | |
| **dns.cache.SRV.x.target** | **domain name string** | **Null** |
| The domain name of the target host. For more information, see RFC 2782. | | |
| **dns.cache.SRV.x.ttl** | **300 to 536870912 (2^29), seconds** | **300** |
| The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and its reset TTL timer again. | | |
| **dns.cache.SRV.x.weight** | **0 to 65535** | **0** |
| A server selection mechanism. For more information, see RFC 2782. | | |

# <efk/>

Use the following tables to configure the enhanced feature key (EFK) feature on your phone:

- Enhanced Feature Key (EFK) Version Parameters
- Enhanced Feature Key (EFK) List Parameters
- Enhanced Feature Key (EFK) Prompt Parameters
- Enhanced Feature Key (EFK) Soft Key Parameters

**Enhanced Feature Key (EFK) Version Parameters**

| *Parameter Name* | *Permitted Values* | *Default* |
|---|---|---|
| **efk.version** | **2 (1 for SIP 3.0 and earlier)** | **2** |
| The version of the EFK elements. For SIP 3.0.x or earlier, **1** is the only supported version. For SIP 3.1 and later, **2** is the only supported version. If this parameter is Null, the EFK feature s disabled. This parameter is not required if there are no `efk.efklist` entries. | | |

In the following table, the registration line x=1-50.

**Enhanced Feature Key (EFK) List Parameters**

| *Parameter Name* | *Permitted Values* | *Default* |
|---|---|---|
| **efk.efklist.x.action.string** | | |
| The action string contains a macro definition of the action that the feature key performs. If EFK is enabled, this parameter must have a value (it cannot be Null). For a list of macro definitions and example macro strings, see the section Understand Macro Definitions. | | |
| **efk.efklist.x.label** | **string** | **Null** |
| The text string used as a label on any user text entry screens during EFK operation. If Null, the Null string is used. Note: If the label does not fit on the screen, the text is shortened and '…' is appended. | | |
| **efk.efklist.x.mname** | | **expanded_macro** |
| The unique identifier used by the speed dial configuration to reference the enhanced feature key entry. Cannot start with a digit. Note that this parameter must have a value, it cannot be Null. | | |
| **efk.efklist.x.status** | **0 or 1** | **0** |
| If 0 or Null, key x is disabled. If 1, the key is enabled. | | |
| **efk.efklist.x.type** | | **invite** |
| The SIP method to be performed. If set to `invite`, the action required is performed using the SIP INVITE method. Note: This parameter is included for backwards compatibility. Do not use if possible. If `efk.x.action.string` contains types, this parameter is ignored. If Null, the default of INVITE is used. | | |

In the following table, the registration line x=1-50.

**Enhanced Feature Key (EFK) Prompt Parameters**

| *Parameter Name* | *Permitted Values* | *Default* |
|---|---|---|
| **efk.efkprompt.x.label**[1] | **string** | **Null** |
| The prompt text that is presented to the user on the user prompt screen. If Null, no prompt displays. Note: If the label does not fit on the screen, the label is shortened and '…' is appended. | | |
| **efk.efkprompt.x.status**[1] | **0 or 1** | **0** |
| If 0, key x is disabled. If 1, the key is enabled. This parameter must have a value, it cannot be Null. Note: If a macro attempts to use a prompt that is disabled or invalid, the macro execution fails. | | |

**Enhanced Feature Key (EFK) Prompt Parameters (continued)**

| *Parameter Name* | *Permitted Values* | *Default* |
|---|---|---|
| **efk.efkprompt.x.type**[1] | **numeric or text** | **text** |
| The type of characters entered by the user. If set to `numeric`, the characters are interpreted as numbers. If set to `text`, the characters are interpreted as letters. If Null, `numeric` is used. If this parameter has an invalid value, this prompt, and all parameters depending on this prompt, are invalid. Note: A mix of `numeric` and `text` is not supported. | | |
| **efk.efkprompt.x.userfeedback**[1] | **visible or masked** | **visible** |
| The user input feedback method. If set to `visible`, the text is visible. If set to `masked`, the text displays as asterisk characters (*), this can be used to mask password fields. If Null, visible is used. If this parameter has an invalid value, this prompt, and all parameters depending on this prompt, are invalid. | | |

[1] Change causes phone to restart or reboot.

**Enhanced Feature Key (EFK) Soft Key Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **efk.softkey.alignleft** | **0 or 1** | **0** |
| Use this parameter to left-align soft keys and remove blank soft keys from the order. By default, this parameter is disabled. Setting this parameter to 1 left-aligns soft keys and removes blank soft keys from the order. Note: This parameter does not work with custom soft keys. | | |

# <exchange/>

The exchange parameter controls aspects of meeting invitations.

**Exchange Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **exchange.meeting.parseOption**[1] | **All, Location, LocationAndSubject, Description, Enum** | **Location** |
| Indicates the field in the meeting invite from which the VMR or meeting number should be fetched. | | |

[1] Change causes phone to restart or reboot.

# <feature/>

The feature parameters listed in the following table control the activation or deactivation of a feature at run time.

**Feature Activation/Deactivation Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **feature.acdAgentAvailable.enabled**[1] | **0 or 1** | **0** |
| If 0, the ACD agent available/unavailable feature is disabled. If 1, the feature is enabled. | | |
| **feature.acdLoginLogout.enabled**[1] | **0 or 1** | **0** |
| If 0, the ACD login/logout feature is disabled. If 1, the feature is enabled. | | |
| **feature.acdPremiumUnavailability.enabled**[1] | **0 or 1** | **0** |
| If 0, the premium ACD unavailability feature is disabled. If 1, premium ACD unavailability feature is enabled, and unavailability reason codes can be used (if the other ACD feature parameters are also be enabled). | | |
| **feature.acdServiceControlUri.enabled**[1] | **0 or 1** | **0** |
| If 0, the ACD service control URI feature is disabled. If 1, the feature is enabled. | | |
| **feature.advancedConference.enabled** | **0 or 1** | **0** |
| Enable or disable advanced conferences and conference controls for ALU advanced conferences.<br>If enabled, conference controls display. If disabled, conference controls do not display. | | |
| **feature.audioVideoToggle.enabled** | **0 or 1** | **0** |
| Applies to the VVX 1500 and VVX camera-enabled VVX 500/501 and 600/601 business media phones. If 0, the audio/video toggle feature is disabled. If 1, the feature is enabled. | | |
| **feature.bluetooth.enabled** | **0 or 1** | **1** |
| VVX 600/601 high-security environments. If 0, the Bluetooth feature is disabled. If 1, Bluetooth is enabled. | | |
| **feature.broadsoftdir.enabled** | **0 or 1** | **0** |
| If 1, the BroadSoft enterprise directory is enabled. If 0, the directory is disabled | | |
| **feature.broadsoftUcOne.enabled** | **0 or 1** | **0** |
| If 1, the BroadSoft UC-One feature is enabled. If 0, the feature is disabled. | | |
| **feature.broadsoft.xsi.AnonymousCallReject.enabled** | **0 or 1** | **0** |
| Displays the Anonymous Call Rejection menu on the phone. If set to 1, the Anonymous Call Rejection menu displays and the user can turn the feature on or off from the phone. If set to 0, the Anonymous Call Rejection menu does not display to users | | |
| **feature.broadsoft.xsi.BroadWorksAnywhere.enabled** | **0 or 1** | **0** |
| Enable or disable the BroadWorks Anywhere feature menu on the phone. If set to 0, the feature menu is disabled does not display. | | |
| **feature.broadsoft.xsi.LineIdblock.enabled** | **0 or 1** | **0** |
| Enable or disable the Line ID Blocking feature menu on the phone. If set to 0, the feature menu is disabled and does not display on the phone. | | |

**Feature Activation/Deactivation Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **feature.broadsoft.xsi.RemoteOffice.enabled** | **0 or 1** | **0** |
| Enable or disable the Remote Office feature menu on the phone. If set to 1, the feature menu is enabled and displays on the phone. | | |
| **feature.broadsoft.xsi.SimultaneousRing.enabled** | **0 or 1** | **0** |
| Enable or disable the Simultaneous Ring Personal feature menu on the phone. If set to 0, the feature menu is disabled and does not display. | | |
| **feature.btoe.enabled** | **0 or 1** | **0** |
| If 0, the better together over ethernet feature is disabled. If 1, the feature is enabled. | | |
| **feature.callCenterStatus.enabled** | **0 or 1** | **0** |
| If 0, the status event threshold capability is disabled. If 1, the status event threshold capability is enabled. | | |
| **feature.callList.enabled**[1] | **0 or 1** | **1** |
| **All locally controlled call lists.** | | |
| **feature.callListMissed.enabled**[1] | **0 or 1** | **1** |
| **The missed calls list.** | | |
| **feature.callListPlaced.enabled**[1] | **0 or 1** | **1** |
| **The placed calls list.** | | |
| **feature.callListReceived.enabled**[1] | **0 or 1** | **1** |
| **The received calls list.** | | |
| If 0, the call list is disabled. If 1, the call list is enabled. To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled. | | |
| **feature.callPark.enabled**[1] | **0 or 1** | **0** |
| If 0, the call park and call retrieve features are disabled. If 1, the features are enabled. | | |
| **feature.callRecording.enabled**[1] | **0 or 1** | **0** |
| Available for devices with a USB port. If 0, the call recording and playback feature is disabled. If 1, the feature is enabled. | | |
| **feature.corporateDirectory.alt.enabled** | **0 or 1** | **0** |
| Enable or disable the global address book service. | | |
| **feature.corporateDirectory.enabled** | **0 or 1** | **0** |
| If 0, the corporate directory feature is disabled. If 1, the feature is enabled. | | |
| **feature.directedCallPickup.enabled**[1] | **0 or 1** | **0** |
| If 0, the directed call pickup feature is disabled. If 1, the feature is enabled. | | |

**Feature Activation/Deactivation Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **feature.directory.enabled** | **0 or 1** | **1** |
| If 0, the local contact directory is disabled. If 1, the directory is enabled. | | |
| **feature.doNotDisturb.enable** | **0 or 1** | **1** |
| Enable or disable do not disturb (DND). When disabled, the DND soft key does not display and the option is removed from the phone’s menu system at Menu > Settings > Features. | | |
| **feature.enhancedCallDisplay.enabled** | **0 or 1** | **0** |
| If 0, the phone may display the protocol at the end of the called party identification (for example, 1234567 [SIP]). If 1, the phone displays the number only (for example, 1234567). | | |
| **feature.enhancedCallPark.allowAudioNotification** | **0 or 1** | **0** |
| Enables and disables the audio notifications for parked calls on private and shared lines. | | |
| **feature.enhancedFeatureKeys.enabled** | **0 or 1** | **0** |
| If 0, the enhanced feature keys feature is disabled. If 1, the feature is enabled. | | |
| **feature.exchange2007.interop.enabled** | **0 or 1** | **0** |
| If 0, interoperability with Exchange 2007 is disabled. If 1, interoperability with Exchange 2007 is enabled.<br>Note: If enabled, `feature.exchangeVoiceMail.enabled` should be disabled. | | |
| **feature.exchangeCalendar.enabled** | **0 or 1** | **Lync = 1**<br>**Generic = 0** |
| For the VVX 500/501, 600/601 and 1500 phones. If 1, the Exchange Calendar feature is enabled as a service. If 0, the feature is disabled. | | |
| **feature.flexibleLineKey.enable** | **0 or 1** | **Lync = 0** |
| Enables and disables the Lync Flexible Line Key feature. | | |
| **feature.forward.enable** | **0 or 1** | **1** |
| Enable or disable call forwarding. When disabled, the Forward soft key does not display and the option is removed from the phone’s menu system at Menu > Settings > Features. | | |
| **feature.genband.E911.enabled** | **0 or 1** | **0** |
| Enable or disable the GENBAND E.911 feature. | | |
| **feature.groupCallPickup.enabled**[1] | **0 or 1** | **0** |
| If 0, the group call pickup feature is disabled. If 1, the SIP-B group call pickup feature is enabled. | | |
| **feature.hoteling.enabled** | **0 or 1** | **0** |
| If 0, Hoteling is disabled. If 1, Hoteling is enabled. | | |
| **feature.intercom.enable** | **0 or 1** | **0** |

**Feature Activation/Deactivation Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Enable or disable the intercom feature. | | |
| **feature.lastCallReturn.enabled**[1] | **0 or 1** | **0** |
| If 0, the last call return feature is disabled. If 1, the feature is enabled. | | |
| **feature.messaging.enabled**[1] | **0 or 1** | **0** |
| If 0, the instant messaging feature is disabled. If 1, the feature is enabled. | | |
| **feature.nonVolatileRingerVolume.enabled** | **0 or 1** | **1** |
| If 0, user changes to the ringer volume are reset to default when the phone reboots. If 1, user changes to the ringer volume are saved and maintained when the phone reboots. | | |
| **feature.nWayConference.enabled** | **0 or 1** | **0** |
| If 0, the n-way conferencing managing feature is disabled; you can hold three-way conferences but the options to manage the conference do not display. If 1, n-way conferencing is enabled, you can hold conferences with the maximum number of parties, and the options to manage the conference display. | | |
| **feature.persistentMute.enabled**[1] | **0 or 1** | **0** |
| Set to 1 to enable the persistent mute feature. When set to 0, mute ends when the active call ends or when the phone restarts. | | |
| **feature.pictureFrame.enabled**[1] | **0 or 1** | **1** |
| For the VVX 500/501, 600/601, and 1500 only. If 0, the digital picture frame feature is disabled. If 1, the digital picture frame feature is enabled. | | |
| **feature.presence.enabled**[1] | **0 or 1** | **0** |
| If 0, the presence feature—including buddy managements and user status—is disabled. If 1, the presence feature is enabled with the buddy and status options. | | |
| **feature.qml.enabled**[1] | **0 or 1** | **0** |
| If 1, the QML viewer is enabled on phone. If 0, the viewer is disabled. Note that the UC-One directory user interface uses QML as the UI framework and the viewer is used to load the QML applications. | | |
| **feature.ringDownload.enabled**[1] | **0 or 1** | **1** |
| If 0, the phone does not download ringtones when it starts up. If 1, the phone downloads ringtones when it starts up. | | |
| **feature.scap.defCallTypeExclusive** | **0 or 1s** | **1s** |
| Controls the default behavior of a Shared Call Appearance call. By default, an outgoing call from the call group is private. After the call is answered, the user needs to press Share soft key to make the call public so that other people on the line can bridge in to the call. | | |
| **feature.scap.HoldRequestUriUserPart** | **string** | **SCAP-Hold** |
| Specifies the Hold request for Shared Call Appearance calls to the ALU server. This value must match the value configured on ALU server for SCA hold request. | | |

**Feature Activation/Deactivation Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **feature.uniqueCallLabeling.enabled** | **0 or 1** | **0** |
| If 0, disables the unique call labeling feature. If 1, enables the unique call labeling feature. Use reg.x.line.y.label to define unique labels. | | |
| **feature.urlDialing.enabled** | **0 or 1** | **1** |
| If 0, URL/name dialing is not available. If 1, URL/name dialing is available from private lines. Note: If enabled, unknown callers are identified on the display by their phone's IP address. | | |
| **feature.usbRear.power.enabled** | **0 or 1** | **1** |
| If 1, power to the rear USB port (port 2) is enabled. If 0, power to the rear USB port is disabled and the phone does not detect USB devices to the rear USB port.<br>Note: This parameter does not apply to VVX 1500 phones. | | |
| **feature.usbTop.power.enabled** | **0 or 1** | **1** |
| If 1, power to the top USB port (port 1) is enabled. If 0, power to the top USB port is disabled and the phone does not detect USB devices to the top USB port. | | |
| **feature.VDP.enabled** | **0 or 1** | **0** |
| If 1, VDP is enabled and the phone displays the Visitor Login soft key. If 0, VDP is disabled and the phone does not display the Visitor Login soft key. | | |

[1] Change causes phone to restart or reboot.

# <genband/>

The following table lists GENBAND E.911 parameters.

**GENBAND Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **genband.E911.location.description** | **String up to 256 characters [platform-specific display size limitations apply]** | **Other** |
| Enter a description of the location of the phone, for example, cubicle 105. Note: Ensure that the description string you provide here is identical to the description you configure on the location server. | | |
| **genband.E911.location.locationID** | **1 to 256** | **0** |
| Enter the location ID corresponding to the location description you entered in `genband.E911.location.description`, for example, 112876. Note: Ensure that the the location ID you enter here is identical to the one you configure on the location server. | | |
| **genband.E911.registration.line** | **0 to 100** | **1** |
| Select the registration line to use to retrieve E.911 location information | | |

# <homescreen/>

The parameters in this section control the display of icons on the phone's Home screen.

**Homescreen Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **homescreen.intercom.enable** | **0 or 1** | **1** |
| Enable or disable the Intercom icon on the device home screen. | | |
| **homescreen.status.enable** | **0 or 1** | **1** |
| Enable or disable the display of the Status menu icon on the Home screen. | | |

# <hoteling/>

The hoteling enhancement to ACD enables agents to use any available host phone by logging in with agent credentials. After logging in, agents have access to their guest profile and ACD settings on the host phone. The hoteling enhancement is independent of the ACD feature, meaning agents can use hoteling whether the premium ACD feature is enabled or disabled. The following table lists parameters you can configure.

**Hoteling Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **hoteling.reg** | **1 – 34** | **1** |
| If ACD is disabled, the phone uses this line registration index for hoteling. If ACD is enabled, this value must be the same as the ACD line registration index. | | |

# <httpd/>

The phone contains a local Web Configuration Utility server for user and administrator features. Note that several of these parameters can be used with Microsoft Skype for Business Server and the parameter values listed in the table Enable Web Configuration Utility have two default states: a generic default value for UC Software 5.1.0 and a different value when the phone is registered with Skype for Business Server. The following table lists the default values for both states where applicable.

The web server supports both basic and digest authentication. The authentication user name and password are not configurable for this release.

**HTTPD (Web Server) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **httpd.enabled**[1] | **0 or 1** | **Generic=1**<br>**Lync=0** |
| If 0, the HTTP server is disabled (the Web Configuration Utility is also be disabled). If 1, the server is enabled. Note: This parameter must be enabled to take screen captures of the phone screen. | | |

**HTTPD (Web Server) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **httpd.cfg.enabled**[1] | **0 or 1** | **Generic=1**<br>**Lync=0** |
| If 0, the Web Configuration Utility is disabled. If 1, the Web Configuration Utility is enabled. | | |
| **httpd.cfg.port**[1] | **1 to 65535** | **80** |
| Port is 80 for HTTP servers. Care should be taken when choosing an alternate port. | | |
| **httpd.cfg.secureTunnelEnabled**[1] | **0 or 1** | **Generic=1**<br>**Lync=0** |
| If 0, the web does not use a secure tunnel. If 1, the server connects through a secure tunnel. | | |
| **httpd.cfg.secureTunnelPort**[1] | **1 to 65535** | **443** |
| The port to use for communications when the secure tunnel is used. | | |
| **httpd.cfg.secureTunnelRequired**[1] | **0 or 1** | **1** |
| If 0, communications to the web server do not require a secure tunnel. If 1, communications do require a secure tunnel. | | |

[1] Change causes phone to restart or reboot.

# <homeScreen/>

The following table lists parameters that configure the phone’s Home screen display.

**Homescreen Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **homeScreen.application.enable** | **0 or 1** | **1** |
| Enable or disable display of the Applications icon on the phone Home screen. | | |
| **homeScreen.calendar.enable** | **0 or 1** | **1** |
| Enable or disable display of the Calendar icon on the phone Home screen. | | |
| **homeScreen.directories.enable** | **0 or 1** | **1** |
| Enable or disable display of the Directories menu icon on the phone Home screen. | | |
| **homeScreen.doNotDisturb.enable** | **0 or 1** | **1** |
| Enable or disable display of the DND icon on the phone’s Home screen. | | |
| **homeScreen.features.enable** | **0 or 1** | **1** |
| Enable or disable display of the Features menu icon on the phone Home screen. | | |

**Homescreen Parameters  (continued)**

| | | |
|---|---|---|
| **homeScreen.forward.enable** | **0 or 1** | **1** |
| Enable or disable display of the call forward icon on the phone Home screen. | | |
| **homeScreen.messages.enable** | **0 or 1** | **1** |
| Enable or disable display of the Messages menu icon on the phone Home screen. | | |
| **homeScreen.newCalls.enable** | **0 or 1** | **1** |
| Enable or disable display of the New Call icon on the phone Home screen. | | |
| **homeScreen.settings.enable** | **0 or 1** | **1** |
| Enable or disable display of the Settings menu icon on the phone Home screen. | | |
| **homeScreen.UCOne.enable** | **0 or 1** | **0** |
| Enable or disable the UC-One Settings icon to display on the Home screen. | | |

# <ind/>

The table Indicator Pattern Parameters lists parameters that enable you to set the patterns, color, and duration of the LED indicators on VVX phones and expansion modules.

In the LED indicator pattern parameters, *x* is the pattern type and *y* is the pattern number. For y, enter a value of 1-20 to indicate the pattern number. For x, enter one of the values in the following table to indicate the LED indicator pattern type.

**LED Indicator Pattern Type**

| Pattern Type | Function |
|---|---|
| powerSaving | Sets the behavior for Message Waiting Indicator when the phone is in Power Saving mode. |
| active | Sets the pattern for line keys during active calls. |
| on | Turns on the LED indicator pattern. |
| off | Turns off the LED indicator pattern. |
| offering | Sets the pattern for line keys during incoming calls. |
| flash | Sets the pattern for line keys during held calls and the Message Waiting Indicator when there are unread voicemail messages. |
| lockedOut | Sets the pattern for line keys when a remote party is busy on a shared line. |
| FlashSlow | Sets the pattern for the Headset key when Headset Memory Mode is enabled. |
| held | Sets the pattern for line keys during a held call. |
| remoteBusyOffering | Sets the pattern for line keys for monitored BLF contacts. |

**Indicator Pattern Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **ind.pattern.x.step.y.state** | **On or Off** | **On** |
| Turns the LED indicator on or off depending on the pattern’s state. | | |
| **ind.pattern.x.step.y.color** | **Red, Green, or Yellow** | **Red** |
| Indicates the color of the line key LED indicators. The Yellow value is only available for VVX 300 and 400 series phones. You cannot set the line key LED indicator to Yellow on VVX 101 and 201 phones or on the VVX Expansion Modules.<br>**Note**: You can only configure the color of the LED indicators for line keys only. You cannot configure the color of the LED indicators for the Headset key or MWI. | | |
| **ind.pattern.x.step.y.duration** | **Positive integer** | **0 = infinite** |
| Enter the duration of the pattern of the LED indicator. | | |

# <key/>

The next table lists parameters that enable you to change the default functions of your phone's keypad keys, a process also known as remapping. If you want to change the default function of a key, you must specify the phone model number, the key you want to change, and a new function for the key.

- For a list of products and their model codes, see the section Product, Model Code, and Part Number Mapping.
- To find the key number, location of the key on each phone model, and default key functions, refer to the section Define the Phone Key Layout.
- For a list of parameter values you can assign as functions to a phone key, refer to the table Keypad Key Functions.

**Caution: Key remapping is not recommended**

Polycom does not recommend remapping or changing the default functions of the keys on your phone.

**Key Parameters**

| *Parameter* | *Permitted Values* |
|---|---|
| **key.x.y.function.prim** | For a list of functions refer to the table Keypad Key Functions |
| Specify a phone model, key number, and function.<br>x can be one of the VVX 300 series, 400 series, 500 series, 600 series, or VVX1500 phones.<br>y can be one key number. | |

[1] Change causes phone to restart or reboot.

The following table lists the functions that are available for phone keys.

**Keypad Key Functions**

| | | | | |
|---|---|---|---|---|
| Answer | Dialpad2 | Handsfree | MyStatus | SpeedDialMenu |
| ArrowDown | Dialpad3 | Headset | Null | Talk |
| ArrowLeft | Dialpad4 | Hold | Offline | Video |
| ArrowRight | Dialpad5 | Home | Redial | VolDown |
| ArrowUp | Dialpad6 | Line2 | Release | VolUp |
| Back | Dialpad8 | Line3 | Select | |
| BuddyStatus | Dialpad9 | Line4 | Setup | |
| CallList | DialpadStar | Line5 | SoftKey1 | |
| Conference | DialPound | Line6 | SoftKey2 | |
| Delete | Directories | Messages | SoftKey3 | |
| Dialpad0 | DoNotDisturb | Menu | SoftKey4 | |
| Dialpad1 | Green | MicMute | SpeedDial | |

## Example Custom Key Configurations

This section provides several custom key configuration examples.

### To remap the volume up key to answer a call on the VVX 300:

- Update the configuration file as follows: `key.VVX300.6.function.prim="Answer"`
- **To remap the volume down key to launch the Settings menu on the VVX 300 using a macro:**
- Update the configuration file as follows:
  - `key.VVX300.7.function.prim="$Msetting$"`
  - `efk.efklist.1.action.string="$FSetup$"`
  - `efk.efklist.1.mname="setting"`
  - `efk.efklist.1.status="1"`

### To remap the Mute key to launch the Forward Menu on the VVX 500 using EFK.

- Update the configuration file as follows: `key.VVX500.18.function.prim="$FDivert$"`

### To remap the Transfer key to phone lock using an EFK macro:

- Update the configuration file as follows: `key.37.function.prim="$FLockPhone$"`

### To remap the Redial key:

» Update the configuration file as follows:

```
key.36.function.prim="http://vanoem02.vancouver.polycom.com:8080/MicroBrowserT
est.html"
```

# <lcl/>

You can configure the language you want the Polycom phone user interface to operate and display in. The phones support both North American and international time and date formats.

**Caution: Use a multilingual XML editor**

Edit the language parameters using a multilingual XML editor. If you do not use an XML editor, some of the language labels in the configuration file and in the language menu on the phone display incorrectly. To confirm whether your editor properly supports these characters, view the language parameter for languages such as Chinese, Japanese, Korean, Russian— for example `lcl.ml.lang.menu.1.label`.

This parameter definition includes:

- The multilingual definitions
- The date and time definitions

# <ml/>

The multilingual parameters listed in the following table are based on string dictionary files downloaded from the provisioning server. These files are encoded in standalone XML format and include several eastern European and Asian languages. The files include space for user-defined languages.

**Multilingual Parameters**

| *Parameter* | *Permitted Values* |
|---|---|
| **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) is 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, set this parameter to `German_Germany`. | |
| **lcl.ml.lang.charset**[1] | **string** |
| The language character set. | |
| **lcl.ml.lang.clock.x.24HourClock** | **0 or 1** |
| This parameter overrides `lcl.datetime.time.24HourClock`. If 1, display time in 24-hour clock mode rather than am/pm. | |
| **lcl.ml.lang.clock.x.dateTop** | **0 or 1** |
| If parameter present, overrides `lcl.datetime.date.dateTop`. If 1, display date above time, otherwise display time above date. | |

**Multilingual Parameters (continued)**

| *Parameter* | *Permitted Values* |
|---|---|
| **lcl.ml.lang.clock.x.format** | **string which includes ‘D’, ‘d’ and ‘M’ and two optional commas** |
| This parameter overrides `lcl.datetime.date.format`:<br>D = day of week d = day M = month. Up to two commas may be included.<br>For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday<br>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.ml.lang.clock.x.longFormat** | **0 or 1** |
| If parameter present, overrides `lcl.datetime.date.longFormat`.<br>If 1, display the day and month in long format (Friday/November), otherwise use abbreviations (Fri/Nov). | |
| **lcl.ml.lang.list**[1] | **a comma-separated list** |
| A list of the languages supported on the phones. | |
| **lcl.ml.lang.menu.x** | **String in the format language_region** |
| **Dictionary file** | |
| **lcl.ml.lang.menu.x.label**[1] | **String in the format nativeLanguageName (abbreviation)** |
| **Phone language menu label** | |
| The phone supports multiple languages. Dictionary files and labels must be sequential (for example, `lcl.ml.lang.menu.1, lcl.ml.lang.menu.2, lcl.ml.lang.menu.3… lcl.ml.lang.menu.N)` The dictionary file cannot have caps, and the strings must exactly match a folder name of a dictionary file (you can find the names in the **VVXLocalization** folder of your software distribution). If you edit these parameters, you need to use a multilingual XML editor that supports Unicode, such as XML Notepad 2007.<br>For example, a dictionary file and label for German is: `lcl.ml.lang.menu.8=“German_Germany” lcl.ml.lang.menu.8.label=“Deutsch (de-de)”` | |

[1] Change causes phone to restart or reboot.

## To add a new language:

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
4. `language_region` parallel to the other dictionary files under the VVXLocalization folder on the provisioning server.
5. Add an `lcl.ml.lang.clock.menu.x` parameter to the configuration file.
6. 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` parameters and set them according to the regional preferences.
7. (Optional) Set `lcl.ml.lang` to be the new `language_region` string.

The basic character support includes the Unicode character ranges listed in the next table.

**Unicode Ranges for Basic Character Support**

| *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 |

# <datetime/>

The parameters listed in the following table configure the date and time display on the phone.

**Date and Time Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **lcl.datetime.date.dateTop** | **0 or 1** | |
| If set to 1, display date above time. If 0, display time above date. | | |
| **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.<br>Up to two commas may be included.<br>For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday<br>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 or 1** | |
| If set to 1, display the day and month in long format (Friday/November), otherwise, use abbreviations (Fri/Nov). | | |
| **lcl.datetime.time.24HourClock** | **0 or 1** | |
| If set to 1, display time in 24-hour clock mode rather than a.m./p.m. | | |

# <license/>

The parameters listed in the next table enable you to configure the feature licensing system.

**Feature License Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **license.polling.time**[1] | **00:00 – 23:59** | **02:00** |
| The time (using the 24-hour clock) to check if the license has expired. | | |

[1] Change causes phone to restart or reboot.

**Note: Removing the installed license**

Once the license is installed on a phone, it cannot be removed.

# <lineKey/>

The parameters listed in the next table are available for the Flexible Line Key Assignment.

**Line Key Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **lineKey.x.category**[1] | **unassigned, line, BLF, speedDial, presence** | **unassigned** |
| The line key category. Set the category to unassigned to leave a blank line key. | | |
| **lineKey.x.index**[1] | **0 to 9999** | **0** |
| For lines, the index for line numbers. For speed dials, the speed dial index. For BLF or presence, 0. For unassigned, the value is ignored. | | |
| **lineKey.reassignment.enabled**[1] | **0 or 1** | **0** |
| If 1, flexible line key assignment is enabled. | | |

[1] Change causes phone to restart or reboot.

# <lldp/>

The parameters listed in the following table enable you to configure settings for LLDP discovery.

**LLDP Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **lldpFastStartCount** | **3 to 10** | **5** |
| Specifies the number of consecutive LLDP packets the phone sends at the time of LLDP discovery. Note that LLDP packets are sent every one second. | | |

# <log/>

The event logging system supports the classes of events listed in the table Logging Levels. Two types of logging are supported:

- level, change, and render
- <sched/>

**Caution: Changing the logging parameters**

Logging parameter changes can impair system operation. Do not change any logging parameters without prior consultation with Polycom Technical Support.

**Logging Levels**

| *Logging Level* | *Interpretation* |
|---|---|
| 0 | Debug only |
| 1 | High detail class event |
| 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

Three formats available for the event timestamp are listed in the next table.

**Event Timestamp Formats**

| | |
|---|---|
| 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 |

## <level/> <change/> and <render/>

This configuration parameter is defined in the following table.

**Logging Level, Change, and Render Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **log.level.change.xxx** | **0 to 6** | **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 acom, ares, app1, bluet, bdiag, brow, bsdir, cap, cdp, cert, cfg, cipher, clink, clist, cmp, cmr, copy, curl, daa, dapi, dbs, dbuf, dhcpc, dis, dock, dot1x, dns, drvbt, ec, efk, ethf, flk, h323, hset, httpa, httpd, hw, ht, ib, key, ldap, lic, lldp, loc, log, mb, mobil, net, niche, ocsp, osd, pcap, pcd, pdc, peer, pgui, pmt, poll, pps, pres, pstn, ptt, push, pwrsv, rdisk, res, rtos, rtls, sec, sig, sip, slog, so, srtp, sshc, ssps, style, sync, sys, ta, task, tls, trace, ttrs, usb, usbio, util, utilm, wdog, wmgr, and xmpp. | | |

**Logging Level, Change, and Render Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **log.level.change.flk** | **0 - 6** | **4** |
| Sets the log level for Lync FLK logs. | | |
| **log.level.change.mr** | **0 - 6** | **4** |
| Initial logging level for the Networked Devices log module. | | |
| **log.level.change.mraud** | **0 - 6** | **4** |
| Initial logging level for the Networked Devices Audio log module. | | |
| **log.level.change.mrmgr** | **0 - 6** | **4** |
| Initial logging level for the Networked Devices Manager log module. | | |
| **log.level.change.prox** | **0 - 6** | **4** |
| Initial logging level for the Proximity log module. | | |
| **log.level.change.ptp** | **0 - 6** | **4** |
| Initial logging level for the Precision Time Protocol log module. | | |
| **log.level.change.sopi** | **0 - 6** | **4** |
| Specify the SOPI service log level for the GENBAND Global Address Book and Personnel Address Book. | | |
| **log.render.file** | **0 or 1** | **1** |
| Set to 1. Polycom recommends that you do not change this value. | | |
| **log.render.file.size** | **positive integer, 1 to 180** | **32** |
| Maximum size of flash memory for logs in Kbytes. When this size is about to be exceeded, the phone uploads all logs that have not yet been uploaded, and erase half of the logs on the phone. The administrator may use web browser to read all logs on the phone. | | |
| **log.render.file.upload.append** | **0 or 1** | **1** |
| If set to 1, use append mode when uploading log files to server.<br>Note: HTTP and TFTP don’t support append mode unless the server is set up for this. | | |
| **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 | | |
| **log.render.file.upload.append.sizeLimit** | **positive integer** | **512** |
| Maximum log file size that can be stored on provisioning server in Kbytes. | | |
| **log.render.file.upload.period** | **positive integer** | **172800** |
| Time in seconds between log file uploads to the provisioning server.<br>Note: The log file is not uploaded if no new events have been logged since the last upload. | | |

**Logging Level, Change, and Render Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **log.render.level** | **0 to 6** | **1** |
| Specifies the lowest class of event rendered to the log files. This is the output filter from the internal memory-based log system.<br>The log.render.level maps to syslog severity as follows:<br>0 SeverityDebug (7)<br>1 SeverityDebug (7)<br>2 SeverityInformational (6)<br>3 SeverityInformational (6)<br>4 SeverityError (3)<br>5 SeverityCritical (2)<br>6 SeverityEmergency (0) | | |
| **log.render.realtime** | **0 or 1** | **1** |
| Set to 1. Polycom recommends that you do not change this value. | | |
| **log.render.stdout** | **0 or 1** | **0** |
| Set to 0. Polycom recommends that you do not change this value. | | |
| **log.render.type** | **0 to 2** | **2** |
| Refer to the table Event Timestamp Formats for timestamp type. | | |

## <sched/>

The phone can be configured to schedule certain advanced logging tasks on a periodic basis. Polycom recommends that you set the parameters listed in the next table in consultation with Polycom Technical Support. Each scheduled log task is controlled by a unique parameter set starting with log.sched.x where *x* identifies the task. A maximum of 10 schedule logs is allowed.

**Logging Schedule Parameters**

| *Parameter* | *Permitted Values* | *Default Value* |
|---|---|---|
| **log.sched.x.level** | **0 to 5** | **3** |
| 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 display in the log. | | |
| **log.sched.x.name** | **alphanumeric string** | |
| Name of an internal system command to be periodically executed. To be supplied by Polycom. | | |
| **log.sched.x.period** | **positive integer** | **15** |
| Seconds between each command execution. 0=run once | | |
| **log.sched.x.startDay** | **0 to 7** | **7** |
| When startMode is abs, specifies the day of the week to start command execution. 1=Sun, 2=Mon, ..., 7=Sat | | |

**Logging Schedule Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default Value* |
|---|---|---|
| **log.sched.x.startMode** | **0 - 64** | |
| Start at an absolute time or relative 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. | | |

# <mb/>

The next table lists parameters that configure the home page, proxy, and size limits used by the microbrowser and browser when selected to provide services. The microbrowser and web browser are supported on the VVX 300 series, 400 series, 500 series, 600 series, and 1500 phones.

**Microbrowser and Web Browser Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **mb.idleDisplay.home** | **Null or any fully formed valid HTTP URL. Length up to 255 characters.** | **Null** |
| The URL for the microbrowser home page that is shown on the idle display microbrowser Home page. For example: http://www.example.com/xhtml/frontpage. If Null, there is no idle display microbrowser. Note that the microbrowser idle display displaces the idle display indicator. | | |
| **mb.idleDisplay.refresh** | **0 or an integer > 5** | **0** |
| The time period in seconds that the microbrowser's idle display refreshes. If set to 0, the idle display microbrowser does not refresh. The minimum refresh period is 5 seconds (values from 1 to 4 are ignored, and 5 is used).<br>Note: If an HTTP Refresh header is detected, it is respected, even if this parameter is set to 0. The refresh parameter is respected only in the event that a refresh fails. Once a refresh is successful, the value in the HTTP refresh header, if available, is used. | | |
| **mb.idleRefresh.onFailure** | **60 - 655350 seconds** | **60 seconds** |
| To reduce requests from the phone when the idle display server is unavailable, specify a delay in seconds the phone sends refresh requests to the idle browser server when unavailable. This delay applies only when the server returns HTTP 5xx errors.<br>To control the refresh times when the server is functioning, `use mb.idleDisplay.refresh`. | | |
| **mb.main.autoBackKey**[1] | **0 or 1** | **1** |
| If 0, the phone does not provide a **Back** soft key; all soft keys are created and controlled by the application. If 1, the phone automatically supplies a **Back** soft key in all main browser screens. The **Back** soft key takes the user back to the previous page in the browser history. | | |
| **mb.main.home** | **Any fully formed valid HTTP URL. Length up to 255 characters.** | **Null** |
| The URL of the microbrowser's home page. For example: `http://www.example.com/xhtml/frontpage/home`. If blank, the browser notifies the user that a blank home-page was used. | | |

**Microbrowser and Web Browser Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **mb.main.idleTimeout** | **0 - 600, seconds** | **40** |
| The timeout, in seconds, for the interactive browser. If the interactive browser remains idle for the defined period of time, the phone returns to the idle browser. If 0, there is no timeout. | | |
| **mb.main.loadWebImages** | **0 or 1** | **1** |
| If 0, disables loading of images in a browser. If 1, images can load in a browser. | | |
| **mb.main.reloadPage** | **0 or 1** | **0** |
| If 0, the microbrowser displays the content of the most recently viewed web page. If 1, the microbrowser loads the URL configured in **mb.main.home** each time the browser is launched. | | |
| **mb.main.statusbar** | **0 or 1** | **0** |
| If 0, the status bar does not display. If 1, the status bar displays and status messages are shown. | | |
| **mb.main.toolbar.autoHide.enabled** | **0 or 1** | **1** |
| If 0, the toolbar displays continually. If 1, the toolbar disappears if not selected. | | |
| **mb.proxy** | **Null or domain name or IP address in the format <address>:<port>** | **Null. Default port = 8080** |
| The address of the HTTP proxy to be used by the microbrowser. If blank, normal unproxied HTTP is used by the microbrowser. | | |
| **mb.ssawc.call.mode** | **active or passive** | **passive** |
| Control the spontaneous display of web content. If set to `passive`, web content is displayed only when requested by the user. If set to `active`, web content is displayed immediately. | | |
| **mb.ssawc.enabled** | **0 or 1** | **0** |
| If 0, spontaneous display of web content is disabled. If 1, spontaneous web content display is enabled. | | |

[1] Change causes phone to restart or reboot.

| | | |
|---|---|---|
| **mr.pair.tls.enabled** | **0 or 1** | **0** |
| If 1, use TLS for communications between the RealPresence Trio 8800 and RealPresence Trio Visual+. If 0, do not use TLS for communications. | | |
| **mr.pair.uid.1** | **String** | **Null** |
| Enter the MAC address (Serial Number [S/N]) of the RealPresence Trio Visual+ with which you want to pair. | | |

# <msg/>

The next table lists parameters you can use to configure the message-waiting feature, which is supported on a per-registration basis.

The maximum number of registrations (x) for each phone model is listed in the table Flexible Call Appearances under the column *Registrations*.

**Message Waiting Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **msg.bypassInstantMessage**[1] | **0 or 1** | **0** |
| This parameter determines what is shown on the phone menu when you press the **Messages** or **MSG** key. If 0, the phone shows the menus Message Center and Instant Messages. If 1, the phone bypasses these menus and goes directly to voicemail. This parameter applies only to phone models that have a Messages or MSG key. | | |
| **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 sends a SUBSCRIBE request to this contact after boot-up. | | |
| **msg.mwi.x.callBackMode** | **contact, registration, disabled** | **registration** |
| The message retrieval mode and notification for registration x. `contact`: a call is placed to the contact specified by `msg.mwi.x.callback. registration`: the registration places a call to itself (the phone calls itself). `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** |
| The contact to call when retrieving messages for this registration if `msg.mwi.x.callBackMode` is set to `contact`. | | |
| **msg.mwi.x.led** | **0, 1** | **1** |
| Where x is an integer referring to the registration indexed by reg.x.<br>If set to 0, the red MWI LED does **not** flash when there are new unread messages for the selected line. When set to 1, the LED flashes as long as there are new unread voicemail messages *for any line* in which this is parameter is enabled. | | |

[1] Change causes phone to restart or reboot.

# <mwi/>

The parameters listed in the next table enable and disable a back light on the phone screen to illuminate when you receive a new voicemail message.

**Message Waiting Indicator Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **mwi.backLight.disable** | **0 or 1** | **0** |
| A back light on the phone screen illuminates when you receive a new voicemail. Set to 0 to disable the back light message alert. Set to 1 to enable. The default is disabled.<br>If `mwi.backLight.disable` is enabled, the backlight is not illuminated on new voice message arrival. By default this parameter is disabled. | | |

# <nat/>

The parameters listed in the next table define port and IP address changes used in NAT traversal. The port changes alter 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 does 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 to port 1234.

**Network Access Translation Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **nat.ip**[1] | **IP address** | **Null** |
| IP address to advertise within SIP signaling - should match the external IP address used by the NAT device. | | |
| **nat.keepalive.interval** | **0 to 3600** | **0** |
| The keep-alive interval in seconds. Sets the interval at which phones sends a keepalive packet to the gateway/NAT device to keep the communication port open so that NAT can continue to function. If Null or 0, the phone does not send out keepalive messages. | | |
| **nat.mediaPortStart**[1] | **0 to 65440** | **0** |
| The initially allocated RTP port. Overrides the value set for `tcpIpApp.port.rtp.mediaPortRangeStart`. | | |
| **nat.signalPort**[1] | **1024 to 65535** | **0** |
| The port used for SIP signaling. Overrides `voIpProt.local.port`. | | |

[1] Change causes phone to restart or reboot.

# <net/>

The parameters listed in this section control the Ethernet interface maximum transmission unit (MTU) on VVX business media phones.

**Ethernet Interface MTU Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **net.interface.mtu** | **800 - 1500** | **1496 (for RealPresence Trio 8800 and RealPresence Trio Visual+)** |
| Configure the Ethernet or Wi-Fi interface maximum transmission unit (MTU) on VVX business media phones or RealPresence Trio solution.<br>Note that this parameter affects the LAN port and the PC port. | | |
| **net.lldp.extendedDiscovery** | **0 to 3600** | **0** |
| Specify the duration of time that LLDP discovery continues after sending the number of packets defined by the parameter `lldpFastStartCount`. Note that LLDP packets are sent every 5 seconds during this extended discovery period. | | |

# <phoneLock/>

The parameters listed in the next table must be enabled if you want to use the Lock soft key.

**Phone Lock Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **phoneLock.Allow.AnswerOnLock** | **0 or 1** | **0** |
| If 1, the phone answers any incoming call without asking to UNLOCK. If 0, the phone asks to UNLOCK before answering. | | |
| **phoneLock.authorized.x.description**<br>**The name or description of an authorized number**<br>**phoneLock.authorized.x.value**<br>**The number or address for an authorized contact** | **String**<br><br>**string** | |
| Up to five (x=1 to 5) authorized contacts that a user can call while their phone is locked. Each contact needs a description to display on the screen, and a phone number or address value for the phone to dial. | | |
| **phoneLock.browserEnabled** | **0 or 1** | **0** |
| If 0, the microbrowser or browser is not displayed while the phone is locked. If 1, the microbrowser or browser is displayed while the phone is locked. | | |
| **phoneLock.dndWhenLocked** | **0 or 1** | **0** |
| If 0, the phone can receive calls while it is locked. If 1, the phone enters Do-Not-Disturb mode while it is locked. Note: The user can change this setting from the phone user interface. | | |
| **phoneLock.enabled**[1] | **0 or 1** | **0** |

**Phone Lock Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| If 0, the phone lock feature is disabled. If 1, the phone lock feature is enabled.<br>Note: To ‘unlock’ the phone remotely (in conjunction with deleting/modifying the overrides files), disable and re-enable this parameter. | | |
| **phoneLock.idleTimeout** | **0 to 65535** | **0** |
| The amount of time (in seconds) the phone can be idle before it automatically locks. If 0, automatic locking is disabled. | | |
| **phoneLock.lockState** | **0 or 1** | **0** |
| The value for this parameter indicates whether the phone is locked or unlocked and changes each time you lock or unlock the phone. If 0, the phone is unlocked. If 1, the phone is locked. Note that the phone stores and uploads the value each time it changes via the `MAC-phone.cfg`. You can set this parameter remotely using the Web Configuration Utility. | | |
| **phoneLock.powerUpUnlocked** | **0 or 1** | **0** |
| Use this parameter to override `phoneLock.lockState`. If 0, the phone retains the value in `phoneLock.lockState`. If 1, you can restart, reboot, or power cycle the phone to override the value for `phoneLock.lockState` in the `MAC-phone.cfg` and start the phone in an unlocked state. You can then lock or unlock the phone locally. Polycom recommends that you do not leave this parameter enabled. | | |

[1] Change causes phone to restart or reboot.

# <powerSaving/>

The power-saving feature automatically turns off the phone's LCD display when not in use. This feature is disabled by default on the VVX 300 and 400 series phones, and enabled by default on the VVX 500 series, 600 series, and 1500 phones. The parameters `powerSaving.userDetectionSensitivity.*` listed in the next table are supported only on the VVX 1500 business media phones.

**Power Saving Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **powerSaving.enable** | **0 or 1** | **VVX 201=0**<br>**VVX 300/301/310/311=0**<br>**VVX 400/401/410/411=0**<br>**VVX 500/501, 600/601, 1500=1** |
| If 0, the LCD power saving feature is disabled. If 1, the feature is enabled. The power-saving feature is disabled by default on the VVX 201 and VVX 300- and 400-series phones, and is enabled by default on the VVX 500 series, 600 series, and 1500.<br>Note that when the phone is in power-saving mode, the LED Message Waiting Indicator (MWI) flashes. To disable the MWI LED when the phone is in power saving mode, set the parameter `ind.pattern.powerSaving.step.1.state.x` to 0 where x=your phone’s model. For example, enter the parameter as `ind.pattern.powerSaving.step.1.state.VVX500` to disable the MWI for your VVX 500 phone. | | |

**Power Saving Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **powerSaving.idleTimeout.offHours** | **1 to 10** | **1** |
| The number of minutes to wait while the phone is idle during off hours before activating power saving. | | |
| **powerSaving.idleTimeout.officeHours** | **1 to 600 minutes** | **480, VVX 1500=10** |
| The number of minutes to wait while the phone is idle during office hours before activating power saving. Note that the default time for VVX 300 series, 400 series, 500 series, and 600 series is 480 minutes. The default time for the VVX 1500 is 10 minutes. | | |
| **powerSaving.idleTimeout.userInputExtension** | **1 to 20** | **10** |
| The minimum number of minutes to wait while the phone is idle—after using the phone—before activating power saving. | | |
| **powerSaving.officeHours.duration.Monday**<br>**powerSaving.officeHours.duration.Tuesday**<br>**powerSaving.officeHours.duration.Wednesday**<br>**powerSaving.officeHours.duration.Thursday**<br>**powerSaving.officeHours.duration.Friday**<br>**powerSaving.officeHours.duration.Saturday**<br>**powerSaving.officeHours.duration.Sunday** | **0 to 24**<br>**0 to 24**<br>**0 to 24**<br>**0 to 24**<br>**0 to 24**<br>**0 to 24**<br>**0 to 24** | **12**<br>**12**<br>**12**<br>**12**<br>**12**<br>**0**<br>**0** |
| The duration of the day's office hours. | | |
| **powerSaving.officeHours.startHour.xxx** | **0 to 23** | **7** |
| The starting hour for the day's office hours, where xxx is one of `monday`, `tuesday`, `wednesday`, `thursday`, `friday`, `saturday`, and `sunday` (refer to `powerSaving.officeHours.duration` for an example). | | |
| **powerSaving.userDetectionSensitivity.offHours** | **0 to 10** | **2** |
| Available on the VVX 1500 only. The sensitivity used to detect the presence of the phone's user during off hours. 10 is the most sensitive. If set to 0, this feature is disabled.<br>The default value was chosen for good performance in a typical office environment and is biased for difficult detection during off hours. | | |
| **powerSaving.userDetectionSensitivity.officeHours** | **0 to 10** | **7** |
| Available on the VVX 1500 only. The sensitivity used to detect the presence of the phone's user during office hours. 10 is the most sensitive. If set to 0, this feature is disabled.<br>The default value was chosen for good performance in a typical office environment and is biased for easy detection during office hours. | | |

# <pres/>

The next table lists parameters you can configure for the presence feature. Note that the parameter `pres.reg` is the line number used to send SUBSCRIBE. If this parameter is missing, the phone uses the primary line to send SUBSCRIBE.

**Presence Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **pres.idleSoftkeys** | **0 or 1** | **1** |
| If 0, the **MyStat** and **Buddies** presence idle soft keys do not display. If 1, the soft keys display. | | |
| **pres.idleTimeout.offHours.enabled** | **0 or 1** | **1** |
| If 0, the off hours idle timeout feature is disabled. If 1, the feature is enabled. | | |
| **pres.idleTimeout.offHours.period** | **1 to 600** | **15** |
| The number of minutes to wait while the phone is idle during off hours before showing the Away presence status. | | |
| **pres.idleTimeout.officeHours.enabled** | **0 or 1** | **1** |
| If 0, the office hours idle timeout feature is disabled. If 1, the feature is enabled. | | |
| **pres.idleTimeout.officeHours.period** | **1 to 600** | **15** |
| The number of minutes to wait while the phone is idle during office hours before showing the Away presence status. | | |
| **pres.reg** | ***1 to 34*** | ***1*** |
| The valid line/registration number that is used for presence. This registration sends a SUBSCIRBE for presence. If the value is not a valid registration, this parameter is ignored. | | |

# <prov/>

The parameters listed in the next table control the provisioning server system for your phones.

**Provisioning Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **prov.autoConfigUpload.enabled** | **0 or 1** | **1** |
| Enable or disable the automatic upload of phone and Web Configuration Utility override configuration files to the provisioning server. By default, per-phone `MAC-phone.cfg` and `MAC-web.cfg` files are automatically uploaded to the provisioning server when a configuration change is made from the phone interface or Web Configuration Utility respectively. When disabled, per-phone override files are not uploaded to the provisioning server. | | |
| **prov.configUploadPath** | **string** | **Null** |
| The directory - relative to the provisioning server - where the phone uploads the current configuration file when the user selects Upload Configuration. If set to Null, use the provisioning server directory. | | |
| **prov.lineMap.cma.x**[1] | **1 to 6** | **1** |
| Used to map the CMA H.323 line to a SIP line. Only x=1 is currently supported. | | |
| **prov.login.automaticLogout** | **0 to 46000** | **0** |
| The time (in minutes) before a non-default user is automatically logged out of the handset. If 0, the user is not automatically logged out. | | |

**Provisioning Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **prov.login.defaultPassword** | **String** | **Null** |
| The login password for the default user. | | |
| **prov.login.defaultOnly** | **0 or 1** | **0** |
| If 1, the default user is the only user who can log in. If 0, other users can log in. | | |
| **prov.login.defaultUser** | **String** | **Null** |
| The username for the default user. If present, the user is automatically logged in when the phone boots up and logged in after another user logs out. | | |
| **prov.login.enabled** | **0 or 1** | **0** |
| If 0, the user profile feature is disabled. If 1, the user profile feature is enabled. | | |
| **prov.login.lcCache.domain** | **0 to 64** | **Null** |
| The user’s sign-in domain name. | | |
| **prov.login.lcCache.user** | **0 to 64** | **Null** |
| The user’s sign-in user name. | | |
| **prov.login.localPassword** | **String** | **123** |
| The password used to validate the user login. It is stored either as plain text or encrypted (an SHA1 hash). | | |
| **prov.login.persistent** | **0 or 1** | **0** |
| If 0, users are logged out if the handset reboots. If 1, users remain logged in when the phone reboots. | | |
| **prov.login.required** | **0 or 1** | **0** |
| If 1, a user must log in when the login feature is enabled. If 0, the user does not have to log in. | | |
| **prov.loginCredPwdFlushed.enabled** | **0 or 1** | **1** |
| If 1, when a user logs in or logs out, the login credential password is reset. If 0, the login credential password is not reset. | | |
| **prov.polling.enabled** | **0 or 1** | **0** |
| If 0, the provisioning server is not automatically polled for upgrades. If 1, the provisioning server is polled. | | |
| **prov.polling.mode** | **abs, rel, random** | **abs** |

**Provisioning Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The polling mode.<br>abs The phone polls every day at the time specified by prov.polling.time.<br>rel The phone polls after the number of seconds specified by prov.polling.period.<br>random The phone polls at random between a starting time set in prov.polling.time and an end time set in prov.polling.timeRandomEnd.<br>Note that if you set the polling period in prov.polling.period to a time greater than 86400 seconds (one day) polling occurs on a random day within that polling period (meaning values such as 86401 would be over 2 days) and only between the start and end times. The day within the period is decided based upon the phones MAC address and does not change with a reboot whereas the time within the start and end is calculated again with every reboot. | | |
| **prov.polling.period** | **integer > 3600** | **86400** |
| The polling period in seconds. The polling period is rounded up to the nearest number of days in absolute and random mode. In relative mode, the polling period starts once the phone boots. In random mode, if this is set to a time greater than 86400 (one day) polling occurs on a random day based on the phone's MAC address. | | |
| **prov.polling.time** | **hh:mm** | **03:00** |
| The polling start time. Used in absolute and random modes. | | |
| **prov.polling.timeRandomEnd** | **hh:mm** | **Null** |
| The polling stop time. Only used in random mode. | | |
| **prov.quickSetup.enabled** | **0 or 1** | **0** |
| If 0, the quick setup feature is disabled. If 1, the quick setup feature is enabled. | | |
| **prov.startupCheck.enabled** | **0 or 1** | **1** |
| If 0, the phone is not provisioned at startup. If 1, the phone is provisioned at start up. All configuration files, licenses, and overrides are downloaded even if the software changes. (The previous behavior was to reboot as soon as the phone determined that software changed.) | | |
| **prov.quickSetup.limitServerDetails** | **0 or 1** | **0** |
| If 1, a screen to enter only user name and password is shown. Other details are taken from ztp/dhcp (option66).<br>If 0, user must provide all the details, for example, DHCP option, server address, server type) in addition to user name and password. | | |
| **prov.usercontrol.enabled** | **0 or 1** | **1** |
| When this parameter is set to 1, the phone displays the software update notification and options, and the user can control the software download. If set to 0, the phone does not displays the software update notification and options, and the phone reboots automatically to update the software. | | |
| **prov.usercontrol.postponeTime** | **15 minutes, 1hour, 2 hours, 4 hours, 6 hours** | **2 hours** |
| Configure a time interval for software update notications. Permitted values for this configuration parameter are 15 min, 1 hour, 2hours, 4 hours and 6 hours using the format HH:MM. If a user configures an invalid value the default value is used. | | |

**Provisioning Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **prov.vdp.accessCode.login** | **string** | ***771** |
| Specify the VDP login service access code. | | |
| **prov.vdp.accessCode.logout** | **string** | ***772** |
| Specify the VDP logout service access code. | | |

[1] Change causes phone to restart or reboot.

# <ptt/>

The PTT (push-to-talk) parameter is used to configure Push-to-Talk features. The parameters in the next table configure the PTT mode and page mode features.

**Push-To-Talk and Group Paging Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **ptt.address** | **multicast IP address** | **224.0.1.116** |
| The multicast IP address to send page audio to and receive page audio from. | | |
| **ptt.callWaiting.enable** | **0 or 1** | **0** |
| If 0, incoming PTT sessions do not produce standard call waiting. If 1, incoming PTT sessions produce standard call waiting behavior on the active audio channel. | | |
| **ptt.compatibilityMode** | **0 or 1** | **0** |
| If 0, the PTT/paging codec used is controlled by the ptt.codec and ptt.pageMode.codec parameters, allowing codecs to be chosen that are not compatible with SpectraLink handset models 8020/8030 or older.<br>If 1, the PTT/paging codec used will always be compatible with SpectraLink handset models 8020/8030 or older, even if other configuration parameters are incompatible. For example, if this parameter is enabled and ptt.codec is set to G.722, the G.726QI codec is used for outgoing PTT audio to maintain compatibility. | | |
| **ptt.emergencyChannel.volume** | **-57 to 0** | **-10** |
| The volume of emergency pages relative to the maximum speakerphone volume of the phone. Positive values are louder than the maximum and negative values are quieter. The gain to use for emergency page/PTT is the maximum termination gain plus this parameter. **Note**: To enter a negative number, press the * key first. | | |
| **ptt.port** | **0 to 65535** | **5001** |
| The port to send audio to and receive audio from. | | |
| **ptt.pageMode.allowOffHookPages** | **0 or 1** | **0** |
| If 0, group pages do not play out on the phone during an active call—except for Priority and Emergency pages. If 1, group pages play out on the handset during an active call. | | |
| **ptt.pageMode.codec** | **G.711Mu, G.726QI, or G.722** | **G.722** |

**Push-To-Talk and Group Paging Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The audio codec to use for outgoing group pages. Incoming pages are decoded according to the codec specified in the incoming message. | | |
| **ptt.pageMode.defaultGroup** | **1 to 25** | **1** |
| The paging group used to transmit an outgoing page if the user does not explicitly specify a group. | | |
| **ptt.pageMode.displayName** | **up to 64 octet UTF-8 string** | **PTT** |
| This display name is shown in the caller ID field of outgoing group pages. If Null, the value from `reg.1.displayName` is used. | | |
| **ptt.pageMode.emergencyGroup** | **1 to 25** | **25** |
| The paging group to use for emergency pages. | | |
| **ptt.pageMode.enable** | **0 or 1** | **0** |
| If 0, group paging is disabled. If 1, group paging is enabled. | | |
| **ptt.pageMode.group.x.available** | **0 or 1** | **1** |
| Make the group available to the user. | | |
| **ptt.pageMode.group.x.allowReceive** | **0 or 1** | **1** |
| If 0, phone cannot receive pages on the specified group. If 1, phone can receive pages on the specified group. | | |
| **ptt.pageMode.group.x.allowTransmit** | **0 or 1** | **1** |
| Allow outgoing announcements to the group | | |
| **ptt.pageMode.group.x.label** | **string** | **ch24: Priority,**<br>**ch25: Emergency, others: Null**<br>**ch1, 24, 25: 1, others: 0** |
| The label to identify the group | | |
| **ptt.pageMode.group.x.subscribed** | **0 or 1** | **1** |
| Subscribe the phone to the group.<br>A page mode group x, where x= 1 to 25. The `label` is the name used to identify the group during pages.<br>If `available` is disabled (0), the user cannot access the group or subscribe and the other page mode group parameters is ignored. If enabled, the user can access the group and choose to subscribe.<br>If `allowTransmit` is disabled (0), the user cannot send outgoing pages to the group. If enabled, the user may send outgoing pages.<br>If `subscribed` is disabled, the phone does not subscribe to the group. If enabled, the phone subscribes to the group. | | |
| **ptt.pageMode.payloadSize** | **10, 20, ..., 80 milliseconds** | **20** |
| The page mode audio payload size. | | |
| **ptt.pageMode.priorityGroup** | **1 to 25** | **24** |

**Push-To-Talk and Group Paging Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The paging group to use for priority pages. | | |
| **ptt.pageMode.transmit.timeout.continuation** | **0 to 65535** | **60** |
| The time (in seconds) to add to the initial timeout (`ptt.pageMode.transmit.timeout.initial`) for terminating page announcements. If this value is non-zero, an **Extend** soft key displays on the phone. Pressing the **Extend** soft key continues the initial timeout for the time specified by this parameter. If 0, announcements cannot be extended. | | |
| **ptt.pageMode.transmit.timeout.initial** | **0 to 65535** | **0** |
| The number of seconds to wait before automatically terminating an outgoing page announcement. If 0, page announcements do not automatically terminate. | | |
| **ptt.pttMode.enable** | **0 or 1** | **0** |
| Enable or disable push-to talk mode. | | |
| **ptt.volume** | **-57 to 0** | **-20** |
| Controls the volume level for pages without changing the volume level for incoming calls. | | |

# <qos/>

These parameters listed in the next table configure trol the following Quality of Service (QoS) options:

- The 802.1p/Q user_priority field RTP, call control, and other packets
- The “type of service” field RTP and call control packets

**Quality of Service (Type-of-Service) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **qos.ethernet.callControl.user_priority**[1] | **0 to 7** | **5** |
| User-priority used for call control packets. | | |
| **qos.ethernet.other.user_priority**[1] | **0 to 7** | **2** |
| User-priority used for packets that do not have a per-protocol setting. | | |
| **qos.ethernet.rtp.user_priority**[1] | **0 to 7** | **5** |
| Choose the priority of voice Real-Time Protocol (RTP) packets. The default priority level is 5. | | |
| **qos.ethernet.rtp.video.user_priority**[1] | **0 to 7** | **5** |
| User-priority used for Video RTP packets. | | |
| **qos.ip.callControl.dscp**[1] | **0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43** | **Null** |

**Quality of Service (Type-of-Service) Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Specify the DSCP of packets. If the value is not null, this parameter overrides the other `qos.ip.callControl.*` parameters. The default value is Null, so the other `qos.ip.callControl.*` parameters are used if no value is entered. | | |
| **qos.ip.callControl.max_reliability**[1] | **0 or 1** | **0** |
| **qos.ip.callControl.max_throughput**[1] | **0 or 1** | **0** |
| **qos.ip.callControl.min_cost**[1] | **0 or 1** | **0** |
| **qos.ip.callControl.min_delay**[1] | **0 or 1** | **1** |
| **qos.ip.callControl.precedence**[1] | **0 -7** | **5** |
| Set the bits in the IP ToS field of the IP header used for call control. Specify whether or not to set the max reliability bit, the max throughput bit, the min cost bit, the min delay bit, and the precedence bits.<br>If 0, the bit in the IP ToS field of the IP header is not set. If 1, the bit is set. | | |
| **qos.ip.rtp.dscp**[1] | **0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43** | **Null** |
| Specify the DSCP of packets. If the value is not null, this parameter overrides the other `qos.ip.rtp.*` parameters. The default value is Null, so the other `quality.ip.rtp.*` parameters are used. | | |
| **qos.ip.rtp.max_reliability**[1] | **0 or 1** | **0** |
| **qos.ip.rtp.max_throughput**[1] | **0 or 1** | **1** |
| **qos.ip.rtp.min_cost**[1] | **0 or 1** | **0** |
| **qos.ip.rtp.min_delay**[1] | **0 or 1** | **1** |
| **qos.ip.rtp.precedence**[1] | **0 -7** | **5** |
| Set the bits in the IP ToS field of the IP header used for RTP. Specify whether or not to set the max reliability bit, the max throughput bit, the min cost bit, the min delay bit, and the precedence bit.<br>If 0, the bit in the IP ToS field of the IP header is not set. If 1, the bit is set. | | |
| **qos.ip.rtp.video.dscp**[1] | **0 to 63 or EF or any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43** | **Null** |
| Allows the DSCP of packets to be specified. If the value is non-null, this parameter overrides the other `qos.ip.rtp.video.*`parameters. The default value is Null, so the other `qos.ip.rtp.video.*` parameters are used. | | |
| **qos.ip.rtp.video.max_reliability**[1] | **0 or 1** | **0** |
| **qos.ip.rtp.video.max_throughput**[1] | **0 or 1** | **1** |
| **qos.ip.rtp.video.min_cost**[1] | **0 or 1** | **0** |
| **qos.ip.rtp.video.min_delay**[1] | **0 or 1** | **1** |
| **qos.ip.rtp.video.precedence**[1] | **0 -7** | **5** |
| Set the bits in the IP ToS field of the IP header used for RTP video. Specify whether or not to set the max reliability bit, the max throughput bit, the min cost bit, the min delay bit, and the precedence bit.<br>If 0, the bit in the IP ToS field of the IP header is not set. If 1, the bit is set. | | |

[1] Change causes phone to restart or reboot.

# <reg/>

This section lists all per-registration parameters you can configure. Per-registration parameters apply to a single unique registered line on a phone. You also have the option of associating each registration with a private array of servers for segregated signaling. To see the maximum number of registered lines all Polycom phones support see the table Flexible Call Appearances.

The tables Registration Parameters and Registration Server Parameters list all line registration and server registration parameters.

**Registration Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **reg.x.acd-login-logout**<br>**reg.x.acd-agent-available** | **0 or 1**<br>**0 or 1** | **0**<br>**0** |
| If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration. | | |
| **reg.x.address** | **string address** | **Null** |
| The user part (for example, 1002) or the user and the host part (for example, `1002@polycom.com`) of the registration SIP URI or the H.323 ID/extension. | | |
| **reg.x.applyServerDigitMapLocally** | **0 or 1** | **0** |
| If 1 and `reg.x.server.y.specialInterop` is set to `lync2010`, the phone uses the dialplan from the Microsoft Lync Server. Any dialed number applies the dial plan locally.<br>If 0, the dialplan from the Microsoft Lync Server is not used. | | |
| **reg.x.advancedConference.maxParticipants** | **0-25** | **3** |
| Sets the maximum number of participants allowed in a push to conference forn advanced conference calls. The number of participants configured must match the number of participants allowed on the ALU CTS. | | |
| **reg.x.advancedConference.pushToConference** | **0 or 1** | **0** |
| Enable or disable push-to-conference functionality.for advanced conferences for ALU. If enabled, users can select multiple contacts when initiating a conference and during an active conference.<br>Note: The values for this parameter must match does configured on the server. | | |
| **reg.x.advancedConference.subscribeForConfEvents** | **0 or 1** | **1** |
| Enable or disable conference participants to receive notifications for conference events. | | |
| **reg.x.advancedConference.subscribeForConfEventsOnCCPE** | **0 or 1** | **1** |
| Enable or disable the conference host to receive notifications for conference events. | | |
| **reg.x.auth.domain** | **string** | **Null** |
| The domain of the authorization server that is used to check the user names and passwords. | | |
| **reg.x.auth.optimizedInFailover** | **0 or 1** | **0** |

**Registration Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The destination of the first new SIP request when failover occurs. If 0, the SIP request is sent to the server with the highest priority in the server list. If 1, the SIP request is sent to the server which sent the proxy authentication request. | | |
| **reg.x.auth.password** | **string** | **Null** |
| The password to be used for authentication challenges for this registration. If the password is non-Null, it overrides the password entered into the Authentication submenu on the Settings menu of the phone. | | |
| **reg.x.auth.userId** | **string** | **Null** |
| User ID to be used for authentication challenges for this registration. If the User ID is non-Null, it overrides the user parameter entered into the Authentication submenu on the Settings menu of the phone. | | |
| **reg.x.auth.useLoginCredentials** | **0 or 1** | **0** |
| If 0, login credentials are not used for authentication to the server on registration x. If 1, login credentials are used for authentication to the server. | | |
| **reg.x.bargeInEnabled** | **0 or 1** | **0** |
| If 0, barge-in is disabled for line x. If 1, barge-in is enabled (remote users of shared call appearances can interrupt or barge in to active calls. | | |
| **reg.x.bridgeInEnabled** | **0 or 1** | **0** |
| Enable or disable the Bridge In feature. | | |
| **reg.x.broadsoft.userId** | **String** | **Null** |
| Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. | | |
| **reg.x.broadsoft.useXspCredentials** | **0 or 1** | **1** |
| Set to 0 if registering lines with a server running BroadWorks R19 SP1 or later. Set to 1 if registering lines with a server running BroadWorks R19 or earlier.<br>If this parameter is disabled, the phones use standard SIP credentials to authenticate. | | |
| **reg.x.broadsoft.xsp.password** | **String** | **Null** |
| Enter the password associated with the BroadSoft user account for the line. Required only when `reg.x.broadsoft.useXspCredentials=1`. | | |
| **reg.x.callsPerLineKey**[1] | **1-8, 1-24** | **24 (for VVX phones)** |
| Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration. This parameter overrides `call.callsPerLineKey`. | | |
| **reg.x.csta** | **0 or 1** | **0** |
| If 0, the uaCSTA (User Agent Computer Supported Telecommunications Applications) feature is disabled. If 1, uaCSTA is enabled (overrides the global parameter `voIpProt.SIP.csta`. | | |
| **reg.x.displayName** | **UTF-8 encoded string** | **Null** |

**Registration Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The display name used in SIP signaling and/or the H.323 alias used as the default caller ID. | | |
| **reg.x.enablePvtHoldSoftKey** | **0 or 1** | **0** |
| Enable or disable the Private Hold soft key for a specific line. Set to 1 to display the PvtHold soft key. This parameter applies only to shared lines. | | |
| **reg.x.enhancedCallPark.enabled** | **0 or 1** | **0** |
| Enables and disables the BroadWorks Enhanced Call Park feature. | | |
| **reg.x.filterReflectedBlaDialogs** | **0 or 1** | **1** |
| If 0, bridged line appearance NOTIFY messages (dialog state change) is not ignored. If 1, the messages are ignored. | | |
| **reg.x.fwd.busy.contact** | **string** | **Null** |
| The forward-to contact for calls forwarded due to busy status. If Null, the contact specified by `divert.x.contact` is used. | | |
| **reg.x.fwd.busy.status** | **0 or 1** | **0** |
| If 0, incoming calls that receive a busy signal is not forwarded. If 1, busy calls are forwarded to the contact specified by `reg.x.fwd.busy.contact`. | | |
| **reg.x.fwd.noanswer.contact** | **string** | **Null** |
| The forward-to contact used for calls forwarded due to no answer. If Null, the contact specified by `divert.x.contact` is used. | | |
| **reg.x.fwd.noanswer.ringCount** | **0 to 65535** | **0** |
| The number of seconds the phone should ring for before the call is forwarded because of no answer. Note: The maximum value accepted by some call servers is 20. | | |
| **reg.x.fwd.noanswer.status** | **0 or 1** | **0** |
| If 0, calls are not forwarded if there is no answer. If 1, calls are forwarded to the contact specified by `reg.x.noanswer.contact` after ringing for the length of time specified by `reg.x.fwd.noanswer.ringCount`. | | |
| **reg.x.gruu** | **0 or 1** | **0** |
| Specify if the phone sends sip.instance in the REGISTER request. | | |

**Registration Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **reg.x.label** | **UTF-8 encoded string** | **Null** |
| The text label that displays next to the line key for registration x.<br>If Null, the label is determined as follows:<br>• If reg.1.useteluriAsLineLabel=1, then the tel URI/phone number/address displays as the label.<br>• If reg.1.useteluriAsLineLabel=0, then the value for reg.x.displayName, if available, displays as the label. If reg.x.displayName is unavailable, the user part of reg.x.address is used.<br>Note that the maximum number of characters for this parameter value is 256; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters you use. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The rules for parameter up.cfgLabelElide determine how the label is truncated. | | |
| **reg.x.lineAddress** | **String** | **Null** |
| The line extension for a shared line. If there is no extension provided for this parameter, the call park notification is ignored for the shared line.<br>Note: This parameter does not apply for private lines. | | |
| **reg.x.lineKeys** | **1 to max** | **1** |
| Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model. | | |
| **reg.x.line.y.label** | | |
| Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when up.cfgUniqueLineLabel=1. Note that if reg.x.linekeys=1, this parameter does not have any effect.<br>x = the registration index number starting from 1.<br>Y = the line index from 1 to the value set by reg.x.linekeys. Specifying a string sets the label used for the line key registration on phones with multiple line keys.<br>If no parameter value is set for reg.x.line.y.label, the phone automatically numbers multiple lines by prepending "<y>_" where <y> is the line index from 1 to the value set by reg.x.linekeys.<br>• The following examples show labels for line 1 on a phone with user registration 1234, where reg.x.linekeys=2:<br>➢ If no label is configured for registration, the labels are "1_1234" and "2_1234".<br>➢ If reg.1.line.1.label=Polycom and reg.1.line.2.label=VVX, the labels display as 'Polycom' and 'VVX'. | | |
| **reg.x.lisdisclaimer** | **string, 0 to 256 characters** | **Null** |
| This parameter sets the value of the location policy disclaimer. For example, the disclaimer may be "Warning: If you do not provide a location, emergency services may be delayed in reaching your location should you need to call for help." | | |
| **reg.x.musicOnHold.uri** | **a SIP URI** | **Null** |
| A URI that provides the media stream to play for the remote party on hold. If present and not Null, this parameter overrides voIpProt.SIP.musicOnHold.uri. | | |

**Registration Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **reg.x.outboundProxy.address** | **IP address or hostname** | **Null** |
| The IP address or hostname of the SIP server to which the phone sends all requests. | | |
| **reg.x.outboundProxy.failOver.failBack.mode** | **newRequests**<br>**DNSTTL registration**<br>**duration** | **duration** |
| The mode for failover failback (overrides `reg.x.server.y.failOver.failBack.mode`).<br>`newRequests` all new requests are forwarded first to the primary server regardless of the last used server.<br>`DNSTTL` the phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.<br>`registration` the phone tries the primary server again when the registration renewal signaling begins.<br>`duration` the phone tries the primary server again after the time specified by `reg.x.outboundProxy.failOver.failBack.timeout` expires. | | |
| **reg.x.outboundProxy.failOver.failBack.timeout** | **0, 60 to 65535** | **3600** |
| The time to wait (in seconds) before failback occurs (overrides `reg.x.server.y.failOver.failBack.timeout`).If the fail back mode is set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again. If 0, the phone does not fail back until a failover event occurs with the current server. | | |
| **reg.x.outboundProxy.failOver.failRegistrationOn** | **0 or 1** | **0** |
| When set to 1, and the reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over. When set to 0, and the reRegisterOn parameter is enabled, existing registrations remain active. This means that the phone attempts failback without first attempting to register with the primary server to determine if it has recovered.<br>Note that `reg.x.outboundProxy.failOver.RegisterOn` must be enabled. | | |
| **reg.x.outboundProxy.failOver.onlySignalWithRegistered** | **0 or 1** | **1** |
| When set to 1, and the reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. When set to `0`, and the reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that h as failed (even though failback hasn’t been attempted or failover hasn’t occurred). | | |
| **reg.x.outboundProxy.failOver.reRegisterOn** | **0 or 1** | **0** |
| This parameters overrides `reg.x.server.y.failOver.failBack.RegisterOn`. When set to 1, the phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server. When set to `0`, the phone won’t attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information. | | |
| **reg.x.outboundProxy.port** | **1 to 65535** | **0** |
| The port of the SIP server to which the phone sends all requests. | | |

**Registration Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **reg.x.outboundProxy.transport** | **DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly** | **DNSnaptr** |
| The transport method the phone uses to communicate with the SIP server.<br>**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.<br>**TCPpreferred** TCP is the preferred transport, UDP is used if TCP fails.<br>**UDPOnly** Only UDP is used.<br>**TLS** If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.<br>**TCPOnly** Only TCP will be used. | | |
| **reg.x.protocol.H323** | **0 or 1** | **0** |
| You can use this parameter for the VVX 500/501, 600/601, and 1500. If 0, H.323 signaling is not enabled for registration x. If 1, H.323 signaling is enabled. | | |
| **reg.x.protocol.SIP** | **0 or 1** | **1** |
| You can use this parameter for the VVX 500/501, 600/601, and 1500. If 0, SIP signaling is not enabled for this registration. If 1, SIP signaling is enabled. | | |
| **reg.x.proxyRequire** | **string** | **Null** |
| The string that needs to be entered in the Proxy-Require header. If Null, no Proxy-Require is sent. | | |
| **reg.x.ringType** | **default, ringer1 to ringer24** | **ringer2** |
| The ringer to be used for calls received by this registration. The default is the first non-silent ringer.<br>If you use the configuration parameters ringer13 and ringer14 on a single registered line, the phone plays SystemRing.wav. | | |
| **reg.x.serverFeatureControl.callRecording** | **0 or 1** | **1** |
| Enable or disable BroadSoft BroadWorks v20 call recording feature for individual phone lines. This per-line parameter overrides values you set for the parameter `voIpProt.SIP.serverFeatureControl.callRecording` which sets the feature for all lines on a phone. | | |
| **reg.x.serverFeatureControl.cf**[1] | **0 or 1** | **0** |
| If 0, server-based call forwarding is not enabled. If 1, server based call forwarding is enabled. This parameter overrides `voIpProt.SIP.serverFeatureControl.cf`. | | |
| **reg.x.serverFeatureControl.dnd**[1] | **0 or 1** | **0** |
| If 0, server-based do-not-disturb (DND) is not enabled. If 1, server-based DND is enabled and the call server has control of DND. This parameter overrides `voIpProt.SIP.serverFeatureControl.dnd.` | | |

**Registration Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **reg.x.serverFeatureControl.localProcessing.cf** | **0 or 1** | **1** |
| If 0 and `reg.x.serverFeatureControl.cf` is set to 1, the phone does not perform local Call Forward behavior. If set to 1, the phone performs local Call Forward behavior on all calls received. This parameter overrides `voIpProt.SIP.serverFeatureControl.localProcessing.cf`. | | |
| **reg.x.serverFeatureControl.localProcessing.dnd** | **0 or 1** | **1** |
| If 0 and `reg.x.serverFeatureControl.dnd` is set to 1, the phone does not perform local DND call behavior. If set to 1, the phone performs local DND call behavior on all calls received. This parameter overrides `voIpProt.SIP.serverFeatureControl.localProcessing.dnd`. | | |
| **reg.x.serverFeatureControl.securityClassification** | **0 or 1** | **0** |
| Enable or disable the visual security classification feature for a specific phone line. | | |
| **reg.x.serverFeatureControl.signalingMethod** | **string** | **serviceMsForwardContact** |
| Controls the method used to perform call forwarding requests to the server. | | |
| **reg.x.server.y.registerRetry.maxTimeout** | | **180 seconds** |
| Set the maximum period of time in seconds that you want the phone to try registering with the server. | | |
| **reg.x.srtp.enable**[1] | **0 or 1** | **1** |
| If 0, the registration always declines SRTP offers. If 1, the registration accepts SRTP offers. | | |
| **reg.x.srtp.offer**[1] | **0 or 1** | **0** |
| If 1, the registration includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE. This parameter applies to the registration initiating (offering) a phone call. If 0, no secure media stream is included in SDP of a SIP invite. | | |
| **reg.x.srtp.require**[1] | **0 or 1** | **0** |
| If 0, secure media streams are not required. If 1, the registration is only allowed to use secure media streams. Any offered SIP INVITEs must include a secure media description in the SDP or the call is rejected. For outgoing calls, only a secure media stream description is included in the SDP of the SIP INVITE, meaning that the non-secure media description is not included. If this parameter set to 1, `reg.x.srtp.offer` is also set to 1, regardless of the value in the configuration file. | | |
| **reg.x.srtp.simplifiedBestEffort** | **0 or 1** | **0** |
| If 0, no SRTP is supported. If 1, negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported. This parameter overrides `sec.srtp.simplifiedBestEffort`. | | |
| **reg.x.strictLineSeize** | **0 or 1** | **0** |
| If 1, the phone is forced to wait for 200 OK on registration x when receiving a TRYING notify. If set to 0, dial prompt is provided immediately when you attempt to seize a shared line without waiting for a successful OK from the call server. This parameter overrides `voIpProt.SIP.strictLineSeize` for registration x. | | |

**Registration Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **reg.x.tcpFastFailover** | **0 or 1** | **0** |
| If 1, failover occurs based on the values of `reg.x.server.y.retryMaxCount` and `voIpProt.server.x.retryTimeOut`. If 0, a full 32 second RFC compliant timeout is used. | | |
| **reg.x.thirdPartyName** | **string address** | **Null** |
| This field must match the `reg.x.address` value of the registration which makes up the part of a bridged line appearance (BLA). It must be Null in all other cases. | | |
| **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.useCompleteUriForRetrieve** | **0 or 1** | **1** |
| This parameters overrides `voipPort.SIP.useCompleteUriForRetrieve`. If set to 1, the target URI in BLF signaling uses the complete address as provided in the xml dialog document.<br>If set to 0, only the user portion of the XML dialog document is used and the current registrar's domain is appended to create the full target URI. | | |

[1] Change causes phone to restart or reboot.

You can list multiple registration servers for fault tolerance. The next table shows how you can list up to four servers by using y=1 to 4. If `reg.x.server.y.address` is not null, all of the parameters in the following table override the parameters specified in `voIpProt.server.*`.

**Registration Server Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **regOnPhone** | **0 or 1** | **Lync = 0** |
| Enables and disables all line keys. If 1 (enabled), the line keys are moved to the expansion module if one is connected. | | |
| **reg.x.server.H323.y.address** | **IP address or hostname** | **Null** |
| Address of the H.323 gatekeeper. | | |
| **reg.x.server.H323.y.port** | **0 to 65535** | **0** |
| 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** | **positive integer** | **3600** |
| Desired registration period. | | |
| **reg.x.server.y.address** | **IP address or hostname** | **Null** |

**Registration Server Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The IP address or host name of a SIP server that accepts registrations. If not Null, all of the parameters in this table override the parameters specified in `voIpProt.server.*`. Notes: If this parameter is set, it takes precedence even if the DHCP server is available. | | |
| **reg.x.server.y.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 attempts to re-register at the beginning of the overlap period. For example, if `expires="300"` and `overlap="5"`, the phone re-registers after 295 seconds (300–5). | | |
| **reg.x.server.y.expires.lineSeize** | **0 to 65535** | **30** |
| Requested line-seize subscription period. | | |
| **reg.x.server.y.expires.overlap** | **5 to 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 tries to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value. | | |
| **reg.x.server.y.failOver.failBack.mode** | **newRequests**<br>**DNSTTL**<br>**registration**<br>**duration** | **duration** |
| The mode for failover failback (this parameter overrides `voIpProt.server.x.failOver.failBack.mode`):<br>• **newRequests** All new requests are forwarded first to the primary server regardless of the last used server.<br>• **DNSTTL** The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.<br>• **registration** The phone tries the primary server again when the registration renewal signaling begins.<br>**duration** The phone tries the primary server again after the time specified by `reg.x.server.y.failOver.failBack.timeout`. | | |
| **reg.x.server.y.failOver.failBack.timeout** | **0, 60 to 65535** | **3600** |
| The time to wait (in seconds) before failback occurs (overrides `voIpProt.server.x.failOver.failBack.timeout`).If the fail back mode is set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again. If 0, the phone does not fail back until a failover event occurs with the current server. | | |
| **reg.x.server.y.failOver.failRegistrationOn** | **0 or 1** | **0** |
| When set to 1, and the reRegisterOn parameter is enabled, the phone silently invalidate an existing registration (if it exists), at the point of failing over. When set to 0, and the reRegisterOn parameter is enabled, existing registrations remain active. This means that the phone attempts failback without first attempting to register with the primary server to determine if it has recovered. | | |
| **reg.x.server.y.failOver.onlySignalWithRegistered** | **0 or 1** | **1** |

**Registration Server Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| When set to 1, and the reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. When set to 0, and the reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred). | | |
| **reg.x.server.y.failOver.reRegisterOn** | **0 or 1** | **0** |
| This parameter overrides `voIpProt.server.x.failOver.reRegisterOn`. When set to 1, the phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server. When set to 0, the phone won't attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information. | | |
| **reg.x.server.y.useOutboundProxy** | **0 or 1** | **1** |
| Specify whether or not to use the outbound proxy specified in `reg.x.outboundProxy.address` for server x. This parameter overrides `voIpProt.server.x.useOutboundProxy` for registration x. | | |
| **reg.x.server.y.port** | **0, 1 to 65535** | **Null** |
| The port of the sip server that specifies registrations. If 0, the port used depends on `reg.x.server.y.transport`. | | |
| **reg.x.server.y.register** | **0 or 1** | **1** |
| If 0, calls can be routed to an outbound proxy without registration. See voIpProt.server.x.register.<br>For more information, see *SIP Server Fallback Enhancements on Polycom Phones - Technical Bulletin 5844* on Polycom Engineering Advisories and Technical Notifications. | | |
| **reg.x.server.y.registerRetry.baseTimeOut** | **10 - 120** | **60** |
| The base time period to wait before a registration retry. Used in conjunction with `reg.x.server.y.registerRetry.maxTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626. This value is used as the 'base' in the exponential backoff algorithm. | | |
| **reg.x.server.y.registerRetry.maxTimeOut** | **60 - 1800** | **60** |
| The maximum time limits the upper value of the exponential backoff timer. Used in conjunction with `reg.x.server.y.registerRetry.baseTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626. | | |
| **reg.x.server.y.retryMaxCount** | **0 to 20** | **3** |
| If set to 0, 3 is used. The number of retries attempted before moving to the next available server. | | |
| **reg.x.server.y.retryTimeOut** | **0 to 65535** | **0** |
| The amount of time (in milliseconds) to wait between retries. If 0, use standard RFC 3261 signaling retry behavior. | | |
| **reg.x.server.y.specialInterop** | **standard,**<br>**lync2010,**<br>**GENBAND** | **standard** |

**Registration Server Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| Specify if this registration should support Microsoft Lync 2010 (lync2010), or GENBAND. | | |
| **reg.x.server.y.subscribe.expires** | **10 – 2147483647 seconds** | **3600 seconds** |
| The phone’s requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. You can use this parameter in conjunction with `reg.x.server.y.subscribe.expires.overlap`. For example, if expires=”300” and overlap=”5”, the phone resubscribes after 295 seconds (300–5). Note that the period negotiated with the server may be different. | | |
| **reg.x.server.y.subscribe.expires.overlap** | **5 – 65535 seconds** | **60 seconds** |
| The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. | | |
| **reg.x.server.y.transport** | **DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly** | **DNSnaptr** |
| The transport method the phone uses to communicate with the SIP server.<br>• **Null** or **DNSnaptr** If `reg.x.server.y.address` is a hostname and `reg.x.server.y.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.server.y.address` is an IP address, or a port is given, then UDP is used.<br>• **TCPpreferred** TCP is the preferred transport; UDP is used if TCP fails.<br>• **UDPOnly** Only UDP is used.<br>• **TLS** If TLS fails, transport fails. Leave port field empty (defaults to `5061`) or set to `5061`.<br>• **TCPOnly** Only TCP is used. | | |

# <request/>

The parameters listed in the following table configure the phone’s behavior when a request for restart or reconfiguration is received.

**Configuration Request Parameter**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **request.delay.type**[1] | **audio, call** | **call** |
| Specify when the phone should process a request for a restart or reconfiguration. If set to `audio`, the request is executed once there is no active audio on the phone—regardless of the call state. If set to `call`, the request should be executed once there are no calls —in any state—on the phone. | | |

[1] Change causes phone to restart or reboot.

# <saf/>

The phone uses built-in sampled audio files (SAF) in wave file format for some sound effects. You can add files downloaded from the provisioning server or from the Internet. Ringtone files are stored in volatile memory which allows a maximum size of 600 kilobytes (614400 bytes) for all ringtones.

The phones support the following sampled audio WAVE (.wav) file formats:

- mono 8 kHz G.711 u-Law   Supported on all phones
- mono L16/8000 (16-bit dynamic range, 8-kHz sample rate)   Supported on all phones
- G.711 A-Law   Supported on all phones
- mono 8 kHz A-law/mu-law   Supported on all phones
- L8/16000 (16-bit, 8 kHz sampling rate, mono)   Supported on all phones
- L16/16000 (16-bit, 16 kHz sampling rate, mono)   Supported on all phones
- L16/32000 (16-bit, 32 kHz sampling rate, mono)   Supported on VVX 500/501, 600/601, and 1500
- L16/44100 (16-bit, 44.1 kHz sampling rate, mono)   Supported on VVX 500/501, 600/601, and 1500
- L16/48000 (16-bit, 48 kHz sampling rate, mono)   Supported on VVX 500/501, 600/601, and 1500

In the following table, *x* is the sampled audio file number.

**Sampled Audio File Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **saf.x** | **Null, valid path name, or an RFC 1738-compliant URL to an HTTP, FTP, or TFTP wave file resource.** | |
| • If Null, the phone uses a built-in file.<br>• If set to a path name, the phone attempt<br>• s to download this file at boot time from the provisioning server.<br>• If set to a URL, the phone attempt<br>• s to download this file at boot time from the Internet.<br>Note: A TFTP URL must be in the format: `tftp://<host>/[pathname]<filename>`, for example: `tftp://somehost.example.com/sounds/example.wav`.<br>Note that to use a welcome sound you must enable the parameter `up.welcomeSoundEnabled` and specify a file in `saf.x`. The default UC Software welcome sound file is `Welcome.wav`. For information, see the section Customize Audio Sound Effects. | | |

The next table defines the phone's default use of the sampled audio files.

**Default Sample Audio File Usage**

| *Sampled Audio File Number* | *Default Use (Pattern Reference)* |
|---|---|
| 1 | Ringer 12 (`se.pat.misc.welcome`) |
| 2 | Ringer 15 (`se.pat.ringer.ringer15`) |
| 3 | Ringer 16 (`se.pat.ringer.ringer16`) |
| 4 | Ringer 17 (`se.pat.ringer.ringer17`) |

**Default Sample Audio File Usage (continued)**

| *Sampled Audio File Number* | *Default Use (Pattern Reference)* |
|---|---|
| 5 | Ringer 18 (`se.pat.ringer.ringer18`) |
| 6 | Ringer 19 (`se.pat.ringer.ringer19`) |
| 7 | Ringer 20 (`se.pat.ringer.ringer20`) |
| 8 | Ringer 21 (`se.pat.ringer.ringer21`) |
| 9 | Ringer 22 (`se.pat.ringer.ringer22`) |
| 10 | Ringer 23 (`se.pat.ringer.ringer23`) |
| 11 | Ringer 24 (`se.pat.ringer.ringer24`) |
| 12 to 24 | Not Used |

# <se/>

The next table lists configurable sound effect parameters. Sound effects are defined by patterns: rudimentary sequences of chord-sets, silence periods, and wave files. You can also configure sound effect patterns in <pat/>and ringtones in <rt/>. The phone uses both synthesized and sampled audio sound effects.

**Sound Effect Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **se.appLocalEnabled**[1] | **0 or 1** | **1** |
| If set to 1, local user interface sound effects such as confirmation/error tones is enabled. | | |
| **se.destination** | **chassis, headset, handset, active** | **1** |
| The transducer or audio device that plays sound effects and alerts. Choose from the `chassis` (speakerphone), `headset` (if connected), `handset`, or the `active` destination. If `active`, alerts play from the destination that is currently in use. For example, if you are in a call on the handset, a new incoming call rings on the handset. | | |
| **se.stutterOnVoiceMail** | **0 or 1** | **1** |
| If set to 1, a stuttered dial tone is used in place of a normal dial tone to indicate that one or more voicemail messages are waiting at the message center. | | |

# <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 instructions shown in the next table.

**Sound Effects Pattern Types**

| *Instruction* | *Meaning* |
|---|---|
| **sampled (n)** | **Play sampled audio file n** |
| Example:<br>se.pat.misc.SAMPLED_1.inst.1.type =”sampled” (sampled audio file instruction type)<br>se.pat.misc.SAMPLED_1.inst.1.value =”2” (specifies sampled audio file 2) | |
| **chord (n, d)** | **Play chord set n (d is optional and allows the chord set ON duration to be overridden to d milliseconds)** |
| Example:<br>se.pat.callProg.busyTone.inst.2.type = “chord” (chord set instruction type)<br>se.pat.callProg.busyTone.inst.2.value = “busyTone” (specifies sampled audio file busyTone)<br>se.pat.callProg.busyTone.inst.2.param = “2000” (override ON duration of chord set to 2000 milliseconds) | |
| **silence (d)** | **Play silence for d milliseconds (Rx audio is not muted)** |
| Example:<br>se.pat.callProg.bargeIn.inst.3.type = “silence” (silence instruction type)<br>se.pat.callProg.bargeIn.inst.3.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)** |
| Example:<br>se.pat.callProg.alerting.inst.4.type = “branch” (branch instruction type)<br>se.pat.callProg.alerting.inst.4.value = “-2” (step back 2 instructions and execute that instruction) | |

In the following table, x is the pattern name, y is the instruction number, and cat is the sound effect pattern category. Both x and y need to be sequential. There are three categories of sound effect patterns that you can use to replace cat in the parameter names: callProg (Call Progress Patterns), ringer (Ringer Patterns) and misc (Miscellaneous Patterns).

**Sound Effects Pattern Parameters**

| *Parameter* | *Permitted Values* |
|---|---|
| **se.pat.callProg.secondaryDialTone.name** | **1-255** |
| Found in region.cfg | |
| **se.pat.callProg.secondaryDialTone.inst.1.type** | **0-255** |
| Found in region.cfg | |
| **se.pat.callProg.secondaryDialTone.inst.1.value** | **0-50** |
| Found in region.cfg | |
| **se.pat.callProg.secondaryDialTone.inst.1.param** | |

**Sound Effects Pattern Parameters (continued)**

| | |
|---|---|
| This is a debug parameter. | |
| **se.pat.callProg.secondaryDialTone.inst.1.atten** | |
| This is a debug parameter. | |
| **se.pat.callProg.secondaryDialTone.inst.1.atten** | **UTF-8 encoded string** |
| Sound effects name, where cat is `callProg`, `ringer`, or `misc`. | |
| **se.pat.cat.x.inst.y.type** | **sampled, chord, silence, branch** |
| Type of sound effect, where cat is `callProg`, `ringer`, or `misc`. | |
| **se.pat.cat.x.inst.y.value** | **String** |
| The instruction: `sampled` – sampled audio file number, `chord` – type of sound effect, `silence` – silence duration in ms, `branch` – number of instructions to advance. `cat` is `callProg`, `ringer`, or `misc`. | |

The next table lists the call progress pattern names and their descriptions.

**Call Progress Tone Pattern Names**

| *Call Progress Pattern Name* | *Description* |
|---|---|
| alerting | Alerting |
| bargeIn | Barge-in tone |
| busyTone | Busy tone |
| callWaiting | Call waiting tone |
| callWaitingLong | Call waiting tone long (distinctive) |
| confirmation | Confirmation tone |
| dialTone | Dial tone |
| howler | Howler tone (off-hook warning) |
| intercom | Intercom announcement tone |
| msgWaiting | Message waiting tone |
| precedenceCallWaiting | Precedence call waiting tone |
| precedenceRingback | Precedence ringback tone |
| preemption | Preemption tone |
| precedence | Precedence tone |
| recWarning | Record warning |
| reorder | Reorder tone |

**Call Progress Tone Pattern Names (continued)**

| *Call Progress Pattern Name* | *Description* |
|---|---|
| ringback | Ringback tone |
| secondaryDialTone | Secondary dial tone |
| stutter | Stuttered dial tone |

The next table lists the ring pattern names and their default descriptions.

**Ringtone Pattern Names**

| Parameter Name | Ringtone Name | Description |
|---|---|---|
| ringer1 | Silent Ring | Silent ring |
| ringer2 | Low Trill | Long single A3 Db3 major warble |
| ringer3 | Low Double Trill | Short double A3 Db3 major warble |
| ringer4 | Medium Trill | Long single C3 E3 major warble |
| ringer5 | Medium Double Trill | Short double C3 E3 major warble |
| ringer6 | High Trill | Long single warble 1 |
| ringer7 | High Double Trill | Short double warble 1 |
| ringer8 | Highest Trill | Long single Gb3 A4 major warble |
| ringer9 | Highest Double Trill | Short double Gb3 A4 major warble |
| ringer10 | Beeble | Short double E3 major |
| ringer11 | Triplet | Short triple C3 E3 G3 major ramp |
| ringer12 | Ringback-style | Short double ringback |
| ringer13 | Low Trill Precedence | Long single A3 Db3 major warble Precedence |
| ringer14 | Ring Splash | Splash |
| ringer15 | Ring16 | Sampled audio file 1 |
| ringer16 | Ring17 | Sampled audio file 2 |
| ringer17 | Ring18 | Sampled audio file 3 |
| ringer18 | Ring19 | Sampled audio file 4 |
| ringer19 | Ring20 | Sampled audio file 5 |
| ringer20 | Ring21 | Sampled audio file 6 |
| ringer21 | Ring22 | Sampled audio file 7 |
| ringer22 | Ring23 | Sampled audio file 8 |

**Ringtone Pattern Names (continued)**

| Parameter Name | Ringtone Name | Description |
|---|---|---|
| ringer23 | Ring24 | Sampled audio file 9 |
| ringer24 | Ring25 | Sampled audio file 10 |

**Note: Silent ring**

Silent ring provides a visual indication of an incoming call, but no audio indication.

Sampled audio files 1 to 10 all use the same built-in file unless that file has been replaced with a downloaded file. For more information, see

The next table lists the miscellaneous patterns and their descriptions.

**Miscellaneous Pattern Names**

| *Parameter Name* | *Miscellaneous pattern name* | *Description* |
|---|---|---|
| instantmessage | instant message | New instant message |
| localHoldNotification | local hold notification | Local hold notification |
| messageWaiting | message waiting | New message waiting indication |
| negativeConfirm | negative confirmation | Negative confirmation |
| positiveConfirm | positive confirmation | Positive confirmation |
| remoteHoldNotification | remote hold notification | Remote hold notification |
| welcome | welcome | Welcome (boot up) |

# <rt/>

Ringtone 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 parameters such as call-waiting and ringer index, if appropriate. The ring class can use one of four types of rings that are defined as follows:

- **Ring** Plays a specified ring pattern or call waiting indication.
- **Visual** Provides a visual indication (no audio) of an incoming call;, no ringer needs to be specified.
- **Answer** Provides auto-answer on an incoming call.
- **Ring-answer** Provides auto-answer on an incoming call after a certain number of rings.

**Note: Use the answer ring type**

The auto answer for an incoming call works only when there is no other call in progress on your phone, including no other calls in progress on phone lines you share or are monitoring. However, if a phone initiates a call on a line you are sharing or monitoring, auto answer on your phone works.

The phone supports the following ring classes:

- default

- **v**isual
- answerMute
- autoAnswer
- ringAnswerMute
- ringAutoAnswer
- internal
- external
- emergency
- precedence
- splash
- custom<*y*> where y is 1 to 17.

In the following table, x is the ring class name.

**Caution: Ringtone parameters do not work after a software downgrade**
If you are using Polycom UC Software 4.0.0 or later and then downgrade to SIP 3.2.3 or earlier, the ringtone parameters are unusable due to configuration parameters name changes in UC Software 4.0.0.

**Sound Effects Ringtone Parameters**

| *Parameter* | *Permitted Values* | *Default Value* |
|---|---|---|
| **se.rt.enabled** | **0 or 1** | **1** |
| If **0**, the ringtone feature is not enabled on the phone. If **1** (default), the ringtone feature is enabled. | | |
| **se.rt.modification.enabled** | **0 or 1** | **1** |
| A flag to determine whether or not to allow user modification (through phone's user interface) of the pre-defined ringtone enabled for modification. | | |
| **se.rt.<ringClass>.callWait** | **callWaiting, callWaitingLong, precedenceCallWaiting** | |
| The call waiting tone to be used for this class of ring. The call waiting should match one defined in the table Call Progress Tone Pattern Names. The default call waiting tone is `callWaiting`. | | |
| **se.rt.<ringClass>.name** | **UTF-8 encoded string** | |
| The answer mode for a ringtone. Used for identification purposes in the user interface. | | |
| **se.rt.<ringClass>.ringer** | **default, ringer1 to ringer24** | |
| The ringtone to be used for this class of ring. The ringer must match one in the table of Ringtone Pattern Names. The default ringer is `ringer2`. | | |
| **se.rt.<ringClass>.timeout** | **1 to 60000 only relevant if the type is set to ring-answer** | |
| The duration of the ring in milliseconds before the call is auto answered. The default is 2000. | | |

**Sound Effects Ringtone Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default Value* |
|---|---|---|
| **se.rt.<ringClass>.type** | **ring, visual, answer, ring-answer** | |
| The answer mode for a ringtone as defined in list earlier in this section. | | |

# <sec/>

The parameters listed in the next table configure security features of the phone.

**General Security Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.tagSerialNo**[1] | **0 or 1** | **0** |
| If 0, the phone does not advertise its serial number (MAC address) through protocol signaling. If 1, the phone may advertise its serial number through protocol signaling. | | |

[1] Change causes phone to restart or reboot.

This parameter also includes:

- <encryption/>
- <pwd/><length/>
- <srtp/>
- <H235/>
- <dot1x><eapollogoff/>
- <hostmovedetect/>
- <TLS/>

## <encryption/>

The next table lists available encryption parameters.

**File Encryption Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.encryption.upload.callLists**[1] | **0 or 1** | **0** |
| The encryption on the phone-specific call lists that is uploaded to the provisioning server.<br>If 0, the call list is uploaded unencrypted regardless of how it was downloaded, the directory replaces whatever phone-specific call list is on the server, even if the file on the server is encrypted.<br>If 1, the call list is uploaded encrypted regardless of how it was downloaded. The file replaces any existing phone-specific call lists file on the server. | | |
| **sec.encryption.upload.config** | **0 or 1** | **0** |

**File Encryption Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The encryption on the phone-specific configuration file created and uploaded to the provisioning server when the user selects **Upload Configuration** from the phone menu.<br>If 0, the file is uploaded unencrypted, and overwrites whatever phone-specific configuration file is on the server, even if the file on the server is encrypted.<br>If 1, the file is uploaded encrypted and replaces any existing phone-specific configuration file on the server. If there is no encryption key on the phone, the file is not uploaded. | | |
| **sec.encryption.upload.dir**[1] | **0 or 1** | **0** |
| The encryption on the phone-specific contact directory that is uploaded to the provisioning server.<br>If 0, the directory is uploaded unencrypted regardless of how it was downloaded, the directory replaces whatever phone-specific contact directory is on the server, even if the file on the server is encrypted.<br>If 1, the directory is uploaded encrypted regardless of how it was downloaded. The file replaces any existing phone-specific contact directory file on the server. | | |
| **sec.encryption.upload.overrides** | **0 or 1** | **0** |
| The encryption on the phone-specific **<MACaddress>-phone.cfg** override file that is uploaded to the server.<br>If 0, the file is uploaded unencrypted regardless of how it was downloaded, the file replaces whatever file was on the server, even if the file on the server is encrypted.<br>If 1, the file is uploaded encrypted regardless of how it was downloaded. The file replaces any existing phone-specific override file on the server. | | |

[1] Change causes phone to restart or reboot.

## <pwd/><length/>

The next table lists configurable password length parameters.

**Password Length Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **sec.pwd.length.admin**[1] | **0-32** | **1** |
| The minimum length for administrator passwords changed using the phone. Use 0 to allow null passwords. | | |
| **sec.pwd.length.user**[1] | **0-32** | **2** |
| The minimum length for user passwords changed using the phone. Use 0 to allow null passwords. | | |

[1] Change causes phone to restart or reboot.

## <srtp/>

As per RFC 3711, you cannot turn off authentication of RTCP. The next table lists SRTP parameters.

**SRTP Parameters**

| *Parameter* | *Permitted values* | *Defaults* |
|---|---|---|
| **sec.srtp.answerWithNewKey** | **0 or 1** | **1** |
| If 0, a new key is not provided when answering a call. If 1, a new key is provided when answering a call. | | |
| **sec.srtp.enable[1]** | **0 or 1** | **1** |
| If 0, the phone always declines SRTP offers. If 1, the phone accepts SRTP offers. Note: The defaults for SIP 3.2.0 was 0 when Null or not defined. | | |
| **sec.srtp.key.lifetime[1]** | **0, positive integer minimum 1024 or power of 2 notation** | **Null** |
| The lifetime of the master key used for the cryptographic parameter in SDP. The value specified is the number of SRTP packets. If 0, the master key lifetime is not set. If set to a valid value (at least 1024, or a power such as 2^10), the master key lifetime is set. When the lifetime is set, a re-invite with a new key is sent when the number or 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 the performance of the phone. | | |
| **sec.srtp.mki.enabled[1]** | **0 or 1** | **Lync = 1**<br>**Generic = 0** |
| If enabled, the phone sends two encrypted attributes in the SDP, one with MKI and one without MKI. If disabled, the phone sends only one encrypted attributed without MKI. | | |
| **sec.srtp.mki.startSessionAtOne** | **0 or 1** | **0** |
| If set to 1, use an MKI value of 1 at the start of an SDP session. If set to 0, the MKI value increments for each new crypto key. | | |
| **sec.srtp.offer[1]** | **0 or 1** | **0** |
| If 1, the phone includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE. This parameters applies to the phone initiating (offering) a phone call. If 0, no secure media stream is included in SDP of a SIP invite. | | |
| **sec.srtp.offer.HMAC_SHA1_32[1]** | **0 or 1** | **0** |
| If 1, a crypto line with the `AES_CM_128_HMAC_SHA1_32` crypto-suite is included in offered SDP. If 0, the crypto line is not included. | | |
| **sec.srtp.offer.HMAC_SHA1_80[1]** | **0 or 1** | **1** |
| If 1, a crypto line with the `AES_CM_128_HMAC_SHA1_80` crypto-suite is included in offered SDP. If 0, the crypto line is not included. | | |
| **sec.srtp.padRtpToFourByteAlignment[1]** | **0 or 1** | **0** |
| Packet padding may be required when sending or receiving video from other video products. If 1, RTP packet padding is needed. If 0, no packet padding is needed. | | |
| **sec.srtp.require[1]** | **0 or 1** | **0** |

**SRTP Parameters (continued)**

| *Parameter* | *Permitted values* | *Defaults* |
|---|---|---|
| If 0, secure media streams are not required. If 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 is rejected. For outgoing calls, only a secure media stream description is included in the SDP of the SIP INVITE, meaning that the non-secure media description is not included. If this parameter set to 1, `sec.srtp.offer` is also set to 1, regardless of the value in the configuration file. | | |
| **sec.srtp.requireMatchingTag**[1] | **0 or 1** | **1** |
| If 0, the tag values in the crypto parameter in an SDP answer are ignored. If 1, the tag values must match. | | |
| **sec.srtp.sessionParams.noAuth.offer**[1] | **0 or 1** | **0** |
| If 0, authentication of RTP is offered. If 1, no authentication of RTP is offered; a session description that includes the `UNAUTHENTICATED_SRTP` session parameter is sent when initiating a call. | | |
| **sec.srtp.sessionParams.noAuth.require**[1] | **0 or 1** | **0** |
| If 0, authentication of RTP is required. If 1, no authentication of RTP is required; a call placed to a phone configured with this parameter must offer the UNAUTHENTICATED_SRTP session parameter in its SDP. If this parameter is set to 1, `sec.srtp.sessionParams.noAuth.offer` is also set to 1, regardless of the value in the configuration file. | | |
| **sec.srtp.sessionParams.noEncrypRTCP.offer**[1] | **0 or 1** | **0** |
| If 0, encryption of RTCP is offered. If 1, no encryption of RTCP is offered; a session description that includes the UNENCRYPTED_SRTCP session parameter is sent when initiating a call. | | |
| **sec.srtp.sessionParams.noEncrypRTCP.require**[1] | **0 or 1** | **0** |
| If set to 0, encryption of RTCP is required. 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 this parameter is set to 1, `sec.srtp.sessionParams.noEncryptRTCP.offer` is also set to 1, regardless of the value in the configuration file. | | |
| **sec.srtp.sessionParams.noEncrypRTP.offer**[1] | **0 or 1** | **0** |
| If 0, encryption of RTP is offered. If 1, no encryption of RTP is offered; a session description that includes the UNENCRYPTED_SRTP session parameter is sent when initiating a call. | | |
| **sec.srtp.sessionParams.noEncrypRTP.require**[1] | **0 or 1** | **0** |
| If 0, encryption of RTP is required. If 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 set to 1, sec.srtp.sessionParams.noEncryptRTP.offer is also set to 1, regardless of the value in the configuration file. | | |
| **sec.srtp.simplifiedBestEffort** | **0 or 1** | **0** |
| If 0, no SRTP is supported. If 1, negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported. | | |

[1] Change causes phone to restart or reboot.

## <H235/>

You can use the parameters listed in the next table with the Polycom VVX 500/501, 600/601, and 1500 business media phones. The H.235 Voice Profile implementation is Polycom HDX compatible. OpenSSL-based Diffie-Hellman key exchange and AES-128 CBC encryption algorithms are used to encrypt the RTP media.

**H.235 Media Encryption Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.H235.mediaEncryption.enabled**[1] | **0 or 1** | **1** |
| If 0, H.235 Voice Profile RTP media encryption is disabled. If 1, H.235 media encryption is enabled and negotiated when such encryption is requested by the far end. | | |
| **sec.H235.mediaEncryption.offer**[1] | **0 or 1** | **0** |
| If 0, media encryption negotiations is not initiated with the far end. If 1 and `sec.H235.mediaEncryption.enabled` is also 1, media encryption negotiations is initiated with the far end; however, successful negotiations are not a requirement for the call to complete. | | |
| **sec.H235.mediaEncryption.require**[1] | **0 or 1** | **0** |
| If 0, media encryption negotiations are not required. If 1 and `sec.H235.mediaEncryption.enabled` is also 1, media encryption negotiations are initiated or completed with the far end, and if negotiations fail, the call is dropped. | | |

[1] Change causes phone to restart or reboot.

## <dot1x><eapollogoff/>

The next table lists configurable parameters.

**802.1X EAP over LAN (EAPOL) Logoff Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.dot1x.eapollogoff.enabled**[1] | **0 or 1** | **0** |
| If 0, the phone does not send an EAPOL Logoff message on behalf of the disconnected supplicant. If 1, the feature is enabled and the phone sends an EAPOL Logoff message on behalf of the disconnected supplicant connected to the phone's secondary (PC) port. | | |
| **sec.dot1x.eapollogoff.lanlinkreset**[1] | **0 or 1** | **0** |
| If 0, the phone software does not reset (recycle) the LAN port link in the application initiation stage. If 1, the LAN port link resets in the application initiation stage. | | |

[1] Change causes phone to restart or reboot.

## <hostmovedetect/>

The next table lists configurable parameters.

**Host Movement Detection Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.hostmovedetect.cdp.enabled**[1] | **0 or 1** | **0** |
| If set to 1, the phone software unconditionally sends a CDP packet (to the authenticator switch port) to indicate a host has been connected or disconnected to its secondary (PC) port. | | |
| **sec.hostmovedetect.cdp.sleepTime**[1] | **0 to 60000** | **1000** |
| If `sec.hostmovedetect.cdp.enabled` is set to 1, there is an x microsecond time interval between two consecutive link–up state change reports, which reduces the frequency of dispatching CDP packets. | | |

[1] Change causes phone to restart or reboot.

# <TLS/>

The next table lists configurable TLS parameters. For the list of configurable ciphers, refer to the table Configurable TLS Cipher Suites.

This parameter also includes:

- <profile/>
- <profileSelection/>.

**TLS Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.TLS.browser.cipherList** | **String** | **NoCipher** |
| The cipher list for browser. The format for the cipher list uses OpenSSL syntax found at: http://www.openssl.org/docs/apps/ciphers.html. | | |
| **sec.TLS.cipherList** | **String** | **“RSA:!EXP:!LOW:!NULL:!MD5:@STRENGTH”** |
| The global cipher list parameter. The format for the cipher list uses OpenSSL syntax found at: http://www.openssl.org/docs/apps/ciphers.html. | | |
| **sec.TLS.customCaCert.x** | **String** | **Null** |
| The custom certificate for TLS Application Profile x (x= 1 to 6). | | |
| **sec.TLS.customDeviceCert.x** | **String** | **Null** |
| The custom device certificate for TLS Application Profile x (x= 1 to 6). | | |
| **sec.TLS.customDeviceKey.x** | **String** | **Null** |
| The custom device certificate private key for TLS Application Profile x (x= 1 to 6). | | |
| **sec.TLS.LDAP.cipherList** | **String** | **NoCipher** |
| The cipher list for the corporate directory. The format for the cipher list uses OpenSSL syntax found here: http://www.openssl.org/docs/apps/ciphers.html. | | |

**TLS Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.TLS.profileSelection.SOPI** | **1 - 7** | **PlatformProfile1** |
| Select the platform profile you want to use. You can choose platform profile 1 - 7. | | |
| **sec.TLS.profile.x.caCert.application7** | **0 or 1** | **1** |
| Enable or disable the ability to choose a CA certificate for the application7 profile. | | |
| **sec.TLS.prov.cipherList** | **String** | **NoCipher** |
| The cipher list for provisioning. The format for the cipher list uses OpenSSL syntax found here: http://www.openssl.org/docs/apps/ciphers.html. | | |
| **sec.TLS.SIP.cipherList** | **String** | **NoCipher** |
| The cipher list for SIP. The format for the cipher list uses OpenSSL syntax found here: http://www.openssl.org/docs/apps/ciphers.html. | | |
| **sec.TLS.SIP.strictCertCommonNameValidation** | **0 or 1** | **1** |
| If 1, enable common name validation for SIP. | | |
| **sec.TLS.SOPI.cipherList** | **1 – 1024 character string** | **NoCipher** |
| Choose a cipher key. | | |
| **sec.TLS.SOPI.strictCertCommonNameValidation** | **0 or 1** | **1** |
| Enable or disable strict common name validation for the URL provided by the server. | | |
| **sec.TLS.syslog.cipherList** | **String** | **NoCipher** |
| The cipher list for syslog. The format for the cipher list uses OpenSSL syntax found here: http://www.openssl.org/docs/apps/ciphers.html. | | |

## <profile/>

Profiles are a collection of related security parameters. The next table lists TLS profile parameters. There are two platform profiles and six application profiles.

**TLS Profile Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.TLS.profile.x.caCert.application1**<br>**Application CA 1** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.application2**<br>**Application CA 2** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.application3**<br>**Application CA 3** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.application4**<br>**Application CA 4** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.application5**<br>**Application CA 5** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.application6**<br>**Application CA 6** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.application7**<br>**Application CA 7** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.platform1**<br>**Platform CA 1** | **0 or 1** | **1** |
| **sec.TLS.profile.x.caCert.platform2**<br>**Platform CA 2** | **0 or 1** | **1** |
| Specify which CA certificates should be used for TLS Application Profile x (where x is 1 to 7). If set to 0, the CA is not used. If set to 1, the CA is used. | | |
| **sec.TLS.profile.x.caCert.defaultList** | **String** | **Null** |
| The list of default CA certificates for TLS Application Profile x (x= 1 to 7). | | |
| **sec.TLS.profile.x.cipherSuite** | **String** | **Null** |
| The cipher suite for TLS Application Profile x (where x is 1 to 7). | | |
| **sec.TLS.profile.x.cipherSuiteDefault** | **0 or 1** | **1** |
| If 0, use the custom cipher suite for TLS Application Profile x (x= 1 to 7). If 1, use the default cipher suite. | | |
| **sec.TLS.profile.x.deviceCert** | **Polycom, Platform1, Platform2, Application1, Application2, Application3, Application4, Application5, Application6, Application7** | **Polycom** |
| The device certificate to use for TLS Application Profile x (x = 1 to 7). | | |

## <profileSelection/>

You can configure the parameters listed in the next table to choose the platform profile or application profile to use for each TLS application.

The permitted values are:

- PlatformProfile1

- PlatformProfile2
- ApplicationProfile1
- ApplicationProfile2
- ApplicationProfile3
- ApplicationProfile4
- ApplicationProfile5
- ApplicationProfile6
- ApplicationProfile7

**TLS Profile Selection Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **sec.TLS.profileSelection.browser** | **a TLS profile** | **PlatformProfile1** |
| The TLS platform profile or TLS application profile (see preceding list) to use for the browser or microbrowser. | | |
| **sec.TLS.profileSelection.LDAP** | **a TLS profile** | **PlatformProfile1** |
| The TLS platform profile or TLS application profile (see preceding list) to use for the Corporate Directory. | | |
| **sec.TLS.profileSelection.SIP** | **a TLS profile** | **PlatformProfile1** |
| The TLS platform profile or TLS application profile (see preceding list) to use for SIP operations. | | |
| **sec.TLS.profileSelection.syslog** | **PlatformProfile1 or PlatformProfile2** | **PlatformProfile1** |
| The TLS platform profile to use for syslog operations. | | |
| **sec.TLS.profileSelection.SOPI** | **a TLS profile** | **PlatformProfile1** |
| The TLS platform profile or TLS application profile (see preceding list) to use for the GENBAND “Subscriber Open Provisioning Interface” (SOPI). | | |

# <softkey/>

You can use the softkey parameters to customize soft keys on the phone interface. Note that `feature.enhancedFeatureKeys.enabled` must be enabled (set to 1) to use the Configurable Soft Key feature.

In the following table listing soft key configuration parameters, x=1 to a maximum number of 10 soft keys.

**Soft Key Customization Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **softkey.feature.basicCallManagement.redundant** | **0 or 1** | **1** |
| This parameter applies only to VVX 300 and 400 series business media phones. Controls the display of the **Hold**, **Transfer**, and **Conference** soft keys. If set to 0 and the phone has hard keys mapped for **Hold**, **Transfer**, and **Conference** functions (all must be mapped), none of the soft keys are displayed. If set to 1, all of these soft keys are displayed. | | |
| **softkey.feature.buddies** | **0 or 1** | **1** |
| If 0, the **Buddies** soft key is not displayed. If 1, the soft key is displayed (if `pres.idleSoftKeys` is set to 1). | | |
| **softkey.feature.callers** | **0 or 1** | **0** |
| If 1, the **Callers** soft key displays on all platforms. If 0, the **Callers** soft key is disabled for all platforms.<br>The default value , the default is 0. | | |
| **softkey.feature.directories** | **0 or 1** | **0** |
| If 1, the **Dir** soft key displays on all platforms. If 0, the **Dir** soft key is disabled for all platforms.<br>The default value is 0. | | |
| **softkey.feature.doNotDisturb** | **0 or 1** | **1** |
| Enable or disable the DND soft key on the phone. | | |
| **softkey.feature.endcall** | **0 or 1** | **1** |
| If 0, the **End Call** soft key is not displayed. If 1, the soft key is displayed. | | |
| **softkey.feature.forward** | **0 or 1** | **1** |
| If 0, the **Forward** soft key is not displayed. If 1, the soft key is displayed. | | |
| **softkey.feature.intercom** | **0 or 1** | **1** |
| Enable or disable the intercom soft key. | | |
| **softkey.feature.join** | **0 or 1** | **1** |
| Join two individual calls to form a conference. If 0, the **Join** soft key is not displayed. If 1, the soft key is displayed. | | |
| **softkey.feature.mystatus** | **0 or 1** | **1** |
| If 0, the **MyStatus** soft key is not displayed. If 1, the soft key is displayed (if `pres.idleSoftKeys` is set to 1). | | |
| **softkey.feature.newcall** | **0 or 1** | **1** |
| If 0, the **New Call** soft key is not displayed when there is an alternative way to place a call. If 1, the **New Call** soft key is displayed. | | |
| **softkey.feature.redial** | **0 or 1** | **0** |
| Enables or disables the display of the Redial soft key on the Home screen.<br>**Note**: The parameter feature.enhancedFeatureKeys.enabled must be set to 1 first to configure this feature, and the parameter efk.softkey.alignleft must be set to 1 to move enabled soft keys into the positions of disabled soft keys. | | |

**Soft Key Customization Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **softkey.feature.split** | **0 or 1** | **1** |
| Split up a conference into individual calls. If 0, the **Split** soft key is not displayed. If 1, the soft key is displayed. | | |
| **softkey.x.action** | **macro action string, 256 characters** | **Null** |
| The action or function for custom soft key x. This value uses the same macro action string syntax as an Enhanced Feature Key. For a list of actions, see Understand Macro Definitions. | | |
| **softkey.x.enable** | **0 or 1** | **0** |
| If 0, the soft key x is disabled. If 1, the soft key is enabled. | | |
| **softkey.x.insert** | **0 to 10** | **0** |
| The position on the phone screen for soft key x. If 0, the phone places the soft key x in the first available position. If 1 or above, the phone places the soft key x in the corresponding position and moves the soft key in that position and the following soft keys one position over to the right.<br>For example, if usoft.1.insert is set to 3, the soft key is displayed on the screen in the third position from the left. If a soft key was already in the third position, that soft key is moved to the fourth position, and the following soft keys are moved to the right by one space<br>Note: If `softkey.x.precede` is configured, this value is ignored. If the insert location is greater than the number of soft keys, the key is positioned last after the other soft keys. | | |
| **softkey.x.label** | **string** | **Null** |
| The text displayed on the soft key label. If Null, the label is determined as follows:<br>• If the soft key performs an Enhanced Feature Key macro action, the label of the macro is used.<br>• If the soft key calls a speed dial, the label of the speed dial contact is used.<br>• If the soft key performs chained actions, the label of the first action is used.<br>• If the soft key label is Null and none of the preceding criteria are matched, the label is blank.<br>Note that the maximum number of characters for this parameter value is 16; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters used. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The phone truncates the beginning of numerical labels (for example, …4567) and truncates the end of alphabetical labels (for example, Abcd…). | | |
| **softkey.x.precede** | **0 or 1** | **0** |
| If 0, soft key x is positioned in the first empty space from the left. If 1, the soft key is displayed before (to the left of) the first default soft key. | | |

**Soft Key Customization Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **softkey.x.use.active**<br>Display in the active call state | **0 or 1** | **0** |
| **softkey.x.use.alerting**<br>Display in the alerting state | **0 or 1** | **0** |
| **softkey.x.use.dialtone**<br>Display in the dial tone state | **0 or 1** | **0** |
| **softkey.x.use.hold**<br>Display in the hold state | **0 or 1** | **0** |
| **softkey.x.use.idle**<br>Display in the idle state | **0 or 1** | **0** |
| **softkey.x.use.park**<br>Display in the parked state | | |
| **softkey.x.use.proceeding**<br>Display in the proceeding state | **0 or 1** | **0** |
| **softkey.x.use.setup**<br>Display in the proceeding state | **0 or 1** | **0** |
| If 0, the soft key is not displayed when the phone is in the call state. If 1, the soft key is displayed when the phone is in the call state. | | |

# <tcpIpApp/>

This parameter includes:

- <dhcp/>
- <dns/>
- <ice/>
- <sntp/>
- <port/><rtp/>
- <keepalive/>
- <fileTransfer/>

## <dhcp/>

The DHCP parameters listed in the next table enable you to configure how the phone reacts to DHCP changes.

**DHCP Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.dhcp.releaseOnLinkRecovery** | **0 or 1** | **1** |
| If 0, no DHCP release occurs. If 1, a DHCP release is performed after the loss and recovery of the network. | | |

## <dns/>

The <dns/> parameters listed in the next table enables you to set Domain Name System (DNS). However, any values set through DHCP have a higher priority and any values set through the <device/> parameter in a configuration file have a lower priority.

**Domain Name System (DNS) Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **tcpIpApp.dns.address.overrideDHCP**[1] | **0 or 1** | **0** |
| If set to 0, a DNS address is requested from the DHCP server. When set to 1, a DNS primary and secondary address is set using the parameters `tcpIpApp.dns.server` and `tcpIpApp.dns.altServer`. | | |
| **tcpIpApp.dns.server**[1] | **IP address** | **Null** |
| The primary server to which the phone directs DNS queries. | | |
| **tcpIpApp.dns.altServer**[1] | **IP address** | **Null** |
| The secondary server to which the phone directs DNS queries. | | |
| **tcpIpApp.dns.domain**[1] | **String** | **Null** |
| The phone's DNS domain. | | |
| **tcpIpApp.dns.domain.overrideDHCP**[1] | **0 or 1** | **0** |
| If set to 0, a domain name is retrieved from the DHCP server, if one is available. If set to 1, the DNS domain name is set using the parameter `tcpIpApp.dns.domain`. | | |

[1] Change causes phone to restart or reboot.

## <ice/>

Parameters in the following table enable you to set the STUN/TURN/ICE feature.

**ICE Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.ice.password** | **String** | **Null** |
| Enter the password to authenticate to the TURN server. | | |
| **tcpIpApp.ice.stun.server** | **String** | **Null** |
| Enter the IP address of the STUN server. | | |
| **tcpIpApp.ice.stun.udpPort** | **1-65535** | **3478** |
| The UDP port number of the STUN server. | | |
| **tcpIpApp.ice.tcp.enabled** | **0 or 1** | **1** |
| If 0, TCP is disabled. If 1, TCP is enabled. | | |
| **tcpIpApp.ice.turn.callAdmissionControl.enabled** | | **1** |

**ICE Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| | | |
| **tcpIpApp.ice.turn.server** | **String** | **Null** |
| Enter the IP address of the TURN server. | | |
| **tcpIpApp.ice.turn.tcpPort** | **1-65535** | **443** |
| The UDP port number of the TURN server. | | |
| **tcpIpApp.ice.turn.udpPort** | **1-65535** | **443** |
| The UDP port number of the TURN server. | | |
| **tcpIpApp.ice.username** | **String** | **Null** |
| Enter the user name to authenticate to the TURN server. | | |

## <sntp/>

The next table lists the Simple Network Time Protocol (SNTP) parameters used to set up time synchronization and daylight savings time. The default values enable and configure daylights savings time (DST) for North America.

Daylight savings time 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 2am.
- Stop DST on the first Sunday in November at 2am.

**Simple Network Time Protocol (SNTP) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.sntp.address** | **Valid hostname or IP address** | **Null** |
| The address of the SNTP server. | | |
| **tcpIpApp.sntp.AQuery** | **0 or 1** | **0** |
| If set to 0, queries to resolve the SNTP hostname are performed using DNS SRV.<br>If set to 1, the host name is queried for a DNS A record instead. | | |
| **tcpIpApp.sntp.address.overrideDHCP** | **0 or 1** | **0** |
| If 0, the DHCP values for the SNTP server address are used. If 1, the SNTP parameters override the DHCP values. | | |
| **tcpIpApp.sntp.daylightSavings.enable** | **0 or 1** | **1** |
| If 0, daylight savings time rules are not applied to the displayed time. If 1, the daylight savings rules apply. | | |
| **tcpIpApp.sntp.daylightSavings.fixedDayEnable** | **0 or 1** | **0** |
| If 0, `month`, `date`, and `dayOfWeek` are used in the DST calculation. If 1, only `month` and `date` are used. | | |

**Simple Network Time Protocol (SNTP) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.sntp.daylightSavings.start.date** | **1 to 31** | **8** |
| The start date for daylight savings time. If `fixedDayEnable` is set to 1, the value of this parameter is the day of the month to start DST. If `fixedDayEnable` is set to 0, this value specifies the occurrence of `dayOfWeek` when DST should start. Set 1 for the first occurrence in the month, set 8 for the second occurrence, 15 for the third occurrence, or 22 for the fourth occurrence. For example, if set to 15, DST starts on the third `dayOfWeek` of the month. | | |
| **tcpIpApp.sntp.daylightSavings.start.dayOfWeek** | **1 to 7** | **1** |
| The day of the week to start DST. 1=Sunday, 2=Monday, … 7=Saturday. Note: this parameter is not used if `fixedDayEnable` is set to 1. | | |
| **tcpIpApp.sntp.daylightSavings.start.dayOfWeek.lastInMonth** | **0 or 1** | **0** |
| If 1, DST starts on the last `dayOfWeek` of the month and the `start.date` is ignored). Note: this parameter is not used if `fixedDayEnable` is set to 1. | | |
| **tcpIpApp.sntp.daylightSavings.start.month** | **1 to 12** | **3 (March)** |
| The month to start DST. 1=January, 2=February… 12=December. | | |
| **tcpIpApp.sntp.daylightSavings.start.time** | **0 to 23** | **2** |
| The time of day to start DST – in 24 hour clock format. 0= 12am, 1= 1am,… 12= 12pm, 13= 1pm, … 23= 11pm. | | |
| **tcpIpApp.sntp.daylightSavings.stop.date** | **1 to 31** | **1** |
| The stop date for daylight savings time. If `fixedDayEnable` is set to 1, the value of this parameter is the day of the month to stop DST. If `fixedDayEnable` is set to 0, this value specifies the occurrence of `dayOfWeek` when DST should stop. Set 1 for the first occurrence in the month, set 8 for the second occurrence, 15 for the third occurrence, or 22 for the fourth occurrence. For example, if set to 22, DST stops on the fourth `dayOfWeek` of the month. | | |
| **tcpIpApp.sntp.daylightSavings.stop.dayOfWeek** | **1 to 7** | **1** |
| The day of the week to stop DST. 1=Sunday, 2=Monday, … 7=Saturday. Note: this parameter is not used if `fixedDayEnable` is set to 1. | | |
| **tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth** | **0 or 1** | **0** |
| If 1, DST stops on the last `dayOfWeek` of the month and the `stop.date` is ignored). Note: this parameter is not used if `fixedDayEnable` is set to 1. | | |
| **tcpIpApp.sntp.daylightSavings.stop.month** | **1 to 12** | **11** |
| The month to stop DST. 1=January, 2=February… 12=December. | | |
| **tcpIpApp.sntp.daylightSavings.stop.time** | **0 to 23** | **2** |
| The time of day to stop DST – in 24 hour clock format. 0= 12am, 1= 1am,… 12= 12pm, 13= 1pm, … 23= 11pm. | | |
| **tcpIpApp.sntp.gmtOffset** | **positive or negative integer** | **0** |
| The offset in seconds of the local time zone from GMT.3600 seconds = 1 hour, -3600 seconds = -1 hour. | | |
| **tcpIpApp.sntp.gmtOffset.overrideDHCP** | **0 or 1** | **0** |
| If 0, the DHCP values for the GMT offset are used. If 1, the SNTP values for the GMT offset are used. | | |

**Simple Network Time Protocol (SNTP) Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.sntp.resyncPeriod** | **positive integer** | **86400** |
| The period of time (in seconds) that passes before the phone resynchronizes with the SNTP server. Note: 86400 seconds is 24 hours. | | |
| **tcpIpApp.sntp.retryDnsPeriod** | **60 – 2147483647 seconds** | **86400** |
| Set a retry period for DNS queries. Note that the DNS retry period you configure is affected by other DNS queries made by the phone. If the phone makes a query for another service such as SIP registration during the retry period you configure and receives no response, the Network Time Protocol (NTP) DNS query is omitted to limit the overall number of retry attempts made to the unresponsive server. If no other DNS attempts are made by other services, then the rety period you configure is not affected. If at any time the DNS server becomes responsive to another service, then NTP also immediately retries its DNS query as well. | | |

## <port/><rtp/>

The parameters listed in the next table enable you to configure the port filtering used for RTP traffic.

**RTP Port Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.port.rtp.filterByIp**[1] | **0 or 1** | **1** |
| IP addresses can be negotiated through the SDP or H.323 protocols. If set to 1, the phone rejects RTP packets that arrive from non-negotiated IP addresses. Note that the H.323 protocol is supported on the VVX 500/501, 600/601, and 1500 phones. | | |
| **tcpIpApp.port.rtp.filterByPort**[1] | **0 or 1** | **0** |
| Ports can be negotiated through the SDP protocol. If set to 1, the phone rejects RTP packets arriving from (sent from) a non-negotiated port. | | |
| **tcpIpApp.port.rtp.forceSend**[1] | **0 to 65535** | **0** |
| Send all RTP packets to, and expect all RTP packets to arrive on, this port. If 0, RTP traffic is not forced to one port. Note: Both `tcpIpApp.port.rtp.filterByIp` and `tcpIpApp.port.rtp.filterByPort` must be set to 1 for this to work. | | |
| **tcpIpApp.port.rtp.mediaPortRangeEnd**[1] | **Default, 1024 to 65485** | **2269** |
| Determines the maximum supported end range of audio ports. | | |
| **tcpIpApp.port.rtp.mediaPortRangeStart**[1] | **even integer 1024 to 65440** | **2222** |
| The starting port for RTP packets. Ports are allocated from a pool starting with this port up to a value of (start-port + 47) for a voice-only phone or (start-port + 95) for a video phone.<br>Note: Ensure that there is no contention for port numbers. For example, do not use 5060 (default port for SIP). | | |
| **tcpIpApp.port.rtp.videoPortRange.enable** | **0 or 1** | **Base profile**<br>**Lync = 1**<br>**Generic = 0** |

**RTP Port Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| If 1, video ports are chosen from the range specified by **tcpIpApp.port.rtp.videoPortRangeStart** and **tcpIpApp.port.rtp.videoPortRangeEnd**.<br>If 0, video ports are also chosen within the range specified by **tcpIpApp.port.rtp.mediaPortRangeStart** and **tcpIpApp.port.rtp.mediaPortRangeEnd**. | | |
| **tcpIpApp.port.rtp.videoPortRangeEnd**[1] | **Default, 1024 to 65535** | **2319** |
| Determines the maximum supported end range of video ports. | | |
| **tcpIpApp.port.rtp.videoPortRangeStart**[1] | **Default, 1024 to 65486** | **2272** |
| Determines the start range for video ports.<br>This is used only when the value of **tcpIpApp.port.rtp.videoPortRange.enable** is **1**. | | |

[1] Change causes phone to restart or reboot.

## <keepalive/>

The parameters listed in the next table enable the configuration of TCP keep-alive on SIP TLS connections; the phone can detect a failure quickly (in minutes) and attempt to re-register with the SIP call server (or its redundant pair).

**TCP Keep-Alive Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.keepalive.tcp.idleTransmitInterval** | **10 to 7200** | **30** |
| The amount of time to wait (in seconds) before sending the keep-alive message to the call server.<br>Note: If this parameter is set to a value that is out of range, the default value is used.<br>Note: On VVX phones and SoundStructure VoIP Interface, this parameter specifies the number of seconds TCP waits between transmission of the last data packet and the first keep-alive message. | | |
| **tcpIpApp.keepalive.tcp.noResponseTransmitInterval** | **5 to 120** | **20** |
| If no response is received to a keep-alive message, subsequent keep-alive messages are sent to the call server at this interval (every x seconds).<br>Note: On VVX phones and SoundStructure VoIP Interface, this parameter specifies the amount of idle time between the transmission of the keep-alive packets the TCP stack waits. This applies whether the last keep-alive was acknowledged or not. | | |
| **tcpIpApp.keepalive.tcp.sip.persistentConnection.enable**[1] | **0 or 1** | **0** |
| If 0, the TCP socket opens a new connection when the phone tries to send any new SIP message and closes after one minute. If 1, the TCP socket connection remains open indefinitely. | | |
| **tcpIpApp.keepalive.tcp.sip.tls.enable** | **0 or 1** | **0** |

**TCP Keep-Alive Parameters (continued)**

| | | |
|---|---|---|
| If 0, disable TCP keep-alive for SIP signaling connections that use TLS transport. If 1, enable TCP keep-alive for SIP signaling connections that use TLS transport. | | |

[1] Change causes phone to restart or reboot.

## <fileTransfer/>

The parameters listed in the next table configure file transfers from the phone to the provisioning server.

**File Transfer Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tcpIpApp.fileTransfer.waitForLinkIfDown** | **0 or 1** | **1** |
| If 1, file transfer from the FTP server is delayed until Ethernet comes back up.<br>If 0, file transfer from the FTP server is not attempted. | | |

# <tones/>

This parameter lists configuration items for available tone resources and includes:

- <DTMF/>
- <chord/>

## <DTMF/>

The parameters listed in the next table configure Dual-tone multi-frequency (DTMF) tone signaling.

**DTMF Tone Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **tone.dtmf.chassis.masking**[1] | **0 or 1** | **0** |
| If 0, DTMF tones play through the speakerphone in handsfree mode. If 1 (set only if `tone.dtmf.viaRtp` is set to 0), DTMF tones are substituted with non-DTMF pacifier tones when dialing in handsfree mode—this is to prevent the tones from broadcasting to surrounding telephony devices or being inadvertently transmitted in-band due to local acoustic echo. | | |
| **tone.dtmf.level**[1] | **-33 to 3** | **-15** |
| The level of the high frequency component of the DTMF digit measured in dBm0; the low frequency tone is two dB lower. | | |
| **tone.dtmf.offTime**[1] | **positive integer** | **50** |
| When a sequence of DTMF tones is played out automatically, this is the length of time in milliseconds the phone pauses between digits. This is also the minimum inter-digit time when dialing manually. | | |
| **tone.dtmf.onTime**[1] | **positive integer** | **50** |

**DTMF Tone Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| When a sequence of DTMF tones is played out automatically, this is the length of time in milliseconds the tones play for. This is also the minimum time the tone plays when dialing manually (even if key press is shorter). | | |
| **tone.dtmf.rfc2833Control**[1] | **0 or 1** | **1** |
| If set to 1, the phone indicates 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 always honor the DTMF format present in the offer since the phone has native support for RFC 2833. | | |
| **tone.dtmf.rfc2833Payload**[1] | **96 to 127** | **Lync = 101**<br>**Generic = 127** |
| The phone-event payload encoding in the dynamic range to be used in SDP offers. | | |
| **tone.dtmf.viaRtp**[1] | **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: If this parameter is set to 0, `tone.dtmf.chassis.masking` should be set to 1. | | |

[1] Change causes phone to restart or reboot.

## <chord/>

Chord-sets are the building blocks of sound effects that used synthesized audio rather than sampled audio. Most call progress and ringer sound effects are synthesized. 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. Chord parameters are listed in the next table.

There are three chord sets: callProg, misc, and ringer. Each chord set has different chord names, represented by *x* in the following table. The chord names are as follows:

For callProg, *x* can be one of the following chords:

- **dialTone**, **busyTone**, **ringback**, **reorder**, **stutter_3**, **callWaiting**, **callWaitingLong**, **howler**, **recWarning**, **stutterLong**, **intercom**, **callWaitingLong**, **precedenceCallWaiting**, **preemption**, **precedenceRingback**, or **spare1** to **spare6**.

For **misc**, *x* can be one of the following chords

- **spare1** to **spare9**.

For **ringer,** *x* can be one of the following chords:

- **ringback**, **originalLow**, **originalHigh**, or **spare1** to **spare19**.

**Chord Parameters**

| Parameter | Permitted Values |
|---|---|
| **tone.chord.callProg.x.freq.y**<br>**tone.chord.misc.x.freq.y**<br>**tone.chord.ringer.x.freq.y** | **0-1600**<br>**0-1600**<br>**0-1600** |
| The frequency (in Hertz) for component y. Up to six chord-set components can be specified (y=1 to 6). | |

**Chord Parameters (continued)**

| Parameter | Permitted Values |
|---|---|
| **tone.chord.callProg.x.level.y**<br>**tone.chord.misc.x.level.y**<br>**tone.chord.ringer.x.level.y** | **-57 to 3**<br>**-57 to 3**<br>**-57 to 3** |
| The level of component y in dBm0. Up to six chord-set components can be specified (y=1 to 6). | |
| **tone.chord.callProg.x.onDur**<br>**tone.chord.misc.x.onDur**<br>**tone.chord.ringer.x.onDur** | **positive integer**<br>**positive integer**<br>**positive integer** |
| The on duration (length of time to play each component) in milliseconds, 0=infinite. | |
| **tone.chord.callProg.x.offDur**<br>**tone.chord.misc.x.offDur**<br>**tone.chord.ringer.x.offDur** | **positive integer**<br>**positive integer**<br>**positive integer** |
| The off duration (the length of silence between each chord component) in milliseconds, 0=infinite. | |
| **tone.chord.callProg.x.repeat**<br>**tone.chord.misc.x.repeat**<br>**tone.chord.ringer.x.repeat** | **positive integer**<br>**positive integer**<br>**positive integer** |
| The number of times each ON/OFF cadence is repeated, 0=infinite. | |

# <up/>

Use the parameters listed in the next table to set user preferences on the phones.

**User Preferences Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **up.25mm** | **1 or 2** | **1** |
| Specify whether to use a mobile phone or a PC to connect to the 2.5mm audio port on a conference phone. Set to 1 if using a mobile phone. Set to 2 if using a PC. | | |
| **up.accessibilityFeatures** | **0 or 1** | **0** |
| VVX 1500 only. If 0, accessibility features are disabled. If 1, the screen background flashes orange for incoming calls. | | |
| **up.analogHeadsetOption** | **0, 1, 2, 3** | **0** |
| The Electronic Hookswitch mode for the phone’s analog headset jack. 0 - no EHS-compatible headset is attached. 1 - a Jabra EHS-compatible headset is attached. 2 - a Plantronics EHS-compatible headset is attached. 3 - a Sennheiser EHS-compatible headset is attached. | | |
| **up.audioMode** | **0 or 1** | **0** |
| Specify whether you want to use the handset or headset for audio. | | |

**User Preferences Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **up.backlight.idleIntensity** | **VVX 300/301/310/311 = 0, 1, 2, 3**<br>**All other phones = 1, 2, 3** | **1** |
| The brightness of the LCD backlight when the phone is idle. 1 – low, 2 – medium, and 3 – high. Note: If this is higher than the active backlight brightness (`onIntensity`), the active backlight brightness is used. | | |
| **up.backlight.onIntensity** | **VVX 300/301/310/311 = 0, 1, 2, 3**<br>**All other phones = 1, 2, 3** | **3** |
| The brightness of the LCD backlight when the phone is active (in use). 1 – low, 2 – medium, 3 – high. | | |
| **up.backlight.timeout** | **5 to 60** | **40** |
| The number of seconds to wait before the backlight dims from the active intensity to the idle intensity. | | |
| **up.basicSettingsPasswordEnabled** | **0 or 1** | **0** |
| If set to 1, a password is required for access to the Basic settings menu on the phone. If set to 0, no password is required to access the Basic settings menu. | | |
| **up.cfgLabelElide** | **None, Right, Left** | **None** |
| Controls the alignment of the line label. When the line label is an alphanumeric or alphabetic string, the label aligns right. When the line label is a numeric string, the label aligns left. | | |
| **up.cfgUniqueLineLabel** | **0 or 1** | **0** |
| Allow unique labels for the same registration that is split across multiple line keys using reg.X.linekeys.<br>Set to 0 to use the same label on all linekeys. Set to 1 to display a unique label as defined by reg.X.line.Y.label.<br>If reg.X.line.Y.label is not configured, then a label of the form <integer>_ will be applied in front of the applied label automatically. | | |
| **up.cfgWarningsEnabled** | **0 or 1** | **0** |
| If 1, a warning is displayed on the phone if the phone is configured with pre-UC Software 3.3.0 parameters. If 0, the warning does not display. | | |
| **up.echoPasswordDigits** | **0 or 1** | **1** |
| If 1, the phone briefly displays password characters before being masked by an asterisk. If 0, the phone displays only asterisks for the password characters. | | |
| **up.em.linkalivecheck.enabled** | **0 or 1** | **0** |
| If 1, the host VVX phone periodically sends ping packets to the expansion modules. If 0, the host VVX phone does not ping the expansion modules. | | |

**User Preferences Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **up.em.smartpaging.enabled** | **0 or 1** | **1** |
| Enable or disable line key assignments and page distribution on the VVX Expansion Modules.<br>If 1, Smart Paging is enabled. If 0, Smart Paging is disabled. Note that the flexible line key configuration overrides Smart Paging for the expansion module, and Smart Paging is disabled for VVX Expansion Modules with a paper display. | | |
| **up.handsetModeEnabled** | **0 or 1** | **1** |
| Enable or disable the handset port. | | |
| **up.handsfreeMode** | **0 or 1** | **1** |
| If 0, the speakerphone is disabled (cannot be used). If 1, the speakerphone is enabled. | | |
| **up.headsetAlwaysUseIntrinsicRinger** | **0 or 1** | **1** |
| If 1, the USB headset uses the intrinsic ringer mixed with DSP ringer when the sound effect destination is the USB headset. | | |
| **up.headsetMode** | **0 or 1** | **0** |
| If 0, handsfree mode is used by default instead of the handset. If 1, the headset is used as the preferred audio mode after the headset key is pressed for the first time, until the headset key is pressed again. | | |
| **up.headsetModeEnabled** | **0 or 1** | **1** |
| If 0, the headset port is disabled and cannot be used. If 1, the headset port is enabled and can be used. | | |
| **up.headset.phoneVolumeControl**[1] | **disable, enable, auto** | **auto** |
| Controls the phone’s behavior when you adjust volume at the headset.<br>• **enable** The phone responds to volume up/down events from the headset by displaying the volume widget in the phone’s user interface and adjusting the phone's internal volume.<br>• **disable** The phone ignores volume up/down events from the headset; pressing the headset's volume controls has no effect on the phone.<br>• **auto** The phone automatically selects which of the above two behaviors to apply based on the type and model of headset that you attach. | | |
| **up.hearingAidCompatibility.enabled** | **0 or 1** | **0** |
| If set to 1, the phone audio Rx (receive) equalization is disabled for hearing aid compatibility. If 0, audio Rx equalization is enabled. | | |
| **up.idleBrowser.enabled** | **0 or 1** | **0** |
| If 0, the idle browser is disabled. If 1, the idle browser is enabled. Note that if `up.prioritizeBackgroundMenuItem.enabled` is 1, you can choose to display the background or the idle browser on the phone menu. | | |
| **up.idleStateView**[1] | **0 or 1** | **0** |
| Sets the default view on the phone.<br>If 0, The call/line view is the default view. If 1, the Home screen is the default view. | | |

**User Preferences Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **up.idleTimeout**[1] | **0 to 65535, seconds** | **40** |
| The number of seconds that the phone can be idle for before automatically leaving a menu and showing the idle display. If 0, there is no timeout and the phone does not automatically exit to the idle display. | | |
| **up.IdleViewPreferenceRemoteCalls**[1] | **0 or 1** | **0** |
| Use this parameter to determine when the phone displays the idle browser.<br>When set to 1, a phone with only remote calls active, for example, on a BLF monitored line, is treated as in the active state and the idle browser does not display.<br>When set to 0, a phone with only remote calls active, for example, on a BLF monitored line, is treated as in the idle state and the idle browser displays. | | |
| **up.lineKeyCallTerminate** | **0 or 1** | **0** |
| If 1, the user can press a line key to end an active call on that line. If 0, the user cannot end a call by pressing the line key (this is the previous behavior). | | |
| **up.localClockEnabled** | **0 or 1** | **1** |
| If 0, the date and time are not shown on the idle display. If 1, the date and time and shown on the idle display. | | |
| **up.manualProtocolRouting** | **0 or 1** | **1** |
| You can use this parameter with the VVX 500/501, 600/601, and 1500 phones. If 1, the user is presented with a protocol routing choice in situations where a call can be placed using either protocol (for example, with SIP and H.323 protocols). If 0, the default protocol is used and the user does not choose. | | |
| **up.manualProtocolRouting.softKeys** | **0 or 1** | **1** |
| You can use this parameter with the VVX 500/501, 600/601, and 1500 phones. Choose whether or not you want to display soft keys that control Manual Protocol Routing options. When Soft Key Control is enabled, you can use soft keys to choose between the SIP or H.323 protocol. When disabled, soft keys for protocol routing do not display. The soft keys are enabled by default. | | |
| **up.mwiVisible**[1] | **0 or 1** | **0** |
| If set is 0, 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.<br>If set to 1, the MWI for lines whose MWI is disabled display (pre-SIP 2.1 behavior), even though MWI notifications have been received for those lines. | | |
| **up.numberFirstCID**[1] | **0 or 1** | **0** |
| If 0, the caller ID display shows the caller's name first. If 1, the caller's phone number is shown first. | | |
| **up.numOfDisplayColumns**[1] | **1, 2, 3, 4** | **VVX 500/501=max 3**<br>**VVX 600/601=max 4** |
| Set the maximum number of columns the VVX 500/501 and 600/601 display. Note that phones display one column when the value is set to 0. The maximum number of columns for the VVX 500/501 is 3. The maximum number of columns for the VVX 600/601 is 4. | | |
| **up.offHookAction.none**[1] | **0 or 1** | **0** |

**User Preferences Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| If 1, when the user lifts the handset, the phone does not seize the line and the ringer continues until the user takes further action. | | |
| **up.oneTouchVoiceMail[1]** | **0 or 1** | **0** |
| If 1, the phone dials voicemail services directly, if available on the call server, without displaying the voicemail summary. If 0, the phone displays a summary page with message counts.<br>Users must press the Connect soft key to dial the voicemail server. | | |
| **up.osdIncomingCall.Enabled** | **0 or 1** | **1** |
| If 1, the full screen popup or OSD for incoming calls displays. If 0, the full screen popup or OSD for incoming calls does not display. | | |
| **up.pictureFrame.timePerImage** | **3 to 300 seconds** | **5** |
| For the VVX 500/501, 600/601, and 1500 only. The number of seconds to display each picture frame image. | | |
| **up.pictureFrame.folder** | **string** | **Null** |
| For the VVX 500/501, 600/601, and 1500 only. 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 this parameter to `images/phone` . | | |
| **up.prioritizeBackgroundMenuItem.enabled[1]** | **0 or 1** | **1** |
| If `up.idleBrowser.enabled` is 1, this parameter can be set to 1 to display a **Prioritize Background** menu to the user. The user can choose whether the phone background should take priority over the idle browser or not. | | |
| **up.screenCapture.enabled[1]** | **0 or 1** | **0** |
| If 0, screen captures are disabled. If 1, the user can enable screen captures from the Screen Capture menu on the phone. Note: when the phone reboots, screen captures are disabled from the Screen Capture menu on the phone. | | |
| **up.screenSaver.enabled** | **0 or 1** | **0** |
| If 0, the screen saver feature is disabled. If 1, the screen saver feature is enabled. If a USB flash drive containing images is connected to the phone, and the idle browser is not configured, a slide show cycles through the images from the USB flash drive when the screen saver feature is enabled. The images must be stored in the directory on the flash drive specified by `up.pictureFrame.folder`. The screen saver displays when the phone has been in the idle state for the amount of time specified by `up.screenSaver.waitTime`. | | |
| **up.screenSaver.type** | **0 or 2** | **0** |
| Choose the type of screen saver to display. If 0, the phone screen saver displays default images. If 2, the phone screen saver displays the idle browser. You can use this parameter with the VVX 300 and 400 series phones. | | |
| **up.screenSaver.waitTime** | **1 to 9999, minutes** | **15** |
| The number of minutes that the phone waits in the idle state before the screen saver starts. | | |
| **up.simplifiedSipCallInfo** | **0 or 1** | **0** |
| If 1, the displayed host name is trimmed for both incoming and outgoing calls and the protocol tag/information is not displayed for incoming and outgoing calls. | | |

**User Preferences Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **up.SLA.ringType** | **default, ringer1 to ringer24** | **ringer2** |
| Specifies a ring type for Shared Line Appearance (SLA) lines. | | |
| **up.status.message.flash.rate** | **2 - 8 seconds** | **2 seconds** |
| Controls the scroll rate of the status bar on VVX 300 and 400 series business media phones. | | |
| **up.transparentLines** | **0 or 1** | **0** |
| If 0, line keys block display of the background image. If 1, line keys are transparent and allow the background image to display behind the line labels. This parameter applies only to the VVX 500/501 and 600/601 business media phones. | | |
| **up.useDirectoryNames**[1] | **0 or 1** | **1** |
| If 0, names provided through network signaling are used for caller ID. If 1, the name field in the local contact directory is used as the caller ID for incoming calls from contacts in the local directory. Note: Outgoing calls and corporate directory entries are not matched. | | |
| **up.warningLevel**[1] | **0 to 2** | **0** |
| If 0, the phone's warning icon and a pop-up message display on the phone for all warnings. If 1, the warning icon and pop-up messages are only shown for critical warnings. All warnings are listed in the Warnings menu. If 2, the phone displays a warning icon and no warning messages.<br>For all the values, all warnings are listed in the warning menu. Access to the Warnings menu varies by phone model:<br>• **VVX 1500** Menu > Status > Diagnostics > Warnings<br>• **VVX 101, 201, 300/301/310/311, 400/401/410/411, 500/501, and 600/601** Settings > Status > Diagnostics > Warnings | | |
| **up.welcomeSoundEnabled**[1] | **0 or 1** | **1** |
| If 0, the welcome sound is disabled. If 1, the welcome sound is enabled and played each time the phone reboots. Note that to use a welcome sound you must enable the parameter `up.welcomeSoundEnabled` and specify a file in `saf.x` as shown in the section <saf/>. The default UC Software welcome sound file is `Welcome.wav`. See the example configuration in the section Customize Audio Sound Effects. | | |
| **up.welcomeSoundOnWarmBootEnabled**[1] | **0 or 1** | **0** |
| If 0, the welcome sound is played when the phone powers up (cold boot), but not after it restarts or reboots (warm boot). If 1, the welcome sound plays each time the phone powers up, reboots, or restarts. | | |

[1] Change causes phone to restart or reboot.

# <upgrade/>

Use the parameters listed in the next table to specify the URL of a custom download server and the Polycom UC Software download server for the phone to check when searching for software upgrades.

**Upgrade Server Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **upgrade.custom.server.url** | **URL** | **Null** |
| The URL of a custom download server. | | |
| **upgrade.plcm.server.url** | **URL** | `http://downloads.polycom.com/voice/software/` |
| The URL of the Polycom UC Software download server. | | |

# <video/>

The parameters in the table are supported on the VVX 500/501, VVX 600/601, and VVX 1500.

This parameter also includes:

- <camera/>
- <codecs/>
- <localCameraView/>

**Video Parameters**

| *Parameter* | *Permitted Values* | *Default* | |
|---|---|---|---|
| **video.allowWithSource** | **0 or 1** | **0** | |
| Restrict when to send video codec negotiation in SDP. Applies only to the VVX 500/501 and VVX 600/601. | | | |
| **video.enable** | **video.allowWithSource** | **Camera Attached** | **Result** |
| 0 | 0 | 0 | no video codecs advertised |
| 0 | 1 | 0 | no video codecs advertised |
| 1 | 0 | 0 | video codecs advertised |
| 1 | 1 | 0 | no video codecs advertised |
| 0 | 0 | 1 | no video codecs advertised |
| 0 | 1 | 1 | no video codecs advertised |
| 1 | 0 | 1 | video codecs advertised |
| 1 | 1 | 1 | video codecs advertised |

**Video Parameters (continued)**

| | | |
|---|---|---|
| **video.autoFullScreen** | **0 or 1** | **0** |
| If 0, video calls only use the full screen layout if it is explicitly selected by the user. If 1, video calls use the full screen layout by default, such as when a video call is first created or when an audio call transitions to a video call) | | |
| **video.autoStartVideoTx** | **0 or 1** | **1** |
| When enabled, video transmission to the far side begins when you start a call. When disabled, video transmission does not begin until you press the **Video > Start Video** soft keys. This parameter controls video sent to the far side. Video from the far side always displays if available, and far side users can control when to send video. | | |
| **video.callMode.default** | **audio or video** | **audio** |
| When the device uses SIP protocol, this parameter allows users to select the outbound call mode. Note that you must enable `feature.audioVideoToggle.enabled="1"` to apply this parameter. Note that when you set this parameter to 'video', the VVX 500/501 and 600/601 display a Video Mode soft key, and the VVX 1500 displays a video icon. | | |
| **video.callRate** | **128 to 2048** | **512** |
| The default call rate (in kbps) to use when initially negotiating bandwidth for a video call. | | |
| **video.dynamicControlMethod** | **0 or 1** | **0** |
| If 1, the first I-Frame request uses the method defined by `video.forceRtcpVideoCodecControl` and subsequent requests alternate between RTCP-FB and SIP INFO.<br>In case of network device problems, you can set the phone to attempt multiple methods of I-frame requests. To set other methods for I-frame requests, refer to the parameter video.forceRtcpVideoCodecControl. | | |
| **video.enable** | **0=Disable, 1=Enable** | **1** |
| If 0, video is not enabled and all calls—both sent and received—are audio-only. If 1, video is sent in outgoing calls and received in incoming calls if the other device supports video.<br>Note: On the VVX 500/501 and 600/601, when you enable video, the G.722.1C codec is disabled. | | |
| **video.forceRtcpVideoCodecControl**[1] | **0 or 1** | **0** |
| If 1, the phone is forced to send RTCP feedback messages to request fast update I-frames along with SIP INFO messages for all video calls irrespective of a successful SDP negotiation of a=rtcp-fb. If 0, RTCP-FB messages depend on a successful SDP negotiation of a=rtcp-fb and are not used if that negotiation is missing.<br>For an account of all parameter dependencies when setting I-frame requests, refer to the section Configure I-Frames. | | |
| **video.iFrame.delay**[1] | **0 to 10, seconds** | **0** |
| When non-zero, an extra I-frame is transmitted after video starts. The amount of delay from the start of video until the I-frame is sent is configurable up to 10 seconds. | | |
| **video.iFrame.minPeriod** | **1 - 60** | **2** |
| After sending an I-frame, the phone always waits at least this amount of time before sending another I-frame in response to requests from the far end. | | |
| **video.iFrame.onPacketLoss** | **0 or 1** | **0** |
| If 1, an I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred. | | |

**Video Parameters (continued)**

| | | |
|---|---|---|
| **video.maxCallRate**[1] | **128 to 2048 kbps** | **768** |
| The maximum call rate allowed. This allows the administrator to limit the maximum call rate that the users can select. If `video.callRate` exceeds this value, this value is used as the maximum. | | |
| **video.quality**[1] | **motion, sharpness** | **NULL** |
| The optimal quality for video that you send in a call or a conference. Use `motion` if your outgoing video has motion or movement. Use `sharpness` or Null if your outgoing video has little or no movement.<br>Note: If `motion` is not selected, moderate to heavy motion can cause some frames to be dropped. | | |

[1] Change causes phone to restart or reboot.

## <camera/>

The settings in the next table control the performance of the camera.

**Video Camera Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **video.camera.brightness** | **0 to 6** | **3** |
| Set brightness level. The value range is from 0 (Dimmest) to 6 (Brightest). | | |
| **video.camera.contrast** | **0 to 4** | **0** |
| Set contrast level.<br>The value range is from 0 (No contrast increase) to 3 (Most contrast increase), and 4 (Noise reduction contrast). | | |
| **video.camera.flickerAvoidance** | **0 to 2** | **0** |
| Set flicker avoidance.<br>If set to 0, flicker avoidance is automatic.<br>If set to 1, 50hz AC power frequency flicker avoidance (Europe/Asia).<br>If set to 2, 60hz AC power frequency flicker avoidance (North America). | | |
| **video.camera.frameRate** | **5 to 30** | **25** |
| Set target frame rate (frames per second). Values indicate a fixed frame rate, from 5 (least smooth) to 30 (most smooth).<br>Note: If `video.camera.frameRate` is set to a decimal number, the value 25 is used. | | |
| **video.camera.saturation** | **0 to 6** | **3** |
| Set saturation level. The value range is from 0 (Lowest) to 6 (Highest). | | |
| **video.camera.sharpness** | **0 to 6** | **3** |
| Set sharpness level. The value range is from 0 (Lowest) to 6 (Highest). | | |

## <codecs/>

The video codecs include:

- <codecPref/>
- <profile/>

## <codecPref/>

The next table lists video codec parameters.

**Video Codec Preference Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **video.codecPref.H261**[1]<br>**video.codecPref.H264**[1]<br>**video.codecPref.H263 1998**[1]<br>**video.codecPref.H263**[1] | **1 to 4** | **4**<br>**1**<br>**2**<br>**3** |
| Specifies the video codec preferences for the VVX 500/501, 600/601, and 1500 phones. Note that the VVX 500/501 and 600/601 support H.263 and H.264 and do not support H.261 or H.263 1998. | | |
| **video.codecPref.H264SVC** | | |

[1] Change causes phone to restart or reboot.

## <profile/>

The next table lists settings for a group of low-level video codec parameters. For most use cases, the default values are appropriate. Polycom does not recommend changing the default values unless specifically advised to do so.

**Video Profile Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **video.profile.H261.annexD**[1] | **0 or 1** | **1** |
| Enable or disable Annex D when negotiating video calls. | | |
| **video.profile.H261.CifMpi**[1] | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating CIF resolution for a video call. You can enter a value between 0-4. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H261.jitterBufferMax**[1] | **(video.profile.H261.jitter BufferMin + 500ms) to 2500ms** | **2000ms** |
| The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size always causes lost packets. This parameter should be set to the smallest possible value that support the expected network jitter. | | |
| **video.profile.H261.jitterBufferMin**[1] | **33ms to 1000ms** | **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 still continues. 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 Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **video.profile.H261.jitterBufferShrink[1]** | **33ms to 1000ms** | **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.H261.payloadType[1]** | **0 to 127** | **31** |
| RTP payload format type for H261 MIME type. | | |
| **video.profile.H261.QcifMpi[1]** | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-4. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H263.CifMpi[1]** | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H263.jitterBufferMax[1]** | **(video.profile.H263.jitterBufferMin + 500ms) to 2500ms** | **2000ms** |
| The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size always causes lost packets. This parameter should be set to the smallest possible value that supports the expected network jitter. | | |
| **video.profile.H263.jitterBufferMin[1]** | **33ms to 1000ms** | **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 still continues. 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[1]** | **33ms to 1000ms** | **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.payloadType[1]** | **0 to 127** | **34** |
| RTP payload format type for H263 MIME type. | | |
| **video.profile.H263.QcifMpi[1]** | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H263.SqcifMpi[1]** | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |

**Video Profile Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **video.profile.H2631998.annexF**[1] | **0 or 1** | **0** |
| Enable or disable Annex F when negotiating video calls. | | |
| **video.profile.H2631998.annexI**[1] | **0 or 1** | **0** |
| Enable or disable Annex I when negotiating video calls. | | |
| **video.profile.H2631998.annexJ**[1] | **0 or 1** | **0** |
| Enable or disable Annex J when negotiating video calls. | | |
| **video.profile.H2631998.annexK**[1] | **0, 1, 2, 3, 4** | **1** |
| Specify the value of Annex K to use when negotiating video calls. You can enter a value between 0-4. To disable, enter ‘0’. The default value is ‘1’. | | |
| **video.profile.H2631998.annexN**[1] | **0, 1, 2, 3, 4** | **1** |
| Specify the value of Annex N to use when negotiating video calls. You can enter a value between 0-4. To disable, enter ‘0’. The default value is ‘1’. | | |
| **video.profile.H2631998.annexT**[1] | **0 or 1** | **0** |
| Enable or disable Annex T when negotiating video calls. | | |
| **video.profile.H2631998.CifMpi**[1] | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H2631998.jitterBufferMax**[1] | **(video.profile.H2631998.jitterBufferMin+ 500ms) to 2500ms** | **2000ms** |
| The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size always causes lost packets. This parameter should be set to the smallest possible value that supports the expected network jitter. | | |
| **video.profile.H2631998.jitterBufferMin**[1] | **33ms to 1000ms** | **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 still continues. 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**[1] | **33ms to 1000ms** | **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**[1] | **96 to 127** | **96** |
| RTP payload format type for H263-1998/90000 MIME type. | | |

**Video Profile Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **video.profile.H2631998.QcifMpi**[1] | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H2631998.SqcifMpi**[1] | **1 to 32** | **1** |
| Specify the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’. The default frame rate divider is ‘1’. | | |
| **video.profile.H264.jitterBufferMax**[1] | **(video.profile.H264.jitterBufferMin + 500ms) to 2500ms** | **2000ms** |
| The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size always causes lost packets. This parameter should be set to the smallest possible value that supports the expected network jitter. | | |
| **video.profile.H264.jitterBufferMin**[1] | **33ms to 1000ms** | **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 still continues. 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**[1] | **33ms to 1000ms** | **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**[1] | **96 to 127** | **109** |
| RTP payload format type for H264/90000 MIME type. | | |
| **video.profile.H264.profileLevel**[1] | **1, 1b, 1.1, 1.2, 1.3, and 2** | **1.3** |
| Specify the highest profile level within the baseline profile supported in video calls. The phone supports the following levels: 1, 1b, 1.1, 1.2, 1.3, and 2. The default level is 1.3.<br>**Note**: VVX 500/501 and VVX 600/601 phones support H.264 with a profile level of 2, and VVX 1500 phones support H.264 with a profile level of 1.3. | | |

[1] Change causes phone to restart or reboot.

## <localCameraView/>

The parameters in the next table configure how the local camera displays on the screen.

**Local Camera View Preferences Parameters**

| *Parameters* | *Permitted Values* | *Default* |
|---|---|---|
| **video.localCameraView.fullscreen.enabled** | **0=Disable, 1=Enable** | **1** |
| Determines whether the local camera view is shown in the full screen layout.<br>If set to 0, the local camera view is not shown. If set to 1, the local camera view is shown. | | |
| **video.localCameraView.fullscreen.mode** | **pip, side-by-side** | **side-by-side** |
| Determines how the local camera view is shown. If set to pip, the local camera view displays as a picture-in-picture with the far end window.<br>If set to side-by-side, the local camera view displays side-by-side with the far end window. | | |

# <voice/>

The parameters listed in the next table configure phone audio.

This parameter includes:

- <aec/>
- <aes/>
- <cn/>
- <codecPref/>
- <handset/>
- <headset/>
- <line/>
- <ns/>
- <volume/>
- <vad/>
- <quality monitoring/>
- <rxQoS/>

**Voice Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voice.txEq.hf.preFilter.enable** | **0 or 1** | **0** |
| If 1 and a narrow band codec is in use, such as G.711mu, G.711A, G.729, or iLBC, a 300 Hz high-pass filter is applied to the transmit audio prior to encoding.<br>Enabling this filter may improve intelligibility to the far end when making narrow band calls through a PSTN gateway in a noisy environment. | | |

**Voice Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voice.txPacketDelay[1]** | **low, normal, Null** | **Null** |
| If set to normal or Null, no audio parameters are changed.<br>If set to low and there are no precedence conflicts, the following changes are made:<br>`voice.codecPref.G722="1"`<br>`voice.codecPref.G711Mu="2"`<br>`voice.codecPref.G711A="3"`<br>`voice.codecPref.<OtherCodecs>=""`<br>`voice.audioProfile.G722.payloadSize="10"`<br>`voice.audioProfile.G711Mu.payloadSize= "10"`<br>`voice.audioProfile.G711A.payloadSize= "10"`<br>`voice.aec.hs.enable="0"`<br>`voice.ns.hs.enable="0"` | | |
| **voice.txPacketFilter[1]** | **0 or 1** | **Null** |
| If 0, no Tx filtering is performed. If 1, narrowband Tx high pass filter is enabled. | | |

[1] Change causes phone to restart or reboot.

## <aec/>

Use these parameters to enable or disable the acoustic echo cancellation (AEC) function for a specified termination.

**Acoustic Echo Canceller (AEC) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.aec.hf.enable** | **0 or 1** | **1** |
| Enable or disable the handsfree AEC function. Note: Polycom recommends that you do not disable this parameter. | | |
| **voice.aec.hs.enable** | **0 or 1** | **1** |
| Enable or disable the handset AEC function. | | |

## <aes/>

Use these parameters to control the speakerphone acoustic echo suppression (AES). These parameters remove residual echo after AEC processing. Because AES depends on AEC, enable AES only when you also enable AEC using `voice.aec.hd.enable`.

**Acoustic Echo Suppression Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.aes.hf.enable** | **0 or 1** | **1** |
| Enable or disable the handsfree AES function.<br>Note: Polycom recommends that you do not disable this parameter. | | |
| **voice.aes.hs.enable** | **0 or 1** | **1** |
| Enable or disable the handset AES function. | | |

## <cn/>

Use these parameters to configure the addition and volume of comfort noise during conferences.

**Comfort Noise Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.cn.hf.enable** | **0 or 1** | **0** |
| If 1, comfort noise is added into the Tx path for hands-free operation. This feature should be used only when users at the far end perceive that the phone has gone "dead" after the near end user stops talking.<br>If 0, no comfort noise is added. | | |
| **voice.cn.hf.attn** | **0 - 90** | **30 (quite loud)** |
| Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hf.enabled` is 1. | | |
| **voice.cn.hd.enable** | **0 or 1** | **0** |
| If 1, comfort noise is added into the Tx path for the headset. This feature should be used only when users at the far end perceive that the phone has gone "dead" after the near end user stops talking.<br>If 0, no comfort noise is added. | | |
| **voice.cn.hd.attn** | **0 - 90** | **30 (quite loud)** |
| Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hd.enabled` is 1. Default value is 30, which is quite loud. | | |
| **voice.cn.hs.enable** | **0 or 1** | **0** |
| If 1, comfort noise is added into the Tx path for the handset. This feature should be used only when users at the far end perceive that the phone has gone "dead" after the near end user stops talking.<br>If 0, no comfort noise is added. | | |
| **voice.cn.hs.attn** | **0 - 90** | **30 (quite loud)** |
| Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hs.enabled` is 1. Default value is 30, which is quite loud. | | |

## <codecPref/>

As of Polycom UC Software 3.3.0, you can configure a simplified set of codec properties for all phone models to improve consistency and reduce workload on the phones. Phone codec preferences are listed in the next table.

If a particular phone does not support a codec, the phone ignores that codec and continue to the codec next in the priority. For example, using the default values, the highest-priority codec on a VVX 310 phone is G.722.1 since that model doesn't support G.722.1C or G.719.

For more information on codecs on particular phones and priorities, see Supported Audio Codecs.

**Voice Codec Preferences Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.codecPref.G711_A** | **0 to 27** | **7** |
| **voice.codecPref.G711_Mu** | | **6** |
| **voice.codecPref.G719.32kbps** | | **0** |
| **voice.codecPref.G719.48kbps** | | **0** |
| **voice.codecPref.G719.64kbps** | | **0** |
| **voice.codecPref.G722** | | **4** |
| **voice.codecPref.G7221.16kbps** | | **0** |
| **voice.codecPref.G7221.24kbps** | | **0** |
| **voice.codecPref.G7221.32kbps** | | **5** |
| **voice.codecPref.G7221_C.24kbps** | | **0** |
| **voice.codecPref.G7221_C.32kbps** | | **0** |
| **voice.codecPref.G7221_C.48kbps** | | **2** |
| **voice.codecPref.G729_AB** | | **8** |
| **voice.codecPref.iLBC.13_33kbps** | | **0** |
| **voice.codecPref.iLBC.15_2kbps** | | **0** |
| **voice.codecPref.Lin16.8ksps** | | **0** |
| **voice.codecPref.Lin16.16ksps** | | **0** |
| **voice.codecPref.Lin16.32ksps** | | **0** |
| **voice.codecPref.Lin16.44_1ksps** | | **0** |
| **voice.codecPref.Lin16.48ksps** | | **0** |
| **voice.codecPref.Siren14.24kbps** | | **0** |
| **voice.codecPref.Siren14.32kbps** | | **0** |
| **voice.codecPref.Siren14.48kbps** | | **3** |
| The priority of the codec. If 0 or Null, the codec is disabled. A value of 1 is the highest priority. If a phone does not support a codec, it treats the setting as if it were 0 and not offer or accept calls with that codec. | | |

## <handset/>

The parameters listed in this section control the level of sidetone on handsets of VVX business media phones.

**Handset Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.handset.st** | **-12 to +12** | **0** |
| Adjust the handset sidetone level from the default in 1 decibel (dB) increments. | | |

## <headset/>

The parameters listed in this section control the level of sidetone on headsets connected to VVX business media phones.

**Headset Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.headset.st** | **-12 to +12** | **0** |
| Adjust the headset sidetone level from the default in 1 decibel (dB) increments. | | |

## <line/>

The following parameters control audio level settings for phone handset and headset.

**Voice Line Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.lineAgc.hs.enable** | **0 or 1** | **0** |
| Enable or disable the line automatic gain control function which automatically boosts the volume of low-level signals and reduces high-level signals on the phone handset. If 0, the line automatic gain control is disabled on the handset. If 1, the line automatic gain control is enabled for the handset. This parameter applies to the VVX 300 series, 400 series, 500 series, and 600 series business media phones. | | |
| **voice.lineAgc.hd.enable** | **0 or 1** | **0** |
| Enable or disable the line automatic gain control function which automatically boosts the volume of low-level signals and reduces high-level signals on the phone headset. If 0, the line automatic gain control is disabled on the headset. If 1, the line automatic gain control is enabled for the headset. This parameter applies to the VVX 300 series, 400 series, 500 series, and 600 series business media phones. | | |

## <ns/>

This section lists noise suppression parameters available with the Acoustic Fence and Polycom NoiseBlock features.

**Noise Suppression Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.ns.hd.enable** | **0 or 1** | **1** |
| Enable or disable noise suppression for headsets. | | |
| **voice.ns.hd.enhanced** | **0 or 1** | **0** |
| Enable or disable Acoustic Fence noise suppression for headsets. Note that voice.ns.hd.enable must also be set to 1 to use this parameter. | | |
| **voice.ns.hd.nonStationary Thresh** | **1 -10** | **8** |
| Increase or decrease the Acoustic Fence noise suppression threshold for headsets. A lower value allows more background sound to enter, and a higher value suppresses background noise. Note that high values can suppress the speaker's voice and impact far-end audio quality. | | |
| **voice.ns.hf.block** | **0 or 1** | **0** |
| If 0, Polycom NoiseBlock technology is disabled. If 1, NoiseBlock is enabled. | | |
| **voice.ns.hs.enable** | **0 or 1** | **1** |
| Enable or disable noise suppression for handsets. | | |
| **voice.ns.hs.enhanced** | **0 or 1** | **0** |
| Enable or disable Acoustic Fence noise suppression for handsets. Note that voice.ns.hs.enable must also be set to 1 to use this parameter. | | |
| **voice.ns.hs.nonStationary Thresh** | **1 -10** | **8** |
| Increase or decrease the Acoustic Fence noise suppression threshold for handsets. A lower value allows more background sound to enter, and a higher value suppresses background noise. Note that high values can suppress the speaker's voice and impact far-end audio quality. | | |

## <volume/>

In some countries, regulations state that a phone’s receiver volume must be reset to a nominal level for each new call. This is the phone’s default behavior. The next able lists parameters that configure the receiver volume to persist across calls each time a user makes changes to the default volume level.

**Volume Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.volume.persist.bluetooth.headset** | **0 or 1** | **0** |
| If 0, the Bluetooth headset are not used for every call. If 1, the Bluetooth headset are used for all calls. | | |

**Volume Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.volume.persist.handset** | **0 or 1** | **0** |
| If 0, the handset receive volume automatically resets to a nominal level after each call. If 1, the volume for each call are the same as the previous call. If set to 1, the handset receive volume persists across calls.<br>If set to 0, the handset receive volume resets to nominal at the start of each call. | | |
| **voice.volume.persist.headset** | **0 or 1** | **0** |
| If 0, the headset receive volume automatically resets to a nominal level after each call. If 1, the volume for each call is the same as the previous call. | | |
| **voice.volume.persist.handsfree** | **0 or 1** | **1** |
| If 0, the speakerphone receive volume automatically resets to a nominal level after each call. If 1, the volume for each call is the same as the previous call. | | |
| **voice.volume.persist.usb.handsfree** | **0 or 1** | **0** |
| If 0, the USB headset is not used. If 1, the USB headset is used. | | |
| **voice.volume.persist.usbHeadset** | **0 or 1** | **0** |
| If 0, the USB headset is not used. If 1, the USB headset is used. | | |

## <vad/>

The parameters listed in the next table configure voice activity detection (silence suppression) feature.

**Voice Activity Detection (VAD) Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.CNControl** | **0 or 1** | **0** |
| Publishes support for Comfort Noise in the SDP body of the INVITE message and includes the supported comfort noise payloads in the media line for audio. If set to 1, either the payload type 13 for 8 KHz sample rate audio codec is sent for Comfort Noise, or the dynamic payload type for 16 KHz audio codecs are sent in the SDP body. | | |
| **voice.CN16KPayload** | **96 to 127** | **122** |
| Alters the dynamic payload type used for Comfort Noise RTP packets for 16 KHz codecs. | | |
| **voice.vad.signalAnnexB**[1] | **0 or 1** | **1** |
| If 0, there is no change to SDP. If 1, Annex B is used and a new line is added to SDP depending on the setting of `voice.vadEnable`.<br>If `voice.vadEnable` is set to 1, add parameter line `a=fmtp:18 annexb="yes"` below `a=rtpmap`… parameter line (where '18' could be replaced by another payload).<br>If `voice.vadEnable` is set to 0, add parameter line `a=fmtp:18 annexb="no"` below `a=rtpmap`… parameter line (where '18' could be replaced by another payload). | | |
| **voice.vadEnable**[1] | **0 or 1** | **0** |

**Voice Activity Detection (VAD) Parameters  (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| If 0, voice activity detection (VAD) is disabled. If 1, VAD is enabled. | | |
| **voice.vadThresh**[1] | **integer from 0 to 30** | **15** |
| The threshold for determining what is active voice and what is background noise in dB. Sounds louder than this value will be considered active voice, and sounds quieter than this threshold will be considered background noise. This does not apply to G.729AB codec operation which has its own built-in VAD function. | | |

[1] Change causes phone to restart or reboot.

## <quality monitoring/>

The next table lists Voice Quality Monitoring (VQMon) parameters.

**Voice Quality Monitoring Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.qualityMonitoring.collector.alert.moslq.threshold.critical**[1] | **0 to 40** | **0** |
| The 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 0 or Null, critical alerts are not generated due to MOS-LQ.<br>For example, a configured value of 28 corresponds to the MOS score 2.8. | | |
| **voice.qualityMonitoring.collector.alert.moslq.threshold.warning**[1] | **0 to 40** | **0** |
| 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 0 or Null, warning alerts are not generated due to MOS-LQ.<br>For example, a configured value of 35 corresponds to the MOS score 3.5. | | |
| **voice.qualityMonitoring.collector.alert.delay.threshold.critical**[1] | **0 to 2000** | **0** |
| Threshold value of one way delay (in ms) that causes phone to send a critical alert quality report. If 0 or Null, critical 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.warning**[1] | **0 to 2000** | **0** |
| Threshold value of one way delay (in ms) that causes phone to send a critical alert quality report. If 0 or 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.enable.periodic**[1] | **0 or 1** | **0** |
| If 0, periodic quality reports are not generated. If 1, periodic quality reports are generated throughout a call. | | |
| **voice.qualityMonitoring.collector.enable.session**[1] | **0 or1** | **0** |
| If 0, quality reports are not generated at the end of each call. If 1, reports are generated at the end of each call. | | |
| **voice.qualityMonitoring.collector.enable.triggeredPeriodic**[1] | **0 to 2** | **0** |

**Voice Quality Monitoring Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| If 0, alert states do not cause periodic reports to be generated. If 1, periodic reports are generated if an alert state is critical. If 2, period reports are generated when an alert state is either warning or critical. Note: This parameter is ignored when `voice.qualityMonitoring.collector.enable.periodic` is 1, since reports are sent throughout the duration of a call. | | |
| **voice.qualityMonitoring.collector.period**[1] | **5 to 90 seconds** | **900 seconds** |
| The time interval between successive periodic quality reports. | | |
| **voice.qualityMonitoring.collector.server.x.address**[1] | **IP address or hostname** | **Null** |
| The server address of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages. Set x to 1 as only one report collector is supported at this time. | | |
| **voice.qualityMonitoring.collector.server.x.outboundProxy.address** | **IP address or FQDN** | **NULL** |
| When configured, this parameter directs SIP messages related to voice quality monitoring to a separate proxy. No failover is supported for this proxy, and voice quality monitoring is not available for error scenarios. | | |
| **voice.qualityMonitoring.collector.server.x.outboundProxy.port** | **0 to 65535** | **0** |
| Specify the port to use for the voice quality monitoring outbound proxy server. | | |
| **voice.qualityMonitoring.collector.server.x.outboundProxy.transport** | **DNSnaptr,<br>TCPpreferred,<br>UDPOnly, TLS,<br>TCPOnly** | **DNSnaptr** |
| Specify the transport protocol the phone uses to send the voice quality monitoring SIP messages. | | |
| **voice.qualityMonitoring.collector.server.x.port**[1] | **1 to 65535** | **5060** |
| The port of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages. Set x to 1 as only one report collector is supported at this time. | | |
| **voice.qualityMonitoring.failover.enable** | **0 or 1** | **1** |
| If 1, the phone will perform a failover when voice quality SIP PUBLISH messages are unanswered by the collector server. If 0, no failover is performed; note, however, that a failover is still triggered for all other SIP messages. This parameter is ignored if `voice.qualityMonitoring.collector.server.x.outboundProxy` is enabled. | | |
| **voice.qualityMonitoring.location** | **Valid location string** | **Unknown** |
| Specify the device location with a valid location string. If you do not configure a location value, you must use the default string ‘Unknown’. | | |
| **voice.qualityMonitoring.rfc6035.enable** | **0 or 1** | **0** |
| If 0, the existing draft implementation is supported. If 1, complies with RFC6035. | | |
| **voice.qualityMonitoring.rtcpxr.enable**[1] | **0 or 1** | **0** |
| If 0, RTCP-XR packets are not generated. If 1, the packets are generated. | | |

[1] Change causes phone to restart or reboot.

## <rxQoS/>

The following table lists the jitter buffer parameters for wired network interface voice traffic and push-to-talk interface voice traffic.

**Voice Jitter Buffer Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voice.rxQoS.avgJitter**[1]<br>**The typical average jitter.** | **0 to 80** | **20** |
| **voice.rxQoS.maxJitter**[1]<br>**The maximum expected jitter.** | **0 to 200** | **160** |
| The average and maximum jitter in milliseconds for wired network interface voice traffic.<br>`avgJitter` The wired interface minimum depth will be automatically configured to adaptively handle this level of continuous jitter without packet loss.<br>`maxJitter` The wired interface jitter buffer maximum depth will be automatically configured to handle this level of intermittent jitter without packet loss.<br>Actual jitter above the average but below the maximum may result in delayed audio play out while the jitter buffer adapts, but no packets will be lost. Actual jitter above the maximum value will always result in packet loss. Note that if legacy `voice.audioProfile.x.jitterBuffer.*` parameters are explicitly specified, they will be used to configure the jitter buffer and these `voice.rxQoS` parameters will be ignored. | | |
| **voice.rxQoS.ptt.avgJitter**[1]<br>**The typical average jitter.** | **0 to 200** | **150** |
| **voice.rxQoS.ptt.maxJitter**[1]<br>**The maximum expected jitter.** | **20 to 500** | **480** |
| The average and maximum jitter in milliseconds for IP multicast voice traffic.<br>`avgJitter` The PTT/Paging interface minimum depth will be automatically configured to adaptively handle this level of continuous jitter without packet loss.<br>`maxJitter` The PTT/Paging interface jitter buffer maximum depth will be automatically configured to handle this level of intermittent jitter without packet loss.<br>Actual jitter above the average but below the maximum may result in delayed audio play out while the jitter buffer adapts, but no packets will be lost. Actual jitter above the maximum value will always result in packet loss.<br>Note: if legacy `voice.audioProfile.x.jitterBuffer.*` parameters are explicitly specified, they will be used to configure the jitter buffer and these `voice.rxQoS` parameters will be ignored for PTT/Paging interface interfaces. | | |

[1] Change causes phone to restart or reboot.

# <voIpProt/>

You must set up the call server and DTMF signaling parameters.

This parameter includes:

- <server/>
- <SDP/>
- <SIP/>
- <H323/>

## <server/>

The next table describes VoIP server configuration parameters.

**VoIP Server Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voIpProt.server.dhcp.available[1]** | **0 or 1** | **0** |
| If 0, do not check with the DHCP server for the SIP server IP address. If 1, check with the server for the IP address. | | |
| **voIpProt.server.dhcp.option[1]** | **128 to 254** | **128** |
| The option to request from the DHCP server if `voIpProt.server.dhcp.available`= 1.<br>Note: If `reg.x.server.y.address` is non-Null, it takes precedence even if the DHCP server is available. | | |
| **voIpProt.server.dhcp.type[1]** | **0 or 1** | **0** |
| Type to request from the DHCP server if `voIpProt.server.dhcp.available` is set to 1.If this parameter is set to 0, IP request address. If set to 1, request string | | |
| **voIpProt.server.x.address** | **IP address or hostname** | **Null** |
| The IP address or hostname and port of a SIP server that accepts registrations. Multiple servers can be listed starting with x=1 to 4 for fault tolerance. | | |
| **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.lineSeize** | **positive integer, minimum 10** | **30** |
| Requested line-seize subscription period. | | |
| **voIpProt.server.x.expires.overlap** | **5 to 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 the server value is less than the configured overlap value. | | |
| **voIpProt.server.x.failOver.failBack.mode** | **newRequests, DNSTTL, registration, duration** | **duration** |
| Specify the failover failback mode.<br>• **newRequests** All new requests are forwarded first to the primary server regardless of the last used server.<br>• **DNSTTL** The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.<br>• **registration** The phone tries the primary server again when the registration renewal signaling begins.<br>• **duration** The phone tries the primary server again after the time specified by `voIpProt.server.x.failOver.failBack.timeout`. | | |

**VoIP Server Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voIpProt.server.x.failOver.failBack.timeout** | **0, 60 to 65535** | **3600** |
| If `voIpProt.server.x.failOver.failBack.mode` is set to duration, this is the time in seconds after failing over to the current working server before the primary server is again selected as the first server to forward new requests to. Values between 1 and 59 will result in a timeout of 60 and 0 means do not fail-back until a fail-over event occurs with the current server. | | |
| **voIpProt.server.x.failOver.failRegistrationOn** | **0 or 1** | **0** |
| When set to 1, and the reRegisterOn parameter is enabled, the phone will silently invalidate an existing registration (if it exists), at the point of failing over. When set to 0, and the reRegisterOn parameter is enabled, existing registrations will remain active. This means that the phone will attempt failback without first attempting to register with the primary server to determine if it has recovered. | | |
| **voIpProt.server.x.failOver.onlySignalWithRegistered** | **0 or 1** | **1** |
| When set to 1, and the reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call will end. No SIP messages will be sent to the unregistered server. When set to 0, and the reRegisterOn and failRegistrationOn parameters are enabled, signaling will be accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred). | | |
| **voIpProt.server.x.failOver.reRegisterOn** | **0 or 1** | **0** |
| When set to 1, the phone will attempt to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling will proceed with the secondary server. When set to 0, the phone won't attempt to register with the second. | | |
| **voIpProt.server.x.port** | **0, 1 to 65535** | **0** |
| The port of the server that specifies registrations. If 0, the port used depends on `voIpProt.server.x.transport.` | | |
| **voIpProt.server.x.protocol.SIP** | **0 or 1** | **1** |
| If 1, server is a SIP proxy/registrar. Note: if set to 0, and the server is confirmed to be a SIP server, then the value is assumed to be 1. | | |
| **voIpProt.server.x.registerRetry.baseTimeOut** | **10 - 120** | **60** |
| The base time period to wait before a registration retry. Used in conjunction with `voIpProt.server.x.registerRetry.maxTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626.<br>If both parameters `voIpProt.server.x.registerRetry.baseTimeOut` and `reg.x.server.y.registerRetry.baseTimeOut` are set, the value of `reg.x.server.y.registerRetry.baseTimeOut` takes precedence. | | |
| **voIpProt.server.x.registerRetry.maxTimeOut** | **60 - 1800** | **60** |

**VoIP Server Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| The maximum time period to wait before a registration retry. Used in conjunction with `voIpProt.server.x.registerRetry.maxTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626.<br>If both parameters `voIpProt.server.x.registerRetry.maxTimeOut` and `reg.x.server.y.registerRetry.maxTimeOut` are set, the value of `reg.x.server.y.registerRetry.maxTimeOut` takes precedence. | | |
| **voIpProt.server.x.subscribe.expires** | **10 – 2147483647 seconds** | **3600 seconds** |
| The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. For example, if expires="300" and overlap="5", the phone resubscribes after 295 seconds (300–5). Note that the period negotiated with the server may be different. | | |
| **voIpProt.server.x.subscribe.expires.overlap** | **5 – 65535 seconds** | **60 seconds** |
| The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. | | |
| **voIpProt.server.x.transport** | **DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly** | **DNSnaptr** |
| The transport method the phone uses to communicate with the SIP server.<br>• **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.<br>• **TCPpreferred** TCP is the preferred transport; UDP is used if TCP fails.<br>• **UDPOnly** Only UDP will be used.<br>• **TLS** If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.<br>• **TCPOnly** Only TCP will be used. | | |
| **voIpProt.server.x.protocol.SIP** | **0 or 1** | **1** |
| If 1, server is a SIP proxy/registrar. Note: if set to 0, and the server is confirmed to be a SIP server, then the value is assumed to be 1. | | |
| **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** | **5 to 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 the server value is less than the configured overlap value. | | |

**VoIP Server Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voIpProt.server.x.expires.lineSeize** | **positive integer, minimum 0 was 10** | **30** |
| Requested line-seize subscription period. | | |
| **voIpProt.server.x.failOver.failBack.mode** | **newRequests, DNSTTL, registration, duration** | **duration** |
| The mode for failover failback.<br>• **newRequests** All new requests are forwarded first to the primary server regardless of the last used server.<br>• **DNSTTL** The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.<br>• **registration** The phone tries the primary server again when the registration renewal signaling begins.<br>• **duration** The phone tries the primary server again after the time specified by `voIpProt.server.x.failOver.failBack.timeout`. | | |
| **voIpProt.server.x.failOver.failBack.timeout** | **0, 60 to 65535** | **3600** |
| If `voIpProt.server.x.failOver.failBack.mode` is set to duration, this is the time in seconds after failing over to the current working server before the primary server is again selected as the first server to forward new requests to. Values between 1 and 59 will result in a timeout of 60 and 0 means do not fail-back until a fail-over event occurs with the current server. | | |
| **voIpProt.server.x.failOver.failRegistrationOn** | **0 or 1** | **0** |
| When set to 1, and the reRegisterOn parameter is enabled, the phone will silently invalidate an existing registration (if it exists), at the point of failing over. When set to 0, and the reRegisterOn parameter is enabled, existing registrations will remain active. This means that the phone will attempt failback without first attempting to register with the primary server to determine if it has recovered. | | |
| **voIpProt.server.x.failOver.onlySignalWithRegistered** | **0 or 1** | **1** |
| When set to 1, and the reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call will end. No SIP messages will be sent to the unregistered server. When set to 0, and the reRegisterOn and failRegistrationOn parameters are enabled, signaling will be accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred). | | |
| **voIpProt.server.x.failOver.reRegisterOn** | **0 or 1** | **0** |
| When set to 1, the phone will attempt to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling will proceed with the secondary server. When set to 0, the phone won't attempt to register with the second. | | |
| **voIpProt.server.x.register** | **0 or 1** | **1** |
| If 0, calls can be routed to an outbound proxy without registration. See `reg.x.server.y.register`.<br>For more information, see *Technical Bulletin 5844: SIP Server Fallback Enhancements on Polycom Phones*. | | |
| **voIpProt.server.x.retryTimeOut** | **0 to 65535** | **0** |
| The amount of time (in milliseconds) to wait between retries. If 0, use standard RFC 3261 signaling retry behavior. | | |

**VoIP Server Parameters (continued)**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **voIpProt.server.x.retryMaxCount** | **0 to 20** | **3** |
| If set to 0, 3 is used. The number of retries that will be attempted before moving to the next available server. | | |
| **voIpProt.server.x.specialInterop** | **standard,**<br>**lcs2005,**<br>**ocs2007r2,**<br>**lync2010, GENBAND, or ALU-CTS** | **standard** |
| Enables server-specific features. | | |
| **voIpProt.server.x.subscribe.expires** | **10 – 2147483647 seconds** | **3600 sec** |
| The phone’s requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. For example, if expires=”300” and overlap=”5”, the phone resubscribes after 295 seconds (300–5). Note that the period negotiated with the server may be different. | | |
| **voIpProt.server.x.subscribe.expires.overlap** | **5 – 65535 seconds** | **60 seconds** |
| The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. | | |
| **voIpProt.server.x.useOutboundProxy** | **0 or 1** | **1** |
| Specify whether or not to use the outbound proxy specified in `voIpProt.SIP.outboundProxy.address` for server x. | | |
| **voIpProt.server.H323.x.address** | **IP address or hostname** | **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 to 65535** | **1719** |
| Port to be used for H.323 signaling.<br>Note: The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP. | | |
| **voIpProt.server.H323.x.expires** | **positive integer** | **3600** |
| Desired registration period. | | |

[1] Change causes phone to restart or reboot.

## <SDP/>

The next table describes Session Description Protocol configuration parameters.

**Session Description Protocol (SDP) Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **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.<br>Note: If the 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.early.answerOrOffer** | **0 or 1** | **0** |
| 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.<br>Note: An SDP offer or answer is not generated if `reg.x.musicOnHold.uri` is set. | | |
| **voIpProt.SDP.iLBC.13_33kbps.includeMode** | **0 or 1** | **1** |
| If set to 1, the phone should include the mode=30 FMTP parameter in SDP offers:<br>If voice.codecPref.iLBC.13_33kbps is set and voice.codecPref.iLBC.15_2kbps is Null.<br>If voice.codecPref.iLBC.13_33kbps and voice.codecPref.iLBC.15_2kbps are both set, the iLBC 13.33 kbps codec is set to a higher preference.<br>If set to 0, the phone should not include the mode=30 FTMP parameter in SDP offers even if iLBC 13.33 kbps codec is being advertised. See the section <codecPref/>. | | |
| **voIpProt.SDP.useLegacyPayloadTypeNegotiation** | **0 or 1** | **0** |
| 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.<br>If set to 0, RFC 3264 is followed for transmit and receive RTP payload type values. | | |
| **voIpProt.SDP.offer.rtcpVideoCodecControl** | **0 or 1** | **0** |
| This parameter determines whether or not RTCP-FB-based controls are offered in Session Description Protocol (SDP) when the phone negotiates video I-frame request methods. Even when RTCP-FB-based controls are not offered in SDP, the phone may still send and receive RTCP-FB I-frame requests during calls depending on other parameter settings. For more information about video I-frame request behavior, refer to video.forceRtcpVideoCodecControl. For an account of all parameter dependencies refer to the section Configure I-Frames.<br>If 1, the phone adds the SDP attribute "a=rtcp-fb" into offers during outbound SIP calls. If 0, the phone does not include the SDP attribute "a=rtcp-fb". | | |

# <SIP/>

The next table describes SIP configuration parameters.

**Session Initiation Protocol (SIP) Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.SIP.acd.signalingMethod**[1] | **0 or 1** | **0** |
| If set to 0, the 'SIP-B' signaling is supported. (This is the older ACD functionality.)<br>If set to 1, the feature synchronization signaling is supported. (This is the new ACD functionality.) | | |

**Session Initiation Protocol (SIP) Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.SIP.alertInfo.x.class** | **see the list of ring classes in <rt/>** | **default** |
| 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 is applied. | | |
| **voIpProt.SIP.alertInfo.x.value** | **string** | **Null** |
| A string to match the Alert-Info header in the incoming INVITE. | | |
| **voIpProt.SIP.allowTransferOnProceeding** | **0, 1, 2** | **1** |
| If set to 0, a transfer is not allowed during the proceeding state of a consultation call.<br>If set to 1, a transfer can be completed during the proceeding state of a consultation call.<br>If set to 2, phones will accept an INVITE with replaces for a dialog in early state. This is needed when using transfer on proceeding with a proxying call server such as openSIPS, reSIProcate or SipXecs. | | |
| **voIpProt.SIP.authOptimizedInFailover** | **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.<br>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.<br>If `reg.x.auth.optimizedInFailover` set to 0, this parameter is checked.<br>If `voIpProt.SIP.authOptimizedInFailover` is 0, then this feature is disabled.<br>If both parameters are set, the value of `reg.x.auth.optimizedInFailover` takes precedence. | | |
| **voIpProt.SIP.CID.sourcePreference** | **ASCII string up to 120 characters long** | **Null** |
| Specify the priority order for the sources of caller ID information. The headers can be in any order.<br>If Null, caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order.<br>The values `From,P-Asserted-Identity, Remote-Party-ID` and `P-Asserted-Identity,From, Remote-Party-ID` are also valid. | | |
| **voIpProt.SIP.compliance.RFC3261.validate.contentLanguage** | **0 or 1** | **1** |
| If set to 1, validation of the SIP header content language is enabled. If set to 0, validation is disabled. | | |
| **voIpProt.SIP.compliance.RFC3261.validate.contentLength** | **0 or 1** | **1** |
| If set to 1, validation of the SIP header content length is enabled. | | |
| **voIpProt.SIP.compliance.RFC3261.validate.uriScheme** | **0 or 1** | **1** |
| If set to 1, validation of the SIP header URI scheme is enabled. If set to 0, validation is disabled. | | |

**Session Initiation Protocol (SIP) Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.SIP.conference.address** | **ASCII string up to 128 characters long** | **Null** |
| If Null, conferences are set up on the phone locally.<br>If set to some value, conferences are set up by the server using the conferencing agent specified by this address. Acceptable values depend on the conferencing server implementation policy. | | |
| **voIpProt.SIP.conference.parallelRefer** | **0 or 1** | **0** |
| If 1, a parallel REFER is sent to the call server. **Note**: This parameter must be set for Siemens OpenScape Centralized Conferencing. | | |
| **voIpProt.SIP.connectionReuse.useAlias** | **0 or 1** | **0** |
| If set to 0, the alias parameter is not added to the via header<br>If set to 1, the phone uses the connection reuse draft which introduces "alias". | | |
| **voIpProt.SIP.csta** | **0 or 1** | **0** |
| If 0, the uaCSTA (User Agent Computer Supported Telecommunications Applications) feature is disabled. If 1, uaCSTA is enabled (If `reg.x.csta` is set, it will override this parameter). | | |
| **voIpProt.SIP.dialog.strictXLineID** | **0 or 1** | **0** |
| If 0, the phone will not look for x-line-id (call appearance index) in a SIP INVITE message, if one is not present. Instead, when it receives INVITE, the phone will generate the call appearance locally and pass that information to other parties involved in the call. | | |
| **voIpProt.SIP.dialog.usePvalue** | **0 or 1** | **0** |
| If set to 0, phone uses a `pval` field name in Dialog. This obeys the draft-ietf-sipping-dialog-package-06.txt draft.<br>If set to 1, the phone uses a field name of `pvalue`. | | |
| **voIpProt.SIP.dialog.useSDP** | **0 or 1** | **0** |
| If set to 0, a new dialog event package draft is used (no SDP in dialog body).<br>If set to 1, for backwards compatibility, use this setting to send SDP in the dialog body. | | |
| **voIpProt.SIP.dtmfViaSignaling.rfc2976**[1] | **0 or 1** | **0** |
| Enable or disable DTMF relays for active SIP calls. Not supported for H.323 calls. If set to 1, DTMF digit information is sent in RFC2976 SIP INFO packets during a call. If set to 0, no DTMF digit information is sent. | | |
| **voIpProt.SIP.dtmfViaSignaling.rfc2976.nonLegacyEncoding**[1] | **0 or 1** | **0** |
| Controls the behavior of the Star and Pound keys used for DTMF relays for active SIP calls. Not supported for H.323 calls.<br>If set to 1, the phone sends an asterisk (*) when the Star key is pressed and a hashtag (#) when the Pound key is pressed.<br>If set to 0, the phone sends 10 when the Star key (*) is pressed and 11 when the Pound key (#) is pressed. | | |
| **voIpProt.SIP.enable**[1] | **0 or 1** | **1** |

**Session Initiation Protocol (SIP) Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| A flag to determine if the SIP protocol is used for call routing, dial plan, DTMF, and URL dialing.<br>If set to 1, the SIP protocol is used. | | |
| **voIpProt.SIP.failoverOn503Response** | **0 or 1** | **1** |
| A flag to determine whether or not to trigger a failover if the phone receives a 503 response. You must use a registration expiry of 66 seconds or greater for failover with a 503 response to work properly. This rule applies both to the phone configuration (`reg.x.server.y.expires` and `voIpProt.server.x.expires`) as well as the 200 OK register response from the server. | | |
| **voIpProt.SIP.header.diversion.enable**[1] | **0 or 1** | **0** |
| If set to 1, the diversion header is displayed if received. If set to 0, the diversion header is not displayed. | | |
| **voIpProt.SIP.header.diversion.list.useFirst**[1] | **0 or 1** | **1** |
| If set to 1, the first diversion header is displayed. If set to 0, the last diversion header is displayed. | | |
| **voIpProt.SIP.header.warning.codes.accept** | **comma separated list** | **Null** |
| Specify a list of accepted warning codes.<br>If set to Null, all codes are accepted. Only codes between 300 and 399 are supported.<br>For example, if you want to accept only codes 325 to 330:<br>`voIpProt.SIP.header.warning.codes.accept=325,326,327,328,329,330`<br>Text will be shown in the appropriate language. For more information, see lcl_ml_lang_menu_x<XREF>. | | |
| **voIpProt.SIP.header.warning.enable** | **0 or 1** | **0** |
| If set to 1, the warning header is displayed if received. If set to 0, the warning header is not displayed. | | |
| **voIpProt.SIP.IM.autoAnswerDelay** | **0 to 40, seconds** | **10** |
| The time interval from receipt of the instant message invitation to automatically accepting the invitation. | | |
| **voIpProt.SIP.intercom.alertInfo** | **Alpha-Numeric string** | **Intercom** |
| The string you want to use in the Alert-Info header. You can use the following characters: '@', '-' ,'_' , '.' .<br>If you use any other characters, NULL, or empty spaces, the call is sent as normal without the Alert-Info header. | | |
| **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 an INVITE. The phone will still respond to a re-INVITE or UPDATE. The phone will not try to re-INVITE or UPDATE even if the remote endpoint asks for it. | | |
| **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.local.port**[1] | **0 to 65535** | **5060** |

**Session Initiation Protocol (SIP) Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| The local port for sending and receiving SIP signaling packets.<br>If set to 0, 5060 is used for the local port but is not advertised in the SIP signaling.<br>If set to some other value, that value is used for the local port and it is advertised in the SIP signaling. | | |
| **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 applies when installing Microsoft Live Communications Server.<br>Note that if any endpoint registered to the same account has MS-forking disabled, all other endpoints default back to non-forking mode. Windows Messenger does not use MS-forking so be aware of this behavior if one of the endpoints is using Windows Messenger. | | |
| **voIpProt.SIP.musicOnHold.uri** | **a SIP URI** | **Null** |
| A URI that provides the media stream to play for the remote party on hold. This parameter is used if `reg.x.musicOnHold.uri` is Null.<br>Note: The SIP URI parameter transport is supported when configured with the values of `UDP`, `TCP`, or `TLS`. | | |
| **voIpProt.SIP.newCallOnUnRegister** | **0 or 1** | **1** |
| If set to 0 , the phone does not generate new Call-ID and From tag during re-registration. | | |
| **voIpProt.SIP.outboundProxy.address** | **IP address or hostname** | **Null** |
| The IP address or hostname of the SIP server to which the phone sends all requests. | | |
| **voIpProt.SIP.outboundProxy.port** | **0 to 65535** | **0** |
| The port of the SIP server to which the phone sends all requests. | | |
| **voIpProt.SIP.outboundProxy.failOver.failBack.mode** | **newRequests, DNSTTL, registration, duration,** | **duration** |
| The mode for failover failback (overrides `voIpProt.server.x.failOver.failBack.mode`).<br>• **newRequests** All new requests are forwarded first to the primary server regardless of the last used server.<br>• **DNSTTL** The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.<br>• **registration** The phone tries the primary server again when the registration renewal signaling begins.<br>• **duration** The phone tries the primary server again after the time specified by `reg.x.outboundProxy.failOver.failBack.timeout` expires. | | |
| **voIpProt.SIP.outboundProxy.failOver.failBack.timeout** | **0, 60 to 65535** | **3600** |
| The time to wait (in seconds) before failback occurs (overrides `voIpProt.server.x.failOver.failBack.timeout`).If the fail back mode is set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again. If 0, the phone will not fail-back until a fail-over event occurs with the current server. | | |
| **voIpProt.SIP.outboundProxy.failOver.failRegistrationOn** | **0 or 1** | **0** |

**Session Initiation Protocol (SIP) Parameters  (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| When set to 1, and the reRegisterOn parameter is enabled, the phone will silently invalidate an existing registration (if it exists), at the point of failing over. When set to 0, and the reRegisterOn parameter is enabled, existing registrations will remain active. This means that the phone will attempt failback without first attempting to register with the primary server to determine if it has recovered.<br>Note that `voIpProt.SIP.outboundProxy.failOver.RegisterOn` must be enabled. | | |
| **voIpProt.SIP.outboundProxy.failOver.onlySignalWithRegistered** | **0 or 1** | **1** |
| When set to 1, and the reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call will end. No SIP messages will be sent to the unregistered server. When set to 0, and the reRegisterOn and failRegistrationOn parameters are enabled, signaling will be accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred). This parameter overrides `voIpProt.server.x.failOver.onlySignalWithRegistered`. | | |
| **voIpProt.SIP.outboundProxy.failOver.reRegisterOn** | **0 or 1** | **0** |
| This parameter overrides the `voIpProt.server.x.failOver.reRegisterOn`. When set to 1, the phone will attempt to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling will proceed with the secondary server. When set to `0`, the phone won't attempt to register with the secondary server, since the phone will assume that the primary and secondary servers share registration information. | | |
| **voIpProt.SIP.outboundProxy.transport** | **DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly** | **DNSnaptr** |
| The transport method the phone uses to communicate with the SIP server.<br>• **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.<br>• **TCPpreferred**    TCP is the preferred transport, UDP is used if TCP fails.<br>• **UDPOnly**    Only UDP will be used.<br>• **TLS**    If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.<br>• **TCPOnly**    Only TCP will be used. | | |
| **voIpProt.SIP.pingInterval** | **0 to 3600** | **0** |
| The number in seconds to send PING message. This feature is disabled by default. | | |
| **voIpProt.SIP.pingMethod** | **PING, OPTIONS** | **PING** |
| The ping method to be used. | | |
| **voIpProt.SIP.presence.nortelShortMode**[1] | **0 or 1** | **0** |
| Different headers sent in SUBSCRIBE when used for presence on an Avaya (Nortel) server. Support is indicated by adding a header `Accept-Encoding: x-nortel-short`. A PUBLISH is sent to indicate the status of the phone. | | |

**Session Initiation Protocol (SIP) Parameters  (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.SIP.requestValidation.digest.realm[1]** | **A valid string** | **PolycomSPIP** |
| Determines the string used for Realm. | | |
| **voIpProt.SIP.requestValidation.x.method[1]** | **Null, source, digest, both, all** | **Null** |
| If Null, no validation is made. Otherwise this sets the type of validation performed for the request:<br>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 | | |
| **voIpProt.SIP.requestValidation.x.request[1]** | **INVITE, ACK , BYE, REGISTER, CANCEL, OPTIONS, INFO, MESSAGE, SUBSCRIBE, NOTIFY, REFER, PRACK, UPDATE** | **Null** |
| Sets the name of the method for which validation will be applied.<br>Note: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases. | | |
| **voIpProt.SIP.requestValidation.x.request.y.event[1]** | **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`.<br>If set to Null, all events will be validated. | | |
| **voIpProt.SIP.requestURI.E164.addGlobalPrefix** | **0 or 1** | **0** |
| If set to 1, '+' global prefix is added to the E.164 user parts in sip: URIs. | | |
| **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`.<br>If set to 1, SIP header names generated by the phone use the short form, for example `f`. | | |
| **voIpProt.SIP.serverFeatureControl.callRecording** | **0 or 1** | **0** |
| Enable or disable the BroadSoft BroadWorks v20 call recording feature for multiple phones. | | |
| **voIpProt.SIP.serverFeatureControl.cf[1]** | **0 or 1** | **0** |
| If set to 1, server-based call forwarding is enabled. Server and local phone call-forwarding are synchronized. If set to 0, server-based call forwarding is not enabled. Requires server-side support of synchronized call forwarding. | | |
| **voIpProt.SIP.serverFeatureControl.dnd[1]** | **0 or 1** | **0** |
| If set to 1, server-based DND is enabled. Server and local phone DND are synchronized.<br>If set to 0, server-based DND is not enabled. Requires server-side support of synchronized do not disturb (DND). | | |

**Session Initiation Protocol (SIP) Parameters  (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.SIP.serverFeatureControl.localProcessing.cf** | **0 or 1** | **1** |
| This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.cf`.<br>If set to 0 and `voIpProt.SIP.serverFeatureControl.cf` is set to 1, call forwarding is performed on the server side only, and the phone does not perform local call forwarding.<br>If set to 1 and `voIpProt.SIP.serverFeatureControl.cf` is set to 1, the phone and the server perform call forwarding.<br>If both `voIpProt.SIP.serverFeatureControl.localProcessing.cf` and `voIpProt.SIP.serverFeatureControl.cf` are set to 0, the phone performs local call forwarding and the `localProcessing` parameter is not used. | | |
| **voIpProt.SIP.serverFeatureControl.localProcessing.dnd** | **0 or 1** | **1** |
| This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.dnd`.<br>If set to 0 and `voIpProt.SIP.serverFeatureControl.dnd` is set to 1, do not disturb (DND) is performed on the server-side only, and the phone does not perform local DND.<br>If set to 1 and `voIpProt.SIP.serverFeatureControl.dnd` is set to 1, the phone and the server perform DND.<br>If both `voIpProt.SIP.serverFeatureControl.localProcessing.dnd` and `voIpProt.SIP.serverFeatureControl.dnd` are set to 0, the phone performs local DND and the `localProcessing` parameter is not used. | | |
| **voIpProt.SIP.serverFeatureControl.missedCalls[1]** | **0 or 1** | **0** |
| If set to 1, server-based missed calls is enabled. The call server has control of missed calls.<br>If set to 0, server-based missed calls is not enabled. | | |
| **voIpProt.SIP.serverFeatureControl.securityClassification** | **0 or 1** | **0** |
| Enable or disable the visual security classification feature for all lines on a phone. | | |
| **voIpProt.SIP.specialEvent.checkSync.alwaysReboot[1]** | **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 even if there has not been a change to software or configuration.<br>If set to 0, the phone will only reboot if necessary. Many configuration parameter changes can be applied dynamically without the need for a reboot. | | |
| **voIpProt.SIP.specialEvent.checkSync.downloadDirectory[1]** | **0 or 1** | **0** |
| If set to 0, the phone only downloads software and configuration updates after receiving a `checksync NOTIFY` message. If set to 1, the phone downloads the updated global and personal directory files along with any software and configuration updates after receiving a `checksync NOTIFY` message. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates. | | |
| **voIpProt.SIP.specialEvent.lineSeize.nonStandard[1]** | **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. | | |

**Session Initiation Protocol (SIP) Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.SIP.strictLineSeize** | **0 or 1** | **0** |
| If set to 1, The phone is forced to wait for a 200 OK response when receiving a TRYING notify.<br>If set to 0, dial prompt is provided immediately when you attempt to seize a shared line without waiting for a successful OK from the call server. | | |
| **voIpProt.SIP.strictReplacesHeader** | **0 or 1** | **1** |
| This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.<br>If set to 1, the phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when `call.directedCallPickupMethod` is configured as `native`.<br>If set to 0, call pick-up requires a call id only. | | |
| **voIpProt.SIP.strictUserValidation** | **0 or 1** | **0** |
| If set to 1, the phone is forced to match the user portion of signaling exactly.<br>If set to 0, the phone will use the first registration if the user part does not match any registration. | | |
| **voIpProt.SIP. supportFor100rel** | **0 or 1** | **1** |
| If set to 1, the phone advertises support for reliable provisional responses in its offers and responses.<br>If set to 0, the phone will not offer 100rel and will reject offers requiring 100rel. | | |
| **voIpProt.SIP.tcpFastFailover** | **0 or 1** | **0** |
| If set to 1, failover occurs based on the values of `reg.x.server.y.retryMaxCount` and `voIpProt.server.x.retryTimeOut`.<br>If 0, a full 32 second RFC compliant timeout is used. See `reg.x.tcpFastFailover` . | | |
| **voIpProt.SIP.tlsDsk.enable** | **0 or 1** | **0** |
| If 0, TLS DSK is disabled. If 1, TLS DSK is enabled. For more information, see Protocol Overview on Microsoft Developer Network. | | |
| **voIpProt.SIP.turnOffNonSecureTransport**[1] | **0 or 1** | **0** |
| If set to 1, stop listening to port 5060 when using AS-SIP enabled. | | |
| **voIpProt.SIP.use486forReject** | **0 or 1** | **0** |
| If set to 1 and the phone is indicating a ringing inbound call appearance, the phone will transmit a 486 response to the received INVITE when the Reject soft key is pressed.<br>If set to 0, no 486 response is transmitted. | | |
| **voipPort.SIP.useCompleteUriForRetrieve** | **0 or 1** | **1** |
| If set to 1, the target URI in BLF signaling will use the complete address as provided in the xml dialog document.<br>If set to 0, only the user portion of the XML dialog document is used and the current registrar's domain is appended to create the full target URI. | | |

**Session Initiation Protocol (SIP) Parameters (continued)**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voipProt.SIP.useLocalTargetUriforLegacyPickup** | **0 or 1** | **1** |
| If set to 1, BLF signaling will use the address as provided in the local target URI in xml dialog document with additional rules based on `voipPort.SIP.useCompleteUriForRetrieve`.<br>If set to 0, the local target uri is not considered and the identity attribute is used with additional rules based on `voipPort.SIP.useCompleteUriForRetrieve`. | | |
| **voIpProt.SIP.useContactInReferTo** | **0 or 1** | **0** |
| If set to 0, the “To URI” is used in the REFER.<br>If set to 1, the “Contact URI” is used in the REFER. | | |
| **voIpProt.SIP.useRFC2543hold** | **0 or 1** | **0** |
| If set to 0, use SDP media direction parameters (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.<br>Note: `voIpProt.SIP.useRFC2543hold` is effective only when the call is initiated. | | |
| **voIpProt.SIP.useRFC3264HoldOnly** | **0 or 1** | **0** |
| If set to 1, and no media direction is specified, the phone uses `sendrecv` compliant with RFC 3264 when negotiating SDP and generates responses containing RFC 3264-compliant media attributes for calls placed on and off hold by either end. If set to 0, and no media direction is specified, the phone enters backward compatibility mode when negotiating SDP and responds using the c=0.0.0.0 RFC 2543 signaling method.<br>Note: `voIpProt.SIP.useSendonlyHold` applies only to calls on phones that originate the hold. | | |
| **voIpProt.SIP.useSendonlyHold** | **0 or 1** | **1** |
| If set to 1, the phone will send a reinvite with a stream mode parameter of “sendonly” when a call is put on hold. This is the same as the previous behavior.<br>If set to 0, the phone will send a reinvite with a stream mode parameter of “inactive” when a call is put on hold.<br>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). | | |

[1] Change causes phone to restart or reboot.

## <H323/>

The parameters listed in the next table are supported only with the Polycom VVX 500/501, 600/601, and 1500 phones.

**H.323 Protocol Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **voIpProt.H323.autoGateKeeperDiscovery**[1] | **0 or 1** | **1** |
| 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.<br>If set to 0, the phone will not send out any gatekeeper discovery messages. | | |

**H.323 Protocol Parameters (continued)**

| | | |
|---|---|---|
| **voIpProt.H323.blockFacilityOnStartH245**[1] | **0 or 1** | **0** |
| If set to 1, facility messages when using H.245 are removed. | | |
| **voIpProt.H323.dtmfViaSignaling.enabled**[1] | **0 or 1** | **1** |
| If set to 1, the phone will use the H.323 signaling channel for DTMF key press transmission. | | |
| **voIpProt.H323.dtmfViaSignaling.H245alphanumericMode**[1] | **0 or 1** | **1** |
| If set to 1, the phone will support H.245 signaling channel alphanumeric mode DTMF transmission.<br>Note: If both alphanumeric and signal modes can be used, the phone gives priority to DTMF. | | |
| **voIpProt.H323.dtmfViaSignaling.H245signalMode**[1] | **0 or 1** | **1** |
| If set to 1, the phone will support H.245 signaling channel signal mode DTMF transmission. | | |
| **voIpProt.H323.enable**[1] | **0 or 1** | **0** |
| A flag to determine if the H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing.<br>If set to 1, the H.323 protocol is used. | | |
| **voIpProt.H323.local.port**[1] | **0 to 65535** | **1720** |
| Local port to be used for H.323 signaling. Local port for sending and receiving H.323 signaling packets.<br>If set to 0, 1720 is used for the local port but is not advertised in the H.323 signaling.<br>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.local.RAS.port**[1] | **1 to 65535** | **1719** |
| Local port for RAS signaling. | | |

[1] Change causes phone to restart or reboot.

# <webutility/>

The parameters listed in the next table specify the download location of the translated language files for the Web Configuration Utility.

**Web Configuration Utility Parameters**

| Parameter | Permitted Values | Default |
|---|---|---|
| **webutility.language.plcmServerUrl** | **URL** | `http://downloads.polycom.com/voice/software/languages/` |
| The download location of the translated language files for the Web Configuration Utility. | | |

# <xmpp/>

The parameters in the following table set the XML streaming protocols for instant messaging, presence, and contact list for BroadSoft features.

**XML Streaming Protocol Parameters**

| *Parameter* | *Permitted Values* | *Default* |
|---|---|---|
| **xmpp.1.auth.domain** | **UTF-8 encoded string** | **Null** |
| Specify the domain name of the XMPP server. | | |
| **xmpp.1.auth.password** | **UTF-8 encoded string** | **Null** |
| Password used for XMPP registration. Specify the password for XMPP registration. | | |
| **xmpp.1.auth.useLoginCredentials** | **0 or 1** | **0** |
| Choose whether or not to use the login credentials provided in the phone's Login Credentials Menu for XMPP authentication. | | |
| **xmpp.1.dialMethod** | **String min 0, max 256** | **SIP** |
| For SIP dialing, the destination XMPP URI is converted to a SIP URI, and the first available SIP line is used to place the call. | | |
| **xmpp.1.enable** | **0 or 1** | **0** |
| Enable or disable XMPP presence. | | |
| **xmpp.1.jid** | **String min 0, max 256** | **Null** |
| Enter the Jabber identity used to register with the presence server, for example: `presence.test2@polycom-alpha.eu.bc.im`. | | |
| **xmpp.1.roster.invite.accept** | **Automatic or prompt** | **prompt** |
| Choose how phone users receive the BroadSoft XMPP invitation to be added to a buddy list. If set to prompt, the phone displays a list of users who have requested to add you as a buddy and you can accept or reject the invitation. | | |
| **xmpp.1.server** | **dotted-decimal IP address, host name, or FQDN** | **Null** |
| Sets the BroadSoft XMPP presence server to an IP address, host name, or FQDN, for example: `polycom-alpha.eu.bc.im`. | | |
| **xmpp.1.verifyCert** | **0 or 1** | **1** |
| Enable or disable verification of the TLS certificate provided by the BroadSoft XMPP presence server. | | |

# Session Initiation Protocol (SIP)

This section describes the basic Session Initiation Protocol (SIP) and the protocol extensions that the current Polycom UC Software supports.

This section contains information on:

- **Basic Protocols**

  All 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 RFCs and Internet drafts, see the section RFC and Internet Draft Support.

## RFC and Internet Draft Support

The following RFCs and Internet drafts are supported. For more information on any of the documents, enter the RFC number at Request for Comments (RFC).

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

**Forking is unsupported**

The following sections of RFC 3265 are not supported:

- Section 3.3.3 Forking
- Section 4.4.9 Handling of forked requests

RFC 3311—The Session Initiation Protocol (SIP) UPDATE Method

RFC 3325—SIP Asserted Identity

RFC 3420—Internet Media Type message/sipfrag

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 SIP request messages listed in the following table are supported.

**Supported SIP Request Messages**

| Method | Supported | Notes |
|---|---|---|
| REGISTER | Yes | |
| INVITE | Yes | |

**Supported SIP Request Messages**

| Method | Supported | Notes |
|---|---|---|
| ACK | Yes | |
| CANCEL | Yes | |
| BYE | Yes | |
| OPTIONS | Yes | |
| SUBSCRIBE | Yes | |
| NOTIFY | Yes | |
| REFER | Yes | |
| PRACK | Yes | |
| 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 | |

# Supported SIP Request Headers

The following table lists the SIP request headers that are supported. Note that 'Yes' in the Supported column indicates that the header is sent and properly parsed.

**Supported SIP Request Headers**

| Header | Supported |
|---|---|
| Accept | Yes |
| Accept-Encoding | Yes |
| Accept-Language | Yes |
| Accept-Resource-Priority | Yes |
| Access-Network-Info | No |
| Access-URL | Yes |
| Alert-Info | Yes |
| Allow | Yes |
| Allow-Events | Yes |
| Authentication-Info | Yes |

**Supported SIP Request Headers**

| Header | Supported |
|---|---|
| Authorization | Yes |
| Call-ID | Yes |
| Call-Info | Yes |
| Contact | Yes |
| Content-Disposition | Yes |
| Content-Encoding | Yes |
| Content-Language | Yes |
| Content-Length | Yes |
| Content-Type | Yes |
| CSeq | Yes |
| Date | Yes (For missed call; not used to adjust the time of the phone) |
| Diversion | Yes |
| Error-Info | No |
| Event | Yes |
| Expires | Yes |
| Flow-Timer | Yes |
| From | Yes |
| In-Reply-To | No |
| Join | Yes |
| Max-Forwards | Yes |
| Min-Expires | Yes |
| Min-SE | Yes |
| MIME-Version | No |
| Missed-Calls | Yes |
| ms-client-diagnostics | Yes |
| ms-keep-alive | Yes |
| ms-text-format | Yes |
| Organization | No |
| P-Asserted-Identity | Yes |

### Supported SIP Request Headers

| Header | Supported |
|---|---|
| P-Preferred-Identity | Yes |
| Priority | No |
| Privacy | No |
| Proxy-Authenticate | Yes |
| Proxy-Authorization | Yes |
| Proxy-Require | Yes |
| RAck | Yes |
| Reason | Yes |
| Record-Route | Yes |
| Refer-Sub | Yes |
| Refer-To | Yes |
| Referred-By | Yes |
| Referred-To | Yes |
| Remote-Party-ID | Yes |
| Replaces | Yes |
| Reply-To | No |
| Requested-By | No |
| Require | Yes |
| Resource-Priority | Yes |
| Response-Key | No |
| Retry-After | Yes |
| Route | Yes |
| RSeq | Yes |
| Server | Yes |
| Session-Expires | Yes |
| SIP-Etag | Yes |
| SIP-If-Match | Yes |
| Subject | Yes |
| Subscription-State | Yes |

**Supported SIP Request Headers**

| Header | Supported |
|---|---|
| Supported | Yes |
| Timestamp | Yes |
| To | Yes |
| Unsupported | Yes |
| User-Agent | Yes |
| Via | Yes |
| voice-missed-call | Yes |
| Warning | Yes (Only warning codes 300 to 399) |
| WWW-Authenticate | Yes |
| X-Sipx-Authidentity | Yes |

# Response Support

Note that 'Yes' in the Supported column indicates that the header is sent and properly parsed. The phone might not generate the response.The SIP responses are listed in the following tables:

- Supported 1xx SIP Responses
- Supported 2xx SIP Responses
- Supported 3xx SIP Responses
- Supported 4xx SIP Responses
- Supported 5xx SIP Responses
- Supported 6xx SIP Responses

## 1xx Responses - Provisional

**Supported 1xx SIP Responses**

| Response | Supported |
|---|---|
| 100 Trying | Yes |
| 180 Ringing | Yes |
| 181 Call Is Being Forwarded | No |
| 182 Queued | No |
| 183 Session Progress | Yes |

## 2xx Responses - Success

**Supported 2xx SIP Responses**

| Response | Supported | Notes |
|---|---|---|
| 200 OK | Yes | |
| 202 Accepted | Yes | In REFER transfer. |

## 3xx Responses - Redirection

**Supported 3xx SIP Responses**

| Response | Supported |
|---|---|
| 300 Multiple Choices | Yes |
| 301 Moved Permanently | Yes |
| 302 Moved Temporarily | Yes |
| 305 Use Proxy | No |
| 380 Alternative Service | No |

## 4xx Responses - Request Failure

All 4xx responses for which the phone does not provide specific support will be treated the same as 400 Bad Requests.

**Supported 4xx SIP Responses**

| Response | Supported |
|---|---|
| 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 |

**Supported 4xx SIP Responses**

| Response | Supported |
|---|---|
| 414 Request-URI Too Long | No |
| 415 Unsupported Media Type | Yes |
| 416 Unsupported URI Scheme | No |
| 420 Bad Extension | No |
| 421 Extension Required | No |
| 423 Interval Too Brief | Yes |
| 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

**Supported 5xx SIP Responses**

| Response | Supported |
|---|---|
| 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

**Supported 6xx SIP Responses**

| Response | Supported |
|---|---|
| 600 Busy Everywhere | No |
| 603 Decline | Yes |
| 604 Does Not Exist Anywhere | No |
| 606 Not Acceptable | No |

## Hold Implementation

The phone supports two 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/> parameters), but both methods are supported when signaled by the remote endpoint

**Hold methods**

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 used 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 <SIP/> and <reg/> 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 (SCA) Signaling

A shared line is an address of record managed by a call server. The server allows multiple endpoints to register locations against the address of record.

Polycom devices support SCA using the SUBSCRIBE-NOTIFY method specified in RFC 6665. 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 endpoints 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 event used is “dialog” for bridged line appearance subscribe and notify.