Polycom® UC Software 4.0.14

For Polycom® SoundPoint® IP Conference Phones, Polycom® SoundStation® IP Conference Phones, Polycom® SoundStation® Duo Conference Phones, Polycom® VVX® 500 and 1500 Business Media Phones, and Polycom® SoundStructure® VoIP Interface Phones

Polycom announces the release of Polycom® Unified Communications (UC) Software, version 4.0.14. This document provides the latest information about this release.

Contents

What's New

Polycom® Unified Communications (UC) Software 4.0.14 is a maintenance release for Open SIP deployments. These release notes provide important information on software updates, phone features, and known issues.

OpenSSL Upgrade

OpenSSL has been upgraded from version 0.9.8zg to version 1.0.2j for Polycom SoundStation IP 6000 conference phones.

Certificate Enhancements

The following certificate has been added in this release.

Certificate Updates

Certificate Common Name	RSA Public Key	Signature Algorithm	Validity Period Start	Validity Period End
QuoVadis Root CA 2 G3 https://www.quovadisglobal.be/Repository/DownloadRootsAndCRL.aspx	4096 bit	sha256WithRSAEncryption	Jan 12 18:59:32 2012 GMT	Jan 12 18:59:32 2042 GMT

The following certificate has been removed in this release.

Certificate Updates

Certificate Common Name	RSA Public Key	Signature Algorithm	Validity Period Start	Validity Period End
C=US, O=VeriSign, Inc., OU=Class 1 Public Primary Certification Authority - G2, OU=(c) 1998 VeriSign, Inc. - For authorized use only, OU=VeriSign Trust Network	1024 bit	sha1WithRSAEncryption	May 18 00:00:00 1998 GMT	May 18 23:59:59 2018 GMT

Introducing the Integrated Updater Application

In Polycom UC software 4.0.x, the Updater files and the UC software files are combined in a single, integrated software package. The Updater application is new for the UC software and replaces the BootROM application incorporated into previous software versions. The name Updater applies to UC software versions 4.0.0 or later, while the name BootROM continues to be used in UC software versions earlier than 4.0.0.

Updater Version

Polycom UC software 4.0.x firmware has new updater/bootRom **Polycom UC Updater 5.0.14.0580/0579 rts5.**

Only Polycom SoundStation IP 6000 and 7000 conference phones require manual procedure to upgrade new updater version **Polycom UC Updater 5.0.14.0580/0579 rts5**.

Updater Version Table

Platforms	App Version	Updater/BootROM Version
IP6000 (Mirage)	4.0.14.0987	5.0.14.0580/0579 (Manual installation)
IP7000 (Onyx)	4.0.14.0987	5.0.14.0580/0579 (Manual installation)

Phone Features and Licenses

As of release 4.0.0, Polycom UC software supports the Productivity Suite, which includes features such as a Corporate Directory, Visual Conference Management, USB Call Recording, Polycom Desktop Connector, and Exchange Calendar. Upgrading to the current Polycom UC software automatically enables the Productivity Suite; no license is required. The Voice Quality Monitoring (VQMon) feature continues to be a licensed and paid product.

For customers using versions of Polycom UC software prior to 4.0.0, Polycom provides a site license for all features in the Productivity Suite, except for the VQMon feature. To get a license for the VQMon feature, select *Download a Non VQMon SITE License* on Polycom Voice Applications. You can also find a trial license for the VQMon feature which enables you access to the feature for a limited time.

Polycom UC software 4.0.14 supports a range of features that are available on the Polycom SoundStation IP conference phone, Polycom SoundStation conference phone, Polycom VVX business media phone, and Polycom SoundStructure VoIP interface phones. The features and licenses required to operate the phones vary by phone model. Refer to the Polycom SoundStation IP Conference Phone Features and Licenses table or the Polycom SoundStation Conference Phone, Polycom VVX Business Media Phone, and Polycom SoundStructure VoIP Interface Phone Features and Licenses, and table to find out which phone features and licenses you require for your phone model.

The following table describes features available for each phone and indicates whether a feature license is required.

- *No* indicates that a phone does not support a feature.
- *Yes* indicates that a phone supports a feature and no license is required.
- *Yes** indicates that the phone requires you to purchase a feature license from Polycom to support a feature.

Polycom SoundStation IP Conference Phone Features and Licenses

Feature	Polycom SoundStation IP Conference Phone 321/331/335	Polycom SoundStation IP Conference Phone 450/550/560	Polycom SoundStation IP Conference Phone 650/670
4-way local conference	No	Yes	Yes
Asian Languages	Chinese only	Yes	Yes
Call Recording	No	No	Yes
Conference Management	No	Yes	Yes
Configurable Soft keys	Yes	Yes	Yes
Customizable UI Background	No	Yes	Yes
Electronic Hookswitch	Yes	Yes	Yes
Enhanced BLF	No	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes
H.323 Video	No	No	No
VQMon	Yes*	Yes*	Yes*

Polycom SoundStation Conference Phone, Polycom VVX Business Media Phone, and Polycom SoundStructure VoIP Interface Phone Features and Licenses

Feature	Polycom SoundPoint IP 5000 Conference Phone	Polycom SoundStation IP 6000 Confrence Phone	Polycom SoundStation IP 7000 Conference Phone	Polycom SoundStation Duo Conference Phone	Polycom VVX 1500 Business Media Phone	Polycom VVX 500 Business Media Phone	Polycom SoundStructure VoIP Interface Phone
Asian Languages	Yes	Yes	Yes	Yes	Yes	Yes	No
Conference Management	No	No	Yes	Yes	Yes	Yes	No
Customizable UI Background	No	No	No	No	Yes	Yes	No
Electronic Hookswitch	No	No	No	No	Yes	Yes	No
Enhanced BLF	No	No	No	No	No	Yes	No

Polycom SoundStation Conference Phone, Polycom VVX Business Media Phone, and Polycom SoundStructure VoIP Interface Phone Features and Licenses

Feature	Polycom SoundPoint IP 5000 Conference Phone	Polycom SoundStation IP 6000 Confrence Phone	Polycom SoundStation IP 7000 Conference Phone	Polycom SoundStation Duo Conference Phone	Polycom VVX 1500 Business Media Phone	Polycom VVX 500 Business Media Phone	Polycom SoundStructure VoIP Interface Phone
Enhanced Feature Keys	No	No	Yes*	No	Yes	Yes	No
H.323 Video	No	No	No	No	Yes*	No	No
Local Call Recording	No	No	No	No	Yes	Yes	No
Pinyin Character Entry	No	No	No	No	Yes*	No	No
VQMon	Yes*	No	No	Yes*	Yes	Yes	No

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC software 4.0.14 release. For more information on using configuration parameters to enable or disable features, see the *Polycom UC Software Administrator Guide* available on the Polycom Support - Voice site.

Configuration File Enhancements for Polycom UC Software 4.0.14

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.browser`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and phone browser. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The microbrowser restarts when there is a change in the browser TLS protocol or TLS cipher settings, and the last web page displayed is not restored. The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.ldap`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Lightweight Directory Access Protocol (LDAP). The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation 7000 conference phones are TLSv1_0 (default) SSLv2v3	No

Configuration File Enhancements for Polycom UC Software 4.0.14 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.sip`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and SIP signaling. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.sopi`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and SOPI. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.webServer`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Web Server. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	Yes

Configuration File Enhancements for Polycom UC Software 4.0.14 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.xmpp`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and XMPP. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.exchangeServices`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Exchanges services. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`device.sec.TLS.protocol.syslog`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Syslog. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No

Configuration File Enhancements for Polycom UC Software 4.0.14 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.sec.TLS.protocol.prov	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and provisioning. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
device.cfg, site.cfg	device.sec.TLS.protocol.dot1x	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and 802.1x authentication. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_0, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 7000 conference phones is TLSv1_0 (default)	Yes
site.cfg	sec.uploadDevice.privateKey	0 (default) - While generating the Certificate Signing Request from the phone, the device private key is not uploaded to provisioning server. 1 - The device private key is uploaded to provisioning server along with the CSR.	No

Security Updates

Please refer to the Polycom Security Center for information about known and resolved security vulnerabilities.

Install

Consider the following installation and update information when using Polycom UC software 4.0.14.

Download the Distribution Files

To download Polycom UC software 4.0.14, 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 and maintain software versions for each model in the same root directory.

For general use, Polycom recommends using the split resource file that corresponds to the phone models for your deployment. To match the correct UC software resource file to your phone model, see the table Understand the Combined ZIP and Split ZIP Files. If you are provisioning your phones centrally using configuration files, download the corresponding resource file and extract the configuration files to the provisioning server, maintaining the folder hierarchy in the ZIP file.

The current build ID for the sip.ld and resource files is **UCS 4.0.14.1580 rts26 G.**

Understand the Combined and Split ZIP Files

To understand the files distributed in the combined ZIP file, refer to the following table.

Understand the Combined ZIP and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
2345-12360-001.sip.ld	SIP application executable for Polycom SoundStation IP 321 conference phones	No	Yes
2345-12365-001.sip.ld	SIP application executable for Polycom SoundStation IP 331 conference phones	No	Yes
2345-12375-001.sip.ld	SIP application executable for Polycom SoundStation IP 335 conference phones	No	Yes
2345-12450-001.sip.ld	SIP application executable for Polycom SoundStation IP 450 conference phones	No	Yes
2345-12500-001.sip.ld	SIP application executable for Polycom SoundStation IP 550 conference phones	No	Yes
2345-12560-001.sip.ld	SIP application executable for Polycom SoundStation IP 560 conference phones	No	Yes
2345-12600-001.sip.ld	SIP application executable for Polycom SoundStation IP 650 conference phones	No	Yes
2345-12670-001.sip.ld	SIP application executable for Polycom SoundStation IP 670 conference phones	No	Yes
3111-30900-001.sip.ld	SIP application executable for Polycom SoundStation IP 5000 conference phones	No	Yes
3111-15600-001.sip.ld	SIP application executable for Polycom SoundStation IP 6000 conference phones	No	Yes
3111-40000-001.sip.ld	SIP application executable for Polycom SoundStation IP 7000 conference phones	No	Yes

(continued)Understand the Combined ZIP and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
3111-19000-001.sip.ld	SIP application executable for Polycom SoundStation Duo conference phones	No	Yes
3111-44500-001.sip.ld	SIP application executable for Polycom VVX 500 business media phones	No	Yes
2345-17960-001.sip.ld	SIP application executable for Polycom VVX 1500 business media phones	No	Yes
3111-33215-001.sip.ld	SIP application executable for Polycom SoundStructure VoIP interface phone	No	Yes
sip.ld	Concatenated SIP application executable	Yes	No
sip.ver	Text file detailing the build-identification(s) for the release.	Yes	Yes
000000000000.cfg	Master configuration template file	Yes	Yes
000000000000-directory~.xml	Local contact directory template file. To apply on a per phone basis, replace the 0s with the MAC address of the phone and remove '~' from the file name.	Yes	Yes
applications.cfg	Contains configuration parameters for microbrowser and browser applications.	Yes	Yes
features.cfg	Contains configuration parameters for telephony features.	Yes	Yes
H323.cfg	Contains configuration parameters for the H.323 signaling protocol.	Yes	Yes
reg-advanced.cfg	Contains configuration parameters for the line and call registration and advanced phone feature settings.	Yes	Yes
reg-basic.cfg	Contains configuration parameters for the line and call registration and basic phone feature settings.	Yes	Yes
region.cfg	Contains configuration parameters for regional and localization settings such as time, date and language.	Yes	Yes
sip-basic.cfg	Contains configuration parameters for the VoIP server, softswitch registration.	Yes	Yes
sip-interop.cfg	Contains configuration parameters for the VoIP server, softswitch registration, and interoperability configuration.	Yes	Yes
site.cfg	Contains configuration parameters that are set on a per site basis.	Yes	Yes
video.cfg	Contains configuration parameters for video connectivity.	Yes	Yes
video-integration.cfg	Contains configuration parameters for Polycom SoundStation IP 7000 conference phone and Polycom® HDX® integration.	Yes	Yes
SoundPointIPWelcome.wav	Start-up welcome sound effect	Yes	Yes

(continued)Understand the Combined ZIP and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
LoudRing.wav	Loud ringer sound effect	Yes	Yes
Warble.wav	Loud ringer sound effect	Yes	Yes

Version History

This following table shows the release history of Polycom® Unified Communications (UC) Software 4.0.14.

Version History

Release	Release Date	Features
4.0.14	July 2019	This release has important field fixes.
4.0.13	October 2017	This release has important field fixes.
4.0.12	June 2017	This release has important field fixes.
4.0.11 Rev K	April 2017	New software change on Polycom VVX 500 business media phones to support new LCD panel on the phones.
4.0.11	June 2016	This release has important field fixes.
4.0.10 Rev D	April 2016	This release has important field fixes.
4.0.10	January 2016	This release has important field fixes.
4.0.9	June 2015	This release has important field fixes.
4.0.8	January 2015	This release has important field fixes.
4.0.7	July 2014	This release has important field fixes.
4.0.6	March 2014	Added support for the improved touchscreen driver for the VVX 500 business media phone.
4.0.5	December 2013	Added support for GENBAND directory, Global Address Book (GAB) and Personal Address Book (PAB). Added support for Multiple Appearance Directory Number-Single Call Appearance (MADN-SCA) for Shared Line Appearance scenarios.
4.0.4 Rev C	June 2013	Added corporate directory under the Directories softkey in the call transfer menu.

Language Support

The Polycom VVX phones' user interface include native support for the following languages:

- Chinese, Traditional
- Chinese, Simplified
- Danish, Denmark

- Dutch, Netherlands
- English, Canada
- English, United Kingdom
- English, United States
- French, France
- German, Germany
- Italian, Italy
- Japanese, Japan
- Korean, Korea
- Norwegian, Norway
- Polish, Poland
- Portuguese, Portugal
- Russian, Russia
- Slovenian, Slovenia
- Spanish, Spain
- Swedish, Sweden

Resolved Issues

The following issues have been resolved for the Polycom UC software 4.0.14 release.

Polycom UC Software 4.0.14 Resolved Issues

Category	Issue Number	Description
Call Management	EN-93086	Polycom phone does not perform Naming Authority Pointer (NAPTR) query for the outbound proxy while validating the incoming request using parameters `voIpProt.SIP.requestValidation.1.method`= “source” and `voIpProt.SIP.requestValidation.1.request`= “INVITE”.
Calling	EN-93139	Polycom VVX business media phone continues to display the placed call list when `feature.callList.enabled` and `feature.callListPlaced.enabled` parameters are disabled.
Calling	EN-82307	Polycom phone goes to hold state immediately after the call is answered, when the phone receives two consecutive UPDATE messages during early dialog, one without a direction attribute and the other with a direction attribute as “recvonly” in the SDP.
Calling	EN-93085	Polycom SoundStation IP conference phone puts the call on hold while dialing to hunt group which has the pre-alerting feature enabled.
Calling	EN-62056	Polycom SoundStation IP conference phone goes to held state immediately when it receives “recvonly” in the initial call answer and fails to go back to active state even after re-negotiation with “sendrecv”.

Polycom UC Software 4.0.14 Resolved Issues

Category	Issue Number	Description
Calling	EN-70647	All VxWorks version 5.5 supported phone freezes while uploading logs to the boot server when the upload frequency is 3-4 seconds.
Configuration	EN-65058	Polycom SoundStation IP 5000 conference phone is unable to fetch the IP address from DHCP server during the software upgrade when the sub-option in Option 43 is incorrectly configured.
General	EN-89543	If the primary server responds with a 503 response, the phone sends delayed Refresh REGISTER to secondary server when the expiry timeout is less than 70 seconds.
General	EN-83605	When Polycom phone receives NOTIFY with terminated before the SUBSCRIBE response, the phone crashes/reboots due to a memory leak.
General	EN-58645	Polycom SoundStation IP conference phone does not offer all the codecs from previous Session Description Protocol (SDP) when resuming a call with `voIpProt.offerFullCodecListUponResume` parameter enabled.
General	EN-85672	When the primary server does not respond, the phone sends REGISTER with new call-id and Cseq 1 to the registered secondary server during fail back.
General	EN-81595	With the re-registration on failover (RRoFO) feature enabled and transport configured as TCPpreferred, the phone do not honor the 401 challenge with authentication header for UDP if TCP connection is not successful.
Hardware	EN-67943	Polycom SoundStation IP conference phone randomly reboots during DNS lookup.
Interoperability	EN-89106	Polycom phone prepends “sip:” while relaying SIPS-URI in Record-Route/Route header.
Interoperability	EN-81177	Polycom SoundStation IP conference phone do not handle “sips:” URL in Route header correctly when user name is present.
Localization	EN-65184	Polycom phones do not display the label properly if Unicode character value for HTML entities is beyond 127.
Network	EN-66337	Polycom SoundStation IP 7000 conference phones fail to send option 60 and 125 in every 7th DHCP request message.
Network	EN-92774	Polycom phone is unable to parse NAPTR response if NAPTR response contains encoded replacement string.
Provisioning	EN-87721	The subscription for server-based call forward and DND fails over TLS.
Provisioning	EN-81581	When the Polycom phone generates Certificate Signing Request (CSR), the phone does not upload a private key to the provisioning server.
Security	EN-121039 EN-121040	Fixed security vulnerabilities.

Polycom UC Software 4.0.14 Resolved Issues

Category	Issue Number	Description
User Interface	EN-84011	In GENBAND environments with an Oracle SBC, the phone reboots when a user presses the Hold, Transfer, or both soft keys at the same time during a call.
User Interface	EN-88155	Polycom VVX business media phone is unable to display Unicode text having Armenian characters.

Known Issues

The following table lists all known issues and suggested workarounds for Polycom UC Software version 4.0.14.

These release notes do not provide a complete listing of all known issues that are included in the software. Issues not expected to significantly impact customers with standard voice and video conferencing environments may not be included. In addition, the information in these release notes is provided as-is at the time of release and is subject to change without notice.

Known Issues

Category	Issue ID	Description	Workaround
Application	VOIP-46997	Camera brightness adjustment does not work between levels 3 to 6 on the Polycom VVX 1500 business media phone.	No workaround currently available.
Application	VOIP-66251	British Telecom Caller ID type is not correctly supported (applies to Polycom SoundStation Duo conference phone).	No workaround currently available.
Audio	EN-76703	While the Polycom RealPresence Group Series system connected to Polycom SoundStation IP 7000 conference phone is in video call, when the connection is lost and then restored again, the far-end audio randomly switches to the Polycom SoundStation IP 7000 conference phone instead of staying on the Polycom RealPresence Group Series system.	No workaround currently available.
Calling	VOIP-53514	H.264 calls to an Polycom HDX 9002 device using an MGC 50 Gateway using H.320 result in lip sync issues (applies to VVX 1500).	Set the call for transcoding on the MGC.
Calling	VOIP-54799	The Polycom VVX 1500 business media phones transmit H.264 QCIF video to Tandberg MXPs in H.323 calls.	Set the video bit rate on the Polycom VVX 1500 business media phone to 512 Kbps to avoid the issue.
Calling	VOIP-63609	Cannot answer a call using the speaker soft key when DND is enabled and `call.rejectBusyOnDnd` is set to zero (applies to SpectraLink 84xx).	No workaround currently available.
Configuration	EN-95403	The Polycom SoundStation IP 6000 and 7000 conference phones idle browser displays SSL connection timeout error after the restart when the idle browser is configured with HTTPS URL.	No workaround currently available.
General	VOIP-89407	In an MADN scenario, observed that the received call list on the phone is not getting updated when a MADN call is answered by a different destination.	No workaround currently available.

Known Issues

Category	Issue ID	Description	Workaround
Hardware	VOIP-37984	Enabling the Idle bit-map on Polycom SoundStation IP 330 and 320 conference phones causes the line key labels and dialed digits to be invisible.	Do not use the idle bit-map on 330/320 phones; instead, set `ind.idleDisplay.enabled=0`.
Network	VOIP-72211	An explicitly trusted Intermediate CA fails TLS verification when it is the issuer of a server certificate.	No workaround currently available.
Network	EN-96231	Phone authentication fails for the first time when using the EAP-TLS method for 802.1x authentication.	No workaround currently available.
Security	VOIP-37175	If configuration files are used to set the SNTP server address, date validity checking on CA certificates will be ignored for https provisioning.	Set the SNTP server address through the phone UI or use DHCP to inform the phone of the SNTP server address.
Security	VOIP-62482	Server certificate serial number is checked against the host name if the outbound proxy is configured.	No workaround currently available.
Software Updates	VOIP-70728	Software upgrade does not work if <partnumber>.xml file is not specified as a part of `upgrade.custom.server.url` configuration value.	Ensure the part-number.xml file is part of the `upgrade.custom.serverurl` configuration value.

Updates to Previous Software Releases

This section lists new, enhanced, and discontinued software features and capabilities in previous software releases.

What's New in Polycom UC Software 4.0.13

Upgrading to or Downgrading from Polycom UC Software 4.0.x

As of UC software 4.0.x, Polycom has changed the process of upgrading and downgrading software on the phones. To upgrade to UC software 4.0.x, you need to install the new Upgrader application. Once you have installed the Upgrader, you can install UC software 4.0.x.

To downgrade from UC software 4.0.x, you need first to install a new application called the Downgrader. Once you have installed the Downgrader, you can install an earlier UC software version.

For detailed instructions on how to upgrade and downgrade your UC software, see Polycom UC Software 4.0.x Upgrade and Downgrade Methods (Engineering Advisory 64731) on Polycom Support.

Introducing the Integrated Updater Application

In Polycom UC software 4.0.x, the Updater files and the UC software files are combined in a single, integrated software package. The Updater application is new for the UC software and replaces the BootROM application incorporated into previous software versions. The name Updater applies to UC software versions 4.0.0 or later, while the name BootROM continues to be used in UC software versions earlier than 4.0.0.

Considerations for Legacy Phones

Polycom UC software 4.0.13 does not include support for the Polycom® SoundPoint® IP 300, 301, 320, 330, 430, 500, 501, 600, 601 conference phones, Polycom® SpectraLink® 84xx series Wireless Telephone, and Polycom® SoundStation® IP 4000 conference phones. These phones are termed legacy products and are supported for critical issue fixes in SIP 3.2.x (Polycom SoundPoint IP 430 conference phone), SIP 3.1.x (Polycom SoundPoint IP 301, 320, 330, 501, 600, 601, 4000 conference phones), and SIP 2.1.x (Polycom SoundPoint IP 300, 500 conference phones). For assistance on provisioning methods that support a mix of legacy and current products, refer to *Maintaining Older Polycom Phones Beyond Their Last Supported Software Release (Technical Bulletin 35311)* on Polycom Support.

Supporting Earlier Software versions

For details on platform support for previous software versions, refer to the UC Software/SIP Software Release Matrix.

Managing Polycom® SoundStation® IP 7000 Conference Phones with Polycom® HDX® Integration

If your phone deployment includes Polycom SoundStation IP 7000 conference phones with Polycom HDX 4000/6000/7000/8000/9000 integration, use Recommended Software Versions for the Polycom SoundStation IP 7000 Conference Phones with Polycom HDX Systems table as a guide to select a recommended software version.

Recommended Software Versions for the Polycom SoundStation IP 7000 Conference Phones with Polycom HDX Systems

Polycom SoundStation IP 7000 conference phone Software Version	Polycom HDX Software Version
3.2.1 or 3.2.2 and BootROM 4.2.0	2.5.0.7, 2.5.0.8
3.2.3 and BootROM 4.2.2	2.6.0, 2.6.0.2, 2.6.1, 2.6.1.3
3.3.1 and BootROM 4.3.1	2.6.1.3, 3.0, 3.0.0.1, 3.0.0.2, 3.0.1, 3.0.2, 3.0.2.1, 3.0.3, 3.0.3.1
4.0.1 and Updater 5.0.1	3.0.3, 3.0.3.1
4.0.2 Rev B and Updater 5.0.1	3.0.4, 3.0.5, 3.1.0, 3.1.0.1
4.0.3 Rev F and Updater 5.0.1	3.1.2, 3.1.1.2, 3.1.1.3
4.0.4 and Updater 5.0.3	3.1.3
4.0.7 and Updater 5.0.4	3.1.6*
4.0.9 and Updater 5.0.8	3.1.9
4.0.10 and Updater 5.0.9	3.1.11
4.0.13 and Updater 5.0.13	3.1.11

Support is available for Polycom HDX version 3.1.2 and above only. Customers are encouraged to upgrade to Polycom HDX version 3.1.2 and above and the corresponding recommended UC Software suitable for the Polycom SoundPoint IP 7000 conference phone.

Managing Polycom® SoundStation® IP 7000 Conference Phones with Polycom® RealPresence® Group Series Integration

If your phone deployment includes Polycom SoundStation IP 7000 conference phones with Polycom RealPresence Group Series 300/500/700 integration, use Recommended Software Versions for the

Polycom SoundStation IP 7000 Conference Phones with Polycom RealPresence Group Series System table as a guide to select a recommended software version.

Recommended Software Versions for the Polycom SoundStation IP 7000 Conference Phones with Polycom RealPresence Group Series System

Polycom SoundStation IP 7000 Conference Phone Software Version	Polycom RealPresence Group Series System Software Version
4.0.2 Rev B and Updater 5.0.1	4.0.0, 4.0.0.1
4.0.3 Rev F and Updater 5.0.1	4.0.1, 4.0.2
4.0.3F and Updater 5.0.1	4.0.1, 4.0.2
4.0.4 and Updater 5.0.1	4.1.1, 4.1.1.1
4.0.5 and Updater 5.0.1	4.1.3,4.1.4, 4.1.5
4.0.7 and updater 5.0.4	4.2.0
4.0.7 and Updater 5.0.5	5.0.2
4.0.11 and Updater 5.0.11	5.1.2, 6.0.0, 6.0.1, 6.1.0, 6.1.1
4.0.13 and Updater 5.0.13	5.1.2, 6.0.0, 6.0.1, 6.1.0, 6.1.1, 6.1.2

Supporting the PinYin Input Mode (XT9 input)

Polycom UC software 4.0.13 supports Chinese character input mode known as PinYin. This feature is available on Polycom® VVX® 1500 business media phones and will require a feature license for activation. Refer to *Using PinYin to Input Chinese Characters on Polycom VVX Business Media Phones (FP 76554)* on Polycom Support to learn more.

Supporting the BroadSoft Hoteling Feature

Polycom UC software 4.0.13 supports Polycom SoundPoint IP 450, 550, 560, 650, and 670 conference phones for use with the BroadSoft Hoteling feature.

Supporting BroadSoft Premium Automatic Call Distribution (ACD)

Polycom UC software 4.0.13 supports Polycom SoundPoint IP 450, 550, 560, 650, and 670 conference phones and Polycom VVX 500 business media phones for use with the Broadsoft Premium ACD.

For detailed information about the ACD feature, see Using Premium Automatic Call Distribution for Call Centers (Feature Profile 76179) on Polycom Support.

Resolved Issues

The following issues have been resolved for the Polycom UC software 4.0.13 release.

Polycom UC Software 4.0.13 Resolved Issues

Category	Issue Number	Description
Calling	EN-32391	When Polycom RealPresence Group Series system attempts to dial multiple calls simultaneously, only one call will connect if the Polycom SoundStation IP 7000 conference phone is plugged in as a microphone.
Calling	EN-32324	The Polycom SoundPoint IP conference phone handsets reject MIME message without a Session Description Protocol (SDP) and sends a 415-unsupported media type error message.
Calling	EN-32607	Calls to Polycom SoundStation IP conference phones do not appear on the missed calls list.
Calling	EN-30015	When the configuration parameter `voIpProt.SIP.RFC3261TimerI` is set to 1, the Polycom VVX business media phone sends 481 response code to second INVITE after rejecting initial INVITE message with 488 response even when the transaction timeout is zero, as per specification for Transmission Control Protocol (TCP) transport.
General	EN-32375	The Polycom VVX business media phone's SSL Client Hello publishes incorrect time for the GMT UNIX Time.
Network	EN-32248	The Polycom SoundStation IP 6000 and 7000 conference phones are not honoring the Maximum Segment Size (MSS) received in TCP SYN packet during the TCP handshake.
Security	EN-41471	Resolved important security vulnerabilities.
Security	EN-35538	The Polycom phone fails to login when registered to server using Built-in certificate.
User Interface	EN-32599	If the Diffie-Hellman (DH) key length is more than 128 bytes, client applications on the Polycom VVX business media phones take longer time to establish a connection.
User Interface	EN-42373	Call Forward No Answer (CFNA) ring count field defaults to incorrect value when the server call forwarding is disabled in the device and enabled on the server.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC software 4.0.13 release. For more information on using configuration parameters to enable or disable features, see the Administrator Guide for Polycom UC software for your software release available on the Polycom Voice Support site.

Configuration File Enhancements for Polycom UC Software 4.0.13

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.browser`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and phone browser. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The microbrowser restarts when there is a change in the browser TLS protocol or TLS cipher settings, and the last web page displayed is not restored. The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.ldap`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Lightweight Directory Access Protocol (LDAP). The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No

Configuration File Enhancements for Polycom UC Software 4.0.13 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.sip`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and SIP signaling. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.sopi`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and SOPI. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.webServer`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Web Server. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No

Configuration File Enhancements for Polycom UC Software 4.0.13 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.xmpp`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and XMPP. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.exchangeServices`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and exchanges services. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
`device.cfg`, `site.cfg`	`device.sec.TLS.protocol.syslog`	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Syslog. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No

Configuration File Enhancements for Polycom UC Software 4.0.13 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.sec.TLS.protocol.prov	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and provisioning. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The permitted values for Polycom SoundStation IP 6000 and 7000 conference phones are TLSv1_0 (default) SSLv2v3	No
device.cfg, site.cfg	device.sec.TLS.protocol.dot1x	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and 802.1x authentication. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3, TLSv1_0, TLSv1_1 and TLSv1_2 The permitted value for Polycom SoundStation IP 6000 and 7000 conference phones is TLSv1_0 (default)	No
sip-interop.cfg	tone.dtmf.rfc2833Payload_OPUS	Sets the Dual Tone Multi Frequency (DTMF) payload required to use the Opus codec. 126 (default) 96 - 127	Yes
sip-interop.cfg	voIpProt.SIP.RFC3261TimerI	0 (default) - Timer I for reliable transport will be fired at five seconds. This parameter does not cause any change for unreliable transport. 1 - Timer I for reliable transport will be fired at zero seconds.	No

Configuration File Enhancements for Polycom UC Software 4.0.13 (continued)

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.cipherList`	Specifies the cipher list for all applications except web server. ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!MD5:!RC4:@STRENGTH (default) String (Maximum of 1024 characters)	No
`site.cfg`	`sec.TLS.webServer.cipherList`	Specifies the cipher list for web server. ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!AES256-SHA:!AES128-SHA:!MD5:!RC4:@STRENGTH (default) String (Maximum of 1024 characters)	No

What's New in Polycom UC Software 4.0.12

The Polycom UC software 4.0.12 release includes the following new features and enhancements:

Important Upgrade Notes and Considerations in Polycom UC Software 4.0.12

This section contains important notes on Polycom hardware and software.

Configuration Files

Since UC software 3.3.0, Polycom has simplified and extensively modified the configuration files, their respective parameters and defaults, and the provisioning methods. Some of the software updates in UC software 4.0.12 may not be compatible with configuration files from software releases prior to UC software 3.3.0. If you are updating to the UC software 4.0.12 from a release prior to 3.3.0, Polycom recommends reading Technical Bulletin 60519: Simplified Configuration Enhancements in Polycom® UC Software 3.3.0 before you deploy the updated software.

If you use a Polycom UC software version prior to 3.3.0 and want to use a Polycom® SoundStructure® VoIP Interface phone, Polycom® VVX® 500 business media phone or Polycom® SoundStation® Duo conference phone, you need to update your existing configuration files as described in Technical Bulletin 60519.

Upgrading to or Downgrading from Polycom UC Software 4.0.x

As of UC software 4.0.x, Polycom has changed the process of upgrading and downgrading software on the phones. To upgrade to UC software 4.0.x, you need to install the new Upgrader application. Once you have installed the Upgrader, you can install UC software 4.0.x.

To downgrade from UC software 4.0.x, you need first to install a new application called the Downgrader. Once you have installed the Downgrader, you can install an earlier UC software version.

For detailed instructions on how to upgrade and downgrade your UC software, see Polycom UC Software 4.0.x Upgrade and Downgrade Methods (Engineering Advisory 64731) on Polycom Support.

Introducing the Integrated Updater Application

In Polycom UC software 4.0.x, the Updater files and the UC software files are combined in a single, integrated software package. The Updater application is new for the UC software and replaces the BootROM application incorporated into previous software versions. The name Updater applies to UC software versions 4.0.0 or later, while the name BootROM continues to be used in UC software versions earlier than 4.0.0.

Considerations for Legacy Phones

Polycom UC software 4.0.12 does not include support for the Polycom® SoundPoint® IP 300, 301, 320, 330, 430, 500, 501, 600, 601 conference phones, Polycom® SpectraLink® 84xx series Wireless Telephone, and Polycom® SoundStation® IP 4000 conference phones. These phones are termed legacy products and are supported for critical issue fixes in SIP 3.2.x (Polycom SoundPoint IP 430 conference phone), SIP 3.1.x (Polycom SoundPoint IP 301, 320, 330, 501, 600, 601, and 4000 conference phones), and SIP 2.1.x (Polycom SoundPoint IP 300 and 500 conference phones). For assistance on provisioning methods that support a mix of legacy and current products, refer to *Maintaining Older Polycom Phones Beyond Their Last Supported Software Release (Technical Bulletin 35311)* on Polycom Support.

Supporting the PinYin Input Mode (XT9 input)

Polycom UC software 4.0.12 supports Chinese character input mode known as PinYin. This feature is available on Polycom® VVX® 1500 business media phones and will require a feature license for activation. Refer to *Using PinYin to Input Chinese Characters on Polycom VVX Business Media Phones (FP 76554)* on Polycom Support to learn more.

Supporting the BroadSoft Hoteling Feature

Polycom UC software 4.0.12 supports the following Polycom phones for use with the BroadSoft Hoteling feature: Polycom SoundPoint IP 450, 550, 560, 650, and 670 conference phones.

Supporting BroadSoft Premium Automatic Call Distribution (ACD)

Polycom UC software 4.0.12 supports the following Polycom phones for use with the BroadSoft Premium ACD: Polycom SoundPoint IP 450, 550, 560, 650, and 670 conference phones and Polycom VVX 500 business media phone.

For detailed information about the ACD feature, see Using Premium Automatic Call Distribution for Call Centers (Feature Profile 76179) on Polycom Support.

Resolved Issues

The following issues have been resolved for the Polycom UC software 4.0.12 release.

Polycom UC Software 4.0.12 Resolved Issues

Category	Issue Number	Found in Release	Description
Calling	VOIP-121789	UCS 4.0.11	The Polycom SoundPoint IP conference phones do not switch back to the local ring tone during blind transfers.
Configuration	VOIP-118008	UCS 4.0.11	The changed settings in the MAC-web.cfg file of the phone does not get uploaded to the path provided in the OVERRIDES_DIRECTORY through Web Configuration Utility.
General	VOIP-118307	UCS 4.0.10	Special characters such as '() {} #' are not supported in the line authentication password field of the phone while entering the user login credentials.
General	VOIP-111907	UCS 4.0.10	When dialing a contact from the corporate directory, if the corporate directory is scrolled multiple times, the Polycom SoundPoint IP 560 and 670 conference phones might reboot.
General	VOIP-122934	UCS 5.2.0, UCS 4.1.1, UCS 4.0.11	If the compatibility view is not set on Polycom SoundPoint IP conference phone, the web pages are not displayed on Internet Explorer 11.
General	VOIP-120400	UCS 4.0.9	The Polycom SoundPoint IP 650 conference phones restart occasionally in an irregular pattern.
General	VOIP-114168	UCS 4.0.8	When dialing a contact selected from the corporate directory search, the Polycom SoundPoint IP450 conference phone restarts automatically within two seconds.
General	VOIP-114347	UCS 4.0.8	When updating the configuration from the phone menu, if `dir.local.readonly` parameter is set to 1 and provisioning server is unreachable, the phone restarts automatically.

Polycom UC Software 4.0.12 Resolved Issues

Category	Issue Number	Found in Release	Description
General	VOIP-123101	UCS 4.0.11	During a call, if the `voIpProt.SIP.musicOnHold.uri` is configured, the Polycom SoundPoint IP conference phone does not send the SDP information in the session refresh INVITE.
General	VOIP-120795	UCS 4.0.11	In a simultaneous ring environment, when an incoming call is received, the call drops.
Network	VOIP-117354	UCS 4.0.10	The phone displays a "Request Failed" error on a successful blind transfer when the Session Description Protocol (SDP) origin session-id and session-version does not match with the previous INVITE request.
Network	VOIP-118668	UCS 4.0.8	During boot-up, if the switch takes time to respond to the phone as a part of the Link Layer Discovery Protocol (LLDP), the Polycom SoundPoint IP conference phone might not pick up the correct VLAN ids from LLDP.
Network	VOIP-124271	UCS 4.0.11	When Link Layer Discovery Protocol (LLDP) packets receive empty port description with Type Length Value being 0, the phone ignores VLAN.
Network	VOIP-122825	UCS 5.4.5, UCS 4.0.4, UCS 4.0.11	When sending Session Description Protocol (SDP) offer in a transfer scenario, if an INVITE without SDP is received, the `sec.srtp.offer` is not considered.
Network	VOIP-121141	UCS 5.4.1, UCS 5.1.3, UCS 5.1.3, UCS 4.0.8	When Maximum Transmission Unit (MTU) is reduced on VVX 1500 phone, high load occurs on Session Border Controller (SBC) due to excessive fragmented packets.
Provisioning	VOIP-120547	UCS 4.0.9	The Polycom SoundPoint IP 6000 conference phone stops responding after a period of time when a Multi Line Telephone Systems (MTLS) connection is used for provisioning periodically.
Provisioning	VOIP-125679	UCS 4.0.8, UCS 4.0.10	The Polycom SoundPoint IP 5000 conference phones send a subscribe message every 30 seconds over the TLS even when the subscription has not expired.
Security	VOIP-122626	UCS 5.4.5	All GlobalSign Root R3 CA certificates are not shown to be supported.
Security	VOIP-124676	UCS 5.5.1	Polycom VVX business media phones do not allow HTTPS provisioning as Let’s Encrypt certificate is not on the list of trusted certificates.
Security	VOIP-121529	UCS 5.5.1	When generating a Certificate Signing Request (CSR) on Polycom VVX business media phones, the phone's private key is sent to the provisioning server along with the CSR.
User Interface	VOIP-123568	UCS 4.0.11	During an incoming call, there is no option to disable the call center status information being displayed.

Polycom UC Software 4.0.12 Resolved Issues

Category	Issue Number	Found in Release	Description
User Interface	VOIP-120740	UCS 4.0.11	When Line Registration is disabled on the phone, a "Line Unregistered" pop-up text message and a warning icon display. Set `up.warningLevel` to 2, available in releases 4.0.12 and later, to suppress the warnings.
Web Interface	VOIP-120618	UCS 4.0.10	The Web Server Configuration feature cannot be enabled on the Polycom SoundStation IP conference phones when disabled through configuration file.
Web Interface	VOIP-126409	UCS 4.1.1	Polycom SoundStation IP 5000 conference phone has a memory leak issue that results in a medium-security vulnerability (CVSSv3 score = 5.4) in the web interface. An independent security researcher, Francis Alexander, has reported this vulnerability to Beyond Security's SecuriTeam Secure Disclosure program, who in turn alerted Polycom.

Configuration File Enhancements

The following table lists the configuration file enhancements that include new or changed parameters for this Polycom UC software 4.0.12.

Configuration File Enhancements

Parameter Template	Permitted Values
`feature.callCenterCallInformation.enable`	1 (default) - Call Center status information displayed while receiving an incoming call. 0 - Does not display Call Center status information. The Call Center information can be viewed by pressing Call Info soft key.
`up.warningLevel`	0 (default) - Displays warning icon and a pop-up message for all warnings. 1 - Displays a warning icon and pop-up messages only for critical warnings. 2 - Does not display any warning icon or pop-up message.
`call.switchToLocalRingbackWithoutRTP`	0 (default) - The phone does not switch back to local ringback if Real-Time Transport Protocol (RTP) is stopped. 1 - The phone switches back to local ringback if RTP is stopped.

Configuration File Enhancements

Parameter Template	Permitted Values
`device.sec.TLS.constTimeFlag.enabled`	1 (default) - Enables constant time flag for BIG NUM calculation during Multi Line Telephone Systems (MLTS) connection. 0 - Disables constant time flag.
`voIpProt.SIP.renewSubscribeOnTLSRefresh`	1 (default) - For an as-feature-event, the SUBSCRIBE message is sent along with the RE-REGISTER when Transport Layer Security (TLS) breaks. 0 - The SUBSCRIBE and RE-REGISTER messages is sent at different times.
`device.net.lldpFastStartCount`	Specifies the number of consecutive Link Layer Discovery Protocol (LLDP) packets the phone sends at the time of LLDP discovery. 5 (default) 3 to 10 LLDP packets are sent every seconds during this extended discovery period.
`net.lldp.extendedDiscovery`	Specifies the duration of time that LLDP discovery continues after sending the number of packets defined by the parameter `device.net.lldpFastStartCount`. 0 (default) 0 to 3600 LLDP packets are sent every 5 seconds during this extended discovery period.

Important Upgrade Notes and Considerations in UC Software 4.0.11 Rev K

Important Update

Impacted: All Polycom® SoundPoint® IP conference phone, Polycom® SoundStation® IP conference phone, and Polycom® VVX® business media phone.

Details: This release includes a critical fix (VOIP-116371) that corrects a problem with Polycom phones manufactured with a serial number or MAC address in the range 64167F as opposed to 0004F2. Without this fix, customers using Polycom phones with the new serial number range will see impaired performance.

Recommendation: Polycom recommends that this release be used for all phones going forward. Phones with older serial numbers or MAC addresses will continue to work correctly with this build with no impact.

What's New in Polycom UC Software 4.0.11 Rev K

This release has a new software change and resolved issues from previous releases.

New Features in 4.0.11 Rev K

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.11 Rev K beside their respective Polycom tracking identification numbers.

New or Enhanced Features

The Polycom UC software 4.0.11 Rev K software version includes the features and functionality of version 4.0.11, with the addition of important field fixes.

New Display Component on Polycom® VVX® 500 Business Media Phones

Polycom VVX 500 business media phones manufactured as of May 2017 are being shipped with the new display component from a secondary component vendor. When the Polycom VVX 500 business media phone encounters an incompatible version of UC software on the provisioning server that does not support the new component, the phone installs the UC software and you may experience a flicker. This release includes a software change that makes it compatible with the new display component.

What's New in Polycom UC Software 4.0.11

This release has a new software change and resolved issues from previous releases.

New Features in 4.0.11

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.11 beside their respective Polycom tracking identification numbers.

New or Enhanced Features

The 4.0.11 software version includes the features and functionality of version 4.0.10, with the addition of important field fixes.

Resolved Issues

The following issues have been resolved for the Polycom UC software 4.0.11 release.

Polycom UC Software 4.0.11 Resolved Issues

Category	Issue No.	Found in Release	Description
Calling	VOIP-115888	4.0.9	Outbound calls from the Multiple Appearance Directory Number (MADN) lines now display the called number.
Calling	VOIP-116267	4.0.9	Phones now use the contact uniform resource identifier (URI) and Tel URI in the request line of the BYE message. When the `reg.1.telUri` parameter is disabled, the phone uses the contact URI and the call ends. When the `reg.1.telUri` parameter is enabled, the phone uses the Tel URI and the call ends.
Calling	VOIP-117314	5.4.2	Ringer and paging, hands-free and Push-to-talk (PTT), and handset and headset loudness in phones are now boosted up to full scale when a low power signal is received. For more details on the configuration parameters added for this issue, see Polycom UC Software 4.0.11 Configuration File Enhancements.
Calling	VOIP-117387	4.0.9	Outbound calls from the MADN lines now show the called number without any issue.
Cisco	VOIP-108644	4.0.8	Polycom SoundStation IP 6000 conference phone no longer displays the “DHCP failed” message when connected to a Cisco® Catalyst® 2960-X switch with ports set to auto negotiate.
Configuration	VOIP-115998	4.0.8 4.0.9	Polycom SoundStation IP 550 conference phones now send network address translation (NAT) keep alive messages to the call server without any issue.
Configuration	VOIP-117275	4.0.10	Polycom SoundStation IP 7000 conference phone performance has been improved by increasing the available free space.
Functionality	VOIP-112880	4.0.9	If the phone auto-answers a click-to-dial call, the phone plays the correct ringtone and not the reboot ringtone.
Functionality	VOIP-116990	4.0.10	Phone works fine and no longer causes issues during an incoming call while the phone is under any of the menus from the Home Screen and the call is ended by the far end before the call is answered.
General	VOIP-113784	4.0.9	Forwarding a call manually no longer causes any issues even if feature.urlDialing.enabled parameter is disabled.

Polycom UC Software 4.0.11 Resolved Issues

Category	Issue No.	Found in Release	Description
General	VOIP-114780	4.0.8 4.0.9 4.0.10	During an incoming page, if the volume is increased in the phone, the phone now retains the increased paging volume.
General	VOIP-117261	5.4.4 4.0.10	During an outage, the phone now falls back to the secondary server and sends the SUBSCRIBE message for registration and presence.
Security	VOIP-113939	5.4.1	The following two StartCom Root CA certificates are added to the Polycom trusted CA bundle: • StartCom Certification Authority • StartCom Certification Authority G2
User Interface	VOIP-114192	4.0.9	The Forward After Rings UI no longer gets over lapped with its value configured.

Configuration File Enhancements

The following table includes configuration file enhancements for this release.

Polycom UC Software 4.0.11 Configuration File Enhancements

Parameter	Permitted Values	Default
`call.shared.preferCallInfoCID`	0 or 1	0
If the value is 0, the Caller-ID information received in the 200 OK status code will not be ignored if the NOTIFY message received with caller information includes display information. If the value is set to 1, the Caller-ID information received in the 200 OK status code will be ignored if the NOTIFY message received with caller information includes display information.		
`voice.handsetHeadset.rxdg.offset`	9 to -12	0
Offsets the RxDg range of the handset and headset by the specified number of decibels.		
`voice.ringerPage.rxdg.offset`	9 to -12	0
Offsets the RxDg range of the ringer and hands-free Page by the specified number of decibels.		
`voice.handsfreePtt.rxdg.offset`	9 to -12	0
Offsets the RxDg range of the hands-free and hands-free Push-to-Talk (PTT) by the specified number of decibels.		
`voIpProt.SIP.newCallOnUnRegister`	0 or 1	1

Polycom UC Software 4.0.11 Configuration File Enhancements

Parameter	Permitted Values	Default
If the value is set to 1, the phone generates a new caller-ID from the “From” tag while re-registering. If the value is set to 0, the phone does not generate a new caller-ID from the “From” tag while re-registering.		
`voIpProt.SIP.looseContact`	0 or 1	0
If the value is 0, the ephemeral port is added to the contact header in the Transport Layer Security connection (TLS) case. If the value is set to 1, the port parameter will not be added to the contact header message or SIP messages.		
`call.autoAnswer.playTone.enable`	0 or 1	1
If call.autoAnswer.playTone.enable is enabled, the auto-answer tone is played. If call.autoAnswer.playTone.enable is disabled, the auto-answer tone is not played.		

Updates to Polycom UC Software 4.0.10

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.10 beside their respective Polycom tracking identification numbers.

Configuration File Enhancements

The following table includes configuration file enhancements for this release.

Polycom UC Software 4.0.10 Configuration File Enhancements

Parameter	Permitted Values	Default
`voIpProt.SIP.dtmfViaSignaling.rfc2976.nonLegacyEncoding`	0 or 1	1
If 1, the phone sends " * " and " # " for " * " and " # " when relaying DTMF digits using the SIP INFO method. If 0, the phone sends "10 " and "11" for " * " and " # " when relaying DTMF digits using the SIP INFO method.		
`sec.TLS.profile.webServer.cipherSuiteDefault`	0 to 1	1
If 0, use the custom cipher suite for web server profile. If 1, use the default cipher suite.		
`voIpProt.server.x.failOver.failBack.mode`	newRequests,DNSTTL, registration, duration	duration
The default value for this parameter is changed to “duration” from “newRequests”.		
`voIpProt.SIP.outboundProxy.failOver.failBack.mode`	newRequests,DNSTTL, registration, duration	duration
The default value for this parameter is changed to “duration” from “newRequests”.		

Polycom UC Software 4.0.10 Configuration File Enhancements

Parameter	Permitted Values	Default
`reg.x.outboundProxy.failOver.failBack.mode`	newRequests,DNSTTL, registration, duration	duration
The default value for this parameter is changed to “duration” from “newRequests”.		
`reg.x.server.y.failOver.failBack.mode`	newRequests,DNSTTL, registration, duration	duration
The default value for this parameter is changed to “duration” from “newRequests”.		
`sec.TLS.LDAP.strictCertCommonNameValidation`	0 or 1	1
If 0, the Common Name validation of the certificate used for LDAPS (LDAP over TLS) connection is skipped. If 1, the Common Name of the certificate used for LDAPS (LDAP over TLS) connection is validated. Note: Changing this parameter value causes the phone to restart.		
`softkey.feature.silence`	0 or 1	1
If 0, the Silence softkey is disabled. If 1, the Silence softkey is enabled.		
`sec.TLS.cipherList`	String	RSA:!EXP:!LOW:!NULL:!MD5:!RC4:@STRENGTH
For this parameter, the default value is changed from "RSA:!EXP:!LOW:!NULL:!MD5:@STRENGTH" to "RSA:!EXP:!LOW:!NULL:!MD5:!RC4:@STRENGTH".		

Resolved Issues

The following issues have been resolved for the Polycom UC software 4.0.10 release.

Polycom UC Software 4.0.10 Resolved Issues

Category	Issue No.	Found in Release	Description
Audio	VOIP-111421	4.0.9	Repeat pressing a DTMF digit by the callee does not causes packet loss, which would have decreased the Mean Opinion Score (MOS) in the Voice Quality monitoring system. The MOS remains constant as expected.
Calling	VOIP-102676	4.0.7	The 3CX call park feature with TCP trunk does not cause one-way audio and the phone is able to unpark call.
Calling	VOIP-102748	4.0.7	Phone uses the blind transfer behavior from EFK soft keys, and sends a HOLD before the REFER message.
Calling	VOIP-104132	4.0.9	If an emergency server is not configured, emergency calls are routed through the registrar server and any calls dialed to configured emergency numbers are treated as emergency calls.

Polycom UC Software 4.0.10 Resolved Issues

Category	Issue No.	Found in Release	Description
Contact Directory	VOIP-110539	4.0.8	The phone does not reboot if a contact is not selected and dialed within two seconds of receiving search results in the Corporate Directory search.
General	VOIP-101787	4.0.5	Phone scrolls the TO and FROM headers in the caller ID in BLF feature with Genband.

General	VOIP-102240	5.3.0	The default for the following parameters is changed from “newRequests” to “duration”.
			`voIpProt.server.x.failOver.failBack.mode`
			`voIpProt.SIP.outboundProxy.failOver.failBack.mode`
			`reg.x.outboundProxy.failOver.failBack.mode`
			`reg.x.server.y.failOver.failBack.mode`

General	VOIP-102354	5.4.0	Mismatch between file system versions due to a flash corruption is fixed. The phone now starts successfully without displaying the Fix soft key.
General	VOIP-102415	4.0.10	A new configuration parameter is added to support the draft-kaplan-dispatch-info-dtmf-package draft for relaying DTMF digits.
General	VOIP-102488	4.0.8	IP 5000 software acquires DHCP address successfully when option 43 uses HEX value.
General	VOIP-103424	4.0.8	When a call server sends a NOTIFY message to the phone with "State : TERMINATED", the phone does not send additional SUBSCRIBE messages once the call is canceled.
General	VOIP-105621	4.0.8	Enabling SSLv3 on the LDAP server and disabling the same on the phone does not cause phone freeze or power off after one day from reboot.
General	VOIP-108864	4.0.9	The phone sends a CANCEL request if the call is ended, even if there has been no response from the server for the original INVITE request.
General	VOIP-109628	4.0.4	The overall Mean Opinion Score (MOS) is accurate even when there are several SSRC changes, for example, when codec changes. The phone triggers a VQMon report when SSRC change is reported by DSP.
General	VOIP-110514	4.0.9	Common Name validation of the certificate used for LDAPS (LDAP over TLS) connection is configurable through configuration file, phone interface, and Web UI.
Inter-operability	VOIP-102162	4.0.9	Improved Polycom® VVX® 1500 business media phone integration with RPRM.

Polycom UC Software 4.0.10 Resolved Issues

Category	Issue No.	Found in Release	Description
Network	VOIP-106112	4.0.8	Phone does not broadcast SIP and RTP packages to PC port and prevents unwanted network load.
Provisioning	VOIP-106847	4.0.8	Phones do not get into a reboot loop when provisioning from an FTP server that has a contact directory with more than 350 contacts.
Provisioning	VOIP-109272	4.0.9	For a call number that ends in “#” or “*”, the phone sends “user=phone” in the invite message.
Security	VOIP-109244	4.0.9	Denial of service vulnerabilities in OpenSSL are fixed.
Security	VOIP-109284	5.4.1	RC4 encryption ciphers are not supported on TCP port 443.
Security	VOIP-111442	4.0.10	Cipher support for the web server is configurable.
User Interface	VOIP-101549	4.0.8	For Polycom® SoundStation® IP 7000 conference phone, the Automatic Gain Control or Noise Suppression sub menus under the administration menu are removed.
User Interface	VOIP-101667	4.0.9	Phones configured to use the custom cipher suite display the cipher list correctly.
User Interface	VOIP-108496	4.0.9	Polycom VVX 1500 business media phone displays the callee name in the Called List when the callee does not answer the call.
User Interface	VOIP-109005	4.0.8	When the Silence softkey is enabled, phone displays this softkey when the phone is ringing. Pressing this key silences the ringing phone. Note: The Silence Softkey is applicable only for Polycom® SoundPoint® IP 650, 670, and 450 conference phones, Polycom VVX 500 and 1500 business media phones and Polycom SoundStation IP 7000 conference phone.
User Interface	VOIP-109418	4.0.9	Phone does not display “Unknown” for incoming call ID unless the calling party address is null. For all other cases, phone displays the destination address for incoming call ID.
User Interface	VOIP-111325	5.4.0	Enable ease of install by customizing Quick Setup to show only username and password fields.

Updates to Polycom UC Software 4.0.9

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.9 beside their respective Polycom tracking identification numbers.

Resolved Issues

The following issues have been resolved for the Polycom UC software 4.0.9 release.

Polycom UC Software 4.0.9 Resolved Issues

Category	Issue No.	Found in Release	Description
Audio	VOIP-92804	4.0.5	To resolve a problem that caused audio issues on the Polycom® SoundStation® Duo conference phone, support has been discontinued for using the Spectralink proprietary G.726QI codec for PTT/Paging. Users should use G.711-mu or G.722 for PTT/Paging instead.
Calling	VOIP-97987	4.0.5	The default value of the parameter `up.simplifiedSipCallInfo` is now correctly set to '1'.
Calling	VOIP-101500	4.0.8	A problem has been resolved that caused the phone to send DTMF characters "*" and "#" incorrectly.
Contact Directory	VOIP-101462	4.0.6	A problem was resolved that prevented a Polycom® VVX® 1500 business media phone from saving directory entries locally when it is registered to a Polycom® RealPresence® Resource Manager system.
General	VOIP-100999	4.0.8	Several unused configuration parameters were removed from the cfgParamDef.xml file
General	VOIP-100887	4.0.8	New parameter `device.net.lldpCapabilitiesRequired.set` has been added for use when the LLDP packet with system capabilities should be set to 0x0000.
General	VOIP-100310	4.0.8	New parameter `ptt.pageMode.group.x.allowReceive` was added to allow the phone to send a page to an unsubscribed group.
General	VOIP-97556	4.0.7	An issue was resolved that caused the Polycom® SoundStation® IP 7000 conference phone to send incorrect information about the software version running on the system. The correct information can be viewed on Polycom® RealPresence® Group Series systems running software versions 5.x and higher.
General	VOIP-97556	4.0.7	A problem with handling the MWI NOTIFY message has been resolved.
User Interface	VOIP-98141	4.0.8	When you do a quick search in the Genband GAB, the Submit soft key does not display correctly.

Polycom UC Software 4.0.9 Resolved Issues

Category	Issue No.	Found in Release	Description
User Interface	VOIP-100146	4.0.8	A problem was resolved that caused SoundPoint systems to display soft keys incorrectly when the flexible line key feature is enabled.
VOIPUser Interface	VOIP-98086	4.0.8	When you edit a directory entry and save changes, the Save soft key now clears the screen correctly.

Configuration File Enhancements

The following table includes configuration file enhancements for this release.

Polycom UC Software 4.0.9 Configuration File Enhancements

Parameter	Permitted Values	Default
`ptt.channel.x.allowReceive`	0 to 1	1
If 0, disables receiving PTT calls on the specified channel. If 1, allows receiving PTT calls on the specified channel.		
`device.net.lldpCapabilitiesRequired.set`	0 to 1	0
If 0, the phone ignores LLDP packets with System Capabilities as 0x0000. If 1, the phone uses LLDP packets with System Capabilities as 0x0000.		
`ptt.pageMode.group.x.allowReceive`	0 to 1	1
If 0, disables receiving page calls on the specified group. If 1, allows receiving page calls on the specified group.		

Updates to Polycom UC Software 4.0.8

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.8 beside their respective Polycom tracking identification numbers.

New or Enhanced Features

The version 4.0.8 software version includes the features and functionality of 4.0.7, with the following additions.

- You can now configure Exchange Calendar alert tones on Polycom® VVX® business media phones. (VOIP-96381)

  Administrators can customize Exchange Calendar alert tones using configuration files only. The parameter `exchange.meeting.reminderType` is an existing parameter that has been updated for the new customization functionality. (Related to VOIP-96381)
- You can now configure the Message Waiting Indicator (MWI) alert tone. (VOIP-94741)

  Administrators can customize MWI tones using configuration files only. The parameter `mwi.reminder.enable` is a new parameter that has been added.

- You can now configure whether applications can use the SSLv3 or SSLv2 protocols. (VOIP-97280)

  Administrators can customize use of these protocols using the parameter `device.sec.TLS.SSLv2v3.enabled`, a new parameter added with this release.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to Polycom UC software 4.0.8 configuration file parameters.

Polycom UC Software 4.0.8 Configuration File Enhancements

Parameter	Permitted Values	Default
`call.advancedMissedCalls.addToMissedList`	0 or 1	0
If 1, remotely handled calls are logged as missed calls. The parameter call.advancedMissedCalls.enabled must also be set to 1. The parameter call.advancedMissedCalls.addToRecievedList is ignored. If 0, remotely handled calls are not logged as missed calls.		
`call.doNotPlayLocalOnProvResponseSdp`	0 or 1	0
If 1, the local ringback tone does not play after 183 with SDP. If 0, the local ringback tone plays immediately after 183 with or without SDP.		
`device.sec.TLS.SSLv2v3.enabled`	0 or 1	0
If 1, applications can use SSLv3 (or SSLv2) protocol. Note that some applications disable both SSLv2 and SSLv3, regardless of this setting. If 0, applications cannot use SSLv3 (or SSLv2) protocol.		
`exchange.meeting.reminderType`	0 - 2	2
If 2, all calendar reminders are audible and visual. If 1, the first reminder is audible, followed by silent visual reminders If 0, all reminders are silent visual reminders		
`mwi.reminder.enable`	0 or 1	0
If 1, a Message Waiting Indicator (MWI) tone plays each time the phone receives an MWI update. If 0, no MWI tone is played.		
`tcpIpApp.keepalive.tcp.sip.persistentConnection.enable`	0 or 1	0
If 0, the TCP Socket connection is closed after 1 minute. When the phone sends a new SIP message, a new connection is opened. If 1, the TCP Socket connection remains open indefinitely.		
`voIpProt.offerFullCodecListUponResume`	0 or 1	1

Polycom UC Software 4.0.8 Configuration File Enhancements

Parameter	Permitted Values	Default
If 1, the phone offers the full codec list in the resume offer. If 0, the phone does not send the full codec list in the resume offer.		

Documentation Corrections

This section includes updates to correct errors inadvertently included in the *UC Software Administrator Guide 4.0.5*. You can find this guide at support.polycom.com.

Omitted Parameter

The following parameter was omitted from the previous version of the *UC Software Administrator Guide*.

Parameter	Permitted Value	Default	Description
`tcpIpApp.sntp.retryDnsPeriod`	60 – 2147483647 seconds	86400	Sets 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 configured retry period but 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 retry 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.

Incorrect Default Values Listed for <device/> Parameters

The default values of the <device/> parameters vary according to the region to which the phone is shipped. The *UC Software Administrator Guide 4.0.5* incorrectly lists the default values for the <device/> parameters as “null”. Instead, the document should not list default values for the <device/> parameters. (VOIP-95919, VESC-4789)

Updates to Polycom UC Software 4.0.7

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.7 beside their respective Polycom tracking identification number.

New or Enhanced Features

There are no new or enhanced features for this release.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to Polycom UC software 4.0.7 configuration file parameters.

Parameter	Permitted Value	Default	Description
`tcpIpApp.sntp.retryDnsPeriod`	60 – 86400 seconds	60	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 retry 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.

Updates to Polycom UC Software 4.0.6

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.6 beside their respective Polycom tracking identification number.

New or Enhanced Features

87581/91102 Added support for the improved touchscreen driver (*applies to Polycom® VVX® 500 business media phone*).

Configuration File Enhancements

There are no changes to the configuration file parameters.

Updates to Polycom UC Software 4.0.5

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.5 beside their respective Polycom tracking identification number.

New or Enhanced Features

83102 Added support for Multiple Appearance Directory Number-Single Call Appearance (MADN-SCA) for Shared Line Appearance Scenarios.

87263/72669 Added support for Global Address Book (GAB).

87264 Added support for Personal Address Book (PAB).

87029/88933 The Message Waiting Indicator (MWI) LED light no longer blinks when the phone is in power-saving mode and has not received any new messages.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to Polycom UC software 4.0.5 configuration file parameters.

Polycom UC Software 4.0.5 Configuration File Enhancements

Parameter	Permitted Value	Default	Description
`Feature.corporateDirectory.alt.enabled`	0 or 1	0	Enables the GAB functionality.
`Sec.TLS.SOPI.strictCertCommonNameValidation`	0 or 1	1	Determines certification validation. When set to 0, there is no certification validation needed and any certification is accepted and trusted.
`reg.x.bargeInEnabled`	Maximum line number registered	0	Determines whether a user can barge in on an active call.
`sec.TLS.SOPI.cipherList`	String (max of 1024 characters)	NoCipher	Defines the cipher list for SOPI transactions.
`dir.local.serverFeatureControl.method`	GENBANDSOPI	<Blank>	Sets GENBANDSOPI to synchronize to the server. If this is not set, no synchronization is done and the standard contact directory is used.
`dir.loca` genband `genbandl.serverFeatureControl.reg`	1 to Maximum line number registered	1	Determines the line that is used to obtain PAB information. Does not change how numbers are dialed. Does not require a reboot –automatically updates at runtime
`msg.mwi.x.led`	0 or 1	1	Enables and disables the LED Message Waiting Indicator on receiving a new message
`mwi.backLight.disable`	0 or 1	0	If set to 1, disables backlight on new voice message arrival.

Polycom UC Software 4.0.5 Configuration File Enhancements

Parameter	Permitted Value	Default	Description
`reg.1.server.1.specialInterop`	GENBAND	standard	Required to enable MADN-SCA on GENBAND.
`sec.TLS.profileSelection.SOPI`	PlatformProfile1, PlatformProfile2 ApplicationProfile1 to ApplicationProfile7	PlatformProfile1	Defines the certificates to be used for SOPI authentication.

Note: Also refer to configuration parameter changes in Polycom UC software 4.0.4, 4.0.3 Rev F, 4.0.2 Rev B and 4.0.2.

Updates to Polycom UC Software 4.0.4

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.4 beside their respective Polycom tracking identification number.

New or Enhanced Features

35157 Added ability to dial '+' character by pressing '*' twice for international dialing.

51255 Added SDP negotiation of "a=rtcp-fb" for RTCP Feedback Messages to request I-Frames.

72440 Added support for MS 2008 Radius Server.

76188 Added ability to process DTMF entry during early media.

78976 Added support for early media followed by local ring back.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the Polycom UC software 4.0.4 configuration file parameters.

Polycom UC Software 4.0.4 Configuration File Enhancements

File	Modification	Parameter	Default	Permitted Values
feature	Added	`reg.X.teluri`	0	0 or 1
feature	Added	`voIpProt.SIP.supportFor100rel`	1	0 or 1
feature	Added	`video.dynamicControlMethod`	0	0 or 1

Polycom UC Software 4.0.4 Configuration File Enhancements

File	Modification	Parameter	Default	Permitted Values
feature	Added	video.forceRtcpVideoCodecControl	0	0 or 1
feature	Added	voIpProt.SIP.useLocalTargetUriForLegacyPickup	0	0 or 1
feature	Added	reg.1.useLocalTargetUriForLegacyPickup	0	0 or 1
feature	Added	voIpProt.SIP.fmtpMustForG7221	0	0 or 1
feature	Added	voIpProt.SDP.offer.rtcpVideoCodecControl	0	0 or 1
feature	Added	tcpIpApp.sntp.Aquery	0	0 or 1
feature	Added	lcl.datetime.date.dateTop	1*	0 or 1
feature	Added	lcl.ml.lang.clock.X.dateTop	1*	0 or 1
feature	Added	pstn.fxsType	CO	0 to 256
feature	Added	dialplan.digitmap.timeOut	3|3|3|3|3|3	0 to 768

Updates to Polycom UC Software 4.0.3 Rev B

This section lists additions, changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.3 Rev B beside their respective Polycom tracking identification number.

Polycom is releasing UC software 4.0.3. Rev B for AT&T only. Please refer to Updates to Previous Software Releases for all available software features and updates.

New or Enhanced Features

There are no new changes or enhanced features from previous software versions.

Updates to Polycom UC Software 4.0.3 Rev F

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.3 Rev F beside their respective Polycom tracking identification number.

If you are currently using Polycom UC software 4.0.3 (build ID 4.0.3.7314), upgrade to Polycom UC software 4.0.3 Rev F (build ID 4.0.3.7562).

New or Enhanced Features

There are no new or enhanced features from previous software versions.

Configuration File Enhancements

There are no changes to the configuration file parameters.

Updates to Polycom UC Software 4.0.3

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.3 beside their respective Polycom tracking identification number.

Polycom UC software 4.0.3 is a limited release that was distributed only to select partners and customers. The build ID for this release was 4.0.3.7314.

New or Enhanced Features

78271 Added support for tel URI in the P-Preferred-Identity header of INVITE messages.

79601/79006/79007 Added support to allow only RFC3264 type SDP media negotiation.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the Polycom UC software 4.0.3 configuration file parameters.

Polycom UC Software 4.0.3 Configuration File Enhancements

File	Modification	Parameter	Modification Description
feature	Added	`voIpProt.SIP.useRFC3264HoldOnly`	1 Phone always uses RFC3264 during hold 0 Phone uses RF3264 or RFC2543 during hold based on the response from far end (default)

Note: Also refer to configuration parameter changes in Polycom UC software 4.0.2 Rev B and 4.0.2.

Updates to Polycom UC Software 4.0.2 Rev B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to the Polycom UC software 4.0.2 Rev B beside their respective Polycom tracking identification number.

New or Enhanced Features

72403 Added support for DHCP renews after loss and recovery of Wi-Fi LAN connection (*applies to SpectraLink 8400*).

76730 Enhanced the digitmap by removing the prepending '+' to the outbound calls and giving the option of configuring the "+" in the dial plan (applies to Lync mode only). Note: For more information on digitmap, refer to the Microsoft Lync Interoperability section.

77038 Added support for early media followed by local ring back.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the Polycom UC software 4.0.2 Rev B configuration file parameters.

File	Modification	Parameter	Modification Description
Wireless	Added	`device.dhcp.releaseOnLinkRecovery`	1 Phone performs a DHCP release on network link recovery (default) 0 Phone does not perform a DHCP release on network link

Note: Also refer to configuration parameter changes in Polycom UC software 4.0.2.

Updates to Polycom UC Software 4.0.2

Polycom distributed UC software 4.0.2 to select partners and customers. It was shipped with Polycom® SoundStation® Duo conference phone. The build-ID for this release was UC software 4.0.2.8017.

New or Enhanced Features

40451/75448 Added support for XT9 PinYin input for Chinese characters (*applies to Polycom® VVX® 1500 business media phone*).

52485/66494 Added support for BroadSoft Hoteling Event Package.

57167/66494/76023 Added support for BroadSoft Call Center Status Event Package.

54576 Added support for the new SpectraLink 8452 Wi-Fi handset with 2D barcode reader.

Configuration File Enhancements

Refer to the following table for a list of new parameters. Note that these configuration parameters are detailed in *Using Premium Automatic Call Distribution for Call Centers (FP 76179)*, which will be made available on Polycom Profiled UC Software Features.

Polycom UC Software 4.0.2 Configuration File Enhancements

File	Modification	Parameter	Modification Description
feature	Added	`feature.callCenterStatus.enabled`	Call feature parameter
feature	Added	`feature.hoteling.enabled`	Call feature parameter
feature	Added	`hoteling.reg`	Call feature parameters
wireless	Added	`barcode.X.Y`	Parameters used to configure the 2D barcode scanner*

*For a detailed description of new parameters specific to the SpectraLink 8400 product family, their properties and values, refer to the Polycom SpectraLink 8400 Series Wireless Telephone Deployment Guide.

Understanding Updates to Polycom UC Software 4.0.1B

There are no functional differences between Polycom UC software 4.0.1B and Polycom UC software 4.0.1. Polycom UC software 4.0.1B was released to include the Polycom® VVX® 500 business media phone and SoundStructure VoIP Interface sip.ld files in the software release package.

Updates to Polycom UC Software 4.0.1

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.1 beside their respective Polycom tracking identification number.

New or Enhanced Features

48734 In a server-based, centralized conference, the phone can now send parallel REFERs without waiting for a 202 Accepted.

67081/70634 Added support for phones to interoperate with a limited set of Microsoft® Lync™ server features (*applies to Polycom® SoundPoint® IP 321, 331, 335, 450, 550, 560, 650, and 670 conference phones; Polycom® VVX® 500 and 1500 business media phones; Polycom® SoundStation® IP 5000 conference phone; Polycom® SoundStation® Duo conference phone; and SpectraLink 84xx*).

67090 Syslog now includes the ability to identify multiple audio streams (*applies to SpectraLink 84xx*).

67594 Added interoperability between the Message Waiting Indicator (MWI) and Microsoft Lync.

68500 The SpectraLink 84xx handsets now display the X-Loader version information in the Phone menu (Menu > Settings > Status > Platform > Phone).

68602 Added support for SSRTP.

68798 Added support for Microsoft SRTP extensions.

69962/70924 Added Microsoft OCS/Lync Presence functionality to phones (*applies to Polycom SoundPoint IP 321, 331, 335, 450, 550, 560, 650, and 670 conference phones; Polycom VVX 500 and 1500 business media phones; Polycom SoundStation IP 5000 conference phone; Polycom SoundStation Duo conference phone; and SpectraLink 84xx*).

70122 Phones now display the Away presence status after a period of user inactivity specified by the following parameters: `pres.idleTimeout.offHours.period`, `pres.idleTimeout.officeHours.period`, `pres.idleTimeout.offHours.enabled`, and `pres.idleTimeout.officeHours.enabled`.

70232 Added a parameter `call.transfer.blindPreferred` to control whether the Transfer soft key on the SpectraLink 84xx should be a consultative transfer or blind transfer.

70614 Added support for Microsoft 2008 Radius (802.1X).

71025 Added new per-registration configuration options for several SRTP parameters: `reg.x.srtp.enable`, `reg.x.srtp.offer`, `reg.x.srtp.require`.

71183 Added missing barcode symbologies (*applies to SpectraLink 84xx*).

71198 Added an option in the Web Configuration Utility for SIP and Provisioning TLS applications to make the Common Name of Subject test configurable.

71424 Updated the presence icon on the phones to be consistent with the Microsoft Lync/OCS style (*applies to SpectraLink 84xx*).

71439 Not including the parameter `oai.userID` in the configuration file or setting the value to NULL both result in the phone using its MAC address to check in into the OAI server (*applies to SpectraLink 84xx*).

71660 Enhanced the Reset to Default option in the Updater to match the option in the application software.

71774 The call forward status on the status bar now displays when Forward – No Answer or Forward –Busy is enabled (*applies to SpectraLink 84xx*).

71997 Added full support for RFC2782 (DNS load balancing).

72074 In the Web Configuration Utility, the Country Code field has been renamed to Regulatory Domain (*applies to SpectraLink 84xx*).

72193 In the phone menu, the Country Code field has been renamed to Regulatory Domain (*applies to SpectraLink 84xx*).

72279 Enabled an EFK to allow a user to invoke the Call Back feature while on hook (*applies to Polycom® VVX® 1500 business media phone*).

72304 The default value for the configuration parameter `up.useDirectoryNames` is now 1 (enabled).

72310/7412 The SpectraLink series handsets can now display the BootL1 version information in Phone menu (Menu > Status > Platform > Phone).

72319 The phone displays a warning icon when the WLAN Network Manager detects an invalid Regulatory Domain request (*applies to SpectraLink 84xx*).

72320 The phone displays a warning triangle when the WLAN Network Manager detects an invalid Regulatory Domain limit setting (*applies to SpectraLink 84xx*).

72367 The phone automatically publishes an Inactive (Idle) presence status after 5 minutes of user inactivity.

72554 Added the ability to configure the `pres.idleTimeout` parameters through the phone menus.

72555 Added the ability to configure the `pres.idleTimeout` parameters through the Web Configuration Utility.

72654 The Exchange Calendaring feature on the SpectraLink handsets has been improved with the following enhancements:

The Calendar icon is shown in the main menu once the calendar is authorized.

The phone displays a Calendar: synchronizing scrolling message in the status bar.

72791 The microbrowser on the Polycom SoundPoint IP conference phones has been functionally improved with the following enhancements:

The audio tag element will inject a Play soft key when in focus.

The user can now specify additional attributes to the audio tag which will be interpreted as a soft key, thus allowing the user to do things such as a Details soft key.

The audio tag will have a descriptive label which will be used as the button label for the audio element in the page. This enables each audio tag to be rendered as a single element in the page with an icon and a descriptive text. The user no longer needs to switch to the text to see the details.

The descriptive label will also be used for the title of the playback screen.

72823 In the media player, the Exit soft key has been renamed to Back.

72824 Playback automatically starts when selecting an audio element from the browser.

73420 When the phone language is set to Japanese, the phone now uses the English AM/PM string for the time/date display (*applies to all Polycom SoundPoint IP conference phone, all Polycom SoundStation IP conference phone, and Polycom SoundStation Duo conference phone*).

73500 The ‘Connect/disconnect from the server’ option has been moved to the Calendar menu (Features > Calendar) in the SpectraLink 84xx handsets.

73510 The Web Configuration Utility language now supports multiple default language labels and help text in English, with the option to add/access other languages.

73669 Updated the 2048-bit Trusted CA Root Certificate List from VeriSign.

73670 Added new VeriSign Intermediate CA certificates.

73671 Added RSA 2048 V3 Root Certificate to Root Store to all phones.

73805 The phone can now display up to 4 Chinese characters in the soft keys (*applies to* Polycom SoundPoint IP 450 conference phone).

73907/74289 Added the ability to automatically upgrade the BootL1 and BootBlock (*applies to SpectraLink 84xx*).

74247 In the Web Configuration Utility, the default available utility languages depend on the platform.

74417 The Updater (BootROM) now supports Basic Authentication with HTTP/HTTPS.

75308 Added the ability to upload encrypted call lists to the provisioning server (*applies to SpectraLink 84xx*).

75469 The volume of PTT audio has been increased and setting the parameter `voice.handsfree.rxag.SL8440=10`, then updating the phone using Update Configuration does not cause the phone to restart (*applies to SpectraLink 84xx*).

Configuration File Enhancements

Refer to Polycom UC Software 4.0.1 Configuration File Enhancements table for a list of enhancements made to the Polycom UC software 4.0.1 configuration file parameters.

The following table includes parameters modified in Polycom UC software 4.0.1. You can find detailed descriptions of the parameters and their values in the Polycom UC Software 4.0.1 Administrators' Guide.

Polycom UC Software 4.0.1 Configuration File Enhancements

File	Modification	Parameter	Modification Description
sip-interop	Added	`call.transfer.blindPreferred`	Call feature parameter
debug	Added	`feature.lyncDebug`	Call feature parameter

site	Added	`reg.x.srtp.enable`
		`reg.x.srtp.offer`
		`reg.x.srtp.require`
			Call feature parameters

wireless	Added	`np.custom1.ringing.toneVolume.usbHeadset`	Notification profiles parameter
wireless	Added	`np.meeting.ringing.toneVolume.usbHeadset`	Notification profiles parameter
wireless	Added	`np.normal.ringing.toneVolume.usbHeadset`	Notification profiles parameter
wireless	Added	`np.silent.ringing.toneVolume.usbHeadset`	Notification profiles parameter
site	Added	`sec.encryption.upload.callLists`	Security parameter
sip-interop	Added	`sec.srtp.mki.length`	Security parameter
sip-interop	Added	`sec.srtp.padRtpToFourByteAlignment`	Security parameter
reg-advanced, site	Added	`up.headset.phoneVolumeControl`	User preferences parameter
debug	Added	`up.headset.AlwaysUseIntrinsicRinger`	User preferences parameter
reg-advanced, site	Added	`up.idleStateView`	User preferences parameter
video	Added	`video.iFrame.delay`	Video parameter

Polycom UC Software 4.0.1 Configuration File Enhancements

File	Modification	Parameter	Modification Description
debug	Added	video.iFrame.period	Video parameter
techsupport	Added	voice.usb.headset.rxdg	Audio parameter
techsupport	Added	voice.usb.headset.txdg	Audio parameter
site	Added	voice.volume.persist.usbHeadset	Audio parameter
sip-interop	Added	voIpProt.SIP.conference.parallelRefer	Call feature parameter
site	Added	webutility.language.plcmServerUrl	Web Configuration Utility parameter
techsupport	Removed	voice.gain.rx.digital.headset.IP_330	Audio parameter
techsupport	Removed	voice.gain.rx.digital.headset.IP_335	Audio parameter
site	Changed	dns.cache.NAPTR.x.ttl dns.cache.SRV.x.ttl dns.cache.A.x.ttl	The maximum value increased from 65535 to 2147483647.
reg-advanced, site	Changed	up.useDirectoryNames	The default value changed from 0 (disabled) to 1 (enabled).
pstn	Changed	up.operMode	The default value changed from 0 to auto.
techsupport	Changed	voice.headset.rxag.adjust.IP_335	The default value changed from -11 to 4 and the maximum value changed from -11 to 90.
techsupport	Changed	voice.headset.rxag.adjust.IP_330	The default value changed from -5 to 4 and the maximum value changed from -5 back to 90.
sip-interop	Changed	VoIpProt.SIP.failoverOn503Response	The default value changed from 1 (enabled) to 0 (disabled).

Updates to Polycom UC Software 4.0.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to Polycom UC software 4.0.0 beside their respective Polycom tracking identification number.

New or Enhanced Features

26549 Enhanced the local missed call feature for shared line appearances. This feature supports RFC 3326 Reason Header.

28514 Enhanced the method of selecting a ring type on the menu screen.

29056 Enhanced the method of notifying the user of unregistered lines.

30251 Added support for non-volatile call lists (*applies to SpectraLink 84xx).*

30887 Added support for 802.1X authentication. Authentication methods include MD-5, EAP-PEAP, EAP-FAST, EAP-TLS, and EAP-TTLS.

32169 The user is now notified with a confirmation when deleting contact information.

33546 Added a host name field to the DHCP registration.

35170 Added support for User Profiles. Users may log into and out of the phone using a server-independent, configuration file-based, authentication method. When successfully authenticated, the user's personal configuration files are applied as well as the user's personal local contact directory and call lists.

35171 Updated most configuration parameters to be updated without the need of a reboot. Only a select number of configuration parameters require a reboot in order to be invoked.

36166 Added the option for the user to allow ringer volume levels to persist after the phone reboots.

38201 The Web-based configuration utility no longer requires the user to submit changes along with a reboot after each page has been modified.

41429 Added the ability in the microbrowser to manage allowable characters into the input field.

41430 Added the ability in the microbrowser for the user to select an item from a list using the dial pad.

44258 Enhanced the API by enhancing the HTTP Push capability by supporting mutual TLS.

44699 Added a Reset to Factory capability.

45777 Added user accessible diagnostic functions ping and traceroute.

47730 The scrolling status bar has been enhanced. The time between scrolled lines has been increased (*applies to Polycom® SoundPoint® IP 3XX conference phone).*

47766 The Trusted CA Pool Management capability has been enhanced. The number of supported customer certificates has been increased to six.

48714 Added ability for the phone to mute the microphone when auto-answering a call.

48750 The Web-based configuration utility now enables the user to configure outbound proxies on a per line basis.

48757 Contacts added to the list are now highlighted and displayed without the need to scroll up or down to view the addition.

50258 Enhanced the method of notifying the user of error and warning indications.

51101 Added the ability to use an Emergency Location Identification Number (ELIN) from LLDP-MED to add a P-Asserted-Identity when using emergency routing:
`dialplan.routing.emergency.preferredSource=[ELIN|Config]` (default ELIN)
`dialplan.routing.emergency.outboundIdentity=xxxxxxxx` (default null)
`dialplan.routing.emergency.outboundIdentity.lldp=xxxxxxxx` (default null).

51471 Added a configuration option to disable the test of subject's CommonName against the registration address (associated with CA management).

52844 Added certificate validation for 802.1X.

53128 Added a configuration option to modify the Backlight timeout duration.

53360 Added the ability to display the phone's current ARP table in the diagnostic menu (*applies to SpectraLink 84xx).*

53908 The web-based configuration utility now offers the ability to configure soft keys and line keys.

54301 The timestamp is now displayed alongside the caller in the Call Lists.

54648 Added HTTPS support in the Updater that was previously called BootROM.

54680 Introduced the ability to import and export local and global configuration files using a PC browser.

54683 The browser-based SW Upgrade button that enables user to upgrade phones with one of multiple compatible software versions is available on the Polycom provisioning server.

54730 Noticeable enhancement from the time the phone is powered up and when it is ready for use.

56150 Added Data Link Layer L2 Discovery between phones and PC.

56187 Added ToID and FromID in SIP Publish packets for VQMon reports.

56274 Added multicast group paging based on the SpectraLink PTT solution.

56942 Configuring Soft key (EFK) settings no longer require a reboot in order to take effect.

57392 Added support to the microbrowser for HTTP proxy authentication (*applies to SpectraLink 84xx).*

57981 Added support for custom device certificates.

58007 Added the ability to revoke certificates used in SSL transactions by using OCSP.

58336 Added SHOULD SDP answer behavior as per RFC 3264.

58507 Enhanced the Web-based configuration and provisioning utility.

60297 Added the ability of random distribution of polling to check for software upgrades.

60907 Added the ability to disable Call Waiting while still allowing further outgoing calls.

61051 Added the ability to display custom soft keys on input forms in the microbrowser.

61138 Added support for incoming TLS connections on the Web server.

61343 Added the ability to disable authentication verification for received SRTP packets.

62671 Added a time-stamped log event indicating when the phone is ready to be used.

63592 Added API calls to the microbrowser for Media Player.

63629 Added Sennheiser EHS configuration menus and options.

64144 The alerting LED and associated line-key animation for second and subsequent incoming calls are now disabled when the Call Waiting feature waiting is disabled.

64243 The API Data push message size limit has been increased to 2048 bytes from 1024 bytes.

64359 Converted the BLA dialog rendering from *No* to *Yes* for user agents that are a remote party to the existing call dialog.

65287 Added the ability to prevent a phone from being provisioned at start-up.

Configuration parameter `prov.startupCheck.enabled` [default = 1 (enabled)]

66212 Added support for setting the syslog server address from DHCP.

66323 Added an administrator operations menu in the Updater to the setup menu: Reboot, Reset Settings, Format File System, and Install BootBlock.

66604 The phone reports connectivity event notifications to an 802.1x enabled switch port when a non-authenticated PC disconnects or reconnects to the phone.

67600 Password and other security entry fields now perform a brief echo of entered characters before being obscured from view.

68110 Added control of available telephony features on the Office Communications Server (OCS) using the `reg.x.telephony` configuration parameter.

68899 Added the ability on the microbrowser to enter two-digit dial pad values for selecting entries in a list.

69225 Added the ability to allow a hard key to be directly assigned an Enhanced Feature Key (EFK) style macro.

69442 Productivity features such as LDAP, Local Call Recording, and Visual Conference Management are enabled without the requirement of a license file. Note that VQMon will remain a licensable feature.

71633 The Reset setting in the Updater menu does not erase the CA and Device Certificates.

Configuration File Enhancements

Certain groups of configuration parameters have been modified in Polycom UC software 4.0.0. In these cases, instead of listing every parameter, the following table will specify a group of related parameters with an abbreviated XML path name ending with (.*).

For example, suppose the following parameters are modified: `device.wifi.enabled`, `device.wifi.ipAddress`, and `device.wifi.ssid`. Since these parameters all begin with `device.wifi`, Polycom UC Software 4.0.0 Configuration File Enhancements table abbreviates these parameters as `device.wifi.*`

Most device parameters have identical parameters ending with `.set`. The `.set` parameters are not included in the following table.

The following table includes parameters changed in Polycom UC software 4.0.0. You can find the new descriptions and values in the Polycom UC Software Administrators Guide.

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Discontinued	`apps.x.Label`	Productivity Applications parameter
Discontinued	`apps.x.Url`	Productivity Applications parameter
Discontinued	`apps.ucdesktop.IP`	Productivity Applications parameter
Discontinued	`apps.ucdesktop.name`	Productivity Applications parameter
Discontinued	`apps.ucdesktop.port`	Productivity Applications parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Discontinued	`device.auth.*` The parameters `device.auth.localAdminPassword` and `device.auth.localUserPassword` have not been removed	Provisioning parameters
Discontinued	`device.dhcp.offerTimeout`	Provisioning parameter
Discontinued	`device.prov.appProvString`	Provisioning parameter
Discontinued	`device.prof.appProvType`	Provisioning parameter
Discontinued	`device.sec.SSL.*`	Provisioning parameters
Discontinued	`device.sec.deviceCertEnabled`	Provisioning parameter
Discontinued	`exchange.server.address`	Productivity Applications parameter
Discontinued	`httpd.lp.port`	Provisioning parameter
Discontinued	`lcl.datetime.date.digitFormatEnable`	User preference parameter
Discontinued	`log.level.change.lp`	Log parameter
Discontinued	`log.level.change.nwmgr`	Log parameter
Discontinued	`log.level.change.sync`	Log parameter
Discontinued	`reg.x.filterReflectedBlaDialogs`	Call feature parameter
Discontinued	`reg.x.server.H323.y.register`	Call feature parameter
Discontinued	`sec.TLS.customDeviceCert.enable`	Security parameter
Discontinued	`sec.dot1x.eapollogoff.pcforcelanlinkreset`	Security parameter
Discontinued	`voIpProt.SDP.useLegacyPayloadTypeNegotiation`	Call feature parameter
Discontinued	`voIpProt.server.H323.x.register`	Call feature parameter
Discontinued	`voice.aec.hd.*` The parameter voice.aec.hd.enable has not been removed	Audio parameters

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Discontinued	`voice.aec.hf.*` The parameter voice.aec.hf.enable has not been removed	Audio parameters
Discontinued	`voice.aec.hs.*` The parameter voice.aec.hs.enable has not been removed	Audio parameters
Discontinued	`voice.aes.hd.duplexBalance`	Audio parameter
Discontinued	`voice.aes.hf.*` The parameter voice.aes.hf.enable has not been removed	Audio parameters
Discontinued	`voice.aes.hs.duplexBalance`	Audio parameter
Discontinued	`voice.gain.rx.digital.ringer.*` The parameter `voice.gain.rx.digital.ringer` has also been removed	Audio parameters
Discontinued	`voice.handset.wideband`	Audio parameter
Discontinued	`voice.rxEq.hf.postFilter.*` The parameter `voice.rxEq.hf.postFilter.enable` has not been removed	Audio parameters
Discontinued	`voice.rxEq.hf.preFilter.*` The parameter `voice.rxEq.hf.preFilter.enable` has not been removed	Audio parameters
Discontinued	`voice.rxEq.hs.postFilter.*` The parameter `voice.rxEq.hs.postFilter.enable` has not been removed	Audio parameters
Discontinued	`voice.rxEq.hs.preFilter.*` The parameter `voice.rxEq.hs.preFilter.enable` has not been removed	Audio parameters
Added	`apps.x.label`	Productivity Applications parameter
Added	`apps.x.url`	Productivity Applications parameter
Added	`apps.push.secureTunnelEnabled`	Productivity Applications parameter
Added	`apps.push.secureTunnelPort`	Productivity Applications parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	apps.push.secureTunnelRequired	Productivity Applications parameter
Added	apps.statePolling.responseMode	Productivity Applications parameter
Added	apps.telNotification.callStateChangeEvent	Productivity Applications parameter
Added	apps.telNotification.userLogInOutEvent	Productivity Applications parameter
Added	apps.ucdesktop.* The parameter apps.ucdesktop.enabled has existed in previous versions	Productivity Applications parameters
Added	bg.color.*	Background parameter
Added	bluetooth.radioOn	Call feature parameter
Added	call.advancedMissedCalls.*	Call feature parameters
Added	call.callWaiting.enable	Call feature parameter
Added	call.localConferenceEnabled	Call feature parameter
Added	callLists.*	Call feature parameters
Added	device.hostname	Provisioning parameter
Added	device.net.dhcpBootServer	Provisioning parameter
Added	device.net.dot1x.*	Provisioning parameters
Added	device.pacfile.*	Provisioning parameters
Added	device.prov.upgradeServer	Provisioning parameter
Added	device.sec.TLS.*	Provisioning parameters
Added	device.usbnet.*	Provisioning parameters
Added	device.wifi.*	Provisioning parameter
Added	dialplan.applyToPstnDialing	Call feature parameter
Added	dialplan.routing.emergency.outboundIdentity	Call feature parameter
Added	dialplan.routing.emergency.outboundIdentity.lldp	Call feature parameter
Added	dialplan.routing.ermgency.preferredSource	Call feature parameter
Added	exchange.meeting.*	Productivity Applications parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	exchange.server.url	Productivity Applications parameter
Added	feature.audioVideoToggle.enabled	Call feature parameter
Added	feature.bluetooth.enabled	Call feature parameter
Added	feature.enhancedCallDisplay.enabled	Call feature parameter
Added	feature.exchangeCalendar.enabled	Call feature parameter
Added	feature.nonVolatileRingerVolume.enabled	Call feature parameter
Added	httpd.cfg.secureTunnelEnabled	Web Configuration Utility parameter
Added	httpd.cfg.secureTunnelPort	Web Configuration Utility parameter
Added	httpd.cfg.secureTunnelRequired	Web Configuration Utility parameter
Added	httpd.ta.secureTunnelEnabled	Web Configuration Utility parameter
Added	httpd.ta.secureTunnelPort	Web Configuration Utility parameter
Added	httpd.ta.secureTunnelRequired	Web Configuration Utility parameter
Added	ind.pattern.blink.*	LED indicator parameter
Added	ind.pattern.flashSlow2.*	LED indicator parameter
Added	lcl.x.pstnCountry	Multilingual parameter
Added	lcl.aidt	Multilingual parameter
Added	lcl.callerId	Multilingual parameter
Added	lcl.callerIdType	Multilingual parameter
Added	lcl.country.* The parameter lcl.country has also been added	Multilingual parameters
Added	lcl.dtmfLevel	Multilingual parameter
Added	lcl.dtmfTwist	Multilingual parameter
Added	lcl.flashTiming	Multilingual parameter
Added	lcl.pstnCountryIndex	Multilingual parameter
Added	lineKey.*	Flexible line key assignment parameters

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	log.level.change.barcd	Log parameter
Added	log.level.change.bluet	Log parameter
Added	log.level.change.clist	Log parameter
Added	log.level.change.daa	Log parameter
Added	log.level.change.dock	Log parameter
Added	log.level.change.drvbt	Log parameter
Added	log.level.change.oaip	Log parameter
Added	log.level.change.ocsp	Log parameter
Added	log.level.change.pdc	Log parameter
Added	log.level.change.pres	Log parameter
Added	log.level.change.pstn	Log parameter
Added	log.level.change.ptt	Log parameter
Added	log.level.change.rtls	Log parameter
Added	log.level.change.tls	Log parameter
Added	log.level.change.wifi	Log parameter
Added	log.render.stdout.* The parameter log.render.stdout has existed in previous versions	Log parameter
Added	mb.main.toolbar.autoHide.* The parameter mb.main.toolbar.autoHide.enabled has existed in previous versions	Microbrowser parameters
Added	messaging.*	SpectraLink instant messaging parameters
Added	np.*	SpectraLink notification profiles parameters
Added	oai.*	SpectraLink Open Application Interface parameters
Added	prov.login.*	Distributed polling parameters
Added	prov.loginCredPwdFlushed.enabled	Distributed polling parameter
Added	prov.polling.timeRandomEnd	Distributed polling parameter
Added	prov.startupCheck.enabled	Provisioning parameter
Added	pstn.*	

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	ptt.*	Paging and push-to-talk parameters
Added	qbc.*	SpectraLink quick barcode connector parameters
Added	qos.ethernet.* The parameters qos.ethernet.callControl.user_priority, qos.ethernet.other.user_priority, qos.ethernet.rtp.user_priority, and qos.ethernet.rtp.video.user_priority exist in previous versions.	SpectraLink QoS parameters
Added	reg.x.auth.domain	Call feature parameter
Added	reg.x.auth.useLoginCredentials	Call feature parameter
Added	reg.x.gruu	Call feature parameter
Added	reg.x.server.y.specialInterop	Call feature parameter
Added	reg.x.server.y.useOutboundProxy	Call feature parameter
Added	reg.x.srtp.enable	Call feature parameter
Added	reg.x.srtp.offer	Call feature parameter
Added	reg.x.srtp.require	Call feature parameter
Added	reg.x.telephony	Call feature parameter
Added	se.pat.misc.customX.*	Sound effects parameters
Added	se.pat.misc.miscX.*	Sound effects parameters
Added	se.rt.answerMute.*	Sound effects parameters
Added	se.rt.autoAnswer.micMute	Sound effects parameter
Added	se.rt.autoAnswer.videoMute	Sound effects parameter
Added	se.rt.customX.micMute	Sound effects parameter
Added	se.rt.customX.videoMute	Sound effects parameter
Added	se.rt.default.micMute	Sound effects parameter
Added	se.rt.default.videoMute	Sound effects parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	se.rt.emergency.micMute	Sound effects parameter
Added	se.rt.emergency.videoMute	Sound effects parameter
Added	se.rt.external.micMute	Sound effects parameter
Added	se.rt.external.videoMute	Sound effects parameter
Added	se.rt.internal.micMute	Sound effects parameter
Added	se.rt.internal.videoMute	Sound effects parameter
Added	se.rt.precedence.micMute	Sound effects parameter
Added	se.rt.precedence.videoMute	Sound effects parameter
Added	se.rt.ringAnswerMute.*	Sound effects parameters
Added	se.rt.splash.micMute	Sound effects parameter
Added	se.rt.splash.videoMute	Sound effects parameter
Added	se.rt.visual.micMute	Sound effects parameter
Added	se.rt.visual.videoMute	Sound effects parameter
Added	sec.TLS.SIP.strictCertCommonNameValidation	Security parameter
Added	sec.TLS.customCaCert.*	Security parameters
Added	sec.TLS.customDeviceCert.*	Security parameters
Added	sec.TLS.customDeviceKey.*	Security parameters
Added	sec.TLS.profile.*	Security parameters
Added	sec.TLS.profileSelection.*	Security parameters
Added	sec.hostMoveDetect.*	Security parameters
Added	sec.srtp.holdWithNewKey	Security parameter
Added	sec.srtp.mki.length	Security parameter
Added	sec.srtp.resumeWithNewKey	Security parameter
Added	softkey.x.insert	Security parameter
Added	tcpIpApp.fileTransfer.waitForLinkIfDown	IP parameter
Added	tone.chord.misc.A3Major.*	Tone parameters
Added	tone.chord.misc.C3Major.*	Tone parameters
Added	tone.chord.misc.Db3Major.*	Tone parameters

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	`tone.chord.misc.E3Major.*`	Tone parameters
Added	`tone.chord.misc.cs12.*`	Tone parameters
Added	`up.25mmRealTime`	User preferences parameter
Added	`up.backlight.timeout.*` The parameter up.backlight.timeout has also been added	User preferences parameter
Added	`up.cfgWarningsEnabled`	User preferences parameter
Added	`up.displayOperMode`	User preferences parameter
Added	`up.headsetOnlyAlerting`	User preferences parameter
Added	`up.hearingAidCompatibility.enabled`	User preferences parameter
Added	`up.hideDateTimeWhenNotSet`	User preferences parameter
Added	`up.multiKeyAnswerEnabled`	User preferences parameter
Added	`up.operMode`	User preferences parameter
Added	`up.pstnSetup`	User preferences parameter
Added	`up.warningLevel`	User preferences parameter
Added	`upgrade.*`	Provisioning parameter
Added	`video.callMode.default`	Video parameter
Added	`video.debug`	Video parameter
Added	`voIpProt.SIP.dialog.strictXLineId`	Call feature parameter
Added	`voIpProt.SIP.IM.autoAnswerDelay`	Call feature parameter
Added	`voIpProt.SIP.mtls.enable`	Call feature parameter
Added	`voIpProt.SIP.pingMethod`	Call feature parameter
Added	`voIpProt.server.x.specialInterop`	Call feature parameter
Added	`voIpProt.server.x.useOutboundProxy`	Call feature parameter
Added	`voice.aec.bt.hd.enable`	Audio parameter
Added	`voice.aec.usb.hf.enable`	Audio parameter
Added	`voice.aes.bt.hd.enable`	Audio parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Added	voice.aes.usb.hf.enable	Audio parameter
Added	voice.agc.bt.hd.enable	Audio parameter
Added	voice.agc.usb.hf.enable	Audio parameter
Added	voice.bt.*	Audio parameters
Added	voice.handset.rxag	Audio parameter
Added	voice.handset.rxdg	Audio parameter
Added	voice.handset.st.	Audio parameter
Added	voice.handset.txag	Audio parameter
Added	voice.handset.txdg	Audio parameter
Added	voice.handsfree.*	Audio parameters
Added	voice.headset.rxag	Audio parameter
Added	voice.headset.rxdg	Audio parameter
Added	voice.headset.st	Audio parameter
Added	voice.headset.txag	Audio parameter
Added	voice.headset.txdg	Audio parameter
Added	voice.ns.bt.*	Audio parameters
Added	voice.ns.usb.*	Audio parameters
Added	voice.ringer.rxag	Audio parameter
Added	voice.rxEq.usb.*	Audio parameters
Added	voice.rxQos.ptt.*	Audio parameters
Added	voice.rxQos.wireless.*	Audio parameters
Added	voice.txEq.usb.*	Audio parameters
Added	voice.usb.*	Audio parameters
Added	voice.volume.persist.bluetooth.headset	Audio parameter
Added	voice.volume.persist.usb.handsfree	Audio parameter
Added	wifi.*	Wi-Fi parameters
Changed	apps.push.messageType	Productivity Applications parameter
Changed	apps.uc.desktop.enabled	Productivity Applications parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Changed	call.autoRouting.preferredProtocol	Call feature parameter
Changed	device.auth.*	Provisioning parameters
Changed	device.cma.mode	Provisioning parameter
Changed	device.dhcp.* The parameter device.dhcp.bootSrvOpt has not been changed	Provisioning parameters
Changed	device.dns.*	Provisioning parameters
Changed	device.em.power	Provisioning parameter
Changed	device.line.*	Provisioning parameters
Changed	device.net.* The parameters device.net.dhcpBootServe, device.net.IPgateway, device.net.subnetMask, and device.net.vlanId have not been changed.	Provisioning parameters
Changed	device.ntlm.versionMode	Provisioning parameter
Changed	device.prov.* The parameters device.prov.password, device.prov.serverName, device.prov.upgradeServer, and device.prov.user have not been changed.	Provisioning parameters
Changed	device.serial.enable	Provisioning parameter
Changed	device.sntp.gmtOffset	Provisioning parameter
Changed	device.syslog.* The parameter device.syslog.serverName has not been changed.	Provisioning parameters
Changed	dialplan.x.digitmap	Call feature parameter
Changed	dialplan.x.routing.server.y.transport	Call feature parameter
Changed	dialplan.digitmap	Call feature parameter
Changed	dialplan.digitmap.timeOut	Call feature parameter

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Changed	dialplan.impossibleMatchHandling	Call feature parameter
Changed	dialplan.removeEndOfDial	Call feature parameter
Changed	dialplan.routing.server.x.transport	Call feature parameter
Changed	dir.H350.dev.attribute.x.type	Directory parameter
Changed	dir.H350.dev.transport	Directory parameter
Changed	dir.H350.group.attribute.x.type	Directory parameter
Changed	dir.H350.group.transport	Directory parameter
Changed	dir.H350.person.attribute.x.type	Directory parameter
Changed	dir.H350.person.transport	Directory parameter
Changed	dir.corp.attribute.x.type	Directory parameter
Changed	dir.corp.transport	Directory parameter
Changed	dir.local.nonVolatile.maxSize	Directory parameter
Changed	dir.local.volatile.maxSize	Directory parameter
Changed	divert.x.autoOnSpecificCaller	Call feature parameter
Changed	divert.x.contact	Call feature parameter
Changed	divert.x.sharedDisabled	Call feature parameter
Changed	divert.busy.*	Call feature parameters
Changed	divert.dnd.*	Call feature parameters
Changed	divert.fwd.x.enabled	Call feature parameter
Changed	divert.noanswer.*	Call feature parameters
Changed	ind.led.x.index	LED indicator parameter
Changed	keypadLock.*	Phone lock parameters
Changed	lcl.ml.lang.clock.x.format	Multilingual parameter
Changed	lcl.ml.lang.list	Multilingual parameter
Changed	lcl.ml.lang.menu.*	Multilingual parameters

Polycom UC Software 4.0.0 Configuration File Enhancements

Modification	Configuration Parameter	Description
Changed	`lcl.ml.lang.tags.*`	Multilingual parameters
Changed	`log.level.change.slog`	Log parameter
Changed	`log.render.file.size`	Log parameter
Changed	`log.render.file.upload.append.limitMode`	Log parameter
Changed	`log.render.file.upload.period`	Log parameter
Changed	`log.render.stdout`	Log parameter
Changed	`log.sched.*`	Log parameters
Changed	`msg.bypassInstantMessage`	Voicemail parameter
Changed	`nat.*` The parameter `nat.keepalive.interval` has not been changed.	IP parameters
Changed	`phoneLock.enabled`	Phone lock parameter
Changed	`pnet.remoteCall.dtmfDuration`	Peer networking parameter
Changed	`powerSaving.enable`	Power saving parameter
Changed	`prov.fileSystem.ffs0.minFreeSpace`	Provisioning parameter
Changed	`prov.polling.mode`	Distributed polling parameter
Changed	`prov.polling.time`	Distributed polling parameter
Changed	`qos.ethernet.*`	Quality of Service parameters
Changed	`qos.ip.callControl.*` The parameters `qop.ip.callControl.dscp.*` have not been changed	Quality of Service parameters
Changed	`qos.ip.rtp.*` The parameters `qos.ip.rtp.dscp.*` and `qos.ip.rtp.video.dscp.*` have not been changed.	Quality of Service parameters
Changed	`reg.x.callsPerLineKey`	Line registration parameter
Changed	`reg.x.outboundProxy.transport`	Line registration parameter

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Modification	Configuration Parameter	Description
Changed	reg.x.ringType	Line registration parameter
Changed	reg.x.server.y.transport	Line registration parameter
Changed	res.finder.minFree	Phone memory parameter
Changed	res.finder.sizeLimit	Phone memory parameter
Changed	res.quotas.background	Phone memory parameter
Changed	res.quotas.bitmap	Phone memory parameter
Changed	res.quotas.cache	Phone memory parameter
Changed	res.quotas.font	Phone memory parameter
Changed	res.quotas.tone	Phone memory parameter
Changed	res.quotas.xmlui	Phone memory parameter
Changed	roaming_buddies.reg	SpectraLink call feature parameter
Changed	roaming_privacy.reg	SpectraLink call feature parameter
Changed	se.destination	Sound effects parameter
Changed	se.pat.callProg.msgWaiting.name	Sound effects parameter
Changed	se.pat.misc.instantMessage.name	Sound effects parameter
Changed	se.pat.misc.localHoldNotification.name	Sound effects parameter
Changed	se.pat.misc.messageWaiting.name	Sound effects parameter
Changed	se.pat.misc.negativeConfirm.name	Sound effects parameter
Changed	se.pat.misc.positiveConfirm.name	Sound effects parameter
Changed	se.pat.misc.remoteHoldNotification.name	Sound effects parameter
Changed	se.pat.misc.welcome.name	Sound effects parameter
Changed	sec.H235.*	Sound effects parameters
Changed	tcpIpApp.keepalive.*	IP parameters
Changed	tcpIpApp.port.*	IP parameters
Changed	up.25mm	User preferences parameter

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Modification	Configuration Parameter	Description
Changed	up.analogHeadsetOption	User preferences parameter
Changed	up.backlight.idleIntensity	User preferences parameter
Changed	up.oneTouchVoiceMail	User preferences parameter
Changed	up.useDirectoryNames	User preferences parameter
Changed	up.welcomeSoundEnabled	User preferences parameter
Changed	up.welcomeSoundOnWarmBootEnabled	User preferences parameter
Changed	video.camera.frameRate	Video parameter
Changed	video.localCameraView.fullScreen.mode	Video parameter
Changed	video.maxCallRate	Video parameter
Changed	video.screenMode	Video parameter
Changed	video.screenModeFS	Video parameter
Changed	voIpProt.H323.dtmfViaSignaling	Call feature parameter
Changed	voIpProt.H323.enable	Call feature parameter
Changed	voIpProt.H323.local.port	Call feature parameter
Changed	voIpProt.SIP.local.port	Call feature parameter
Changed	voIpProt.SIP.outboundProxy.transport	Call feature parameter
Changed	voIpProt.SIP.specialEvent.lineSeize.nonStandard	Call feature parameter
Changed	voIpProt.server.x.transport	Call feature parameter
Changed	voIpProt.server.dhcp.option	Call feature parameter
Changed	voice.audioProfile.Lin16.16ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.32ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.44_1ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.48ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.8ksps.payloadSize	Audio parameter

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Modification	Configuration Parameter	Description
Changed	`voice.codecPref.iLBC.*`	Audio parameters
Changed	`voice.gain.rx.analog.*`	Audio parameters
Changed	`voice.gain.rx.digital.*`	Audio parameters
Changed	`voice.gain.tx.analog.*`	Audio parameters
Changed	`voice.gain.tx.digital.*`	Audio parameters
Changed	`voice.handset.rxag.adjust.*`	Audio parameters
Changed	`voice.handset.sidetone.adjust.*`	Audio parameters
Changed	`voice.handset.txag.adjust.*`	Audio parameters
Changed	`voice.headset.sidetone.adjust.*`	Audio parameters
Changed	`voice.headset.txag.adjust*`	Audio parameters
Changed	`voice.ns.hd.*` The parameter `voice.ns.hd.enable` has not been changed.	Audio parameters
Changed	`voice.ns.hf.signalAttn`	Audio parameter
Changed	`voice.ns.hf.silenceAttn`	Audio parameter
Changed	`voice.ns.hs.signalAttn`	Audio parameter
Changed	`voice.ns.hs.silenceAttn`	Audio parameter

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San Jose, CA 95002
USA

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