RELEASE NOTES

Software 4.1.6 Rev R | May 2016 | 3804-11530-416J

Polycom® UC Software 4.1.6 Rev R

Applies to the Polycom® VVX® 300/310, VVX® 400/410, VVX® 500, and VVX® 600 Business Media Phones with VVX® Camera and VVX® Expansion Module Support

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General

Polycom® Unified Communications (UC) software 4.1.6 Rev R is a general release for all open SIP platforms or for Microsoft® Lync® Server 2010 and Microsoft® Lync® Server 2013. Polycom UC software 4.1.6 Rev R supports the following phone platforms:

- Polycom® VVX® 300/310 business media phone
- Polycom® VVX® 400/410 business media phone
- Polycom® VVX® 500 business media phone
- Polycom® VVX® 600 business media phone

This release also provides support for the following VVX accessories:

- Polycom® VVX® Camera
- Polycom® VVX® Expansion Module

Note: End User License Agreement

Read the terms and conditions in the End User License Agreement for Polycom VVX software products before you start using UC software 4.1.6 Rev R.

Note: Registering with Lync Server

If you are using Polycom UC software 4.1.6 Rev R with Microsoft Lync Server, you can only register one line. You cannot use UC software 4.1.6 Rev R to register multiple lines on a phone registered with Lync Server.

These release notes provide important information on software updates, phone features, feature licenses, and known issues. In addition, these release notes refer to previous UC software versions to assist administrators who are updating to UC software 4.1.6 Rev R release from an earlier software release.

What's in These Release Notes?

The Polycom UC software 4.1.6 Rev R Release Notes contain the following sections:

- General Read this section to understand how the changes in UC software 4.1.6 Rev H affect Polycom hardware, deployment, and configuration of the software.
- What's New for Polycom UC Software 4.1.6 Rev H? This section lists new, enhanced, and discontinued software features.
- Updates to Previous Software Releases This section lists enhanced and discontinued software features in previous software releases.
- Known Issues and Suggested Workarounds This section lists existing known issues and suggests workarounds, if available.
- Reference Documents This section lists all documents relevant to these release notes.

Important Upgrade Notes and Considerations in UC Software 4.1.6 Rev R

Important Update

Impacted: All Polycom SPIP, SSIP and VVX Business Media Phones.

Details: This release includes a critical fix (VOIP-116370) that corrects a problem with Polycom phones manufactured with 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.

Understanding Phone Features and Licenses

The phone features and licenses required to operate a feature vary by phone model. Use this section to find out which phone features and licenses you require for your phone model.

The following table lists features available for each phone model and indicates whether or not a feature license is required.

Use the table VVX Series Features and Licenses if you are deploying VVX 300/310, VVX 400/410, VVX 500, or VVX 600 business media phones. In the following table, No indicates that a phone does not support a feature, Yes indicates a phone supports a feature and no license is required, and Yes* indicates that the phone requires a feature license to support a feature.

VVX Series and SoundStructure VoIP Interface Features and Licenses

Feature	*VVX 300/310*	*VVX 400/410*	*VVX 500*	*VVX 600*	*SoundStructure VoIP Interface*
VQMon	Yes*	Yes*	Yes (Audio only)	Yes (Audio only)	No
Call Recording	Yes	Yes	Yes	Yes	No
Conference Management	Yes	Yes	Yes	Yes	No
Electronic Hookswitch	Yes	Yes	Yes	Yes	No
Enhanced Feature Keys	Yes	Yes	Yes	Yes	No
Customizable UI Background	Yes	Yes	Yes	Yes	No
Asian Languages	Yes	Yes	Yes	Yes	No

Feature	*VVX 300/310*	*VVX 400/410*	*VVX 500*	*VVX 600*	*SoundStructure VoIP Interface*
Enhanced BLF	Yes	Yes	Yes	Yes	No

*Feature License needs to be purchased from Polycom.

Downloading the Distribution Files

You can download UC software 4.1.6 Rev H using the combined or the split file in ZIP file format. For general use, Polycom recommends using the split resource file that corresponds to the phone model(s) for your deployment. Use the table Understanding the Split ZIP Resource Files to match the correct UC software resource file to your phone model. 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 **4.1.6.5734**

Downloading the Combined ZIP File

Use the table Understanding the Combined ZIP File as a reference guide to each of the files distributed in the combined ZIP file and a brief description of each file.

Understanding the Combined ZIP File

Distributed Files	*File Purpose and Application*
sip.ld	Concatenated SIP application executable
sip.ver	Text file detailing build-identification(s) for the release
000000000000.cfg	Master configuration template file
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.
applications.cfg	Contains configuration parameters for microbrowser and browser applications
features.cfg	Contains configuration parameters for telephony features
H323.cfg	Contains configuration parameters for the H.323 signaling protocol
reg-advanced.cfg	Contains configuration parameters for line and call registration and advanced phone feature settings
reg-basic.cfg	Contains configuration parameters for line and call registration and basic phone settings

Distributed Files	*File Purpose and Application*
region.cfg	Contains configuration parameters for regional and localization settings such as time and date and language
sip-basic.cfg	Contains configuration parameters for the VoIP server and softswitch registration
sip-interop.cfg	Contains configuration parameters for the VoIP server, softswitch registration, and interoperability configuration
site.cfg	Contains configuration parameters that are set on a per site basis
video.cfg	Contains configuration parameters for video connectivity
video-integration.cfg	Contains configuration parameters for SoundStation IP 7000 and Polycom HDX integration
SoundPointIP-dictionary.xml	Includes native support for the following languages: • Chinese, Traditional (for IP 321, 331, 335, 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) • Chinese, Simplified (for IP 321, 331, 335, 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) • Danish, Denmark • Dutch, Netherlands • English, Canada • English, United Kingdom • English, United States • French, France • German, Germany • Italian, Italy • Japanese, Japan (for IP 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) • Korean, Korea (for IP 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) • Norwegian, Norway • Polish, Poland • Portuguese, Portugal • Russian, Russia • Slovenian, Slovenia • Spanish, Spain • Swedish, Sweden
SoundPointIPWelcome.wav	Start up welcome sound effect
LoudRing.wav	Loud ringer sound effect

Distributed Files	*File Purpose and Application*
Warble.wav	Loud ringer sound effect

Understanding the Split ZIP Resource Files

Polycom recommends using the split ZIP file to shorten upgrade times. Use the table Understanding the Split ZIP Resource Files to find the split resource file for your phone model.

Understanding the Split ZIP Resource Files

Resource Files	*File Purpose and Application*
3111-46135-002.sip.ld	SIP application executable for VVX 300
3111-46161-001.sip.ld	SIP application executable for VVX 310
3111-46157-002.sip.ld	SIP application executable for VVX 400
3111-46162-001.sip.ld	SIP application executable for VVX 410
3111-44500-001.sip.ld	SIP application executable for VVX 500
3111-44600-001.sip.ld	SIP application executable for VVX 600
sip.ver	Text file detailing the build-identification(s) for the release
000000000000.cfg	Master configuration template file
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.
applications.cfg	Contains configuration parameters for microbrowser and browser applications
device.cfg	Contains Network Configuration device parameters
features.cfg	Contains configuration parameters for telephony features
firewall-nat.cfg	Contains Firewall parameters. Typical MS LYNC Environment
lync.cfg	Contains LYNC specific configuration parameters
pstn.cfg	Contains parameters for PSTN Use
reg-advanced.cfg	Contains configuration parameters for the line and call registration and advanced phone feature settings
reg-basic.cfg	Contains configuration parameters for the line and call registration and basic phone feature settings
region.cfg	Contains configuration parameters for regional and localization settings such as time and date and language

Resource Files	*File Purpose and Application*
sip-basic.cfg	Contains configuration parameters for the VoIP server, soft switch registration
sip-interop.cfg	Contains configuration parameters for the VoIP server, soft switch registration, and interoperability configuration
site.cfg	Contains configuration parameters that are set on a per-site basis
SoundPointIP-dictionary.xml	Includes native support for the following languages: • Chinese, Traditional (for IP 321, 331, 335, 450, 550, 560, 650, IP 5000, Duo) • Chinese, Simplified (for IP 321, 331, 335, 450, 550, 560, 650 IP 5000, Duo) • Danish, Denmark • Dutch, Netherlands • English, Canada • English, United Kingdom • English, United States • French, France • German, Germany • Italian, Italy • Japanese, Japan (for IP 450, 550, 560, 650, IP 5000, Duo) • Korean, Korea (for IP 450, 550, 560, 650, IP 5000, Duo) • Norwegian, Norway • Polish, Poland • Portuguese, Portugal • Russian, Russia • Slovenian, Slovenia • Spanish, Spain • Swedish, Sweden
SoundPointIPWelcome.wav	Start-up welcome sound effect
LoudRing.wav	Loud ringer sound effect
Warble.wav	Loud ringer sound effect

What's New for Polycom UC Software 4.1.6 Rev H?

Polycom UC software 4.1.6 Rev H is a general release for all open SIP platforms or for Microsoft Lync Server 2010 and Microsoft Lync Server 2013. Note that if you are using Polycom UC software 4.1.6 Rev H with Microsoft Lync Server, you can only register one line. You cannot use UC software 4.1.6 Rev H to register multiple lines on a phone with Lync Server.

Polycom UC software 4.1.6 Rev H supports the following phone platforms:

- Polycom VVX 300/310 business media phone
- Polycom VVX 400/410 business media phone
- Polycom VVX 500 business media phone
- Polycom VVX 600 business media phone

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.6 Rev H beside their respective Polycom tracking identification number.

New or Enhanced Features

No New enhancements

Enhanced Capabilities

92541 Open SSL libraries are updated and TLS Heartbleed Open SSL Vulnerability is now fixed

Configuration File Enhancements

No Configuration Parameter changes

Known Issues and Suggested Workarounds

The following issues are known to be present in the current release. The issues will be reviewed for possible fixes in a future release if no reasonable workaround is available.

Known Issues and Suggested Workarounds for UC Software 4.1.6 Rev H

No outstanding issues

Known Issues and Suggested Workarounds for Previous UC Software

26615 Subnet mask forces all packets through gateway when not using DHCP and when using the wrong subnet mask for the network class in use. For example, using 192.168.X.X addresses with a 255.255.0.0 subnet mask. Exists in SIP 1.4.x.

Workaround: Use the correct subnet mask.

26920 Centralized conference fails due to RTP port opening too slowly in some cases.

Workaround: No workaround is currently available.

30086 Boot servers running explicit FTPS are not supported.

Workaround: Use implicit FTPS or HTTPS.

30371 Pattern generator for tones does not work well in the case of a single repeating chord.

Workaround: Start the pattern with a short period of silence followed by the desired initial chord. Loop back to the desired initial chord instead of the initial silence.

33445 LCS Presence and dialing from Buddy Lists does not work across federations.

Workaround: To dial contacts across federations, program a speed dial with the SIP URI of the contact. There is no workaround for watching Federated Buddy status from the phone.

37175 If configuration files are used to set the SNTP server address, date validity checking on CA certificates will be ignored for https provisioning.

Workaround: Set the SNTP server address through the phone UI or use DHCP to inform the phone of the SNTP server address.

37273 If the custom idle display and idle browser features are both enabled, the phone UI displays incorrectly.

Workaround: Do not set `ind.idleDisplay.enabled=1` and enable the Idle Browser at the same time.

37984 Enabling the Idle bit-map on SoundPoint IP 330 and 320 phones causes the Line Key labels and dialed digits to be invisible.

Workaround: Do not use the idle bit-map on 330/320 phones; instead, set `ind.idleDisplay.enabled=0`.

41993 Scrolling through the Corporate Directory may not return complete results if results contain Unicode character values > 127 (server does not support sorting).

Workaround: Start the search in a different location or avoid use of Unicode characters >127 in directories.

42027 In certain scenarios, the time-stamping in log files of a SoundStation IP 7000 that is used as a secondary/slave device is incorrect.

Workaround: As of SIP 3.1.0 the occurrence of this issue only relates to the treatment of Daylight savings Time settings.

44764 SRTP processing may cause performance degradation with certain video/audio codec combinations on the VVX 1500.

Workaround: If SRTP is being used, limit the video bit rate to 384 Kbps.

46997 Camera brightness adjustment does not work between levels 3 to 6 on the VVX 1500.

Workaround: No workaround is currently available.

48905 Jitter parameter is not correctly computed on the SoundStation IP 6000/7000 as per RFC3550.

W*orkaround:* No workaround is currently available.

52141 Daisy-chained SoundStation IP 7000 phones sometimes become stuck during software upgrade.

Workaround: Pressing any key on the phone will continue the upgrade.

52142 Video connections with CounterPath Eyebeam client on the VVX 1500 do not work if H.263-1998 codec is selected. This was experienced with Eyebeam version 1.5.19.5 build 52345.

Workaround: Try using a different codec. Try other versions of Eyebeam client as some do work.

53514 H.264 calls to an HDX9002 device using an MGC 50 Gateway using H.320 result in lip sync issues (*applies to VVX 1500*).

Workaround: Set the call for transcoding on the MGC.

54027 The receiving phone does not re-invite with a new key at the half-life of the key life-time.

Workaround: Ensure that both ends use the same key life time so that the sending phone will initiate a key re-negotiation.

54028 Key changes do not function correctly when multiple crypto suites are enabled.

Workaround: Configure a single crypto suite on the phone.

54321 The VVX 1500 does not receive video (does receive audio) when calls are initiated from a Tandberg C20 (running 2.0.0.191232) device using SIP.

Workaround: No workaround is currently available.

54799 The VVX 1500 transmits H.264 QCIF video to Tandberg MXPs in H.323 calls.

Workaround: Set the video bit rate on the VVX 1500 *to* 512 Kbps to avoid the issue.

54976 H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway using encrypted media (offered but not required) results in distorted audio and no video on the VVX 1500.

Workaround: Configure system for encryption required.

54977 H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway result in lip sync issues on the VVX 1500.

Workaround: No workaround is currently available.

59812 Blind transfer to a URL is not successful on the SoundStation IP 7000. Eventually, the URL soft key becomes unavailable.

Workaround: No workaround is currently available.

61091 The configuration parameter `tcpIpApp.port.rtp.forceSend=1024` works only for the SoundStation IP 6000, 7000 and VVX 1500. It does not function properly for SoundPoint IP phones.

Workaround: No workaround is currently available.

62387 Adding a new line registration to a phone with BLF causes the notifications (ringing) for the BLF line to display on the previous line. Introduced in UC software 3.3.1

Workaround: A phone reset will resolve the issue.

62482 Server certificate Serial Number is checked against the host name if the outbound proxy is configured.

Workaround: No workaround is currently available.

63123 Instead of initiating a new call, an attendant phone plays a reorder tone when a BLF line key is pressed for the second time.

Workaround: No workaround is currently available.

63527 The phone sends out INVITE and CANCEL if no provisional response is received.

Workaround: No workaround is currently available.

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*).

Workaround: No workaround is currently available.

66251 British Telecom Caller ID type is not correctly supported (*applies to SoundStation Duo*).

Workaround: No workaround is currently available.

68815 The phone does not send a *CallState=CallConference* notification when a conference is established (*applies to all SoundPoint IP and SpectraLink 84xx*).

Workaround: No workaround is currently available.

69502 Confirm Click-to-dial text does not appear on the SoundPoint IP 331 phone when SNTP fails.

Workaround: Configure SNTP.

69552 Music on hold (MOH) call dialog does not get terminated when there is an update from the MOH server.

Workaround: End the call to restore normal state.

69735 When the phone is registered with a H.323 line, DTMF digits are not sent in the Tel URI call with Ext and Postd options (*applies to VVX 500, 1500*).

Workaround: No workaround is currently available.

69898 Quick search bar on the SoundPoint IP 321, 330, 331, and 355 only accept 15 characters when the corporate directory is configured.

Workaround: No workaround is currently available.

70480 When the phone is using the Polycom Desktop Connector, the keyboard arrow keys do not support active and inactive call navigation (*applies to VVX 500*).

Workaround: No workaround is currently available.

70728 Software Upgrade does not work if <*partnumber*>.xml file is not specified as a part of `upgrade.custom.server.url` configuration value.

Workaround: Ensure the part-number.xml file is part of the `upgrade.custom.serverurl` configuration value.

71386 Soft keys URIs do not function when the phone is in the Enter Number screen (*applies to VVX 1500*).

Workaround: No workaround is currently available.

71800 Users cannot change the user password using the Web Configuration Utility.

Workaround: Use the phone's user interface to change the user password.

72082 The phones do not detect a server certificate status change from REVOKED to GOOD until the phone is rebooted (*applies to SoundPoint IP 321, 331, 450, 550, 560, 650, and 670, and SoundStation 5000.*)

Workaround: No workaround is currently available.

72211 An explicitly trusted Intermediate CA fails TLS verification when it is the issuer of a server certificate.

Workaround: No workaround is currently available.

72242 The phone is not able to connect to radius server when configured with EAP method as PEAP and inner authentication as GTC (applies to VVX 500).

Workaround: Recommend to use Cisco ACS server 5.1 or higher.

72299 When the SoundPoint IP 450, 560, 650 phones are registered with BLA lines, they continue to display remote hold appearances even after the remote BLA resumes the call.

Workaround: No workaround is currently available.

72387 After pressing the Transfer soft key, the remote BLA line does not show remote hold status when `call.shared.exposeAutoHolds` is set to 1.

Workaround: No workaround is currently available.

72601 The SoundPoint IP 33x phones fail to dial authorized call when in the Phone Locked state.

Workaround: No workaround is currently available.

72677 When a NOTIFY message with a higher version is sent, the phone re-subscribes to the server and gets a NOTIFY with the correct version, but fails to update the dialog with the state (*applies to SoundPoint IP 450/560/650*).

Workaround: No workaround is currently available.

72898 Hard key external URL mapping requires EFK enabled on the SoundPoint IP 650.

Workaround: Enable EFK using configuration files.

73015 The LifeSize Team 220 behaves incorrectly by remaining in a connecting state when there is a call from VVX 1500 over H323.

Workaround: No workaround is currently available.

74120 Plantronics Audio 646 DSP USB headset volume control does not work (*applies to VVX 500*).
Workaround: Adjust the volume using the volume keys on the phone.

74533 A phone configured with a Synergy call server displays the incorrect caller ID on the UI for an incoming call (*applies to VVX 1500*).
Workaround: No workaround is currently available.

74650 In an active audio-only call between a PC client and a VVX 1500, the far-end video never starts on the PC client when a user presses *Add Video*.
Workaround: No workaround is currently available.

74763 The MKC5 key to upload logs does not work (*applies to SoundStation Duo*).
Workaround: No workaround is currently available.

74901 When the lock feature is enabled after phone reboot, the emergency/authorized call list is not displayed when the user tries to place a call using headset/speaker key.
Workaround: No workaround is currently available.

74958 When DND is enabled, the phone is missing the call forward message Fwd:<*number*> (*applies to VVX 500, 1500, and SpectraLink 84xx*).
Workaround: No workaround is currently available.

75157 A phone configured with a Synergy call server displays the incorrect soft keys after a “Conference service unavailable” error is shown. This exists in UC software 3.3.3.
Workaround: No workaround is currently available.

75195 Hold/Transfer/Conference does not display when the parameter `softkey.feature.basicCallManagement.redundant = 0` (*applies to SoundStation Duo*).
Workaround: No workaround is currently available.

75229 A phone configured with a Synergy call server displays the local conference UI when establishing a centralized conference using the Join soft key.
Workaround: No workaround is currently available.

75427 The Unified Call Appearance List (UCAL) filtered view times out to the default UCAL view when a user scrolls the filtered list and does not change the focus (*applies to VVX 500*).
Workaround: No workaround is currently available.

75591 In the Lync environment, when the user logs out, the phone does not logout all the user login credential-dependent applications.
Workaround: No workaround is currently available.

75661 The multi-key combination shortcuts for uploading logs and rebooting the phone sometimes do not work (*applies to VVX 500*).
Workaround: No workaround is currently available.

75671 When parking a call from the Favorites menu, the call park input dialog (where users enter a park extension) disappears (*applies to VVX 500*).
Workaround: No workaround is currently available.

75759 Numeric data entered using the dialpad on the phone browser cannot be deleted using the dialpad.

Workaround: Use the virtual keyboard.

75778 Using Microsoft Lync, if a user dials an invalid extension, the entry is sometimes not logged in the Placed Calls call list.

Workaround: No workaround is currently available.

75869 Changing the local contact directory search option from first name to last name and vice versa causes the Restart and Save soft keys to disappear on the phone.

Workaround: Exiting and re-entering the directory.

75898 Pressing the App hard key on the phone and trying to dial the highlighted/focused SIP/Tel URI does not work with the microbrowser (*applies to VVX 1500 and VVX 500*).

Workaround: No workaround is currently available.

76468 In the Premium ACD, Call Agent state changes from *Unavailable* to *Available* after a phone reboot. This could be an interoperability issue with the call server.

Workaround: No workaround is currently available.

76522 In the hoteling call center feature, the phone does not display the status of the call center when there is a special character in the call center name.

Workaround: The call center administrator can set the call center name.

76655 Using a star (*) in the dial string on the SoundStation IP 7000 causes the phone to send the star as a dot (.) to the HDX video end points.

Workaround: Two stars (**) should be used.

76753 Removing a BLF line from the server causes the speed dial icon to disappear.

Workaround: Restart or reboot the phone, and the icon will re-appear.

76881 On a shared call, the reorder tone is not played to the user when a Resume attempt fails.

Workaround: No workaround is currently available.

76977 Adding a new registration line changes the BLF-monitored lines label from first/last name to its extension number.

Workaround: Reboot the phone.

77039 When PTT is enabled, sender name/ID, updated through the parameter `reg.x.displayname,` does not get updated during the PPT call.

Workaround: No workaround is currently available.

77076 When the XT9 input mode is enabled, the phone displays unmatched UIMA-focused items in the 1st position during XT9 (PinYin) input.

Workaround: No workaround is currently available.

77195 Reboots occur occasionally if the roaming contacts exceeds 100 on the SoundPoint IP and 200 on VVX phones.

Workaround: No workaround is currently available

78232 During a remote conference pickup on a shared line, the phone does not display the call appearance and call indicator.

Workaround: No workaround is currently available.

78340 Sending several MWI NOTIFY messages within a few seconds of each other may cause the phone to reset.

Workaround: Avoid sending multiple MWI messages close together.

79634 During paging, the receiving phone displays the MAC address of the sender instead of the caller ID.

Workaround: Try restarting the phone.

79735 Changing the language of the phone from German to any other language (other than English) may result in a display of diacritic letters (*applies to VVX 500 and SoundPoint 331*).

Workaround: Try changing the language to English first.

80212 In a Lync environment, when the corporate directory and parameter `dir.corp.sortcontrol` are enabled, the contact search does not fetch any contacts.

Workaround: Set the parameter `dir.corp.sortcontrol=0.`

80227 The phone does not display the saved name of the contact in the local contact directory.

Workaround: Use the full URI while adding the contacts in the local contact directory.

81272 When the held call is transferred to a CX 600 phone, the call will be established as a one-way call on the far end.

Workaround: Try hold/resume on the CX 600 to establish a two-way call.

81315 The call logs of the first user are available on the phone when a new user logs in without signing out the first user.

Workaround: No workaround is currently available.

82030 When the Calendar is configured on the phone and the active directory credentials are changed by the user/admin, the phone fails to register to Lync server.

Workaround: The user needs to register the phone manually with the correct credentials.

82043 When a Lync profile is used along with the bootserver, any changes performed to the MAC.cfg file using XML notepad and uploaded to the phone will cause the phone to deregister. The xml notepad adds an extra space in the certificate which makes the certificate invalid and causes the phone to deregister.

Workaround: Use VI editor or Edit Plus editor.

82212 Immediately answering a call on a phone which is outside the enterprise (remote worker/federation scenario) when the UDP is blocked by a firewall, may result in a reboot (*applies to SoundPoint IP 321/331).*

Workaround: No workaround is currently available.

82302 In a CAC (Call Admission Control) scenario, when a call transfer fails from the phone to remote Lync client, the phone is unable to resume the call**.**

Workaround: Try doing consultative transfer.

82401 The call order widget disappears on the phone screen after scrolling through five of the maximum number of calls (24).

Workaround: No workaround is currently available.

82873/82877 The phone fails to update its presence state when it is trying to dial out the emergency call number 911.

Workaround: No workaround is currently available.

83101 In a federated environment, when the UDP traffic is blocked on the firewall, the phone may fail to connect the calls.

Workaround: No workaround is currently available.

83157 The phone does not display the protocol field for the local contacts.

Workaround: No workaround is currently available.

83875 In a conference call scenario, the first phone connected to the conference does not transmit video when joined in a H.323 video conference call to a Cisco SX20 IMCU (*applies to VVX 500 and VVX 600*)**.**

Workaround: No workaround is currently available.

83884 The VVX phone displays a gray image when a video call is established with Grandstream Video phone using DMA server.

Workaround: Use a bit rate of 512 Kbps or 384 Kbps, or reduce the packet size to 1200.

83888 In a conference call scenario, the first phone connected to the conference does not transmit video when joined in a H.323 conference call to a HDX 8006 at a bit rate of 768 Kbps.

Workaround: Use any other bit rate except 768 Kbps, i.e., 384, 512, 1024, etc.

83887/83889 VSX displays a blank or reduced image in a video call with VVX when the phone is transmitting at a bit rate of 384 Kbps or 786 Kbps.

Workaround: Use H.263 video codec with a bit rate greater than 1500 Kbps.

84061 In a call center scenario, the phone does not display the call center information on the default screen when the VVX Camera is attached.

Workaround: Press the call center info soft key to retrieve the related information.

84125 The phone is unable to switch the call mode from audio-video to audio only in SIP protocol when auto routing is enabled and `feature.audioVideoToggle.enabled=1` is set (*applies to VVX 500 and VVX 600*).

Workaround: Select SIP protocol manually from the protocol menu to switch phone from video mode to audio only mode.

84450 An incorrect pop-up error message "DHCP failed" displays on the phone's screen, instead of "Duplicate IP", when the phone detects a duplicate IP.

Workaround: No workaround is currently available.

84598 When a Lync user saves contacts locally on the phone, the contacts are displayed on the screen even after the user signs out and a second user signs in.

Workaround: Reboot the phone after the second user signs in.

85011 In a Push-to-Talk (PTT)/Paging scenario and when navigating with the arrow keys, the active page does not go to the waiting state when the Talk soft key is released.

Workaround: Browse through the menu again after releasing the Talk soft key.

85154 An error pop-up message does not display on the phone's screen when the user tries to play an unsupported media file on the phone's microbrowser.

Workaround: No workaround is currently available.

84795 A pop-up message covers the details view of the contacts on the phone when the user tries to add a contact to favorites (*applies to VVX 300/310*).

Workaround: No workaround is currently available.

84774 Calls are displayed in the Call logs menu according to the logging time.

Workaround: No workaround is currently available.

84692 The sign-in pop-up message takes slightly longer (~30s) to display when a Lync user reboots the phone after a few contacts (~15) are pinned to "frequent contacts" (*applies to VVX 300/310*).

Workaround: No workaround is currently available.

85606 Setting DND presence state from the "UC-One Application" or "My status" menu doesn't set the local DND to ON.

Workaround: No workaround is currently available.

86172 Adding or deleting or editing the BroadSoft directory contact from the phone is not available.

Workaround: No workaround is currently available.

88647 Incoming and outgoing URL call information is not truncated in the call logs on phone's screen.

Workaround: No workaround is currently available.

88659/87975 The phone does not properly display a high resolution background image.

Workaround: No workaround is currently available.

89151 In a hot dial scenario, the phone does not displays '**' as '+' even though the parameter `call.internationalDialing.enabled` is set to 1.

Workaround: No workaround is currently available.

89451 The mute key on the phone does not mute audio when the phone is locked and a call is placed to an authorized number.

Workaround: No workaround is currently available.

89778 In a BLF scenario, an active page on the phone is placed on hold state when you answers an incoming call from a monitored user.

Workaround: No workaround is currently available.

89575/89549/89542/88844/89384 Warning messages for the VVX Expansion modules, dial pads, and favorite contacts do not display in the phone's configured language.

Workaround: No workaround is currently available.

89994 In a power deficit scenario, the phone displays only one alert message, 'Rear port USB device is powered down,' even though both the ports are disabled when you try to power three VVX Expansion Modules and one low power USB device and one high power USB device are connected.

Workaround: No workaround is currently available.

90002 The phone does not detect the USB device when the phone reboots after connecting to three VVX Expansion Modules with one low power USB device and one high power USB device.

Workaround: No workaround is currently available.

90165 Powering of the VVX Expansion Module is delayed when the modules are connected to the phone during an active call.

Workaround: No workaround is currently available.

90260 After failing over to the first backup server, the subsequent server failures do not result in failovers to the next server.

Workaround: No workaround is currently available.

90400 The 'RTP Auth and Encrypt' value is shown as 3 instead of 1 even if the parameter `sec.srtp.sessionParams.noAuth.require` is set to 0.

Workaround: No workaround is currently available.

90440 The phone does not display the message 'More results found. Refine Search' message when there are more than 8 entries in the list for the search criteria.

Workaround: No workaround is currently available.

90465 In a BLF and hunt group hybrid scenario, the LED lights on the phone and expansion modules freeze the phone needs to reboot to unfreeze the lights.

Workaround: Increase the duration between placing and ending.

90494 In a BLF scenario, the monitored line key LED on the monitoring phone does not blink when there are 12 SRTP conference calls on the monitored phone.

Workaround: No workaround is currently available.

90495 Loading a website that runs on JavaScript on the phone's microbrowser increases the memory usage of the phone.

Workaround: No workaround is currently available.

90537 In a shared call scenario, the phone occasionally displays the remote shared line calls on the idle screen without pressing the line key.

Workaround: No workaround is currently available.

90563 The Conference and Transfer Dialer screen does not display properly when a phone is connected with a VVX Expansion Module and a VVX Camera with 34 registered lines and 23 active calls.

Workaround: No workaround is currently available.

90693 When the far end has an active call in progress and transfers the call, the phone does not mute the audio, and you can still hear the audio of the active call (Applies to VVX 400 and VVX 410).

Workaround: Place the call on hold and resume the call.

90689 When entering characters in the phone's microbrowser, by default the first character entered displays as a numeric character despite the key that is pressed.

Workaround: No workaround is currently available.

90629 The phone places a call after you exit the Line Key Info menu by pressing the line key contact twice on expansion module.

Workaround: No workaround is currently available.

90800 The phone displays the message 'Network Authentication Failure' when 802.1X and MD5 authentication is enabled and tries to change the authentication method from MD5 to PEAPv0/MSCHAPv2 after a reboot.

Workaround: No workaround is currently available.

Updates to Previous Software Releases

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

Understanding Updates to UC Software 4.1.6

Polycom UC software 4.1.6 is a general release for all open SIP platforms or for Microsoft Lync Server 2010 and Microsoft Lync Server 2013. Note that if you are using Polycom UC software 4.1.6 with Microsoft Lync Server, you can only register one line. You cannot use UC software 4.1.6 to register multiple lines on a phone with Lync Server.

Polycom UC software 4.1.6 supports the following phone platforms:

- Polycom VVX 300/310 business media phone
- Polycom VVX 400/410 business media phone
- Polycom VVX 500 business media phone
- Polycom VVX 600 business media phone

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.6 beside their respective Polycom tracking identification number.

New or Enhanced Features

62657 Added support for VVX Color Expansion Module and VVX Expansion Module with a paper display.

68444 Added support for Power Management feature (*applies to VVX 500 and VVX 600*)

70524 Added support for Flexible line key assignment

88387/89097 Added support for a digit map timeout for on-hook and off-hook dialing when the normalization is done from the server side.

89367 Added support for the phone to send an ACK and play the reorder tone for 3XX responses

89372/82177 Added support for French/AZERTY keyboard

Enhanced Capabilities

90576/89388 In a Lync 2013 deployment, the phone now successfully transfers the gateway calls without any issue.

90099/86823 The phone now generates the log messages when the parameter `mb.idleRefresh.onFailure` value is set below the default value (<60)

90035 Improved `setsockopt` log levels for `setsockopt` calls

89901/89770 The phone now follows RFC3261 while trying to call back a voicemail from the placed call list**.**

89841/89493/89843 Line key LED indication is now optimized for various incoming call scenarios when the far end terminates a call.

89822 The phone now displays the Private name without any URI on the screen when it places/receives a call from an anonymous number.

89800/89840 P-asserted-identity of the LLDP-MED now includes the INVITE with the required ELIN provided identity and no longer includes any extra characters.

89764/89765/89188 In a Hunt group BLF setup, the phone's performance is now optimized for various scenarios.

89756 Selecting a background image of the phone from the local PC using the phone's web interface is now working as expected.

89752/89785/89740 The phone now performs a SRV lookup for emergency dialing when more than one emergency call servers are configured and failover happens to the second server.

89644/83939/87155/87148/88833/87796/89343/89203/88583/88492 Stability of the upgrade process on the phones has been optimized at different customer environments.

89640/88816/89637 The phone now establishes a three-way conference call successfully and no longer causes a reboot when server recording is enabled on one of the phones in the conference call (*applies to VVX 500*).

89611 In a hold-resume scenario, the phone now sends re-invite with an SDP containing audio and rejects the video when the parameter `video.enable` is set to 0.

89448/90115 The phone now takes the precedence over local configuration files when the NTP server settings are changed from the web interface.

89395/89495 Off-hooking and on-hooking the phone's handset in a quick succession (~1sec) is now working as expected.

89353/89351 Multiple issues regarding Group Call Pick up have been optimized.

89284/89496 Using 'Click to call feature' on BroadSoft or ACME packet platforms no longer causes a reboot on the phone (*applies to VVX 1500*).

89229/90271 The phone no longer lit up the message waiting indicator (MWI) when the user press any registered key greater than 6 (*applies to VVX 500 and VVX 600)*.

89142/88953/89267 The paging functionality on the phone is now improved when a parked call is present on the phone.

89191/88219 The phone now generates the unique 'To' when a call is placed to the ring group and all the phones in the group are rebooted simultaneously.

89097 Off-hook dialing using the dial plan timeout now works without any issues with Lync 2013 normalization rules.

89061 The phone now transmits the prack messages via configured TLS medium.

89033/89312 Multi Key Combos (MKC) are now optimized to work in different customer environments.

89005 Adding contacts to the local directory with an apostrophe (') from the BroadSoft directory is now available.

89007 The phone now contacts the NTP server for updates before establishing an SSL connection via HTTPS.

88983 Navigating through the phone's DHCP menu no longer causes a reboot when the language on the phone is set to Japanese (*applies to VVX 300/310/400 and 410*).

88956/88521 Using the Directed Call Park enhanced feature key on the phone is now optimized.

88954 BLF functionality has been optimized to use transfer and conference functionalities for BLF contacts.

88935 The phone now downloads the mac.cfg file successfully and no longer causes a reboot when HTTP provisioning is used.

88932/88773/89201 In a group call pickup scenario, the Group option is now available after selecting the Pickup soft key when the `feature.groupCallPickup.enabled` is set to 1.

88894/88998/89079 The phone now sends the correct option from the DHCP sub option 125.

88871/88872/86090 In a shared call scenario, the shared registration configured on the first line of the phone is selected by pressing the New Call soft key when there are multiple lines registered on the phone and the other party of the shared line is busy.

88819/88403 In a network outage scenario, the phone no longer sends the invite messages after pressing the Cancel soft key.

88497/7891/87062 The phone's screen now updates without any issues while using hook switch in quick succession.

88406/87794 In a BLF Scenario, the phone now displays only the number in the placed call list when monitoring phone calls on a monitored line using speed dials.

88819/88403 In a network outage scenario, the phone no longer sends the invite messages after pressing the Cancel soft key.

88873/88401 The phone now sends a cancel message to the SIP server when the call is placed and ended before the SIP dialog reaches the server.

88806 Error pop-up strings for the PIN authentication failure has been improved.

88803 The phone now displays the default home view by pressing the Home button after the language is set to Japanese.

88774 The option to select the idle browser as the phones screen saver from the phone's web interface is no longer available.

88630 The phone now sends the negotiated frame rate and profile level when a video call is placed to a mobile device soft client.

88481 In a hunt group scenario, the phone's stability has been improved when the phone receives more than 2 incoming calls simultaneously (applies to VVX 300/310/400/410)

88410 The phone now plays the ringback tone when you try to call a mobile number with a SIP-URI header from the call logs.

88312 The phone now displays menus, e.g., ring type list, network diagnostics etc., in the correct configured language.

88168/88396 In a Lync environment, the phone now updates the configured location correctly to the emergency 911 operator when the LIS subnet is configured on the Lync server, and the user signs in with a Pin authentication (*applies to VVX 300/310*).

88259 Setting the soft key positions on the phone using the parameter `Softkey.x.insert` parameters has been improved.

88190/88635 The LDAP directory no longer appends the "*" attribute in the search bar when the parameter `dir.corp.attribute.4.addstar` is set to 0.

88161/89187 The phone no longer displays the call center status notification on the phone screen when `feature.callCenterStatus.enabled` is set to 0.

88045 Contacts in the local Contact Directory can be sorted either by first name or last name by enabling or disabling the parameter `dir.search.field`.

88043/88008 The phone now successfully changes the state from Do Not Disturb to Available when the user changes his/her state by pressing the DND soft key.

88006 The Contact Directory on the phone now successfully displays all the previously added contacts without any issue when the call list feature is disabled.

87891/87062/88497 The phone's screen now updates without any issues while using hook switch in quick succession.

87825 The phone now successfully sends the DHCP request message for renewing the lease without turning on the screen and playing a notification tone when the NTP is not configured on the DHCP server.

87706/88857 Double tapping the "*" key twice in a quick succession now converts into '+' when the `call.internationalDialing.enabled` is set to 0.

87611 The phone's screen now turns off according to the configuration when the dial tone times out after the configured time and the parameters `powerSaving.idleTimeout.userInputExtension=1` and `call.dialtoneTimeOut=180` are configured.

87152 In the DHCP parameter request list, the phone now successfully includes VLAN information under Option 55.

86772 Soft key strings on the phone's microbrowser now format correctly for French.

86113 The phone now removes the BroadWorks favorite contacts from the speed dial menu on the phone's idle screen when the BroadSoft Directory is disabled by setting the parameter `feature.broadsoftdir.enabled` to 0.

85853 Phones are now optimized for different customer environments for using the idle browser and no longer causes any frequent reboots by generating the core dumps.

84781/89022 On a shared registration scenario, the phone now displays the calling party number on the phone's screen when you long press the line key after a Consultative Transfer.

83747/89589 The phone successfully answers a second incoming call when using Jabra Pro 9465 USB headset and no longer causes the call to drop.

81514/89062 In a Lync deployment, the phone now fetches the location information from the available resources when it receives incomplete location information from LLDP.

81327/86291 Changing the static BLF line key configurations on the web interface no longer affects the configured soft keys on the phone.

Configuration File Enhancements

See the following table for a list of all enhancements made to the UC software 4.1.6 configuration file parameters.

UC Software 4.1.6 Parameter Enhancements

Parameter	*Permitted Value*	*Default*	*Description*
`Diags.dumpcore.enabled`	0 or 1	1	The phone no longer generates the core dump during a reboot/crash when set to 0.
`voIpProt.SIP.subscribe.expires`	10 to 2147483647	3600	Requests the subscription period in seconds. Minimum value is 10 and default value is 3600. Note: The period negotiated with the server may be different.
`mb.idleRefresh.onFailure`	60 to 6553500	60	To reduce the idle browser requests on failure
`up.numOfDisplayColumns`	1 to 4	3 (VVX 500) 4 (VVX 600)	Limits the number of keys/contacts a VVX 500 or VVX 600 phone can display. 1 column displays when parameter value is 0 or 1. 2 columns display when parameter value is 2. 3 columns display when parameter value is 3. 3 columns (for VVX500) and 4 columns (for VVX 600) display when parameter value is 4.
`prov.autoConfigUpload.enabled`	0 or 1	1	Enables/disables the automatic update of the configuration files to the provisioning server when changes are made via phone or web interface.
`tcpIpApp.port.rtp.videoPortRange.enable`	0 or 1	0	Sets a separate port for the video stream.
`tcpIpApp.port.rtp.videoPortRangeStart`	1024 to 65486	2272	Sets the port number for transmitting the video frames when `tcpIpApp.port.rtp`.videoPortRangeStart.enabl`e is enabled.`
`tcpIpApp.port.rtp.mediaPortRangeStart`	1024 to 65486	2222	Sets the port number for transmitting the media frames

Parameter	*Permitted Value*	*Default*	*Description*
`up.IdleViewPreferenceRemoteCalls`	0 or 1	0	If set to 1, the call appearance displays for the remote party (SCA or BLF). The phone switches to idle view after there is no activity on remote party lines. On phone, the call appearance displays for remote party's call status and the phone switches to idle view after the remote party answers the call.
`dialplan.conflictMatchHandling`	0 or 1	0 for generic profile 1 for Lync profile	If the value is set to '0' when the digits entered match a digitmap, the digits are dialed immediately even though there are conflicting digitmaps. When the digits entered are matching more than one digitmap, the timeout is considered before dialing the digits.
`dialplan.x.conflictMatchHandling`	0 or 1	0 for generic profile 1 for Lync profile	If the value is set to '0, when the digits entered match a digitmap, the digits are dialed immediately even though there are conflicting digitmaps. When the digits entered are matching more than one digitmap, timeout is considered before dialing the digits.
`lineKey.reassignment.enabled`	0 or 1	0	To configure the Flexible Line Key assignment feature on the expansion modules, you must set the parameter `lineKey.reassignment.enabled` to `1` to enable the reassignment of line key functions.
`log.level.change.em`	0 to 6	4	To set the EM log levels
`log.level.change.pwr`	0 to 6	4	To set the power log levels

Parameter	*Permitted Value*	*Default*	*Description*
`lineKey.x.category`	BLF, Line Favorites, Presence, or Unassigned	Unassigned	Defines categories you can assigned to line key x where x defines the location of a physical line button. For example, VXX 600 + 3 LCD EMS = 16 + 252 = 268 lines **BLF** or **Presence** `lineKey.x.index` can only be set to 0, which automatically assigns line keys to contacts. **Line** `lineKey.x.index` contains the registration index, from 1 to 34, but automatic assignment is not supported. **Favorites** `lineKey.x.index` contains the favorites' index, ranging from 1 to 9999, but automatic assignment is not supported. **Unassigned** Nothing can be assigned to the line.
`lineKey.x.index`	0 to 9999	0	Defines `lineKey.x.category` **BLF** or **Presence** `lineKey.x.index` can only be set to 0, which automatically assigns line keys to contacts. **Line** `lineKey.x.index` contains the registration index, from 1 to 34, but automatic assignment is not supported. **Favorites** `lineKey.x.index` contains the favorites' index, ranging from 1 to 9999, but automatic assignment is not supported.

Understanding Updates to UC Software 4.1.5

Polycom Unified Communications (UC) software 4.1.5 is a general release for all open SIP platforms or for Microsoft Lync Server 2010 and Microsoft Lync Server 2013. Note that if you are using Polycom UC software 4.1.5 with Microsoft Lync Server, you can only register one line. You cannot use UC software 4.1.5 to register multiple lines on a phone with Lync Server.

Polycom UC software 4.1.5 supports the following phone platforms:

- Polycom VVX 300/310 business media phone
- Polycom VVX 400/410 business media phone
- Polycom VVX 500 business media phone
- Polycom VVX 600 business media phone

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.5 beside their respective Polycom tracking identification number.

New or Enhanced Features

84966 Added support to handle URIs up to 256 characters in length.

85344 Added a Lync status menu on the phone's web interface.

Enhanced Capabilities

72893 Date format options on the phone's web interface are improved when the Chinese language is configured.

76468 In a Hoteling scenario, the phone now fetches the previous status of the guest after a reboot.

76769/82121 Calls without SIP DISPLAY INFO now log under the missed or received call lists.

79064/84264 A call initiated from the phone now successfully failover to the second server when the primary server fails during the call initiation.

79638/84307 The phone no longer plays the message waiting tone when all pending pages are received.

79723/80958 The phone now plays 183 early media and 180 local ringback independently and one after the other.

80852/84844 The phone's touch screen responsiveness after a call or after using an application is improved (*applies to VVX 1500*).

81327/84602 Changing the static BLF line key configurations on the web interface no longer causes any effect on the configured soft keys.

81422 The caller name with quotes (') or double quotes (") now displays on the caller screen without any issue.

82044 The phone's menu is now optimized to hide all the soft key impressions of the previous menu.

82319/83255/83806/83834/82286/86548 The phone now displays all the new pop-ups and strings in the configured language instead of the default language.

82781 In a call forward scenario with Lync deployment, the phone now shows the details of both the caller and the call forwarder.

83619/86104/87146 The phone now sends 'A' as the standard query to the DNS server when the phone uses IPv4.

82630/80609/82970 The phone no longer goes to the DND active state after a reboot when the DND feature is disabled on the server, and the parameter `reg.x.serverFeatureControl.dnd` is set to 1.

83708 The phone now successfully answers an incoming call when it is transferred through the Exchange Auto Attendant.

83776 The phone's idle browser no longer displays the "invalid host name" error after a reboot or restart.

83806/84304/86642/87028/88324/88325 The "Clear browsing data" and "Return to calls" strings on the phone's menu display in the configured language.

83854 In a BLF scenario, the New Call soft key now displays on the incoming call screen of the monitoring user when there is an incoming call to one of the monitored users.

84140/84834 When the phone sends a TCP SYN request and does not receive any response, the phone now switches to the failover server.

84161/80529/87856 The selection of ringtone patterns in multiple scenarios is optimized.

84271/82209 The date and time are now arranged successfully on the phone screen using the parameters `lcl.ml.lang.clock.1.dateTop` and `lcl.datetime.date.dateTop`.

84471/84341 The graphical user interface response on the phone is improved when the phone is used for a long time.

84514 The phone's web interface now accepts a blank admin password.

84534/82075 The phone is now able to park and retrieve the call successfully from the park orbit when the parameters `reg.1.lineKeys` and `reg.1.callsPerLineKey` are set to 1*.*

84643 In a Lync environment, the phone now correctly normalizes the emergency E911 dial plan.

84645 In a call forward scenario, the phone now displays an error message when the server rejects the call forward setting.

84664 The phone now returns to the idle screen when the user presses any termination key without initiating a call from the directory menu on the dialer screen.

84717 The web browser now displays the volume index bar on the phone's screen when the user tries to increase the volume for a media file.

84799 The phone now answers an incoming call automatically, without any issues, when the auto answer is configured.

84803 In a BLF scenario, the monitoring phone can now establish calls when one of its monitored lines is on a call and `attendant.resourceList.1.type="automata"`, `attendant.behaviors.display.spontaneousCallAppearances.normal="0"` are set.

84811/83432 The line label now displays the configured Chinese characters in the call logs screen of the phone.

84815 While using the Polycom Desktop Connector, the '<<' soft key is now available on the phone's screen when the user tries to edit the Desktop User option.

84823 Audio quality on the speaker is improved when the user mutes and unmutes the speaker while receiving a page.

84905 The More soft key no longer displays on the ACD queue status pop-up when the user acts as an agent.

84909 Answering an incoming call from E911 is now available when the phone is in a locked state.

84928/85674 Call logs of the phone now update the PSTN calls properly without any issue.

84931 The power saving state on the phone is now optimized while using the BroadSoft Event Hoteling package (*applies to VVX 500*).

84974 Enabling call forward using the phone's web interface is now available.

85034 Pressing the Select hard key in the phone's corporate directory now displays the details of the contact on the phone's screen.

85105 The Hold/Mute keys are now non-functional when the user tries to place a call in a locked state.

85109/84725/84428/84892 The subnet mask now correctly updates on the phone's screen and the serial port when the DHCP sends an option 43 vendor class.

85373 In a Lync environment, the phone now retains the registration when federated contacts (~60) are added and receives 504 server timeout errors from the server.

85403 In a Lync call server deployment, the phone now adds the '+' prefix automatically for emergency E911 dialing, irrespective of the dial plan.

85408/87180 The ringback for outbound PSTN calls is now optimized and no longer produces any double ringback.

85651 In the ACD scenario, the phone now retains the ACD icon when the server-based call forward is enabled.

85675/85676/85369 In a Lync environment, the user now successfully signs in when the root certificate is downloaded using DHCP option 43.

85711/86579 In a Lync environment, users now successfully sign in using Pin-Authentication when the pin contains leading zeros.

85728/86157 In a shared call scenario, the LED or line status on the phone now changes and shows a solid red color when the incoming call is answered by one of the remote destinations.

85794/86721 The phone now sends a publish request to the VQMon server when the VQMon server is configured with the fully qualified domain name (FQDN).

85947/84435/86548 Strings and phrases in various language directories are improved, and the unwanted spaces are removed.

85962/85984/86764 The phone now displays both the Start and Ping soft keys when the user tries to trace an IP from the phone's menu for the second time immediately after cancelling the first attempt.

86054 The phone now successfully receives the ICMP message "Fragmentation needed" without any failover.

86064 During auto discovery, the phone now includes the IP address by comparing the domain name in the sign-in address with the domain name in the DCHP option 120.

86075 Sending a voicemail from the phone is now available without any issue when the parameters `msg.mwi.1.callBackMode="contact"` and `msg.mwi.1.callBack="voicemail"` are configured (*applies to VVX 400/410*).

86206 The phone now displays only the Conf soft key in the active call state when the parameter `softkey.feature.basicCallManagement.redundant` is set to 0.

86237 The phone gets a grace period of an extra 25% of the validity date of the TLS-DSK certificate and does not get de-registered during that period.

86261 The phone now establishes a two-way audio path when the BroadWorks Anywhere extensions are configured as mobile or PSTN numbers.

86276/86767 The phone no longer updates the directory.xml file when the presence of the monitored buddy changes.

86277 The phone now honors the 302 moved temporary request and performs the corresponding DNS lookup without any issue.

86425/84572 The phone's status menu now updates the AOR details with the primary server when there are two active servers for a single registered number.

86568 The user can successfully disable the call forward no answer from the phone's screen when the parameter `reg.1.serverFeatureControl.cf` is set to 1.

86575/85968/84650/83991 The phone now successfully downloads the updates from the mac-directory.xml file and modifies the speed dial list accordingly.

86623 The phone now plays the defined custom ringtone for a directory contact.

86636 In a call center environment, the LED status on the phone now updates correctly for an incoming call on a shared line (*applies to VVX 300/310 and VVX 400/410*).

86709/85370 The call logs menu on the phone will now not get updated for the missed calls when the parameter `call.advancedMissedCalls.reasonCodes=487` is set *(applies to VVX 500 and VVX 600).*

86730/86869/87504 The phone now transfers a call successfully when the language is set to German, Japanese, Chinese, Taiwanese, and Korean.

86772 Soft key strings on the phone's microbrowser now format correctly for French.

86873/86256/86569/85368/85957 In a Lync deployment, the phone's stability is improved for different network environments.

86877/84668 When there are no entries present on the phone's dialer screen, the Send soft key no longer displays (*applies to VVX 400/410*).

86908/86909/87873 In a Lync environment, the phone will dial the entered number according to the `dialplan.1.lyncdigitmap.timeout` parameter.

87025 A power saving menu is now available on the phone's web interface (*applies to VVX 300/310 and VVX 400/400*).

87026 A password field entry page now displays on the phone's screen when the user tries to answer an incoming call in the locked state.

87029 The MWI LED no longer blinks when the phone is in power saving mode and has not received any new message.

87031 The phone now retains the configured language even after a reboot.

87032 In a BLF scenario, the phone shows only the LED activity and does not show the call appearance when `attendant.behaviors.display.spontaneousCallAppearances.normal` is set to 0.

87059 Sending a visual voicemail from the phone is now available when the VVX camera is not attached, and the `video.enable` parameter is set to 1.

87063 The phone now accepts the MIME message without a SDP and no longer sends a 415 error message.

87075 Placing an emergency call from an unregistered phone is now available from all the call terminations.

87076 The enhanced feature key $Tinvite now works and no longer causes the phone to reboot.

87092 The phone now establishes a two-way audio when the video is disabled, and a BroadWorks Anywhere extension is configured.

87096 The Log-in soft key is now available on the phone after a software upgrade without changing the configuration.

87112 Upgrading the touch screen firmware displays only the visual indicators and no longer shows the progress bar.

87114 Accessing the phone menu in the locked state is no longer available even when the parameter `up.idleStateView` is set to 1.

87115 The phone is now able to perform Transfer or Conference functions when an incoming call is answered in a locked state by entering the password.

87151 Enhanced feature keys like Blind Transfer and Voice Mail Transfer now work without any issue.

87220 Contact information of the caller is no longer shown on the phone's screen when caller ID is disabled on the server.

87394/87599 In a call center environment, the phone no longer produces any beep sound when one of its monitored phones has its handset lifted, or receives a call, or goes off hook using any termination.

87440/87098 The digit map value now displays as empty on the phone's web interface when the `reg.1.applyServerDigitMapLocally` is set to 0.

87453/87632/87262 The phone now establishes a two-way audio path without any issues after successfully establishing a number of (~50) PTT or paging calls.

87560/87422 The phone's microbrowser invokes a backspace soft key (for non-touch based phones) or an on-screen key board (for touch based phones) when the user tries to enter numeric values.

87679 The phone no longer displays the "Please try again" message when the parameters `reg.1.serverFeatureControl.cf="1"`, `reg.1.server.1.expires="10"` are set.

87792 The phone lock feature is now redesigned. The phone will not return to the locked state after answering a call by entering the password from the locked phone.

87890 The Ethernet port information of the phone now displays correctly when the LLDP mode is enabled.

87891/87062 The phone's screen now updates without any issue while using hookswitch in quick succession.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC software 4.1.5 configuration file parameters.

UC Software 4.1.5 Parameter Enhancements

Parameter	*Permitted Values*	*Default*	*Description*
`up.numOfDisplayColumns`	0 to 4	3 (VVX 500), 4 (VVX 600)	4 lines display when parameter value is 0 or 1. 8 lines display when parameter value is 2. 12 lines display when parameter value is 3. 12 lines (for VVX 500) and 16 lines (for VVX 600) display when parameter value is 4.
`dialplan.1.lyncdigitmap.timeout`	1 to 99	3	Entered digits in the dial screen will dial out automatically after the defined timeout.
`video.allowWithSource`	0 or 1	0	If set to 1 and camera is detached, no video codecs are advertised in SDP. If set to 1 and camera is attached, video codecs are advertised in SDP. If set to 0 and camera is attached or detached, video codecs are advertised in SDP (default).

Understanding Updates to UC Software 4.1.4

Polycom Unified Communications (UC) software 4.1.4 is a general release for all open SIP platforms and is compliant with Microsoft® Lync® Server 2010. Note that if you are using Polycom UC software 4.1.4 with Microsoft Lync Server, you can only register one line. You cannot register multiple lines on a phone with Lync Server.

Polycom UC software offers support for the following phone platforms:

- VVX 300/310 business media phones
- VVX 400/410 business media phones
- VVX 500 business media phones
- VVX 600 business media phones

Note: Visual Voice Mail Not Supported

Visual Voice Mail support for the Metaswitch server is not fully qualified in this release and is currently unavailable for these platforms.

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.4 beside their respective Polycom tracking identification number.

New or Enhanced Features

71881 Added support for VVX 300/310 and VVX 400/410.

Enhanced Capabilities

No changes were made from the previous UC software release.

Understanding Updates to UC Software 4.1.3 Rev G

Polycom Unified Communications (UC) software 4.1.3 Rev G is a general release for all open SIP servers including Microsoft Lync Server and is available on the following endpoint platforms:

- Polycom VVX 500 business media phone
- Polycom VVX 600 business media phone
- Polycom SoundStructure VoIP Interface

Note: VVX Camera Support

UC software 4.1.3 Rev G offers video support for the VVX Camera for all the open SIP servers except Microsoft Lync. UC software 4.1.3 Rev G does support all other features of Microsoft Lync.

Web Info: Polycom® BroadSoft UC-One Features and Configuration Details

For information on the Polycom BroadSoft UC-One features, available for the VVX 500 and VVX 600 business media phones, and configuration procedures, see Feature Profile 84393: Using the Polycom BroadSoft UC-One Application on Polycom VVX Business Media Phones on the Polycom Support web site.

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.3 Rev G beside their respective Polycom tracking identification number.

New or Enhanced Features

79861 Added support for BroadSoft directory search, presence and favorites.

84273 A video mute icon is now displayed on the phone's local video pane when the VVX camera shutter is closed.

Enhanced Capabilities

85148/84908 In the locked state, the phone no longer initiates the call to the emergency 911 upon off-hooking the handset twice in quick succession, i.e. within 2 seconds.

84954 SoundStructure VoIP Interface Studio now shows the correct status when the SoundStructure VoIP Interface card is restarted from the web configuration page.

84943/84991 The phone now supports long URIs up to 256 characters in length.

84651 In a BLF scenario, the phone is now able to answer an incoming call using a headset when one of the monitored users is in an active/altering state (*applies to VVX 500*).

84608/84751 The parameter `voIpProt.SIP.supportFor100rel` is introduced for handling 1XX provisional responses.

84439 When both call servers fail, the phone first tries the primary server. It then fails over to the secondary server and back to the primary server. This loop continues until the phone registers with one of the servers.

84407/84390 In a conference call scenario, audio path loss is no longer observed when a Lync PC client is re-added to a conference call with CX 600 and a video-enabled VVX phone (*applies to VVX 500 and VVX 600*).

84299 Call appearance details on the studio UI console are now displayed when a call is made from a private line (*applies to SoundStructure VoIP Interface*).

84221 Establishing a federated call using the long URI from the phone contacts or quick dialed list is now available without any issue.

84138/84139 After a call, the presence status of the user is now updated correctly on the phone when the parameter `reg.1.callsPerLineKey` is set to 1.

84019 Line key is now updated with a 10 digit number immediately after a successful Lync registration (*applies to SoundStructure VoIP Interface*).

83945 The phone is now able to auto answer an incoming call with multiple call-info headers without any issue when the parameters `voIpProt.SIP.alertInfo.1.value="auto-answer"` and `voIpProt.SIP.alertInfo.1.class="autoAnswer"` are set.

83910 Phone now displays video on the phone correctly when it receives video at bit rates greater than 512 Kbps from a “Real Presence Mobile” client.

83781 The phone lock option has been removed (*applies to SoundStructure VoIP Interface*).

83776 The idle browser on the phone appears automatically after a reboot/restart without displaying an "Invalid Hostname" message.

83564/84405 On the Russian SKUs, only the SIP signaling is encrypted during in-band provisioning, leaving the media unencrypted.

83460 The phone is now able to resume an outbound PSTN held call successfully when the user dials the PSTN number without any area code.

83326 Scrolling through the contact directory’s menu using * or # is now working.

83308 During DND, the phone’s missed call lists are now correctly updated.

83069 Music on Hold (MOH) no longer disconnects when the phone is using SRTP traffic.

83022 The phones now send SIP INFO packet with a new line between content attributes.

82915 Added the configuration parameter `tcpIpApp.sntp.AQuery`. If set to 1, the phone will send a DNS query directly for NTP requests.

82319/83255/83806/83834 The phone now displays all the new pop-ups and strings in the configured language instead of the default language.

81639 The phone is now able to handle the bit rate mismatch during a call for G.722.1 codec by setting the parameter `voIpProt.SIP.fmtpMustForG7221` to 1.

81554 In a call center scenario, the phone now displays the correct status and soft keys on the phone's screen after a reboot.

81327 Changing the static BLF line key configurations on the web interface no longer affects the configured soft keys.

80578 Enabling the exchange calendar feature no longer causes problems when connecting to the hosted exchange server.

80293 The “Service Unavailable” message is no longer displayed on the phone’s screen when a paging soft key is enabled and then disabled.

80125 Quickly pressing ACD Sign-In/Sign-Out keys is now working properly and does not take the phone to the service unavailable state.

79846 The phone now retains the server-based DND even after the phone is locked.

77877 Missed/received calls are now logged in the respective configured list when the parameters `call.advancedMissedCalls.enabled and call.advancedMissedCalls.addToReceivedList` are set.

77097 The phone’s user interface is now improved to handle 486busy response.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC software 4.1.3 Rev G configuration file parameters.

UC Software 4.1.3 Rev G Parameter Enhancements

Parameter	*Permitted Values*	*Default*	*Description*
`feature.qml.enabled`	0 or 1	0	If 1, the QML viewer is enabled on phone. If 0, the viewer is disabled. The viewer is used to load the QML applications.
`feature.broadsoftdir.enabled`	0 or 1	0	If 1, the BroadSoft Enterprise directory is enabled. If 0, the directory is disabled.
`feature.broadsoftUcOne.enabled`	0 or 1	0	If 1, the BroadSoft UC-One feature is enabled. If 0, the feature is disabled.
`dir.broadsoft.xsp.address`	Dotted-decimal IP address or hostname or FQDN	Null	The IP address or hostname of the BroadSoft XSP server. For example, http://xsp-idc1.asia.polycom.com.
`dir.broadsoft.xsp.username`	String	Null	The username used to authenticate the BroadSoft XSP server.
`dir.broadsoft.xsp.password`	String	Null	The password used to authenticate the BroadSoft XSP server.
`feature.presence.enabled`	0 or 1	0	To enable the BroadSoft Presence
`dir.broadsoft.qml.home`	String	"file:////usr/local/apps/BSD/BSD.qml"	
`xmpp.1.auth.password`	True/false	True (1)	Password used for XMPP registration. When provisioned from CMA, the value is set to the CMA account password.
`xmpp.1.dialMethod`	String min 0, max 256	sip	For "sip" dialing, the destination XMPP URI is converted to a SIP URI, and the first available SIP line is used to place the call.
`xmpp.1.enable`	True/false	False(0)	Flag to determine if XMPP presence is enabled
`xmpp.1.jid`	String min 0, max 256	“”	Jabber identity used to register with presence server

Parameter	*Permitted Values*	*Default*	*Description*
`xmpp.1.jid`	Duplicate		
`xmpp.1.server`	String min 0, max 256	“”	Presence server IP or FQDN
`xmpp.1. verifyCert`	True/false	True(1)	Flag to determine if the phone should verify the TLS certificate provided by the XMPP presence server.

Understanding Updates to UC Software 4.1.3

Polycom Unified Communications (UC) software 4.1.3 is a general release for all open SIP servers and offers video support for the VVX Camera on the following phone platforms:

- VVX 500 business media phone
- VVX 600 business media phone

UC software 4.1.3 does not support Lync video on Microsoft Lync Server. It does support all other features on Microsoft Lync Server available in UC software 4.1.2.

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.3 beside their respective Polycom tracking identification number.

New or Enhanced Features

62655 Added support for the Polycom VVX Camera.

82204 Added video support (SIP and H323) on VVX 500 and VVX 600 business media phones.

Enhanced Capabilities

84089 The phone does now not show the incoming call appearance pop-up for the second time when the call is answered after the initial pop-up has timed out.

83752 The phone’s UI/display is improved when switching between active call and other menus.

83725/81683 The phone is now able to transfer the location information to E911 successfully using NTLM credentials in MS Lync Server.

83716 In a BLF scenario, phone is now able to retain the monitored user when the registration switches between private and shared.

83513 The phone is now equipped to display the E911 disclaimer pop-up when set on the Lync server.

83273 Muting and unmuting the call by double-tapping the mute area on Jabra 9465 USB headset/touch control is now working as expected.

82898 Changing the SIP line to a “SIP + H.323” line using the phone’s web interface during an active call does not cause any call drop or reboot.

82282 Calendar parser on the phone is now able to correctly identify the 'dial in numbers' for conference calls.

82149 In a BLF scenario, phone no longer displays the call appearance of the monitored parties when the parameter `attendant.behaviors.display.spontaneousCallAppearances.normal="0"` is set.

82073 Answering a call using EHS is now available on the phone when the BLF status is active for any monitored user.

82051/81016 Blind transfer from the contact directory/recent call list/favorites is now working as expected.

82005 When the URL dialing is disabled, and the user tries to transfer an active call made to a BLF-monitored line, the user will no longer see the prompt screen.

81541 In a call center scenario, the guest user is now able to sign out from the host phone in a single attempt when the guest has already signed in or out once.

81504 In a call center scenario, phone now updates the previous state of the user after a line re-registration is performed.

81439 In a call center scenario, phone no longer displays “Back” soft key during an incoming call to an ACD line.

81245 Incoming call pop-up will no longer flash on the phone’s screen when the call is answered after 10 seconds.

80959/79673/79242 Provisioning the phone using the SIP-TLS option is now functioning as expected.

79915 Local contact search on the phone now displays all the contacts when there is no search criterion.

79330 The phone now allows the STP traffic from the phone’s PC port.

76769 Missed and received calls are now updated on the phone when the “SIP DISPLAY INFO” portion is missing in the initial SIP INVITE.

75937 The phone’s browser now updates the content properly during scrolling when the toolbar is in a hidden state.

74871 In a PTT call, a beep sound is now played on the phone from the first call, indicating the start of voice transmission.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC software 4.1.3 configuration file parameters.

UC Software 4.1.3 Parameter Enhancements

Parameter	*Permitted Values*	*Default*	*Description*
`call.autoAnswer.H323`	0 or 1	0	If 0, auto-answer is disabled for H.323 calls. If 1, auto-answer is enabled for all H.323 calls.

Parameter	*Permitted Values*	*Default*	*Description*
`call.autoAnswer.videoMute`	0 or 1	0	If 0, video begins transmitting (video Tx) immediately after a call is auto-answered. If 1, video transmission (video Tx) is initially disabled after a call is auto-answered.
`call.autoRouting.preference`	line or protocol	line	If set to line, calls are placed via the first available line, regardless of its protocol capabilities. If the first available line has both SIP and H.323 capabilities, the preferred protocol will be used (call.autoRouting.preferredProtocol). If set to protocol, the first available line with the preferred protocol activated is used, if available. If not available, the first available line will be used. Note: Auto-routing is used when manual routing selection features (up.manualProtocolRouting)are disabled.
`call.autoRouting.preferred Protocol`	SIP or H323	SIP	If set to SIP, calls are placed via SIP if available, or via H.323 if SIP is not available. If set to H323, calls are placed via H.323 if available, or via SIP if H.323 is not available.
`call.enableOnNotRegistered`	0 or 1	1	If 1, users can make calls when the phone is not registered. If 0, calls are not permitted without registration. Note: Setting this parameter to 1 allows VVX phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured.
`feature.audioVideoToggle. enabled`	0 or 1	0	If 0, the audio/video toggle feature is disabled. If 1, the feature is enabled.
`reg.x.protocol.H323`	0 or 1	0	If 0, H.323 signaling is not enabled for registration x. If 1, H.323 signaling is enabled.
`reg.x.server.H323.y.address`	dotted-decimal IP address or hostname	Null	Address of the H.323 gatekeeper.
`reg.x.server.H323.y.port`	0 to 65535	0	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used.
`reg.x.server.H323.y.expires`	positive integer	3600	Desired registration period.

Parameter	*Permitted Values*	*Default*	*Description*
`sec.H235.mediaEncryption.enabled`	0 or 1	1	If 0, H.235 Voice Profile RTP media encryption will be disabled. If 1, H.235 media encryption will be enabled and negotiated when such encryption is requested by the far end.
`sec.H235.mediaEncryption.offer`	0 or 1	0	If 0, media encryption negotiations will not be initiated with the far end. If 1 and sec.H235.mediaEncryption.enabled is also 1, media encryption negotiations will be initiated with the far end. However, successful negotiations are not a requirement for the call to complete.
`sec.H235.mediaEncryption.require`	0 or 1	0	If 0, media encryption negotiations will not be required. If 1 and sec.H235.mediaEncryption.enabled is also 1, media encryption negotiations will be initiated or completed with the far end; if negotiations fail, the call will be dropped.
`sec.srtp.lifetime`	0, positive integer minimum 1024 or power of 2 notation	0	The lifetime of the master key used for the cryptographic parameter in SDP. The value specified is the number of SRTP packets. If 0, the master key lifetime is not set. If set to a valid value (at least 1024, or a power such as 2^10), the master key lifetime is set. When the lifetime is set, a re-invite with a new key will be sent when the number or SRTP packets sent for an outgoing call exceeds half the value of the master key lifetime. Note: Setting this parameter to a non-zero value may affect the performance of the phone. If you are using a VVX Camera on a VVX 500 and 600 phone, set to 2^31.
`up.manualProtocolRouting`	0 or 1	1	If 1, the user is presented with a protocol routing choice in situations where a call can be placed using either protocol (for example, with SIP and H.323 protocols). If 0, the default protocol is used, and the user does not choose.

Parameter	*Permitted Values*	*Default*	*Description*
`up.manualProtocolRouting.softKeys`	0 or 1	1	Choose whether you want to display soft keys that control Manual Protocol Routing options. When Soft Key Control is enabled, you can use soft keys to choose between the SIP or H.323 protocol. When disabled, soft keys for protocol routing will not display. The soft keys are enabled by default.
`video.autoFullScreen`	0 or 1	0	If 0, video calls only use the full screen layout if it is explicitly selected by the user. If 1, video calls use the full screen layout by default, such as when a video call is first created or when an audio call transitions to a video call.
`video.autoStartVideoTx`	0 or 1	1	When enabled, video transmission to the far side begins when you start a call. When disabled, video transmission does not begin until you press the Video > Start Video soft keys. This parameter controls video sent to the far side. Video from the far side will always be displayed if it is available, and far side users can control when to send video.
`video.callMode.default`	audio or video	audio	Allows the user to select the mode to use when using SIP protocol only.
`video.callRate`	128 to 2048	512	The default call rate (in kbps) to use when initially negotiating bandwidth for a video call.
`video.camera.brightness`	0 to 6	3	Set brightness level. The value range is from 0 (dimmest) to 6 (brightest).
`video.camera.contrast`	0 to 4	0	Set contrast level. The value range is from 0 (no contrast increase) to 3 (most contrast increase), and 4 (noise reduction contrast).
`video.camera.flickerAvoidance`	1 to 2	1	Set flicker avoidance. If set to 1, 50Hz AC power frequency flicker avoidance (Europe/Asia). If set to 2, 60Hz AC power frequency flicker avoidance (North America).

Parameter	*Permitted Values*	*Default*	*Description*
`video.camera.frameRate`	5 to 30	25	Set target frame rate (frames per second). Values indicate a fixed frame rate, from 5 (least smooth) to 30 (most smooth). Note: If video.camera.frameRate is set to a decimal number, the value 25 is used.
`video.camera.saturation`	0 to 6	3	Set saturation level. The value range is from 0 (lowest) to 6 (highest).
`video.camera.sharpness`	0 to 6	3	Set sharpness level. The value range is from 0 (lowest) to 6 (highest).
`video.codecPref.H264` `video.codecPref.H263`	1 to 4	1 3	Specifies the video codec preferences.
`video.dynamicControlMethod`	0 or 1	0	If 1, the first I-Frame request uses the method defined by video.forceRtcpVideoCodecControl, and subsequent requests alternate between RTCP-FB and SIP INFO.

`video.enable`
`video.enable for VVX 500`
`video.enable for VVX 600`
	0=Disable, 1=Enable	0
		0
		0
			If 0, video is not enabled and all calls—both sent and received—are audio only. If 1, video is sent in outgoing calls and received in incoming calls if the other device supports video.

`video.forceRtcpVideoCodecControl`	0 or 1	0	If set to 1, the phone is forced to send RTCP feedback messages to request fast update I-frames for all video calls (the phone includes a=rtcp-fb in the SDP). If 0, RTCP feedback messages are not forced.
`video.iFrame.delay`	0 to 10, seconds	0	When non-zero, an extra I-frame is transmitted after video starts. The amount of delay from the start of video until the I-frame is sent is configurable up to 10 seconds. Use a value of 2 seconds if you are using this parameter in a Microsoft Lync environment.
`video.iFrame.minPeriod`	1 - 60	2	After sending an I-frame, the phone will always wait at least this amount of time before sending another I-frame in response to requests from the far end.

Parameter	*Permitted Values*	*Default*	*Description*
`video.iFrame.onPacketLoss`	0 or 1	0	If 1, an I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.
`video.localCameraView.fullscreen.enabled`	0 or 1	1	Determines whether the local camera view is shown in the full screen layout. If set to 0, the local camera view is not shown. If set to 1, the local camera view is shown.
`video.localCameraView.fullscreen.mode`	pip, side-by-side	pip	Determines how the local camera view is shown. If set to pip, the local camera view displays as a picture-in-picture with the far end window. If set to side-by-side, the local camera view displays side-by-side with the far end window.
`video.maxCallRate`	128 to 2048 kbps	768	The maximum call rate allowed. This allows the administrator to limit the maximum call rate that the users can select. If video.callRate exceeds this value, this value will be used as the maximum.
`video.profile.H263.CifMpi`	1 to 32	1	Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-4. To disable, enter ‘0’.
`video.profile.H263.jitterBufferMax` `video.profile.H264.jitterBufferMax`	(video.profile.H263.jitter BufferMin + 500ms) to 2500ms (video.profile.H264.jitter BufferMin + 500ms) to 2500ms	2000ms	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.
`video.profile.H263.jitterBufferMin` `video.profile.H264.jitterBufferMin`	33ms to 1000ms	150ms	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads and larger than the expected short-term average jitter.

Parameter	*Permitted Values*	*Default*	*Description*
`video.profile.H263. jitterBufferShrink` `video.profile.H264. jitterBufferShrink`	33ms to 1000ms	70ms	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (33 ms) to minimize the delay on known good networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).
`video.profile.H263. payloadType` `video.profile.H264. payloadType`	96 to 128 96 to 128	34 109	RTP payload format type for H263 MIME type. RTP payload format type for H264/90000 MIME type.
`video.profile.H264. profileLevel`	1, 1b, 1.1, 1.2, and 1.3 (1 to 2)	1.3	Specify the highest profile level within the baseline profile supported in video calls. The phone supports the following levels: 1, 1b, 1.1, 1.2, 1.3. The default level is 1.3. For more information, refer to ITU-T H.264.
`video.profile.H263.QcifMpi`	1 to 32	1	Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’.
`video.profile.H263.SqcifMpi`	1 to 32	1	Specify the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter ‘0’.
`video.quality1`	motion, sharpness	Null	The optimal quality for video that you send in a call or a conference. Use motion if your outgoing video will have motion or movement. Use sharpness or Null if your outgoing video will have little or no movement. Note: If motion is not selected, moderate to heavy motion can cause some frames to be dropped.

Parameter	*Permitted Values*	*Default*	*Description*
`video.screenMode`	normal, full, crop	normal	The screen mode for the video window shown in non-full screen mode. If set to normal or Null, the entire view is displayed and horizontal or vertical black bars may appear on the edges to maintain the correct aspect ratio. If set to full, the entire view is stretched linearly and independently to fill the video frame. If set to crop, black bars are not shown, the image is re-sized and enlarged to cover the entire video frame, and parts of the image that do not fit in the display are cropped (removed).
`video.screenModeFS`	normal, full, crop	normal	The screen mode for the video window shown in full screen mode. If set to normal or Null, the entire view is displayed and horizontal or vertical black bars may appear on the edges to maintain the correct aspect ratio. If set to full, the entire view is stretched linearly and independently to fill the screen. If set to crop, black bars are not shown, the image is re-sized and enlarged to cover the entire screen, and parts of the image that do not fit in the display are cropped (removed).
`voIpProt.H323.autoGateKeeperDiscovery`	0 or 1	1	If set to 1, the phone will attempt to discover an H.323 gatekeeper address via the standard multicast technique, provided that a statically configured gatekeeper address is not available. If set to 0, the phone will not send out any gatekeeper discovery messages.
`voIpProt.H323.blockFacilityOnStartH245`	0 or 1	0	If set to 1, facility messages when using H.245 are removed.
`voIpProt.H323.dtmfViaSignaling.enabled`	0 or 1	1	If set to 1, the phone will use the H.323 signaling channel for DTMF key press transmission.
`voIpProt.H323.dtmfViaSignaling.H245alphanumericMode`	0 or 1	1	If set to 1, the phone will support H.245 signaling channel alphanumeric mode DTMF transmission. Note: If both alphanumeric and signal modes can be used, the phone gives priority to DTMF.

Parameter	*Permitted Values*	*Default*	*Description*
`voIpProt.H323.dtmfViaSignaling.H245signalMode`	0 or 1	1	If set to 1, the phone will support H.245 signaling channel signal mode DTMF transmission.
`voIpProt.H323.enable`	0 or 1	0	A flag to determine if the H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing. If set to 1, the H.323 protocol is used.
`voIpProt.H323.local.port`	1 to 65535	1720	Local port to be used for H.323 signaling. Local port for sending and receiving H.323 signaling packets. If set to 0, 1720 is used for the local port but is not advertised in the H.323 signaling. If set to some other value, that value is used for the local port and is advertised in the H.323 signaling.
`voIpProt.H323.local.RAS.port`	1 to 65535	1719	Local port for RAS signaling.
`voIpProt.server.H323.x.address`	dotted-decimal IP address or hostname	Null	Address of the H.323 gatekeeper. Note: Only one H.323 gatekeeper per phone is supported. If more than one is configured, only the first is used.
`voIpProt.server.H323.x.port`	0 to 65535	1719	Port to be used for H.323 signaling. Note: The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP.
`voIpProt.server.H323.x.expires`	positive integer	3600	Desired registration period.

Understanding Updates to UC Software 4.1.2 Rev B

Polycom Unified Communications (UC) software 4.1.2 Rev B is a general release for the VVX 500 and VVX 600 products with all supported SIP servers including Microsoft Lync Server 2010 and qualified compatible Microsoft Lync Server 2013. For Microsoft Lync Server, this release supports a single registered line.

Note: UC Software 4.1.x - New and Enhanced Features

Microsoft Lync users can refer to Understanding Updates to UC Software 4.1.0 Rev B for information on new and enhanced features and capabilities available with Lync-compatible UC software.

This section lists all changes, additions, removals, enhancements, and configuration file parameter changes to UC software 4.1.2 Rev B beside their respective Polycom tracking identification number.

New or Enhanced Features

82450 The phone now sends the SIP INVITE and REGISTER messages in the MS-Subnet header.

Enhanced Capabilities

81683 The phone now fetches the Location Information Service by using either NTLM credentials or the Lync-signed user certificate.

83754 Restarting the phone after a network outage automatically registers and signs in the user.

83564 On the Russian SKUs, only the SIP signaling is encrypted during in-band provisioning, leaving the media unencrypted.

84105 Frequent multiple Sign in- Sign out no longer causes any reboot on the phone.

84218 The phone no longer caches the call history of the previously signed user.

84221 The phone can now dial to a federated contact with a long URI from the quick dial list.

84298 The phone calls now comply with the enterprise Lync server bandwidth policy.

84305 The risk of accidentally dialing an emergency number in the locked state is reduced. The user now has to select a number from the authorized contact list and press the send button to initiate a call.

Understanding Updates to UC Software 4.1.2

Polycom Unified Communications (UC) software 4.1.2 is a general release for the VVX 500 and VVX 600 products with all supported SIP servers including Microsoft Lync Server 2010. For Microsoft Lync Server, this release supports a single registered line.

Note: UC Software 4.1.X - New and Enhanced Features

Microsoft Lync users can refer to Understanding Updates to UC Software 4.1.0 Rev B for information on new and enhanced features and capabilities available with Lync-compatible UC software.

New or Enhanced Features

62651 Added Support for the Polycom VVX 600 product.

82042 Added support for BroadSoft Hoteling Event Package.

80220 Added support for Pin Authentication on MS Lync.

82162 Added support for Verizon (Baltimore CA 2048) certificate.

Enhanced Capabilities

82101 The “mute” hard key and mute display on the phone while initiating a conference call are now synchronized.

81878 On the phone’s virtual keyboard the input mode key will no longer overlap with other keys on the screen.

81536 On the contact entry/edit screen, the default encoding “Abc” mode is changed to “abc” by pressing the “Caps” key on the phone virtual keyboard.

80790 The phone no longer deregisters when trying to install an unknown certificate.

80761 When using Polycom Desktop Connector (PDC) the “Alt + numeric” keys are configured to off hock the lines accordingly.

79968 Users can now scroll up/down in the phone voice recording menu while deleting recorded files.

79680 The phone displays a pop-up message when there is a low bandwidth detected while hosting a centralized conference.

79625 When the phone tries to park a call using blind transfer and fails, the call is retained back between the caller and called phone.

78256 In a Lync environment, contacts screen now displays 5 soft keys.

75808 When the phone is configured with one or more BLF lines, the first incoming call to the BLF lines during prompt stage is displayed on main display and the second stacks on top.

74607 Soft key label is corrected as “select” under the unavailable reason codes in ACD menu.

73618 Changing the phone web UI language reflects on all the contents except ringtones.

72197 Clicking the “check for updates” button the phone web UI now populate all the available software versions from the Polycom hosted server in the combo box.

72055 Enhanced the pop-up message to “*Failed to fetch available software from the Polycom Hosted server. Please try again later or contact your Network Administrator if the problem persists*” when trying to connect to the Polycom hosted server from the phone web UI.

64091 Resetting the local configuration on the phone will no longer reboot the phone.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC software 4.1.2 configuration file parameters.

Web Info: Understanding Polycom Configuration Parameters

You can find detailed descriptions of these parameters and their values in the Polycom UC Software 4.1.0 Administrators’ Guide on the support web site.

UC Software 4.1.2 Parameter Enhancements

File	*Modification*	*Parameter*	*Modification Description*
Feature	Added	`reg.X.auth.usePinCredentials` `="0" or “1”` X= line, e.g.1,2,3, etc.	“0” or “1” to enable or disable the Pin authentication. This will be disabled by default and enabled when the Base Profile is set to ‘Lync’.
Feature	Added	`reg.X.auth.useLoginCredentials="0" or “1”` X= line, e.g.1,2,3, etc.	To enable or disable the login credentials. The treatment of this parameter has not changed by the PIN Authentication Feature.
Feature	Added	`reg.X.auth.loginCredentialType="usernameAndPassword"` X= line, e.g.1,2,3, etc.	To enable the type of login credentials. Below are the available options: 1. loginCredentialNone 2. usernameAndPassword 3. extensionAndPIN The default value is loginCredentialNone.

Understanding Updates to UC Software 4.1.0 Rev B

Polycom UC software 4.1.0 Rev B is for use only with Microsoft® Lync® Server using a single registered line.

UC software 4.1.0 Rev B supports the following Polycom devices:

- SoundPoint IP 321, 331, 335, 450, 550, 560, and 650 (The SoundPoint IP 670 is not supported.)
- SoundStation IP 5000 conference phones
- SoundStation Duo conference phones
- VVX 500 business media phone

Note: UC Software 4.1.0 Rev A with SpectraLink 84xx Series Wireless Handsets

If you are using SpectraLink 84xx series wireless handsets, you must use UC software 4.1.0 Rev A. For limitations and enhancements to UC software 4.1.0 Rev B, see the section Understanding UC Software 4.1.0 for the SpectraLink 84xx and VVX 500Understanding UC Software 4.1.0 for the SpectraLink 84xx and VVX 500 in this document.
For further details about support for SpectraLink series, see SpectraLink 8400 Wireless Telephones on the Polycom Support web site.

New or Enhanced Features

78286 Added support for enabling/disabling auto discovery per line

74716 Add support for dialing by extension

Enhanced Capabilities

80554 When the phone language is set to any non-English language and locked, the language on the phone is preserved even after a reboot.

80519 When a new user is added to an existing IM conversation on a PC client, the added user's phone does not ring anymore when there is a join request.

80144 On the private line, the phone remains quiet when the ring type is set to silent.

80081 Enable/disable of Polycom desktop connector (PDC) from the phone's web user interface work as per the design.

79955 The phone automatically installs the CA certificate and registers to Lync sever after reboot, when the certificate is deleted from the phone menu.

79916 The line labels are now updating after performing update configuration on the phone.

79797 The message waiting indicator (MWI) no longer blinks after the user signs out of the phone.

79225 The phone logs out the non-default user automatically as per the set time in the automatic logout parameter.

79070 Enabling login credentials of the user on the phone's web user interface automatically disable the user domain id/credentials on the phone.

77192 Adding a '+' sign to the Line Identification address via the phone's Web Configuration Utility is now displayed correctly.

Configuration File Enhancements

For information on enhancements to UC software 4.1.0 Rev B configuration files, see Configuration File Enhancements for UC software 4.1.0.

For details on all UC software 4.1.0 configuration files and parameters, refer to the *Polycom UC Software 4.1.0 Administrators' Guide* on the Polycom PartnerConnect web site.

Understanding UC Software 4.1.0 for the SpectraLink 84xx and VVX 500

Polycom UC software 4.1.0 (Rev A) was released to support only the SpectraLink 84xx series wireless handsets and the VVX 500 business media phones. UC software 4.1.0 Rev B supports the VVX 500. If you are using SpectraLink 84xx series wireless handsets, you must use UC software 4.1.0 Rev A. This section explains enhanced features and capabilities of UC software 4.1.0 Rev A.

UC software 4.1.0 is a software upgrade for phones that are qualified to deliver direct interoperability for Lync.

New or Enhanced Features

42163 Added support for simplified best-effort SRTP.

66597 Added support for Microsoft STUN/TURN/ICE.

68649 Added support for Lync Certificate Provisioning using MS Web Ticket.

68652 Added support for Microsoft E911.

68653 Added support for Lync Call admission control.

68802 Added support for Lync In-band provisioning.

68803 Added support for Lync server address discovery.

68654 Added support for Lync Media bypass.

69089 Added support for Lync Private incoming line.

69094 Added support to switch over to local ring when early media fails.

69096 Added support for Lync dial plans.

69106 Added support for Branch office resiliency (BOR) feature.

70673 Added support for alternative call forwarding identities.

74216 Added support for video synchronization with Lync client.

74567 Added support for Microsoft Web Ticket Client Protocol.

74510 Added ability to route all outbound requests via Outbound Proxy Server with different callee and caller URI domains.

74557 Added support for a manual configuration re-sync with the Lync provisioning server.

74616 Added support for extension-based dialing.

74894 Added ability to retrieve Lync server root certificate automatically using DHCP Option 43.

75141 Added Base profile menu option for easy out-of-the-box experience.

77281 Added support for Lync mode on phones (Lync Base profile).

Enhanced Capabilities

79321 Significant improvements made on the battery threshold levels (*applies to SpectraLink 8400*).

79063 The phone now updates the presence status information for all watched buddies (*applies to VVX 500).*

78107 The phone now displays a warning message upon reaching the maximum number of buddies.

78100 On the phone Presence - Idle timeout settings, changing the *office hours* timeout value will not change the *Off hours* timeout value.

78087 When a call is made from a SpectraLink Wi-Fi phone, the receiving phone now displays the complete number of the caller.

77089 In a Lync environment, any changes to the buddy presence state are now immediately reflected on the presence icons and status.

76995 In a Lync environment, phone call list details are no longer uploaded to the provisioning server when setting the configuration parameter `feature.callList.enabled=”0”` (*applies to VVX 500*).

76981 The phone now displays call forwarding icon and forwards calls correctly when the function is enabled from the phone web user interface.

76446 Calls to PSTN network now work correctly, even when Media Bypass is enabled on the PSTN gateway.

76261 The phone password field for Lync configuration now remains empty when no password is set, thereby allowing the user to set a new password.

76194 Incoming calls now first display as a pop-up before being minimized to call appearance/filter view on the phone.

76028 The Polycom Desktop Connector (PDC) has been improved and now connects to the user's laptop/PC after installation.

75874 Only the internal headset ring tone is played on Jabra PRO 9450 headsets (*applies to VVX 500*).

75872 The phone ringer now functions correctly while user is attaching/detaching a USB headset during an incoming call.

75743 Corrected the usage of a configured outbound proxy server address for Voice quality monitoring feature on phones.

75715 Local directory can now save contacts searched from the phone's corporate directory, even when the local directory is disabled.

75694 Adding/removing a USB headset no longer affects phones configured with EHS headsets (Jabra, Plantronics or Sennheiser).

75674 URL dialing is now possible between unregistered phones.

75643 "Call list display" now functions correctly when the configuration parameter `feature.callList.enabled="0"`.

75605 Contacts containing long information fields (first name, last name, etc.) in detail view now display correctly on the phone.

75431 PTT and Paging feature has now been enhanced to user iLBC codec.

75355 Improved the synchronization of contacts and addition/deletion between the Lync MOC client on PC and Phone IM.

75245 Buddy presence status is now updated when the phone is disconnected from network.

74901 When the lock feature is enabled, after a phone reboot, the emergency/authorized call list displays when the user tries to place a call using the headset/speaker key.

74888 Host status display during a multiparty Lync conference call now functions correctly.

74175 When the phone presence state is set to Be Right Back, it no longer changes to Offline when the phone is left idle for a long period of time.

73797 Caller details now display correctly for participants in a Lync consultative transfer call.

72518 On the Lync soft client, only mobile platforms are now categorized as mobile.

70723 The phone now fetches the correct available software from the Polycom provisioning server when the parameter `upgrade.plcm.server.url` is set correctly.

57864 Changes to SRTP parameters now take effect immediately, without rebooting the phone.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC software 4.1.0 configuration file parameters.

Web Info: Find Detailed Descriptions of Parameters and Values

You can find detailed descriptions of these parameters and their values in the *Polycom UC Software 4.1.0 Administrators' Guide* on the Polycom PartnerConnect web site.

UC Software 4.1.0 Parameter Enhancements

File	*Modification*	*Parameter*
feature	Added	`apps.telNotification.appInitializationEvent`
feature	Added	`apps.telNotification.networkUpEvent`
feature	Added	`apps.telNotification.uiInitializationEvent`
feature	Added	`apps.telNotification.taInitializationEvent`
feature	Added	`apps.telNotification.uiInitializationEvent`
feature	Added	`device.baseProfile device.baseProfile.set`
feature	Added	`dialplan.applyToForward`
feature	Added	`dialplan.x.applyToForward`
feature	Added	`locInfo.x.A1`
feature	Added	`locInfo.x.A3`
feature	Added	`locInfo.x.country`
feature	Added	`locInfo.x.HNO`
feature	Added	`locInfo.x.HNS`
feature	Added	`locInfo.x.label`
feature	Added	`locInfo.x.LOC`
feature	Added	`locInfo.x.NAM`
feature	Added	`locInfo.x.PC`
feature	Added	`locInfo.x.POD`
feature	Added	`locInfo.x.PRD`
feature	Added	`locInfo.x.RD`

File	*Modification*	*Parameter*
feature	Added	locInfo.x.STS
feature	Added	log.level.change.afe
feature	Added	log.level.change.ice
feature	Added	log.level.change.loc
feature	Added	log.level.change.tickt
feature	Added	log.level.change.xml
feature	Added	np.custom1.ringing.privateLine.tonePattern
feature	Added	np.custom1.ringing.privateLine.vibration
feature	Added	np.meeting.ringing.privateLine.tonePattern
feature	Added	np.meeting.ringing.privateLine.vibration
feature	Added	np.normal.ringing.privateLine.tonePattern
feature	Added	np.normal.ringing.privateLine.vibration
feature	Added	np.silent.ringing.privateLine.tonePattern
feature	Added	np.silent.ringing.privateLine.vibration
feature	Added	prov.login.lcCache.domain
feature	Added	prov.login.lcCache.user
feature	Added	reg.x.dialPlanName
feature	Added	reg.x.lisdisclaimer
feature	Added	reg.x.lync.autoProvisionCertLocation
feature	Added	reg.x.ringType.privateLine
feature	Added	reg.x.serverAutoDiscovery
feature	Added	reg.x.srtp.simplifiedBestEffort
feature	Added	sec.srtp.simplifiedBestEffort
feature	Added	softkey.feature.simplifiedSignIn
feature	Added	tcpIpApp.ice.mode
feature	Added	tcpIpApp.ice.password
feature	Added	tcpIpApp.ice.realm
feature	Added	tcpIpApp.ice.username

File	*Modification*	*Parameter*
feature	Added	`tcpIpApp.ice.stun.passwordServer`
feature	Added	`tcpIpApp.ice.stun.server`
feature	Added	`tcpIpApp.ice.stun.udpPort`
feature	Added	`tcpIpApp.ice.tcp.enabled`
feature	Added	`tcpIpApp.ice.turn.server`
feature	Added	`tcpIpApp.ice.turn.tcpPort`
feature	Added	`tcpIpApp.ice.turn.udpPort`
feature	Added	`voice.page.handsfree.rxag`
feature	Added	`voice.ptt.handsfree.rxag`

Understanding Updates to UC Software 4.0.2 Rev B

This section details updates to UC software 4.0.2 Rev B.

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).*

77038 Added support for early media followed by local ringback.

Enhanced Capabilities

73667 The syslog counter on the SpectraLink 8400 phones are updated accordingly when audio packets are received.

75557 On the web interface of the phone, the logging module parameter Wi-Fi Manager log value is updated in the field help section when the mouse is over parameter name as well as its value as (*applies to SpectraLink 8400*).

76057 When a phone is registered with a single shared line, the Join soft key is no longer displayed inappropriately when the monitored phone puts the call on hold.

76084 In a shared BLF line scenario, the monitoring phone no longer resumes the call from the monitored phone without playing any busy tone.

76095 In a shared BLF line scenario, phone now displays the incoming call status of the monitored phone (*applies to VVX 500*).

76131 The phone default call appearance eliminates the display of remote on hold calls (*applies to VVX 500*).

76321/76515 The phone does not reboot when multiple URI's are pushed to the phone in frequent intervals. *(Customer issue ID VESC-1635, VESC-1650).*

76427 The phone displays only the incoming call received from a line that is also being monitored using BLF (used to display this as two calls in the list) (*applies to VVX 500*).

76467 Improved Call list display for caller ID with long names (*applies to VVX 1500*).

76591 The phones display the correct Asian language fonts for the Lync contacts.

76679 Unauthorized request for configuration files using phone web interface is now restricted.

76754 The Filtered call view of a BLF monitored line is shown properly when the phone is in off-hook state.

76862 During an active BLF call session, frequent pressing of the BLF key on the origination phone no longer causes the caller ID information to be blanked out.

76889 The PPT key can now be configured as a Speed dial key on the SpectraLink 8400 phones.

76911 The dialer screen UI on the phone is refreshed when an incoming ringing call is terminated.

76934 On-hook dialing work as expected when the phone has an incoming call and the remote party ends the call.

76939 The phone highlights the last incoming call as per the order in call appearance screen (*applies to VVX 500*).

76940 Incorrect soft key options no longer displayed on the BLF monitoring phone when there is an incoming call in certain scenarios.

76944 Auto Answer now works correctly when alert information header carrying the string within the angle brackets '< >' is received.

76946 The phone current draw is optimized as per its state (*applies to SpectraLink 8400*).

76947 The audio/ sound effect termination is always on the dock station when there is an active call on the phone (*applies to SpectraLink 8400*).

76949 The browser application on the phone times out as expected (*applies to SpectraLink 8400*).

76976 The phone no longer inadvertently goes off-hook on line1 instead of line2 when the user presses the second line key while lifting the handset with call hold on line1.

77099 Manual configuration of the IP address no longer causes a '*network is down' message to be displayed* (*applies to SoundStation IP 6000, 7000*).

77135 Addressed some Directed call pickup failures in certain situations.

77173 Turning the backlight OFF on the phone no longer sends the browser to the session list (*applies to SpectraLink 8400*).

77226 When the phone is registered to a Lync line and another call server, the Lync contact presence subscription is now correctly sent through the Lync registered line.

77363 Addressed the issue that could cause a reboot when the user has many buddy contacts configured (*applies to VVX 500*). *(Customer* issue ID VESC-1763)

77396 Addressed a phone reboot issue in a certain ACD/call center configuration (*Found in UC software 3.3.2*).

77549 Addressed an issue relating to use of the 'Join' key which was displayed even when only one call was in progress (*applies to SoundPoint IP 33x, SoundStation Duo and SpectraLink 8400 models*).

77626 Two way audio between the phones is now working as expected after resuming the call from MOH when `voIpProt.SIP.musicOnHold.uri` or `reg.X.musicOnHold.uri` is used for the address of the MoH server.

77749 The registration failing issue with Lync server front end due to error in Subject Alternative Name (SAN) validation implementation is addressed.

77830 Addressed an issue where when the phone is configured with certain languages options, attempting to select LLDP or CDP using Ethernet caused a reboot (*applies to SoundStation IP 6000*).

Configuration File Enhancements

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

UC Software 4.0.2 Rev B Parameter Enhancements

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 UC software 4.0.2.

Note: Ensuring Digitmap Compatibility between UC Software Versions 4.0.1 and 4.0.2 Rev B

The existing digitmap `dialplan.digitmap = [2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT` in the UC software 4.0.1 release where the phones were automatically pre-pending a + to outbound calls is now removed in UC software 4.0.2 Rev B. For UC software 4.0.2 Rev B to be backwards compatible to UC software 4.0.1, the digitmap should be `dialplan.digitmap=RR+R[2-9]11|0T|RR+R011xxx.T|RR+R[0-1][2-9]xxxxxxxxx|RR+R[2-9]xxxxxxxxx|RR+R[2-9]xxxT`, or if the digitmap is to apply for a certain line, use `dialplan.1.digitmap=RR+R[2-9]11|0T|RR+R011xxx.T|RR+R[0-1][2-9]xxxxxxxxx|RR+R[2-9]xxxxxxxxx|RR+R[2-9]xxxT`.

Understanding Updates to UC Software 4.0.2 (Limited Release)

This section lists the additions, changes, removals, enhancements and configuration file parameter changes to UC software 4.0.2.

Note: UC software 4.0.2 is a limited release that was distributed only to select partners and customers and is shipping with SoundStation Duo. 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 VVX 1500*).

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.

Enhanced Capabilities

69469 The display name with special character < or > causes phones to respond with a bad request.

73614 An unintentional touch on the phone initiates a call automatically *(applies to VVX 500).*

73946 On the Trapeze/Juniper infrastructure, when multiple SpectraLink phones are involved in a call, one or more phones may lose wireless connectivity.

74292 The Bluetooth radio can now be activated on SpectraLink 8400 phones.

74427 On a redirected call, the phone now sends a PRACK (acknowledgement) message.

75299 The dialpad key presses are now captured when hot dialing from the idle browser screen *(applies to VVX 500).*

75419 The ADHOC conference call now works when there is a + sign, for example, (SIP:voip+world@voipworld.com) in the Sip URI contact header.

75632 Slowness in dialing with large number of contacts on the phone is resolved *(applies to VVX 500).*

75686 The up and down arrow keys are now available on the phone during call transfer *(applies to SoundPoint IP 450.)*

75726/75716 The phone numbers dialed using the auto complete remembers the line info on which the calls are placed earlier *(applies to SpectraLink 8400, VVX 500 and VVX 1500).*

75811 Reassigning the line keys preserves the presence information.

75888 A scrolling status message is now displayed when a line is unregistered on the phone.

75945 When the phone is off hook, auto dialing remember the line information *(applies to SpectraLink 8400, VVX 500 and VVX 1500).*

75954 The overlap of idle browser on the call list screen is now set *(applies to VVX 1500).*

75949/76258 The SoundStation Duo phone complies with clause 5.5.1.4.1 of Australian spec S002 of the Australian analog telephony specification as per clause 5.5.1.4.1 of Australian spec S002.

76141 The call lists on the VVX 1500 phone get updated as required.

76171 The phone no longer continuously reboots on reassignment of the line keys.

76229 A break down observed on the phone monitoring a BLF contact is now fixed. *(Customer issue ID VESC-1670)*

76315 The pop-up message *Error Line: Unregistered* will no longer appear as a result of an absence of a register request.

76316 The default hookflash timing is set on the SoundStation Duo when the country is set to Australia.

76329 Soft keys on the VVX 500 are now responsive to touch.

76376 An unregistered pop-up no longer appears the on SoundStation Duo in PSTN mode.

76379 Double quotes appended to the calling party display name on a shared line are now removed.

76408 Shared lines will continue to ring when another phone with the same shared line answers an incoming call on another line appearance.

76420 After reboot, the phone will correctly display incoming calls of a monitored BLF contact even on the first call.

76422 All the incoming call appearances on a BLF monitored phone are displayed when the monitoring phone cancels the call to the BLF contact.

76423 In a BLF monitored scenario, multiple calls to a monitored phone display the incoming call appearance and call counter appropriately.

76428 The phone will properly display incoming call appearance after terminating a call on phone using the End Call soft key.

76535 The phone will now correctly display a warning icon and a pop-up message for unregistered H.323 lines. *(applies to VVX 1500)*

76600 The blank call appearance on the monitoring BLF enabled phone is set.

76601/76685 In a multi-party BLF enabled call, the widget displaying the call appearance, counter, icons, and the indicator is updated with the appropriate incoming and outgoing call status.

76690 The phone now functions normally with the call appearance of the monitored BLF contacts.

76694 The phone will no longer crash when a monitored line ends the call that is associated with the remote call appearance screen.

76838 On a PSTN line the invocation of redial is restricted by pressing the # key after entering the digits *(applies to SoundStation Duo).*

76849 On a PSTN line pressing the # key is now restricted to send the dialed number only once *(applies to SoundStation Duo).*

76875 The phone now picks up the call forwarding settings from the override file after a reboot.

Configuration File Enhancements

Refer to the following table for a list of new parameters. Note that these configuration parameters are detailed in *Feature Profile 76179: Hoteling and ACD*, which will be made available on Polycom Profiled UC Software Features.

UC Software 4.0.2 Parameter 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 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 VVX 500 and SoundStructure VoIP Interface sip.ld files in the software release package.

Understanding Updates to UC Software 4.0.1

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to the 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 SoundPoint IP 321, 331, 335, 450, 550, 560, 650, 670, VVX 500, 1500, SoundStation IP 5000, SoundStation Duo, 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 SoundPoint IP 321, 331, 335, 450, 550, 560, 650, 670, VVX 500, 1500, SoundStation IP 5000, SoundStation Duo, and SpectraLink 84xx*).

70122 The 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 VVX 1500*).

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

72310/74129 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 SoundPoint 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 SoundPoint IP, all SoundStation IP, and SoundStation Duo*).

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 SoundPoint IP 450*).

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*).

Enhanced Capabilities

55237 Dialing a semicolon using on-hook dialing no longer displays the off-hook dialing dialog (*applies to VVX 1500*).

61038 The phone no longer becomes unresponsive to hard key and touch screen presses for several seconds (*applies to VVX 1500 phones provisioned with CMA using UC software 3.3.1*).

66864 Using a dial plan containing #, when a user dials #1#2#, the phone now sends out an invite message containing %231%232%23 (*applies to SoundStation Duo*).

68356 The phones can now fragment packets when instructed to by an ICMP message (*applies to SoundStation IP 6000 and 7000*).

68501 When using Exchange Calendaring, the passcode now enters automatically (*applies to SpectraLink 84xx*).

68835 The phone can now properly sniff EAP type frames (*applies to VVX 1500*).

69020 SoundPoint IP phones capable of downloadable fonts now correctly display certain Czech characters.

69540 A call dropped by the other party no longer displays as a held call.

69558 An Avaya 10x0 and a VVX 1500 can successfully establish a video call.

69882 In PSTN only mode, a received call is now recorded in the *Received Calls* call list when the call is finished (*applies to SoundStation Duo*).

69883 In PSTN only mode, an incoming call is now recorded in the *Missed Calls* call list when the call is not answered (*applies to SoundStation Duo*).

70228 The phone no longer reboots when attempting a conference using an SCA line (*applies to SoundPoint 321, 331*).

70542 The registered line icon and BLF icon are no longer corrupted in the SpectraLink 84xx handsets.

70944 A dialed number no longer overwrites a registered line's label if the label is very long (*all SoundStation phones*).

71041 The phones can now play audio from the Lync voicemail system (*applies to SpectraLink 84xx*).

71348 Calls between a VVX and RMX no longer cause the VVX to crash and reboot.

71368 Remote shared line activity no longer affects local phone presence.

71433 The phones crash after loading `se.pat.callProg.dialTone` parameters and pressing the New Call button (*applies to SoundPoint IP 450, 650, and SpectraLink 84xx*).

71604 The configuration parameter `sec.TLS.SIP.strictCertCommonNameValidation` can be updated without requiring a phone reboot.

71616 The conference feature can now properly handle a 480 response to a BroadSoft SCA line seize SUBSCRIBE.

71763 The phone no longer sends a re-INVITE to the conference server after sending a REFER for each leg of the conference (*applies to SoundPoint 321, 331*).

71764 Sennheiser and Jabra headsets can now go off-hook after switching to a different headset type (*applies to SoundPoint IP 335, 450, 550, 650*).

71802 Warning icons and records are now removed after the error condition is removed (*applies to all SoundPoint IP*).

71829 An EFK has been created to display the corporate directory Advanced Find menu from the idle screen (*applies to VVX 1500*).

71910 The text 'Enter password' in the Advanced menu is now translated when switching phone languages.

72113 The Polycom Quick Barcode Connector icon now appears and disappears for both multiple and single endpoint modes (*applies to SpectraLink 84xx*).

72250 The phone no longer reboots when queued messages are accessed on the phone (*applies to SpectraLink 84xx*).

72368 URL Dialing from the call list is now fully disabled when `feature.urlDialing.enabled=0` (*applies to SpectraLink 84xx*).

72442 The SoundPoint IP 321 and 331 phones now display the correct call x/y widget when `filterReflectedBlaDialogs=0`.

72298 The Join soft key is no longer missing after establishing the maximum number of calls on all the lines using when using a Sylantro call server (*applies to SoundPoint IP 650 and VVX 1500*).

72469 When `feature.urlDialing.enabled` is set to 0, the phone accepts contact entries with a contact number longer than 10 digits (*applies to SpectraLink 84xx*).

72739 The phones can now play the audio files using the microbrowser (*applies to all SoundStation phones*).

72800 Lines registered to a Microsoft Lync 2010 server now display in the Ring Type menu (Menu > Settings > Basic > Ring Type).

72820 The VVX 1500 phone can be registered with up to 29 lines, but a max of 24 can be displayed.

72822 The phone no longer gets into a bad state (which required an auto-reboot) upon receiving two consecutive 401 to a line-seize SUBSCRIBE during conference initiation.

72837 The speed dial icon no longer disappears after a Reset Local Config or Reset Web Config option is selected (*applies to SoundPoint IP 321, 331, 331C, 335, 335C*).

72895 The phone override file is no longer created on the provisioning server when values are not changed through the phone (*applies to VVX 1500*).

72907 Improved the phone's conference call management when a cell phone is connected to the 2.5mm port (*applies to SoundStation Duo and SoundStation IP 7000*).

72961 Bellcore Caller ID detection in PSTN mode works reliably now (*applies to SoundStation Duo*).

72996 A conference call between three parties now successfully connect all parties after there is a no response to line seize SUBSCRIBE.

73027 A Plantronics Savi740 EHS headset no longer has intermittent pairing issues with a VVX 1500.

73054 Call appearances are now displayed correctly when pressing the New Call soft key while there is an incoming call (*applies to SoundStation Duo*).

73084 Key in Keypad Diagnostics menu is now translated on SoundPoint IP 321, 331, and 335 phones for all languages.

73145 Regulatory Domain Error when radio set to 802.11a and band1 is set to P6 (*applies to SpectraLink 84xx*).

73153 Hot dial window now disappears after auto answering a call (*applies to SoundPoint IP 450, 650*).

73183 The phone now displays the error message 'Network link is down' when the DHCP server is down (*applies to SoundPoint IP 335, 450, 560, 670*).

73195 The bootloader menu for WEP has the correct spelling of Encryption (*applies to SpectraLink 84xx*).

73227 The phone UI now displays the proper network parameters such as IP address, subnet mask, and IP gateway when DHCP is enabled (*applies to SoundStation Duo*).

73238 Changing the options in the directory search, then logging into the phone no longer causes the phone to restart (*applies to SoundPoint IP 321, 331, 331C, 335, 335C*).

73247/74912 In the Quick Setup menu, user entry fields are now set to numeric as default.

73259 Simultaneous incoming and monitored BLF calls are now both displayed on the unified call appearance list (*applies to SoundPoint IP 450, 550, 560, 650, 670*).

73263 Ending a PTT or page no longer causes the active call appearance to disappear for 10 seconds (*applies to SoundPoint IP 450, 550, 560, 650, 670*).

73264 A phantom call appearance no longer displays when there is an active BLF monitored call and the phone has another call appearance.

73290 The User Profiles Log In soft key is no longer missing when the feature presence is enabled (*applies to SoundPoint IP 670*).

73401/74688 With intercom configured, the handset now rings once and incoming calls are answered automatically (*applies to SpectraLink 84xx*).

73405 The Missed Calls list no longer displays on a peer shared line if the call was barged in from a remote shared line (*applies to SoundPoint IP 321, 331, and 335*).

73409 In the Web Configuration Utility, all instances of the text 'extension module' have been replaced with 'expansion module'.

73419 When the IP 670 is configured for 34 lines using three expansion modules, a large contact directory, LDAP, idle browser, and a microbrowser, it no longer crashes due to a lack of memory.

73459 The User Profiles feature is now fully supported on the SoundStation Duo.

73527 If barge-in is enabled on a shared line, remote active calls will not appear.

73561 When using the Web Configuration Utility, the phone now updates the line key icons properly when static BLF is configured (a*pplies to SoundPoint IP 450, 550, and 650*).

73697 The flash timing is now correct for France and Singapore (*applies to SoundStation Duo*).

73749 Plugging in (or in and out) a 2.5mm mobile phone or PC cable no longer causes the phone to reboot (*applies to SoundStation Duo*).

73910 The phone's LCD contrast no longer turns darker after upgrading to UC software 4.0.0 (*applies to SoundPoint IP 450*).

74018 The phone no longer locks up and reboots when there are a large number of incoming calls (*applies to SoundPoint IP 650*).

74045 The phone can now boot up to the idle screen properly after being issued a check sync to enable the paging feature soft key.

74050 The phone no longer segfaults on boot due to language dictionary files (*applies to all SoundPoint IP*).

74060 OAI PT Select Connections are now accepted by phone before an OAI call is answered by the Start key (*applies to SpectraLink 84xx*).

74167 In the Web Configuration Utility, the authentication password can no longer be seen in clear text when opening the line page source code.

74259 In PSTN mode, the phone now displays the date and time information on the idle screen after a reboot (*applies to SoundStation Duo*).

74431 The phone no longer generates a beep sound when a monitored user goes off-hook (*applies to SoundPoint 650*).

74956 The phone will no longer go into an INVITE loop and reboot if it receives a 503 response to its initial INVITE message (*applies to all SoundPoint IP*).

74991 A call between a VVX 1500 and CX series phone can now be properly resumed after it has been held for longer than 30 seconds.

75151 Audio files are now directly downloaded to the ramdisk.

75194 The phones now use an outbound proxy when an outgoing call's URI domain is different from the caller's domain.

75334 Fail over on a 503 response can now be disabled.

75458 The maximum values for the DNS TTL parameters in the static cache have changed to 2147483647.

75485 The default input type for the Unavailable Code field is now numeric.

75491 Ampersand characters are now escaped properly and are no longer being stripped off of a URL (*applies to VVX 500, 1500*).

75743 The Voice Quality monitoring feature now uses an outbound proxy server address for a SIP Publish.

75600 The call appearance for an outgoing call no longer displays the *transport* string.

75618 Users can now place a call from the Placed Calls call list when the original call was placed using a Click-To-Dial Refer message with Refer-To: header as sip:*number* ext. *number @IPaddress*; transport=TCP (*applies to SpectraLink 84xx*).

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the UC software 4.0.1 configuration file parameters.

Web Info: Parameters Changed in UC Software 4.0.1

The following table includes parameters modified in 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.

UC Software 4.0.1 Parameter 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
debug	Added	`video.iFrame.period`	Video parameter
techsupport	Added	`voice.usb.headset.rxdg`	Audio parameter

File	*Modification*	*Parameter*	*Modification Description*
techsupport	Added	`voice.usb.headset.txdg`	Audio parameter
site	Added	`voice.volume.persist.usb Headset`	Audio parameter
sip-interop	Added	`voIpProt.SIP.conference. parallelRefer`	Call feature parameter
site	Added	`webutility.language.plcm ServerUrl`	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.failoverOn503 Response`	The default value changed from 1 (enabled) to 0 (disabled).

Understanding Updates to UC Software 4.0.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to 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 SoundPoint IP 3xx).*

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 phones 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.

Enhanced Capabilities

31158 Instant messages can now be sent if `msg.bypassInstantMessage=1`. The phone menu will no longer be bypassed after pressing the Messages button.

38407 After reboot, the phones now transmit a TCP message to the outbound proxy address.

39249 Distortion on sound from a gateway call back has been removed (*applies to SoundPoint IP 330, 550, 560, and 650*).

40103 The file name of a file copied to a full USB stick full no longer displays (*applies to SoundPoint IP 650 and 670*).

40393 The backlight adjustment has been adjusted to work correctly when the incoming call times out.

41662 The buzzing sound heard on the far end user in hands free mode when a call is answered while ringing has been removed (*applies to SoundPoint IP 550, 650, and 670*).

41897 The sound heard on the phone when attempting to cancel a conference or transfer has been removed.

42427 On the phone microbrowser, the thin line over the data field has been removed.

42442 The Select hard key has been made functional in the speed dial menu (*applies to SoundPoint IP 330*).

43822 When there is an active call, the backlight now adjusts properly.

43846 The menu widget now scales to the correct size of the menu (*applies to SpectraLink 84xx*).

43864 The Soft key, line key and status widgets can now be scaled (*applies to SpectraLink 84xx*).

44981 The phone now seizes the correct line for speed dial when `call.stickyAutoLineSeize.onHookDialing=1`.

45411 The SoundPoint IP 550 and 650 speaker phone volume can now be adjusted via gain settings for Rx audio.

45806 An unsupported format message no longer appears when trying to play a short WAV file.

45889 Lighten and Darken soft keys now display for the selected background image only in the Background menu (*applies to SoundPoint IP 450, 550, 560, and 650*).

45900 The time and date now display on the multiple call appearances screen (*applies to SoundPoint IP 450*).

46134 The phones now play a default ringtone when the ringtone size is larger than the tone quota or the ringtone is not in the cache.

46170 On the phone menu, the Local Directory or Corporate Directory cursor can now reach the end of the highlighting bar (*applies to SoundPoint IP 450*).

46773 On the microbrowser, the backspace soft key is now made available when the user enters a character to the input box (*applies to SoundPoint IP 320, 330, 430, and 450*).

48153 In the phone menu password settings, deleting a character before the character timeout now clears the last asterisk symbol.

48217 When `ramdisk.nBlocks=0` is set, `ramdisk.nBlocks` no longer consumes extraneous memory.

48753 The XML dictionary download no longer fails when the dictionary file size exceeds the defined size.

50234 The phone no longer crashes while starting a native application (*applies to SpectraLink 84xx*).

50633 When the user enters text for a contact in the Contact Directory, the backspace soft key now appears before the character selection widget for the first character has closed (*applies to SoundPoint IP 330 and 331*).

50735 After a conference call, redial now dials the last dialed number (*applies to SoundPoint IP 330 and 335*).

50745 Pressing the hookswitch toggle quickly no longer creates a phone and headset mismatch.

50766 During a save confirmation screen, if there is a missed call, the contact can now be saved. (*SoundPoint IP 330, 331, and 335*).

50788 When the user tries to play a recorded audio file from Menu/Status/Diagnostics/Test Hardware/Audio Diagnostics on different termination points (Handsfree/Headset/Handset), the icons are now updated (*applies to SoundPoint IP 450*).

50972 SIP call format is no longer missing for URL dialing (*applies to SoundPoint IP 330 and 335*).

51238 In the Install Custom CA Certificate menu on the phone, the 1/A/a soft key is no longer missing (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).

51301 A loud ring has been removed from the speaker when canceling a conference call or switching between calls.

51493 When call forward is enabled on the second registered line, phones now display the text string Call Forward Enabled on the top of the UI (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).

51751 When there are multiple calls waiting, dropping one remote party now plays the call waiting ring on the originating phone.

51767 The phone no longer crashes when trying to add a large number of contacts.

51930 The phone now shows *Connecting:* instead of *Transfer to:* on the top UI when initiating a transfer (*applies to SoundPoint IP 330 and 335*).

51994 When the contact is highlighted in the Contact Directory, pressing the character selection widget no longer erases the highlight bar (*applies to SoundPoint IP 330, 331, and 335*).

51996 Trying to dial an invalid speed dial number no longer causes slight corruption in the phone UI (*applies to SoundPoint IP 330, 331, and 335*).

52006 The call waiting tone no longer changes to a single beep when a double beep is configured on the phone.

52007 When a call is automatically disconnected at the far end phone after time out, the current active call no longer goes on hold inadvertently.

52103 The phones no longer display the first line label in scrolling status bar when it is visible (*applies to SoundPoint IP 330, 331, and 335*).

52112 In the phone Contact Directory menu, when the 1/A/a soft key is pressed to change the capitalization of the letters, the encoding no longer resets to ASCII (*applies to SoundPoint IP 450, 560, and 670*).

52219 In the phone Audio Diagnostics Menu, pressing the left arrow to exit now restores the previously selected termination state (*applies to SoundPoint IP 450, 560, and 670*).

52270 Backlight values now match the phone menu option and override file parameters.

52276 When a call is answered automatically while a recorded WAV file is playing the audio player screen now exits properly (*applies to SoundPoint IP 650*).

52380 When the phone lines are configured to call server and presence server respectively, the presence information now displays on the first line as well as the presence line.

52489 Character encoding is no longer available while entering the contact information in the contact directory (*applies to SoundPoint IP 430, 550, 650, and 670*).

52552 The momentarily observed UI corruption/flickering during an incoming call with a long caller ID has been resolved (*applies to SoundPoint IP 330, 331, and 335*).

52560 Menu titles now fit in the phone UI (*applies to SoundPoint IP 330, 331, and 335*).

52688 Enhanced the `mb.main.idleTimeout` parameter behavior.

52692 The character entry mode now shows consistently when entering passwords in the phone menu (*applies to SoundPoint IP 450, 560, and 670*).

52734 The password field now clears after entering an invalid password (*applies to SoundPoint IP 330, 331, and 335*).

52766 The speaker/headset key LED no longer remains ON when the active call is on hold while there is an incoming call on the same line (*applies to SoundPoint IP 550, 560, 650, and 670*).

52789 The phone settings menu now has an appropriate label for the menu item in Handsfree Mode.

52917 In the phone Install Custom CA Cert menu, the character selection widget no longer remains visible when closed (*applies to SoundPoint IP 330, 331, and 335*).

52920 When a custom CA cert URL is unreachable, an appropriate message now displays on the phone.

52945 Inappropriate characters no longer display in the quick search menu (*applies to SoundPoint IP 330, 331, and 335*).

52948 When long text is entered in the quick search menu, the text now moves forward and displays the last character (*applies to SoundPoint IP 330, 331, and 335*).

53055 When composing a new instant message and if there is a new incoming instant message, the UI no longer becomes corrupted (*applies to SoundPoint IP 560 and 670*).

53101 VQMon values displayed on the SQmediator are now the same as the single SIP-Publish packet values.

53138 Instant messaging strings now have spaces in between words in all instances.

53142 Extra spaces at the beginning and end of the phone labels have been removed.

53235 Extra spaces in the phone exit menu have been removed.

54147 The phone conference screen titles are no longer truncated when using certain languages (*applies to SoundPoint IP 330, 331, and 335*).

54449 The phone no longer displays an error message when trying to edit a long contact number.

54553 When the user tries to use two lines to dial, the user no longer sees a scrolling message instead of Enter Number text (*applies to SoundPoint IP 450, 650, and 670*).

54913 In a BLF scenario, when the monitored phone places a call to another phone, the Dialog Event Package no longer contains repeated remote identity when its INVITE has received an initial 407 or 401 response.

55639 When the enhanced BLF feature is enabled on an active call, the call state no longer changes in the call appearance for monitored users (*applies to SoundPoint IP 650*).

55655 Pressing the message hard key now leads the user to voicemail (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).

55669 When the phone is configured as TCP only, and the phone receives a REFER in UDP, the phone now sends an INVITE in TCP.

55681 When transport is set to TCP, Refer-Based Click-To-Dial now works when the phone has an active call.

55852 When the phone is in the setup menu, pressing * key now always moves a page up (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).

55957 When the BLF feature is enabled, the remote call appearance screen now properly times out and does not wait until the call is ended by the monitored user.

57538 In the phone menus, field names no longer truncate when the user tries to make edits (*applies to SpectraLink 84xx*).

57625 Applications are now loading as per the order specified (*applies to SpectraLink 84xx*).

58860 When the forward feature is enabled, the number of rings set now matches the actual ring cadences.

59086 When the phone is configured to an external server like CMA, the phone clock format (12 hrs – 24 hrs) does not get affected until the phone reboots.

59202 When the phone loses an active call on hold, pressing resume no longer drops the call.

59285 The phone no longer shows a delete (<<) soft key when there are no numbers to delete (*applies to SoundPoint IP 330, 331, and 335*).

59355 The phone no longer logs error messages when it is unable to connect to any of the telephony notification URLs.

59463 When using the phone's Web configuration, the phone no longer restarts when updating telephony notification event or URL.

59478 In the phone digit map, segments longer than 40 characters no longer truncated to 40 when applied.

61100 Added the ability to override complex audio codec instance count definition for each individual codec type.

61553 The phone no longer crashes when trying to split a conference service which is unavailable.

63190 The macro $FServerACDSignIn$ now works when configuring the soft key using EFK to exercise the `ServerACDSignIn` function.

63582 The excessively long boot time resulting from FTP errors and failures has been noticeably decreased (*applies to SpectraLink 84xx*).

64389 Added the ability to set the correct TLS Profile using the Updater and/or Application UI menus.

64455 There is no longer a delay between the time the Push URL is sent to the phone and the time it takes the browser to execute the fetch URL (*applies to SpectraLink 84xx*).

64464 The phones no longer wait to auth/re-associate to AP until AP starts the full security exchange (*applies to SpectraLink 84xx*).

64693 The payload settings specified by the phone are now used by the receiving phone.

64932 When the server side Call Forward No Answer (CFNA) is enabled, the user no longer has the option to select the number of rings on the local phone.

65081 The user is now able to navigate on the phone menus having select options available (*applies to SpectraLink 84xx*).

65082 The radio performance has been improved to reduce the reported number of missed packets and high retry rates (*applies to SpectraLink 84xx*).

65255 Selected options on the menus no longer disappear when selected from the phone navigation right hard key (*applies to SpectraLink 84xx*).

65275 When persistence login is enabled for default, the user log out no longer reboots the phone.

65309 Incorrect soft key options no longer show up when certain selections are invoked in the menu/UI (*applies to SpectraLink 84xx*).

65337 The phones now provision correctly via HTTPS.

65622 In case of emergency failover, the correct information with Invite request is routed (*applies to SoundPoint IP 450, 650, and 335*).

66582 In the quick setup menu, the unwanted Ok soft key no longer appears while changing the Boot server options (*applies to SpectraLink 84xx*).

66975 The voicemail icon now displays when there is a voicemail notification on the phone (*applies to SpectraLink 84xx*).

67928 In the phone buddy list, the pop-up message *Contact already exists* has been resolved.

68345 The phone dial pad keys now wake/light up when the phone comes back from the power save mode (*applies to SpectraLink 84xx*).

68533 A reboot is no longer necessary to change the parameter `voIpProt.SIP.conference.address` in the configuration files.

69372 During an IM chat session, the phone now displays all messages using the same quick note string (*applies to SpectraLink 84xx*).

69373 When provisioning TLS applications, the CommonName of a server certificate is now configurable with the new configuration option.

69374 User interface changes have been made in the Application menu and BootROM menu for SIP and Provisioning applications respectively to make CommonName configurable.

69929 When the BLF feature is enabled, the monitored BLF line status now updates as busy after the attendant phone transfers a call to that line (*applies to SoundPoint IP 650*).

69965 When the soft key is configured with capital HTTP in the URL, it no longer tries to dial instead of using the phone microbrowser.

70258 Resetting the phone to factory settings no longer clears the password (*applies to SpectraLink 84xx*).

70458 Text on the input fields and soft keys are now properly displayed.

70547 Download Custom CA Cert screen now disappears after saving a certificate (*applies to all SoundPoint IP*).

70633 The phone no longer mishandles 403 responses to first REFER sent to phone to centralized conference server.

70732 The links on the microbrowser page are no longer hidden when the links are highlighted and moves to another link.

70908 When using user profiles, the phone no longer powers off at the user login prompt (*applies to SpectraLink 84xx*).

71046 When the phone is in a conference call and a roaming attempt is made, the access point no longer rejects the reassociation request (*applies to SpectraLink 84xx*).

71103 When the phone is configured as per call center mode, call hold is no longer treated as a missed call (*applies to SoundPoint IP 650*).

71156 When there is an active call between two parties, accepting another call no longer makes the phone vibrate continuously when the mode is set to Ring and Vibrate (*applies to SpectraLink 84xx*).

71913 Registered lines now display on Expansion modules (*applies to SoundPoint IP 650, 670*).

Configuration File Enhancements

Certain groups of configuration parameters have been modified in 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`, the following table abbreviates these parameters as `device.wifi.*`

Note: Parameters with .set

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

Web Info: Parameters Changed in UC Software 4.0.0

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

UC Software 4.0.0 Parameter 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
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

Modification	*Configuration Parameter*	*Description*
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.useLegacyPayloadType Negotiation`	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
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

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
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.*	
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 existed 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

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
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
Added	`tone.chord.misc.E3Major.*`	Tone parameters
Added	`tone.chord.misc.cs12.*`	Tone parameters
Added	`up.25mmRealTime`	Default 1 Used to configure whether a mobile phone or a PC is connected to the 2.5 mm port in the conference phones. min="1" max="2"
Added	`up.backlight.timeout.*` `The parameter up.backlight.timeout has also been added`	User preferences parameters
Added	`up.cfgWarningsEnabled`	User preferences parameter
Added	`up.displayOperMode`	Default 0 Phone can display the status i.e., Up/Down when it is working on PSTN mode(Applicable to SoundStation duo).
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

Modification	*Configuration Parameter*	*Description*
Added	up.pstnSetup	Default 0 User can setup the PSTN line by selecting the country when the phone initially boots up
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
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

Modification	*Configuration Parameter*	*Description*
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.*	Wifi parameters
Changed	apps.push.messageType	Productivity Applications parameter
Changed	apps.uc.desktop.enabled	Productivity Applications parameter
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.dhcpBootServer, device.net.IPgateway, device.net.subnetMask, and device.net.vlanId have not been changed	Provisioning parameters
Changed	device.ntlm.versionMode	Provisioning parameter

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
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
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

Modification	*Configuration Parameter*	*Description*
Changed	`up.25mm`	User preferences parameter
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
Changed	`voice.codecPref.iLBC.*`	Audio parameters
Changed	`voice.gain.rx.analog.*`	Audio parameters

Modification	*Configuration Parameter*	*Description*
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

Understanding Updates to UC Software 3.3.2

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to UC software version 3.3.2 beside their respective Polycom tracking ID number.

New or Enhanced Features

56249 Added user confirmation on the phone before placing outgoing calls as part of the click to dial behavior.

64548 Added missed-call synchronization. When local call lists are disabled on the phones, Missed calls notifications are sent from the call server to the respective users.

66464 Provides simplified display option by removing protocol tag and host details (*applies to VVX 1500D*).

66466 When the parameter `up.simplifiedSipCallInfo` is enabled, the caller ID will not display the host name for incoming and outgoing calls and the protocol information will not be shown.

66624 Geographical redundancy enhancements.

68836 Added support for Sennheiser EHS headset to the phone menus and configuration.

70475 Extended the `dialplan.digitmap` String to support up to 100 from 30 segments.

Enhanced Capabilities

27469 Local Conferencing is no longer disabled if G.729 is in the Codec preference list (*applies to SoundStation IP 4000*).

27777 The phone now plays a local hold reminder tone (*applies to SoundStation IP 4000*).

34454 If the microbrowser is enabled and it is refreshed too frequently and the pages contain large images, the phone no longer locks up.

34743 A phone no longer freezes when it receives a check-sync if the resources on the phone are heavily used by a downloaded wave files or by a large/complex microbrowser pages.

39630 When using the SoundPoint IP 330/320 phone with LCS2005, blocking a roaming buddy from the privacy list no longer prevents the user from viewing the blocked buddy's status.

45247 The SoundPoint IP 430 no longer reboot when viewing microbrowser pages and the internal memory is being used for other function/operations.

47827 Voice Quality Monitor feature on SoundPoint IP now uses the correct units for Jitter in SIP PUBLISH VQSession Reports.

49324 When dialing 99* from the phone with an integrated Polycom HDX, the * is no longer changed to a dot on the HDX (*applies to SoundStation IP 7000*).

51904 When a Polycom HDX system is configured for SIP using UDP, it now makes a video connection with the VVX 1500 phone.

52592 The phone now provisions when using the combined **sip.ld** file and a TFTP provisioning server that does not support the bulk size option (*applies to SoundStation IP 6000*).

52782 Resolved some video issues that occur when VVX 1500 phones are bridged on Polycom HDX and VSX MCUs.

55910 The phone no longer freezes/stops while preparing to boot up (*applies to SoundPoint IP 430*).

57838 When there is a call between two SoundStation IP 7000 phones along with a HDX system, the HDX9004 system added a video call to HDX9002. There is no longer a hold between the SoundPoint IP 7000 phones.

58177 When attempting to do a blind transfer from a PSTN line to an internal extension, three beeps are no longer heard after pressing the Send soft key. Cancelling the operations enables the call transfer.

60086 The SoundPoint IP 650 phones now send an Off-Hook or On-Hook notification when set to Auto Answer.

60131 Reassignment of the speed dial keys now functions properly (*applies to SoundStation IP 5000, 6000, and 7000*).

60186 Call connection bandwidth between the HDX and VVX 1500 now synchronizes when the VVX 1500 is in CMA provisioning mode.

60255 A noticeable high-frequency flicker has been removed from the display when an update for BLA remote hold/resume status occurs (*applies to SoundPoint IP 650/670*).

60729 The phone now honors a BLA NOTIFY when the version number in the message body has increased by more than 1.

60733 Can now establish a local conference bridge by using a speed-dial key, BLF line key or via call lists.

60984 The `dir.local.contacts.maxNum` parameter no longer accepts 0 as stated in the Administrators' Guide; `dir.local.contacts.naxNum` accepts 1 to 99 OR 1 to 9999.

61013 On the VVX 1500, the Call Rate value can no longer be set higher than Max Call Rate value when configured using the Web Configuration Utility.

61067 On the phone interface menu, pressing the Back soft key in the Authentication menu now restores the menu title correctly.

61089 A pop-up no longer appears when adjusting the ringer volume while the call is on hold. Hands-free Volume no longer appears instead of Ringer Volume (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

61145 The `dialplan.digitmap` uses up to a maximum of 767 characters. The last character is no longer truncated.

61147 The phone no longer reboots when a GET request is sent to the phone to `/TA/getParam?paramName=reg.1.ringType` (*applies to SoundPoint IP 331, 335, 450, 550, 560, 650, 670 and SoundStation IP 5000*).

61316 The message waiting indicator on the VVX 1500 shows up on the correct line after the registrations are moved to different line keys.

61955 An RTP audio delay is no longer detected when calling or receiving calls from a PSTN line.

62389 When the VVX 1500 phone is left idle for an extended period of time and a call is made between the CMA users, the pop-up on the phone user interface is no longer delayed for both incoming and outgoing H.323 calls.

62450 When the value in configuration parameter `mb.idleDisplay.home` is set to point to a URL with an image, the phones idle display no longer shows a break in the border located at the bottom-left corner.

62664 The VVX 1500 phones no longer drop the domain information when sending call setup to the gatekeeper.

62675 Calls placed with the VVX 1500 using the URL dialing now show up in the missed calls list.

62687 Disabling local call forward now stops the phone from forwarding calls.

62974 Local call forwarding rules are now disabled when the feature is disabled.

63070 Call Forward CF messages such as Call Forward *destination:Fwd:<number>* now display when DND is active.

63262 When using the SoundPoint 650 to dial a call using the Out of Dialog REFER based method, the user no longer needs to press the Speakerphone key twice in order to terminate the call.

63277 During a call, a macro which is set to simulate a soft key press no longer displays an error (*applies to the VVX 1500*).

63452 When URL dialing is disabled on the phone, transferring call initiated by BLF speed dial key no longer prompts again for the URL.

63746 During fail-back attempt, line icon shows as unregistered.

64036 Caller ID on the VVX 1500 now works properly when both Contact Directory Matching and Chinese characters are enabled.

64190 Enhanced video quality during conferences with RMX 1500.

63850/64430 Initial dialog event NOTIFY after subscribe has the correct version.

64859 When a call is made using SRTP and TLS, the far end can now hear any audio even when the SRTP packet sequence counter rolls over to zero.

64896 When Call Forwarding is on, the phone no longer updates the display before the server has confirmed operation via NOTIFY.

65014 When a VVX 1500 phone is provisioned using CMA 5.3, LDAP directory searches now return meeting rooms names when searching the CMA directory.

65207 Resolved a memory leak on the phones in specific call server environments when a call is answered from a hunt group.

65288 When server side DND is enabled, an incoming call from a white list phone number can now be picked.

65345/64862 One of the callers is no longer dropped when trying to set up a conference between PSTN users.

65617 When Auto answer is enabled, the phone no longer sends a 180 response to the server.

65754 The XML string <*key key.25.VVX 1500.function.prim=null/*> no longer disables the Menu soft key or the Menu hard key (*applies to the VVX 1500*).

65758 An extra space has been removed from each side of an umlauted character in the microbrowser idle display: G Ä rtner instead of GÄrtner.

65979 The SoundPoint IP phone quick setup menu user name entry has been set to numeric characters as the default.

66027 When the phone lock feature is enabled, trying to dial any number will no longer dial the emergency number.

66106 When a call is made from the phone from the dial pad and pressing the speakerphone key it correctly selects line 2 instead of line 1.

66217 Directed Call Pick-up soft key now works properly (*applies to the SoundPoint IP 650 and VVX 1500*).

66554 During a video call on the VVX 1500, the phone no longer drops video momentarily when there are periodic offer less re-invites/ session refresh messages from server.

66593 When trying to dial an extension of four digits by pressing only two digits, a prompt displays on the phone to enter more digits. After entering the other two digits, it no longer appends to the earlier two digits thereby resulting in a failed call (*applies to SoundPoint IP 3xx running UC software3.3.x*).

66621/66619 Created configuration parameters that allow phones to perform fail-over when a 503 response is received.

66625 The phone no longer sends three extra registration requests to primary proxy during fail-over.

66626 DNSTTL no longer counts down during fail-back that fails. TTL is reset after re-registering to secondary.

66666 The SoundStation IP 7000 phone is now being provisioned via FTP when the Windows 2003 server path MTU Discovery is disabled.

66964 Local call forward behavior is now working, as mentioned in the Admin Guide for the parameter `voIpProt.SIP.serverFeatureControl.localProcessing.cf` .

67455 The phone no longer crashes during registration with TLS (*applies to SoundPoint IP 650/670*).

67622 The phone displays the correct caller name in the user interface where there is a call from a group pickup number.

67633 Added support for the Zero-Touch Provisioning (ZTP) feature. Note that feature should not be enabled unless the phone has been registered for use with the associated provisioning system offered by your service provider.

67641 Pressing the Directory soft key on the VVX 1500 phone no longer redirects to the Advanced Find screen automatically.

67642 When using the VVX 1500 to dial contacts from the corporate directory for which the dialing entries are not filled appropriately, the phones no longer displays the attributes and no longer skips null values.

67753 The phone no longer plays ringback after the call is timed out.

67867 The phone seizes the correct line after transferring an incoming call to the line when going off-hook.

67966 When a call is made between two VVX 1500s the answering VVX 1500 which comes up after first reboot no longer plays a noise pattern on the screen before playing video.

68063 The phone no longer reboots when DHCP failover occurs.

68195 Directed Pickup using star codes, for example, *200,*300 now works on the SoundPoint IP 650.

68267 When the BLF feature is enabled on the SoundPoint IP 650 and there is an active call, a ringtone heard in low sound within the handset has been removed.

68344 Request Validation feature no longer rejects requests from another (second) server listed in the configuration files.

68376 The VVX 1500 phone no longer reboots when the DND button is pressed repeatedly during the idle state.

68382 The phone shows the correct time when the IP address on the NTP server is 12 digits.

68446 The parameter `call.hold.localReminder.startDelay` value is now honored when the value is less than 60 seconds.

68476 The SoundPoint IP 331 phone is now bootable when set with Option 60, ASCII String and DHCP Server Option 43.

69166 Placing a call is on hold in which RFC 3264 directionality attributes are present no longer results in failure of terminating music on hold session.

69421 The VVX 1500 phones no longer fail to pass special character $ in the password via DHCP Option 66.

69671 The phones accept VLAN ID from DHCP option 129.

70027/62203 Conference with Genband CS2000 now works properly.

70988 The SoundPoint IP 550, when powered by external AC power, no longer reboots when certain audio plays on full volume.

70233 When URL dialing is disabled, the user is no longer prompted for a URL when attempting to transfer calls after using directed call pickup of a monitored BLF line.

70317 During an active IVR call, if there is a second incoming call to the same line, the phone now sends DTMF.

70456 The SoundPoint IP 450 phones now show the complete text on the microbrowser screen.

71071 The inbound caller ID on the SoundPoint IP 33x now displays for new calls during an active call on the phone. Observed this in version 3.2.5 and 3.3.1

71328 Modifying the phone username via phone Web user interface no longer changes the password from numeric values to ???? (four question marks).

71947 When the phone lock feature is enabled on the SoundPoint IP 650, outbound calls can no longer be dialed using line key, handset, headset and speakerphone key.

72766 When the configuration parameter `device.set` is set on the SoundPoint IP 650/670 with a new boot server IP address, it now forces the phone to reboot.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the UC software version 3.3.2 configuration file parameters.

UC Software 3.3.2 Parameter Enhancements

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
`call.clickToDial.referBased.userConfirm`	added	accessType		Admin
		callback		DefaultCbNone
		cfgParamType		param
		default		0
		help		
		templates		sip-interop
		type		Bool
`call.shared.notifyTransferHoldAsActive`	added	accessType		Admin
		callback		DefaultCbRestart

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		cfgParamType		param
		default		0
		help		
		max		1
		min		0
		templates		hidden
		type		Bool
`device.prov.ztpEnabled`	added	callback		DefaultCbRest art
		cfgParamType		param
		default		
		help		
		max		256
		min		0
		templates		device, site, new
		type		String
`device.prov.ztpEnabled.set`	added	callback		DefaultCbRest art
		cfgParamType		param
		default		0
		help		
		templates		device, site, new
		type		Bool
`dialplan.x.routing.server.y.transport`	change d	enum		transport
		max	256	
		min	0	
		type	String	Enum

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
`dialplan.routing.server.x.transport`	changed	enum		transport
		max	256	
		min	0	
		type	String	Enum
`dir.H350.dev.transport`	changed	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	
		type	String	Enum
`dir.H350.group.transport`	changed	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	
		type	String	Enum
`dir.H350.person.transport`	changed	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	
		type	String	Enum
`dir.corp.transport`	changed	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		type	String	Enum
`mb.main.toolbar.autoHide.enabled`	added	accessType		User, Admin
		callback		CbMicroBrowser
		cfgParamType		param
		default		1
		help		Enable/Disable browser tool bar auto hide feature
		templates		applications
		type		Bool
`pnet.joinOnAutoAnswer`	added	accessType		Admin
		callback		DefaultCbNone
		cfgParamType		param
		default		0
		help		
		templates		video-integration
		type		Bool
`reg.x.outboundProxy.failOver.onlySignalWithRegistered`	added	callback		CbReg
		cfgParamType		paramArray
		defaultAll		1
		help		
		numReplace1Max		Const_NumLineReg
		templates		reg-advanced
		type		Bool
`reg.x.outboundProxy.transport`	changed	enum		transport

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		max	256	
		min	0	
		type	String	Enum
`reg.x.server.y.failOver.onlySignalWithRegistered`	added	callback		CbReg
		cfgParamType		param2DArray
		defaultAll		1
		help		
		numReplace1Max		Const_NumLineReg
		numReplace2Max		Const_NumServers
		templates		site
		type		Bool
`reg.x.server.y.transport`	changed	enum		transport
		max	128	
		min	0	
		type	String	Enum
`sec.srtp.newContextOnResume`	added	callback		DefaultCbRestart
		cfgParamType		param
		default		1
		help		
		templates		hidden
		type		Bool
`up.simplifiedSipCallInfo`	added	accessType		Admin
		callback		DefaultCbNone
		cfgParamType		param
		default		0

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		help		
		templates		new
		type		Bool
`voIpProt.SIP.dialog.strictVersionValidation`	added	accessType		Admin
		callback		CbSipStack
		cfgParamType		param
		default		1
		help		
		templates		new
		type		Bool
`voIpProt.SIP.failoverOn503Response`	added	callback		CbSipStack
		cfgParamType		param
		default		1
		help		
		templates		sip-interop
		type		Bool
`voIpProt.SIP.outboundProxy.failOver.onlySignalWithRegistered`	added	callback		CbReg
		cfgParamType		param
		default		1
		help		
		templates		sip-interop
		type		Bool
`voIpProt.SIP.outboundProxy.transport`	changed	enum		transport
		max	256	
		min	0	
		type	String	Enum
	added	accessType		Admin

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
voIpProt.SIP.serverFeatureControl.missedCalls		callback		DefaultCbRestart
		cfgParamType		param
		default		0
		help		
		templates		sip-interop
		type		Bool
voIpProt.server.x.failOver.onlySignalWithRegistered	added	callback		CbReg
		cfgParamType		paramArray
		defaultAll		1
		help		
		numReplace1Max		Const_NumServers
		templates		sip-interop
		type		Bool
voIpProt.server.x.transport	changed	enum		transport
		max	256	
		min	0	
		type	String	Enum
voice.gain.rx.digital.headset.IP _330	added	callback		DefaultCbRestart
		cfgParamType		param
		default		-12
		help		
		max		1000
		min		-1000
		templates		techsupport
		type		SInt

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
voice.gain.rx.digital.headset.IP _335	added	callback		DefaultCbRestart
		cfgParamType		param
		default		-9
		help		
		max		1000
		min		-1000
		templates		techsupport
		type		SInt
voice.gain.vol.ringer.gain.adjust	added	callback		DefaultCbRestart
		cfgParamType		param
		default		-9
		help		
		max		1000
		min		-1000
		templates		new
		type		SInt
voice.headset.rxag.adjust.IP _335	added	callback		DefaultCbRestart
		cfgParamType		param
		default		-11
		help		
		max		-11
		min		-1000
		templates		new
		type		SInt

Understanding Updates to UC Software 3.3.1F

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to UC software version 3.3.1F beside their respective Polycom tracking ID number.

Enhanced Capabilities

66743 The phones may be vulnerable to Denial of Service attacks when used in certain configurations. Sending HTTP GET requests with a broken authorization header can produce a device restart under certain circumstances in certain models of phones. For full details, refer to Technical Bulletin 66743.

Understanding Updates to UC Software 3.3.1

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to UC software version 3.3.1 beside their respective Polycom tracking ID number.

New or Enhanced Features

50065 Added support to the VVX 1500 for CMA presence.

52476 Added support for Premium extensions to server synchronized ACD feature.

55061 Added support for the Team Function feature. This feature extends the compatibility of statically configured Busy Lamp Field (BLF) to operate in a system requires the use of two URIs one for call operations and another one to subscribe for notification of dialog events. It also provides Ringing Indication and a Directed call pick-up capability in a system that does not generate RFC 4235 compliant dialog-info+xml documents.

58888 Added the ability to trigger a reboot (or configuration update) from the microbrowser. For example, *<softkey index=3 label=Reboot action=Action:UpdateConfig />*

59000 The phones now ignore BLA dialog documents (via NOTIFY) that are reflected to User Agents that are party to the dialog.

60306 The server certificate Serial Number SN is now verified against the server/proxy's A record domain names if the SRV record domain does not match the SN.

61343 The phones now provide a configurable parameter that allows the verification of the authentication tag to be disabled for received SRTP packets. The purpose of this is to allow system administrators to resolve defects in other endpoints where the authentication tag is not computed correctly. Supported parameter: `sec.srtp.noAuthRxRTP`

61389 During the 802.1x - EAPOL Logoff, the phone will recycle the LAN link (e.g. it will bring it down and up in an interval of one second) upon detecting a PC link down event. This shall force the 802.1X switch to refresh the authorized port state and start to send request for identity challenge messages. The associated configuration parameter is `sec.dot1x.eapollogoff.pcforcelanlinkreset` with values 0 - Never recycle LAN link and 1 - The phone will unconditionally recycle the LAN link upon detecting PC link down event.

61861 Corporate Directory LDAP initialization supports the bind authentication.

62115 The phones now display the full text strings of the Phone Lock feature (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

62259 The phones now display the Call Forward destination on Idle Display.

62775 The toolbar slide-out option is now configurable on the VVX 1500. The associated configuration parameter is `mb.main.toolbar.autohide.feature`

1 feature is enabled (default)

0 feature is disabled. The Autohide enable/disable buttons are no longer visible to the user in the toolbar.

Enhanced Capabilities

44337 Configured characters ;, /, ?, &, =, ~, %, \ are now escaped in INVITE messages.

55794 The SoundStation IP 6000 and 7000 phones no longer reboot upon receiving a call with incorrect SRTCP indices.

56491 As of SIP 3.2.x, the screen no longer displays the IP address of the server when disabling the Call Forwarding feature using a # code.

59824 The phone now changes all of the menu option labels into the selected language.

59843 On the VVX 1500, the caller ID correctly displays during an active call after switching (exchanging) valid logon credentials between two phones.

60015 The phone no longer sends RTP media for 2.4 seconds after call is declined with 603 Decline.

60175 When using the Contact Directory speed dial, the left and right arrow keys no longer increment and decrement the index unexpectedly (applies *to SoundPoint IP 320, 321, 330, 331, and 335*).

60514 The user password can now be changed by an administrator if the old password is unknown.

60572 An EFK soft key no longer requires at least one valid entry in <*efk.efklist /*> configuration in order to be enabled.

60645 The VVX 1500 phone no longer resets to previous values in the Edit contact menu when the mode is changed from Tel to Url.

60761 The Transfer and Conference soft keys on the SoundStation IP 5000 and 6000 are no longer absent upon the 8th active outgoing call.

60788 When operating with a sip X server, there is now music on resume from a double Music On Hold (MOH) between two phones.

60814 The Login soft key on the SoundStation IP 7000 now displays when `feature.acdLoginLogout.enabled` is set.

60831 Ringback tone no longer continues to play for 30 seconds after the SoundStation IP 6000 phone sends a BYE message.

60848 After invoking the Update Configuration menu option, the phone now returns to the idle screen.

60897 The Custom Ringer Types menu no longer uses the file name rather than configured name.

61030 The Buddy Watch presence now works on the VVX 1500 phone after it boots initially with `voIpProt.H323.enable=1`.

61031 Active call now has a timer when attempting to transfer or conference the call (*applies to SoundPoint IP 450, 550, 560, 650, 670*).

61041 The Call Server Configuration Menu now displays Options (1, 2, …) within the Menu items (*applies to the VVX 1500*).

61042 The Directed call pick-up feature now works when the SUBSCRIBE message expires=0.

61046 The Saved Certificate prompt is now shown when a new CA certificate is downloaded (*applies to SoundPoint IP 320, 321, 330, 331, 335, and SoundStation 5000 and 7000*).

61088 The VVX 1500 phone no longer freezes and reboots after making a call with `tcpIpApp.port.rtp.forceSend=1024`.

61090 The configuration parameter `voIpProt.SIP.musicOnHold.uri` is now updated after a configuration change.

61095 While dialing a URL using the on-screen keyboard on the VVX 1500, the first entered character is no longer unexpectedly deleted.

61102 The Handset or Speaker icon no longer appears instead of the Ringer icon when you adjust the ringer volume while the SoundStation 500 and 6000 phones are idle.

61104 With a shared line configured on the VVX 1500 phone, activity on the remote shared line no longer causes the idle browser content to cycle off then on.

61114 The VVX 1500 phone boots-up with the DHCP VLAN 256 DVD option.

61115 Can now answer calls after a configuration update is invoked.

61242 The configuration parameter `voIpProt.SIP.useCompleteUriForRetrieve` updates after a configuration change.

61246 The `voIpProt.SIP.allowTransferOnProceeding` XML schema lists the correct type as stated in the Administrators' Guide.

61273 Joining calls into a local conference when 1 leg is a remotely held BLA line now maintains audio between both remote users.

61314 The number of characters for custom names has been extended to 127 from 12.

61367 When dialing a number with a + sign, e.g. +492101099210, user=phone is now added to the To header.

61677 The VVX 1500 phone no longer escapes the % character as %25 when is present in the destination of a call.

61723 The VVX 1500 phone is no longer missing the first string *<?xml version=1.0 encoding=utf-8 ?>* in FAST UPDATE requests.

61779 The phone no longer reboots spontaneously from the idle state or in-use state.

61904 A call is placed with the correct signaling protocol on the VVX 1500 when the line is configured as dual line protocol.

62036 The SoundPoint IP 320 and 330 phones now send DTMF RTP EVENTS when receiving a second incoming call during an active primary call.

62114 The user can now unlock the VVX 1500 phone after the phone is locked with a password containing letters.

62325 Chinese characters no longer cause the VVX 1500 phone to become unresponsive to user requests.

62333 The correct Chinese characters are displayed in the reboot menu on the VVX 1500.

62417 When an off-hook event is received from the headset base station, the phone no longer sends three events to the base station in DHSG headsets and platforms.

62453 The phone no longer displays a ghost call appearance labeled Unknown Party if a remote party is held while the reorder tone is played locally.

62490 Enabling the Screen capture function with `httpd.enabled=0` no longer causes the phone to freeze and reboot.

62576 The phone now reboots in order to pick up new **sip.ld** file after an Update Configuration is invoked from the menu.

62642 The phone no longer plays a dial tone and RTP media when resuming on a call held at another phone.

62704 BLA presence now recovers properly on the monitoring phone when the LAN cable is disconnected and then re-connected.

62906 The phone correctly provisions using the HTTPS protocol option when using a server certificate with an older MD2 digest message algorithm.

63076 The phones with BLA lines are now able to establish more than 10 outgoing calls.

63214 The phone no longer reboots if it receives more REFERs that `reg.x.callsPerLineKey` is configured for.

64742 Using a dial plan containing #, when a user goes off-hook and dials #1#2#, the phone now sends out an invite message containing %231%232%23.

Configuration File Enhancements

Refer to the following table for a list of changes in configuration parameters. The list applies only to the changes made since UC software 3.3.0.

The configuration files, their respective parameters and defaults, as well as the provisioning methods have been simplified but extensively modified starting from UC software 3.3.0.

Before installing the software, it is highly recommended that you first familiarize yourself with the changes outlined in the Administrators Guide for the Polycom® UC Software – 3.3.0 and Technical Bulletin 60519.

UC Software 3.3.1 Parameter Enhancements

Configuration Parameter	*Action*
`attendant.resourceList.x.callAddress`	added
`attendant.resourceList.x.proceedingIsRecipient`	added
`device.cma.disableTlsForDebug`	added
`device.cma.disableTlsForDebug.set`	added

Configuration Parameter	*Action*
`dir.H350.dev.bindOnInit`	added
`dir.H350.group.bindOnInit`	added
`dir.H350.person.bindOnInit`	added
`dir.corp.bindOnInit`	added
`feature.acdPremiumUnavailability.enabled`	added
`reg.x.filterReflectedBlaDialogs`	added
`sec.dot1x.eapollogoff.pcforcelanlinkreset`	added
`sec.srtp.noAuthRxRTP`	added
`voIpProt.SIP.allowTransferOnProceeding`	changed

Understanding Updates to UC Software 3.3.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to UC software version 3.3.0 beside their respective Polycom tracking ID number.

New or Enhanced Features

23335 Configuration parameter values can now be updated at run-time.

24111 Enhanced the user interface for selecting a distinctive ringtone associated with a contact in the local directory. You can now review the ring name and play the ringtone before accepting and associating the ringtone for specific contacts.

23394 Configuring parameters are now self-contained (default parameter values) and the configuration process is more fault-tolerant.

35245 Line key behavior (configurable) has changed such that keys can now be used to hang-up/terminate calls as well as establishing calls. The associated configuration parameter is `up.lineKeyCallTerminatetype=Booldefault=0 min=0 max=1`.

38826 Added configuration parameters to expand the range of ports as well as to randomize port selection for the purpose of downloading configuration files to the phone using TCP connections.

48138 Added support for dynamic support of G.729AB and iLBC codecs. G.729AB / iLBC (*applies to SoundPoint IP 320, 321, 330, 331, 335, 450, 550, 560, 650, 670).*

48526 Simplified selection of codec configuration preferences. See Technical Bulletin 60519 for more information. This change is not backward compatible to configuration files used with previous software releases.

48690 Added the ability for users to lock the phone and restrict its access from unauthorized users. Users must enter a PIN in order to access and use the phone. Refer to Quick Tip 57215 for more information regarding this feature and configuration.

49658 Added configuration parameter to allow the phone to obtain Caller ID from the From header instead of the P-Asserted-Identity segment. The associated configuration parameter is `voIpProt.SIP.CID.sourcePreference` = P-Asserted-Identity, Remote-Party-ID, or From.

50067 Local contact directory now matches the Polycom CMA products style and user experience.

50151 Removed redundant levels of abstraction associated with arrays in configuration files.

50644 Enhanced the visual indicator of incoming calls on the VVX 1500 for the hearing impaired. Upon receiving an incoming call, the phone will ring and the display will flash on and off with a bright orange and white screen. This visual indicator can be seen even when the display is viewed at an indirect angle. The associated configuration parameter is `up.accessibilityFeatures=1`.

51121 RAM disk configuration parameters have been optimized.

51314 Added a configuration option to allow for minimal latency in order to meet JITC requirements. The associated configuration parameter is `voice.txPacketDelay`. Normal or NULL (default) = no change to Tx latency; low = low delay

51523 Added the ability to scroll horizontally caller ID information (if it is truncated when the number of characters in the caller ID string exceeds the capacity of the display).

51446 Added configuration parameters supporting TLS cipher suites.

51594 Digit map replacements no longer need to be reflected in the placed calls list.

51725 Added support for G.719 audio codec in H.323 calls (*applies to VVX 1500*).

51979 Added support for asymmetric audio codecs.

52253 Configuration parameter values modified by an administrator logon credential using the phones Web server are not permitted to be altered by user level logon credentials.

52493 Added support for MD4 encryption key (OpenSSL).

52532 The phones no longer invoke a reboot during the uploading of override files as a result of an unresponsive provisioning server (after a timeout).

52864 Enhanced the user experience of confirming a Local Directory Search (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

53021 Added support for NTLM version 2 authentication [via XMPP, LDAP and HTTP(s)] for use with CMA.

53023 Edit fields have been expanded to display additional content (*applies to VVX 1500*).

53231 Added a configuration parameter to control the behavior of terminating a 3-way conference by the conference initiator. Options now include either terminating all conference legs or allowing the other parties to stay connected.

53417 Implemented a slider bar on the VVX 1500 for adjusting levels in various menu screens.

53703 Added the ability for phones to send an 802.1x EAPOL Logoff message on behalf of an attached PC when the PC is disconnected from the data port.

53932 Presence and BLF is supported on Avaya CS2100 soft switches.

54037 Attempting a Transfer/Conf of a held party while in active call is now consistent with all phones.

54045 Registration parameters can now be modified and activated without requiring the phone to restart or reboot.

54098 Added the ability to automatically upgrade the BootBlock section of the BootROM.

54167 The BootROM and application software versions may now be obtained by using the on-board Web interface.

54301 A timestamp displays in Call Lists alongside the Caller ID.

54308 The navigation keys can now be used as a spin box control (ability to select values using the up and down arrow keys) for numeric fields (*applies to SoundPoint IP 320, 321, 330, 331, 335*).

54678 The phones can now be deployed with a pre-set language. This supports out-of-the box localization.

54928 Added a new API Telephony Event (XML) which is sent to the attached application upon a successful line registration with a PBX.

55028 The maximum size of the contact directory contact field has been increased to 128 to accommodate complex dialing scenarios.

55040/57981 Added the ability for administrators to install custom device certificates. The administrator can add private and public keys (certificate) via TLS links.

55068 Added support for Null Ciphers to be used with TLS Authentication.

55318 The Advanced LDAP Search screen now supports languages other than English.

55334 Added the ability for the tool bar on the VVX 1500 to hide automatically.

55490 The configuration Web interface has been expanded to include parameters associated with security.

55508 When a precedence call is offered to the phone, it now rings with a corresponding precedence ringtone.

55509 When a precedence outgoing call is initiated, a precedence style ring-back tone is generated.

55510 The DSCP Differentiated Services Code Point levels for standard and precedence level calls are aligned.

55513 The current precedence level of a call is now displayed.

55546 The following diacritic letters and ligature are now supported (language option selection) and can be displayed without having to change the character encoding scheme ä, ö, ü / Ä, Ö, Ü ß.

55745 The phones now generate a MLPP resource-priority Header based on the dialed number.

55985 The SoundPoint IP 7000 now displays the LogOut soft key when configured to be enabled.

56666/56668 Added dynamic codec switching.

56790 Enhanced the computation of jitter buffer parameters based on received Quality of Service QoS and expected payload size values.

56944 Enhanced the ability for application developers to implement changes to the phone's configuration. Configuration parameters can be modified via the Web interface. The Enhanced method also eliminates the need to reboot the phone in order to register the changes.

57504 A new Warble.wav file is available which can be configured as an audible ringer for incoming calls. This file will generate a loud ringer tone for phones deployed in areas with a high ambient noise background.

58103 The default maximum call data rate has been increased to 768 Kbps from 512 on the VVX 1500.

58156 The user video call rate setting parameter value options have been shortened on the VVX 1500.

58758 Enhanced the rendering performance of the browser on the VVX 1500.

58764 Added the ability of uploading configuration files representing the phones current set of configured parameter values to the provisioning server.

59307 Added a diagnostic menu option that enables the display of configuration file statistics.

60316 Added an option in the user interface that allows the user to invoke the phone to force it to re-configure itself based on newly administered configuration file parameter values.

60353 Custom ring classes (`se.rt`) can now be set to a maximum value of 17.

60363 Custom ringer chords (`tone.chord.ringer.spareX`) can now be set to a maximum value of 19.

Discontinued Features

50200/53590 Removed configuration parameters that are no longer required for custom bit-mapped graphic indicator icons.

56209 Removed support for the SoundPoint IP 430.

59917 Removed support for the animated idle display images (static idle display images are still supported).

Enhanced Capabilities

33425 On the SoundStation IP 7000, users can now reply to instant messages.

42509 Can now invoke the speed dial list using the Up Arrow key when first call is kept on hold (*applies to VVX 1500*).

43660 URL addresses on the SoundPoint IP 330 are now saved in the call list entry. When the phone receives a URL call from SPIP @xxx.xxx.xxx.xxx, the phone now saves the incoming URL call address into call list entry.

44034 On the SoundPoint IP 330, the cursor now blinks in hot dial prompt.

44278 The phone number now displays correctly on a line key when the number of digits exceeds 10.

44478 Configurable soft key features now work on the VVX 1500.

44889 The Polycom bitmapped logo now displays on the SoundPoint IP 330 idle screen.

45013 The phones no longer reboot after a check-sync request when a call is held and a new call is initiated and then cancelled.

47135 Casing of current encoding indication at title bar now match corresponding soft keys on the VVX 1500.

47542 The URL entry field on the VVX 1500 allows for 32 instead of only 28 characters.

48228 The Contact Directory now has a functional <*New Entry*> option and a correct Navigation Cluster Guide NCG while dialing (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

48257 The default background image on the VVX 1500 now displays after the following sequence of events: select an image file, followed by selecting an invalid image (file not found) and select the default background image.

48463 Can now view JPEG images with file extensions .jpe or .jfif (*applies to VVX 1500*).

48701 The touch-screen no longer disables during keypad diagnostics (*applies to VVX 1500*).

48776 Scrolling in the Ethernet menu no longer causes the selected highlighted item to be positioned at the bottom of the screen (*applies to VVX 1500*).

48840 Pressing the Slower and Faster soft keys no longer causes the update cursor to advance immediately.

49331 Audio is no longer lost when disabling the hands-free mode while on a speakerphone call (*applies to VVX 1500*).

50812 Changes to configuration options are no longer lost without warning if you exit from the Settings menu without passing through confirmation dialog.

50855 An error message is no longer shown when a contact is saved with an empty contact number (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

50920 The phones display a correct contact upon pressing the speed dial line key while editing the contact entry (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

50922 Dial plan now applies after editing a call list item and attempting to dial the number.

50969 The language displayed for a Missed call notification now changes when the option is changed to another language setting (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

50997 Upon pressing a line key, the phone now dials the stored hot dial number (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

51152 Back arrow now works as back-space when in the Display and Touch Screen Diagnostics or Media Statistics screens (*applies to VVX 1500*).

51237 In the Server Menu, the Server Password option no longer accepts digits instead of characters as the default (*applies to SoundPoint IP 320, 321, 330, 331, and 335*)

51656 Interactive microbrowser will now timeout if `mb.main.idleTimeout > 600.`

51664 The VVX 1500 phone no longer enters LCD Power-down mode in 3 to 4 minutes instead of the time set by `powerSaving.userDetectionSensitivity.officeHours=0`.

51669 After both SIP and H.323 Call Server parameters in Admin Settings are reconfigured on the VVX 1500, only one dialog method is now offered to exit. A reboot is no longer required.

51947 Can now delete the URL on the VVX 1500 by selecting it right-to-left and pressing the backspace key.

51993 Cancelling the deletion of a contact no longer appends an ellipsis to that contacts entry in the list (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

52212 The phone now restarts while another extension on a shared line is in use. The phone no longer thinks it is active on a call.

52374 Options in the Forwarding menu are no longer appended with an ellipsis after returning from the selected option.

52438 Typing in a fully filled field no longer prevent the cursor from advancing and overwriting existing content (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

52447 After placing 21 encrypted calls on hold, the VVX 1500 phone no longer locks-up and reboots at the 22nd multiple encrypted calls.

52590 The Add Video soft key is no longer accessible when flashing the POTS line to make a second POTS call. While playing dial tone for second POTS call, pressing the Add Video soft key and dialing a video number no longer causes the HDX to lock-up and reboot (*applies to SoundStation IP 7000*).

52629 The phones accept tel URIs as tel://number and tel:number.

52655 Upon disabling the directory, saving a contact from the corporate directory to the directory file no longer causes the saved contact to reuse the speed dial index starting from 1.

52690 The Add Phone soft key no longer appears while the Call Type is set to Conference-SIP and the phone is rebooted without a network connection (*applies to SoundStation IP 7000*).

52772 In the Corporate Directory, when `sortControl=1`, a quick search on multiple searchable attributes no longer causes the entry list to display items that are not starting from the beginning.

52851 Cancelling the Directory Search configuration change no longer appends an ellipsis to menu item label (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

52895 Enabling the Call Forwarding feature without entering a contact number no longer causes it to fail (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

52896 The Forwarding status field in the Forward menu option screen now correctly corresponds to the actual call forwarding status (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

52945 In the Corporate Directory, when performing a quick search, the Select/Submit indicator in the Navigation Cluster Guide displays correctly.

53066 When hot dialing on the VVX 1500, the white screen no longer flashes after pressing the Dial soft key, termination key, and dual line key.

53104 When an attempt to change the language option fails, the list of available language options are now sorted correctly.

53447 Initiating a URL based hot-dial by pressing the # or * key on the VVX 1500 no longer causes an invalid character to be inserted in the SIP URL.

53679 The Back soft key on the VVX 1500 is no longer always present in the APP menu screen.

53953 In the LDAP feature on the VVX 1500, the Scroll icons for navigating up and down pages now display when the last contact in the search list is reached.

54131 The SoundStation IP 5000 phone now displays the Volume control while the ringer volume is being adjusted in Quick Search mode.

54175 The Swedish Group soft key is no longer truncated; the visible portion translates properly.

54219 The phones can now establish a link when connected to some switches when both phone and switch are configured for 100Mbits/Full Duplex (*applies to SoundPoint IP 560 670*).

54292 On the VVX 1500, the information in the Line status menu accurately reflects the Gatekeeper address in use by the phone.

54343 Added the ability to save changes to text or IP entry fields while in the Admin Settings menu after viewing the Web browser on the VVX 1500.

54356 The Delete key on the VVX 1500 does not dismiss the character selection control or prevents character entry in the browser.

54614 The VVX 1500, upon originating a conference call, no longer shows a blank black screen instead of the No Video - crossed out camera image, while the call is on hold.

54616 When DND is enabled on the VVX 1500 on both SIP and H.323 lines, a SIP call no longer generates a busy tone and an H.323 call will no longer generate a re-order tone.

54617 While listening to a fast busy tone, if an incoming call is offered, the speaker LED is no longer turned off even though the fast busy tone is still present.

54638 Opening and closing the Web browser on the VVX 1500 no longer resets ABC/abc/123 and encoding soft keys.

54656 The phone now displays an x/y indicator when multiple calls are active if the Time and date display is disabled.

54720 Placing the handset on-hook no longer unexpectedly closes the Audio Diagnostics menu on the VVX 1500.

54727 Invoking the Abc/ASCII entry mode on the VVX 1500 capitalizes entered letters properly in the Corporate Directory search field.

54735 Upon pressing the VIDEO key on the VVX 1500, the focus now changes to Active Conference pane.

54756 The VVX 1500 phone can display the dialing screen when an alpha character is configured and entered into the contact field followed by a call to the specified contact.

54757 The Call Timer on the VVX 1500 displays a correct duration value.

54834 The VVX 1500 no longer connects with audio only when an MGC IVR Video Welcome Slide is used.

54876 Inter-digit DTMF signaling interval now matches the `tone.dtmf.offTime` setting.

54949 An unassigned soft key no longer operates as a Dir soft key on the SoundStation IP 7000.

54966 The Lin16.16ksps codec now engages if it is the only supported codec.

54988 The user is now able to make additional changes to the selected item in the Prioritize Background menu after making an initial selection (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).

54993 The SoundStation IP 7000 phone no longer displays Enter name instead of Enter URL in the Install Custom CA Cert menu.

54995 Pressing the # key while in an idle call state no longer displays the character in dial screen.

55001 The Backspace soft key no longer shows at left edge of a dialed SIP URL (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55002 The user is now able to press and hold the Backspace soft key to clear the contents of the dialing fields (*applies to SoundPoint IP 320, 321, 330, 331, and 335).*

55005 The user Interface no longer becomes corrupted when change the language while hot-dialing (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55014 Soft keys no longer disappear from a shared line when a hold/resume operation is performed on another remote shared line (*applies to SoundStation IP 5000 and 6000*).

55017 Auto Reject now functions as expected when the feature is enabled through the Contact Directory when an Alpha-character is present in the CONTACT field (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55039 The Navigation Control Group (NCG) Indicator no longer shows the right-pointing arrow even when there are no calls in the call lists.

55053 The Page up arrow now functions correctly when the Server menu is highlighted and the DHCP client is set to disabled.

55063 It is no longer possible to select a disabled menu item using the * key, resulting in a non-functional Edit soft key.

55094 When taking the phone off-hook and dialing the # key, the # no longer displays in the Enter more digits field (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55129 The Transfer and Conference soft keys on the SoundStation IP 7000 now display upon an 8th active call.

55139 The user can now view the full date when the phone is configured for Norwegian language Norsk (no-no) (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55271 The Please Enter a Contact pop-up no longer shows up unexpectedly when adding/editing contacts.

55287 The phone no longer drops the incorrect call if the user selects (on the phone UI) a held call and then attempts to terminate the active call (e.g. by placing the handset on hook).

55297 An audio and video conference call appears as a single video call (*applies to the SoundStation IP 7000*).

55313 The Routing soft keys on the VVX 1500 now display correctly when the Call Park feature is enabled.

55339 Resuming a conference while running the Slide Show application on the VVX 1500 no longer causes the user interface to become dysfunctional.

55340 The user can no longer launch picture frame on the VVX 1500 while a recording is in progress causing the USB busy icon to disappear.

55375 The Outgoing Call control interface on the VVX 1500 now displays when the Speed Dial Contact Enhanced Feature Key macro fails to execute.

55423 The correct soft keys and user interface displays on the VVX 1500 after exiting the screen that was previously opened from the icon in the status bar; while hot dialing digits.

55457 When the dual protocol line is registered only to the gatekeeper and not to the SIP server on the VVX 1500, this no longer causes hot-dialed SIP URL to call via H323 and dialog options to appear when a hot dial URL call is attempted.

55477 SRTP Key renewal now occurs during local conference calls.

55478 DHCP VLAN Discovery (DVD) no longer reports as not active when it actually is.

55485 The Camera Settings Save soft key on the VVX 1500 no longer loses its context-sensitivity upon second visit to the menu option.

55514 Calling into a Video Server on the VVX 1500 no longer causes the phone to connect the audio portion of call but does not establish a video connection.

55560 On occasion, the VVX 1500 phone no longer displays an incorrect call duration timer value while on an H.323 call to an RMX-2000.

55641 The Y-axis auto-scaling of the Network Load graph on the VVX 1500 is now accurate.

55697 The phone no longer rejects calls with 486 if `NOTIFY:Alerting` is received before the INVITE and `reg.x.lineKeys` and `reg.x.callsPerLineKey` is set to 1.

55644 The VVX 1500 now uses the correct routing protocol when dialing an LDAP contact from the on-hook state via termination.

55907 On the VVX 1500, typing a or # no longer causes the on-screen keyboard to unexpectedly close and discard any edits.

55911 Changing the text entry mode no longer causes the backspace soft key to disappear (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55929 Pressing the down arrow no longer affects a change on the Navigational Cluster Guide (NCG) (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

55964 The VVX 1500 phone no longer seizes the only unregistered share line using the New Call soft key, speaker and headset function key.

56046 The default value of the Sound Effect Destination parameter setting is now removed from the override file when a new value is selected from the menu option.

56057 The phones no longer de-registered upon receiving a large number of NOTIFY messages for watch buddy enabled contacts.

56147 Adding Contacts to the SoundPoint IP 550 and 670 that are longer than 10 characters or numbers are no longer truncated on the idle screen.

56156 The abc/ASCII string no longer remains in the title bar on the VVX 1500 after leaving edit mode for a menu item.

56161 Emergency numbers matched against `dialplan.routing.emergency.x.value` are now sent to servers listed in `dialplan.routing.emergency.x.server.y`.

56168 When adding more than 7 characters and/or digits to a local contact directory entry on the SoundPoint IP 450, the characters and/or numbers no longer overlap on the idle screen, when they should be truncated.

56294 The `Dutch_Netherlands` localization now displays the correct default 24 hour time format in SIP 3.2.x.

56289 Label text is no longer drawn past the edge of the speed dial label on the display next to the key (*applies to SoundPoint IP 550, 560, 650, 670*).

56333 The SoundStation IP 5000 phone no longer reboots automatically when lease time expires after disabling and enabling the DHCP server.

56334 The phone no longer plays a short burst of ringtone upon switching initiating a call sequence of transfer, conference initiation, and then cancels.

56338 The SoundStation IP 7000 phone no longer reboots when the user presses the Manage soft key during an 8-way MP call plus 1 audio EP conference.

56381 The conference phones now accept a DHCP offer that do include the terminating END (0xFF) option (*applies to SoundPoint IP 5000, 6000, 7000*).

56401 The Admin password length in the boot menu and Menu > Settings > Advanced menu now match.

56678 Local contact directory entries now store up to 32 characters instead of only 31.

56708 The SIP URL dialing field accepts up to 32 characters instead of 256.

56787 The phone no longer plays a short tone while retrieving a parked call using an incorrect contact.

56788 The contact field of the local contact directory now accepts 128 characters instead of only 32 (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

56797 The SoundPoint IP 450 Admin Settings sub-menus correctly display the titles in a white background box.

56809 The configuration parameter `voice.audioProfile.Lin16.48ksps.payloadType` has a default value of 118 instead of 119.

56827 The soft keys associated with Conference Remote Pickup NewCall, Transfer, and Conf soft keys on the SoundStation IP 7000 are no longer missing when the conference call is split.

56868 Published CDP power values in TB 48152 now match actual measured consumption.

57146 The phone no longer freezes and reboots when it receives an INVITE message with special characters in the FROM header and the call is placed on hold.

57368 The second contact in the Local Contact Directory on the SoundStation IP 5000 is no longer highlighted when it is selected.

57369 The Contact entry in the Local Contact Directory no longer takes a long time to display (*applies to SoundStation IP 6000 and 7000*).

57398 The phone no longer displays *Please enter a contact* pop-up message after adding a contact in the local contact directory.

57443 The display on the SoundStation IP 6000 and 7000 no longer flickers while making an outgoing call.

57597 Using the VVX 1500 phone with an HDX, the phone now transmits video upon resuming a SIP call.

57615 The Autohide feature on the VVX 1500 now functions when PIN is pressed while the tool bar is sliding down out of view.

57849 The phone now acquires the correct VLAN using LLDP after a bootup.

57863 The phone accepts a DHCP END (0xFF) option in a DHCP INFORM response.

57958 In the fail-over feature, while re-registering, there is no longer a 32 second delay before sending INVITE to the third server.

58023 A call into a 3COM VCX audio conference server when using the VVX 1500 no longer causes the phone to reboot.

58172 Hot-dial numbers no longer disappear from the screen if there is an incoming call during the outgoing hot-dialing state (*applies to SoundStation IP 5000 and 6000*).

58197 After upgrading from 3.0.0 to 3.1.3 Rev C, there is no longer a delay in the audio signal when answering a call using the speakerphone.

58296 H.323 digit-map no longer routes files when the `reg.1.lineKeys` configuration parameter has a value of greater than 1 and `reg.1` is assigned a SIP number (a*pplies to VVX 1500*).

58362 Initiating a URL hot-dial call by pressing the # or * key on the VVVX 1500 no longer causes the Enter URL dialog to pop-up with the # character already inserted into the field, even though the # character is not a valid SIP URL character.

58464 A Contact can now be saved from a Corporate Directory search result into a local directory. This is as a result of not checking the correct attribute such as SIP vs. H.323.

58498 Within the re-registration on fail-over feature, Subscribe now triggers the fail-over. The phone now sends the register request to the second server after received an ICMP from the primary server.

58509 Within the Re-registration on failover feature, the phone no longer sends an extra Register request to primary server after the first fail-over.

58520 Resolved a uni-directional Video Streaming interoperation issue with Siemens Video Desktop Client ODC (*applies to VVX 1500*).

58574 The SoundPoint IP 650 phone now validates an existing registration when it is registered with a BroadSoft server.

58619 The line no longer becomes unregistered when an invalid name and password is entered from the menu options on the phone. The line becomes unregistered until the phone is rebooted.

58782 The phone sets the Call Control 802.1Q Priority correctly when using TCP. The value is set correctly when using UDP.

58785 The VVX 1500 phone now appends the MAC address to HTTP user agent headers when configured to do so.

58787 The VVX 1500 phone no longer reboots immediately after making a call to an RMX when the Camera Target Frame Rate is set to minimum.

58874 When using TCP preferred transport, the phone now resends a 200 OK message after answering a call without receiving an ACK.

58906 The phone now clears its BLA state table when receiving a NOTIFY message with state = full after a SUBSCRIBE message.

58907 The VVX 1500 phone now sends an INVITE SIP packet when the configuration parameter `msg.mwi.1.callBack=voicemail` and the user presses the Messages key.

58908 With BootROM 4.2.1.0334, the VVX 1500 phone no longer sends a truncated Option 60 message.

58913 The phone no longer reboots when pressing the Messages key while Message Waiting Indicator is disabled. When the phone has more than one registration and `msg.mwi.1.callBackMode=disabled` and `msg.mwi.2.callBackMode=disabled`, the phone no longer freezes when the Messages key is pressed. The phone will no longer respond to any further key presses.

59129 The Centralized Conference feature no longer fails when a URI is incorrectly assigned to `voIpProt.SIP.conference.address`.

59262 A conference notification no longer causes the phone to lock-up and then reboot.

59308 A retransmitted INVITE message no longer results in a 400 response.

59430 Calls received from a mobile to the VVX 1500 no longer cause the phone to display SIP +86…@. The @ should not be displayed.

59561 The VVX 1500 phone no longer displays an incorrect time after the configuration parameter `tcpIpApp.sntp.daylightSavings.enable` is set to disabled.

59737 The Line Label now displays on the top line of the screen when using the HTML idle display micro-browser page (*applies to SoundPoint IP 320, 321, 330, 331, and 335*)

59777 When using NN# speed dial feature, the title no longer displays Directory instead of Speed Dial (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

59949 Idle bitmap graphic is no longer displayed on the bottom of the screen so that only half of the display is utilized when `ind.idleDisplay.mode=2 or 3` (a*pplies to SoundPoint IP 320, 321, 330, 331, and 335*).

59954 The phone no longer locks up and reboots when a Re-INVITE message within same dialog is sent to the phone immediately after sending a CANCEL message for the initial INVITE.

59967 When an incorrect CA certificate is installed, the phone will not attempt to retry a TLS handshake (a*pplies to LDAP on the VVX 1500*).

60013 The phone no longer locks up and reboots when accessing the contact directory if `dir.local.readonly=1` (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

60126 Gateways no longer reject an INVITE message when `reg.1.csta=1`. The INVITE includes the header *Acceptapplication/sdp/application/csta+xml*.

60145 The SoundPoint IP 650 phone now correctly presents 2 BLA call appearances. The 2nd call appearance now correctly indicates a remotely held line, when it is not.

60264 When a BLA line is showing the dialing screen, remote call appearances no longer display when the remote BLA line resumes a call (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).

60266 When a phone is in dialing screen, if a remote SCA/BLA line holds and resumes, the dialing icon no longer changes between the animation arrow and the termination (speaker) icon. The termination icon displays continuously and no longer changes (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

60267 Can now change a checked item twice in the Prioritize Background menu (*applies to SoundPoint IP 550, 560, 650, and 670*).

60340 The Join soft key on the SoundPoint IP 650 no longer displays on a phone with a BLA line when there is only one call on the phone.

60650 The idle browser on the VVX 1500 no longer alternates between current content and earlier content when it the display is refreshed.

62621 SoundPoint IP 321 and 331 phones running SIP 3.2.3.3122 and configured for HTTPS no longer display the error messages *Alert: Fatal, Description, Decode Error.*

Configuration File Enhancements

Note that the configuration files, their respective parameters and defaults, as well as the provisioning methods have been simplified but extensively modified compared to previous releases.

Before installing the software, it is highly recommended that you first familiarize yourself with the changes outlined in the Administrators Guide for the Polycom® UC Software – 3.3.0 and Technical Bulletin 60519: Simplified Configuration Enhancements in Polycom® UC Software 3.3.0.

Understanding Updates to SIP 3.2.5

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.5 beside their respective Polycom tracking ID number.

New or Enhanced Features

59000 The phones now ignore BLA dialog documents sent within NOTIFY messages that are reflected to User Agents that are party to the dialog.

62939 Various enhancements to the Geo-Redundancy (multiple server fail-over support) feature. For full details, refer to the list of documents in Reference Documents.

64359 Bridged Line Appearance BLA line dialog rendering is now converted from No to Yes on User Agents that are a remote party to the dialog.

Enhanced Capabilities

54219 The SoundPoint IP 560 and 670 phones now establish a data link when connected to some switches when both phone and switch are configured for 100Mbits/Full Duplex.

57570 A fail-over is now performed as a result of a SIP Response code 503.

60851 Dialing using the Speaker or Headset key no longer drops the initial call appearance.

60973 Entering a username and password using the Quick Setup (QSetup) soft key followed by a request to save, now automatically invokes the phone to reboot the phone in order to the changes to be applied.

61248 After configuring a phone with 3 line registrations, while the 2nd line is on hold, if a user hot-dials using the speaker/headset termination key, the phone no longer inadvertently seizes line 3 to dial out.

61283 The phone no longer incorrectly sends a NOTIFY with *<param pname=+sip.rendering pvalue=no />* when a user attempts to place a conference call on hold and the phone receives a 400 Bad request.

61541 When a user attempts to place a conference on hold and the phone receives a 400 Bad request, the phone correctly sends a NOTIFY with I=no. This no longer causes the incorrect presence, on the other Bridged Line Appearance line, to be displayed.

62206 The phone no longer displays Service Unavailable upon lifting the handset and pressing the Line 2 key (a*pplies to SoundPoint IP 320, 321, 330, 331, and 335*).

62226 The phones no longer join a conference after receiving a 403 Forbidden from the switch.

62383 A held call on a Bridged Line Appearance with remote phones is now presented (*applies to SoundPoint IP 601*).

62567 SoundPoint IP 3xx phones monitoring each other in a 2x2 BLA configuration are now able to pick up held calls.

62621 SoundPoint IP 3xx phones configured for HTTPS no longer display the error messages *Alert: Fatal, Description, Decode Error*.

62642 The phones no longer play a dial tone as well as RTP audio when resuming a call held at another phone.

62643 When the user presses both line keys (Line 1-hold and Line 2-Active call) simultaneously on the SoundPoint IP 3xx, the active call on Line 2 is no longer dropped.

62669 Multiple phones no longer try to resume a held Bridge Line Appearance BLA line at the same time.

62672 Directed Call Pickup DCP or Group Call Pickup feature (using soft keys instead of *53 and *54 feature access codes) no longer fail when the user enters an account code. The account code is not appended to the user portion of the URI.

62855 Invoking either the Group Call Pickup or Directed Call Pickup feature, using its corresponding soft key, now functions properly. The display shows Unknown and the call is not picked up (*applies to SoundPoint IP 3xx*).

62902 The phone now accepts inbound SIP requests from an RROFO (Geo-redundancy) server that is not registered with that phone.

62926 The Resume soft key on the SoundPoint IP 3xx is now presented when the line key is pressed continuously while the line is in a remote held call state. This occurs when the line is configured as `callsPerLineKey=1`.

63099 The phones monitoring Bridged Line Appearance BLA line, configured for one call per line, can now pick up the held call after the call on a BLA line has been put on hold using the Transfer/Conference key.

63280 Regarding Geo-redundancy RROFO, calls on hold are now released when pressing the Resume soft key after the IP BE fail-over occurs while using the geo-redundancy feature. The user no longer needs to press the End Call soft key to complete the intended result.

63388 If a phone's SIP lines are not registered with a call server, and the Emergency Call Routing Feature is enabled (by configuring the `dialplan.routing.emergency.x.value` and `dialplan.routing.emergency.x.server.y` parameters) dialing the configured emergency number will now work when you use on-hook dialing and when URL Dialing is enabled.

63536 The Redial feature functions correctly after invoking an outgoing call accompanied with an account code.

63631 The counting down aspect of the Geo-redundancy RROFO-DNSTTL feature no longer fails during fail-back. The Time-To-Live TTL timer should be reset after re-registering to the secondary server.

63704 Regarding Geo-redundancy RROFO, the phone no longer sends three extraneous registration requests to the primary proxy server during a fail-over.

64093 Regarding Geo-redundancy RROFO, a fail-over using either the Conference or Transfer feature now stops a consultative call when the primary call is terminated.

64212 Invoking the Call Park feature on the SoundPoint IP 3xx with the soft key now functions correctly when the soft key is configured as 1 line and 1 call per line.

64219 The SoundPoint IP 3xx phone sends a proper hold NOTIFY message after a consultative transfer is canceled when the configuration parameter `notifyTransferHoldAsActive` is disabled.

64274 In an attempt to resume a held call, the held call is no longer terminated when the user inadvertently seizes two line keys simultaneously.

64327 In an attempt to answer an incoming call, the user no longer inadvertently presses 2 line keys. The user is no longer connected to both lines one with an incoming caller and the other with dial tone.

64340 The indicator, on a Bridged Line Appearance BLA line that is monitoring other lines, no longer remains on continuously after the monitored phone performs the following sequence transfer > split > endcall > resume > hold.

64356 The display on the SoundPoint IP 3xx showing a remote call appearance now times out properly when the user presses continuously a BLA line key followed by pressing a down arrow key while there are multiple calls on hold on the remote BLA.

64360 The state of the indicator of a BLA line appearance is now properly reported after the phone receives an INVITE containing replaces.

64762 When special characters in the FROM field are received, they no longer prevent the SoundPoint IP 430 phone from displaying Caller ID information.

64862 Joining an internal extension with an external PSTN call no longer causes one call to drop.

65119 When a Bridged Line Appearance BLA line is presented in a dialing screen, the remote call appearance now displays when the remote BLA line resumes a call.

65207 A slow memory leak no longer occurs in the SIP stack due to the receipt of hunt group INVITE containing replaces with phones using ADTRAN switches.

65368 When the configuration parameter `signalWithUnregistered=0`, the phone now always ignores all of the messaging traffic.

65842 Call waiting tone no longer continues to play after an inbound call has been forwarded and answered by the PSTN.

67178 Centralized conference can now be established when `reg.1.lineKeys` is 5 or greater (*applies to all SoundPoint IP*).

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP 3.2.5 configuration file parameters.

Software Version 3.2.5 Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
phone1	added	`reg.n.server.m.failOver.onlySignalWithRegistered`	N/A	Null
phone1	added	`reg.n.outboundProxy.failOver.onlySignalWithRegistered`	N/A	Null
phone1	added	`reg.n.filterReflectedBlaDialogs`	N/A	Null
sip	added	`voIpProt.server.n.failOver.onlySignalWithRegistered`	N/A	Null
sip	added	`voIpProt.SIP.CID.sourcePreference`	N/A	Null
sip	added	`voIpProt.SIP.failoverOn503Response`	N/A	1
sip	added	`voIpProt.SIP.outboundProxy.failOver.onlySignalWithRegistered`	N/A	Null
sip	added	`call.localConferenceEnabled`	N/A	1

Understanding Updates to SIP 3.2.4B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.4B beside their respective Polycom tracking ID number.

Enhanced Capabilities

66743 The phones may be vulnerable to Denial of Service attacks when used in certain configurations. Sending HTTP GET requests with a broken authorization header can produce a device restart under certain circumstances in certain models of phones. For full details, refer to Technical Bulletin 66743.

Understanding Updates to SIP 3.2.4

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.4 beside their respective Polycom tracking ID number.

Enhanced Capabilities

59308 A retransmitted INVITE message causes a 400 Bad Response reply. This is in violation of RFC 3261 section 17.2.1.

65207 A consistent but slow memory leak occurs as a result of receiving INVITE messages containing replaces.

65435/65725 With reference to IEC 60268-1, the default and maximum values for the headset and headphone audio levels have been adjusted to ensure compliance with the IEC 60268-1 TUV safety requirements (*applies to SoundPoint IP/VVX 1500*).

65660 The BootBlock may become corrupted as a result of accessing unprotected section of flash memory.

Understanding Updates to SIP 3.2.3

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.3 beside their respective Polycom tracking ID number.

New or Enhanced Features

43099 Added support for the SoundStation IP 5000 conference phone.

43297 Sound effects can now be played out of a destination based on user configuration. The available destinations are: *chassis*, *handset*, *headset* or *active*. The default is *chassis*.

45462 All SoundPoint and SoundStation phones now comply with retry-after instructions embedded in SIP Response codes 500 and 503 as part of REGISTER and other requests.

50739 On a multi-leg conference on the SoundStation IP 7000, when the End Call soft key or the On Hook hard key is pressed, the conference phone will ask the user if the entire call should terminate. A negative response will guide the user to the conference manage menu to allow the user to terminate the individual legs of the call. The dialog only appears for multi-leg conference calls.

51753 Enhanced the appearance on the SoundPoint IP 450 of anti-aliased characters.

51940 All SIP phones now have a fail-over feature that enables phones to re-register before diverting SIP signaling to an alternate server. This feature will be formally released and documented in a future release.

54041 Format of DHCP Option 60 Data is now configurable and added support for Option 125 as per RFC 3925.

54983 Internal IP address of the VVX 1500 phone (instead of an alias) is no longer being sent in the Facility Message.

55524 Logs no longer display Cant set 802.1Q VLAN ID for TCP protocol messages at default when running on a VLAN.

56272 Network Configuration DHCP sub menu now supports Option 60 format. The new options include setting either RFC 3925 Binary [default] or ASCII String.

Enhanced Capabilities

45188 The minimum acceptable amount of free RAM has been increased on the SoundPoint IP 320, 330, and 430 in order that functions such as ringtones are not affected.

47897 The Back soft key works when a user tries to exit from Instant Message menu.

52119 VVX 1500 phones no longer reboot during G.729 packet loss concealment such as when the remote phone is placed on hold.

52787 The configuration parameter `voIpProt.SIP.requestValidation.x.method=source` does works with DNS SRV Static Cache

53473 When the SoundStation IP 7000 is used with an HDX, the parameter `voice.volume.persist.handsfree=0` also affects the HDX.

54549 Changes in the display color palette on the SoundPoint IP 450 no longer cause contrast problems.

54751 SIP INVITE messages can be sent when dialing a number containing the period character.

54832 The phone enables user to add more than 32 characters in Hot Dial screen (*applies to VVX 1500, 321, 325, 330, 331, and 335*).

54867 In the Contact Directory, the text fields scroll to the left to reveal the first character as you move the text cursor left (*applies to SoundPoint IP 321, 325, 330, 331, and 335*).

54908 An unexpected colon has been removed in the scrolling status line during an incoming call (*applies to SoundPoint IP 321, 325, 330, 331, and 335*).

55099 In a long SRTP conference, steering video on the VVX 1500 between active and inactive no longer causes the video leg to fail.

55120 Dialing numbers in the Contact Directory no longer opens contacts for editing (*applies to SoundPoint IP 550, 560, 650, and 670*).

55296 On the VVX 1500, the dial pad widget displays when attempting to conference or transfer a held call while in a ringback state.

55378 The VVX 1500 phone can invoke LCD power down mode after a remote end places the call on hold.

55415 The phone enables the user to enter more characters than it is capable of saving in the Contact Directory fields.

55420 The VVX 1500 phone can play back video after a SIP re-INVITE message is sent to an RMX meeting room.

55560 The VVX 1500 phone displays correct call timer values while in an H.323 call to an RMX-2000.

55618 Switching to Katakana characters before the character selection widget times out no longer produces random characters that on occasion cause the phone to malfunction (*applies to SoundPoint IP 450, 550, 560, 650, 670; SoundStation 5000 and 7000*).

55844 Proceeding outgoing call state on one line is adversely affected by an outgoing call on another line (*applies to* SoundPoint IP 321, 325, 330, 331, and 335).

55884 The displays on a SoundPoint IP 650 with expansion modules no longer freeze during a consultative transfer.

56032 SoundPoint IP 650 phones with two expansion modules no longer reboot while monitoring continuous BLF traffic.

56488 In packets sent from the client, the Parameter Request List option no longer contains two duplicate requests for the options Router (3) and Domain Name (15) (*applies to SoundStation IP 6000 and 7000*).

56641 The phone no longer ignores the LLDP broadcast from a switch. It will default to the data VLAN instead of the voice VLAN. There is a LOSS of LINK during the boot process causing LLDP to fail (*applies to SoundStation IP 6000 and 7000*).

56836 After dialing and then adjusting the volume, lifting the handset no longer dials the last hot-dialed number immediately (*applies to SoundPoint IP 550, 560, 650, and 670*).

57133 The SoundPoint IP 321, 330, and 331 phones can display a customer supplied logo.

57457 The LoudRing.wav audio file has been distributed in release 3.2.2.

57796 Invalid Message-Summary Event no longer results in invalid MWI notification.

57849 The SoundPoint IP 330 and 550 phones can acquire the correct VLAN via LLDP.

58024 The Hold function on the VVX 1500D no longer fails in a specific customer scenario.

58024 The Hold function on the VVX 1500D no longer fails in a specific customer scenario.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP 3.2.3 configuration file parameters.

Software Version 3.2.3 Parameter Enhancements

File	*Change*	*Configuration Parameter*	*New Value*
phone1	added	`reg.n.server.1.failOver.reRegisterOn`	
phone1	added	`reg.n.server.1.failOver.failBack.mode`	
phone1	added	`reg.n.server.1.failOver.failBack.timeout`	
phone1	added	`reg.n.server.2.failOver.reRegisterOn`	
phone1	added	`reg.n.server.2.failOver.failRegistrationOn`	
phone1	added	`reg.n.server.2.failOver.failBack.mode`	
phone1	added	`reg.n.server.2.failOver.failBack.timeout`	
phone1	added	`reg.n.outboundProxy.failOver.reRegisterOn`	
phone1	added	`reg.n.outboundProxy.failOver.failRegistrationOn`	
phone1	added	`reg.n.outboundProxy.failOver.failBack.mode`	
phone1	added	`reg.n.outboundProxy.failOver.failBack.timeout`	
phone1	added	`reg.n.useCompleteUriForRetrieve`	1

File	*Change*	*Configuration Parameter*	*New Value*
sip	added	voIpProt.server.1.failOver.reRegisterOn	
sip	added	voIpProt.server.1.failOver.failRegistrationOn	
sip	added	voIpProt.server.1.failOver.failBack.mode	
sip	added	voIpProt.server.1.failOver.failBack.timeout	
sip	added	voIpProt.server.2.failOver.reRegisterOn	
sip	added	voIpProt.server.2.failOver.failRegistrationOn	
sip	added	voIpProt.server.2.failOver.failBack.mode	
sip	added	voIpProt.server.2.failOver.failBack.timeout	
sip	added	voIpPort.SIP.useCompleteUriForRetrieve	1
sip	added	voIpProt.SIP.outboundProxy.failOver.reRegisterOn	
sip	added	voIpProt.SIP.outboundProxy.failOver.failRegistrationOn	
sip	added	voIpProt.SIP.outboundProxy.failOver.failBack.mode	
sip	added	voIpProt.SIP.outboundProxy.failOver.failBack.timeout	
sip	added	voIpProt.H323.blockFacilityOnStartH245	0
sip	added	se.destination	chassis
sip	added	voice.codecPref.IP _5000.G711Mu	2
sip	added	voice.codecPref.IP _5000.G711A	3
sip	added	voice.codecPref.IP _5000.G729AB	4
sip	added	voice.codecPref.IP _5000.G722	1
sip	added	voice.codecPref.IP _5000.iLBC.13_33Kbps	
sip	added	voice.codecPref.IP _5000.iLBC.15_2Kbps	
sip	added	voice.gain.rx.analog.chassis.IP _5000	0
sip	added	voice.gain.rx.analog.ringer.IP _5000	0
sip	added	voice.gain.rx.digital.chassis.IP _5000	11
sip	added	voice.gain.rx.digital.ringer.IP _5000	-12
sip	added	voice.gain.tx.analog.chassis.IP _5000	0
sip	added	voice.gain.tx.digital.chassis.IP _5000	15
sip	added	voice.aes.hf.duplexBalance.IP _5000.0	10

File	*Change*	*Configuration Parameter*	*New Value*
sip	added	`voice.aes.hf.duplexBalance.IP _5000.1`	9
sip	added	`voice.aes.hf.duplexBalance.IP _5000.2`	8
sip	added	`voice.aes.hf.duplexBalance.IP _5000.3`	7
sip	added	`voice.aes.hf.duplexBalance.IP _5000.4`	6
sip	added	`voice.aes.hf.duplexBalance.IP _5000.5`	5
sip	added	`voice.aes.hf.duplexBalance.IP _5000.6`	4
sip	added	`voice.aes.hf.duplexBalance.IP _5000.7`	3
sip	added	`voice.aes.hf.duplexBalance.IP _5000.8`	2
sip	added	`voice.ns.hf.IP _5000.enable`	1
sip	added	`voice.ns.hf.IP _5000.signalAttn`	-6
sip	added	`voice.ns.hf.IP _5000.silenceAttn`	-9
sip	added	`voice.rxEq.hf.IP _5000.preFilter.enable`	1
sip	added	`voice.rxEq.hf.IP _5000.postFilter.enable`	0
sip	added	`voice.txEq.hf.IP _5000.preFilter.enable`	0
sip	added	`voice.txEq.hf.IP _5000.postFilter.enable`	1

Understanding Updates to SIP 3.2.2

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.2 beside their respective Polycom tracking ID number.

New or Enhanced Features

41450 Change of the real time operating system (*applies to VVX 1500*).

43760 H.323 signaling protocol support for video (*applies to VVX 1500*).

43862 Support for Webkit browser to replace the XHTML browser (*applies to VVX 1500*).

45172 Support for iLBC audio codec (*applies to VVX 1500*).

47173 Support for H.261 video codec (*applies to VVX 1500*).

48557 Max video bit rate defaults to 384 Kbps (*applies to VVX 1500*).

48743 Upgraded curl library to version 7.19 (*applies to VVX 1500*).

48961 Support for H.235 security (*applies to VVX 1500*).

49069 Added support for iLBC audio codec (applies *to SoundStation IP 6000 and 7000*).

49079 Support for mutual TLS authentication (*applies to VVX 1500*).

49277 Support for LLDP protocol (*applies to VVX 1500*).

49430 Added ITU-T G.719 vocoder (*applies to VVX 1500*)

50125 Outgoing calls support dual (SIP /H.323) protocols (*applies to VVX 1500*).

51084 Support for video fast update request via RTCP, RFC 5104 (*applies to VVX 1500*).

52944 Menu support applicable to H.323 usage (*applies to VVX 1500*).

53849 Formalized support for DTMF via SIP INFO (initially supported in SIP 3.2.0).

54025 Increased maximum size of contact directory to 128 to facilitate complex dialing scenarios.

54239 Added user accessible menu option to select the video call rate. Default configured using the configuration parameter `video.callRate` (*applies to VVX 1500*).

Discontinued Features

52522 Removed Launchpad Feature (*applies to VVX 1500*).

Enhanced Capabilities

44782 Improved phone UI response when a local conference is active (*applies to VVX 1500*).

44980 The phone falls back to configured video codec configuration for Tx video when incoming signaling lacks codec modifiers (*applies to VVX 1500*).

47023 Text font no longer randomly changes (*applies to VVX 1500*).

47476 Using the XML API, when the user is inside an XHTML Form Field, the Submit soft key displays properly.

47768 CDP power usage advertisement matches the peak power conditions (*applies to SoundPoint IP 450*).

48175 EFK feature can establish conference calls (a*pplies to VVX 1500*).

48784 Soft keys are restored after rejecting a call from within the Applications UI context (*applies to VVX 1500*).

48857 Recording (R) no longer stops or reboots phone in various high load scenarios such as (a) recording during SRTP conference call, or (b) recording while browsing the application menu during non-SRTP conference call (*applies to VVX 1500*).

48921 Digit key presses are no longer missed in certain scenarios (*applies to VVX 1500*).

50152 Change non-null sticky primary filter, search (filtered) bar remains on old data (*applies to VVX 1500*).

50192 Media Statistics menu displays correctly for several languages (*applies to VVX 1500*).

50286 Pressing page down key # does not move entry list after pressing page up key * in quick search menu (*applies to VVX 1500*).

50531 The SoundStation IP 7000 phone can startup without network connection when using the PIC cable.

50624 The phone sends a 603 Decline message when an inbound call times out.

51141 A small number on the left side of the scrolling status bar has been removed.

51449 Out of Dialog Refer based dialing on the VVX 1500 no longer fails. SDP on INVITE from VVX is missing media attributes, generating a 606 response.

51533 Backlight intensity change updates appropriately in Overrides configuration file.

51605 VVX 1500 phones correctly handle back-to-back Push requests.

51643 Japanese displays properly on the SoundStation IP 6000 and VVX 1500.

51753 Display text on the SoundPoint IP 450 looks clearer.

51959 Handling of Hold re-Invites is correct after one-touch blind transfer to full park orbit.

51965 HTTP request messages are directed to proxy.

52164 Hot-dial on the VVX 1500 works in headset mode.

52360 Auth Password field can no longer be viewed in Web configuration page.

52365 The phones can easily transition from LLDP to CDP.

52370 Removing Ethernet cable from the SoundStation IP 7000 no longer un-mutes the muted phone.

52376 The parameter `daylightSavingsTime` can now be disabled. Introduced in SIP 3.2.0 (*applies to SoundStation IP 6000 and 7000*).

52381 The Retrieve, Directed, and Group soft keys no longer disappear after entering some digits. This occurs when using the Call Park/Pick-Up feature using SIP signaling. Introduced in SIP 3.2.0 (*applies to SoundPoint IP 430, 450, and SoundStation 6000*).

52415 When using enhanced BLF, ringtones are no longer suppressed when a user is parked.

52568 The SoundStation IP 7000 phone plays DTMF tone with the default configuration.

52580 Delayed DTMF audio feedback is heard when conferencing third POTS end while using the SoundStation IP 7000 User Interface.

52656 The VVX 1500 phone supports transcoding of video codecs that are not included in the far-ends capability set

52678 Using the quick/AdvFind search on full last name in the Corporate Directory no longer misses some entries.

52709 License menu displays Active to indicate a license with no expiry date.

52770 Message-summary SUBSCRIBE is sent when `reg.x.type=shared.`

52836 The phone no longer enables the user to enter more than the maximum allowed (32) characters in hot dial and contact directory operations.

52860 Split soft key no longer displays while transferring calls if the call per line limit is reached.

52883 When a call is placed to a shared line, the ringer for an IP 650 no longer stutters when the call is picked up at another station.

52943 LLDP reported power usage in logs indicates appropriate power consumption.

52950 Packet Loss and Burst Gap Loss metrics too high when calling IVR, caused by valid gap in audio sent from IVR.

52963 The SoundPoint IP 320, 321, 330, and 331 phones no longer reboot when the user presses NN# from idle screen to invoke Contact Directory entry screen for NN speed dial index.

52971 The phone no longer reboots when the `efkprompt` label is longer than 32 characters.

52977 The Directory soft key on the VVX 1500 does not disappear after selecting Blind transfer mode.

53007 VQMon on the VVX 1500 phone computes RFactor and MOS quality scores for the G7221C codec.

53034 SUBSCRIBE for BLA with expires: 0 received from server is recognized as terminating the subscription

53254 VVX 1500 enables users to change Auth Password for SIP Lines through the phone's user interface.

53598 Side-tone disappears after a call hangs up on headset using GN9350e with EHS.

53656 Part number in Phone Status menu displays proper part number.

53855 When a phones extension has an underscore in the name, followed only by numbers, the underscore is no longer removed in SIP signaling and the device can be found.

53917 The phone no longer reboots in a certain scenario when using the Join key.

53944 SoundPoint IP 320, 330, 321, and 331; SoundStation IP 7000: Phone displays Dir soft key in Korean and Slovenian languages.

53946 SoundPoint IP 550, 560, 650, and 670 phones no longer randomly display the time and date behind a custom idle display.

53975 The phones will send a SUBSCRIBE message in a certain scenario when using SCA with barge in enabled.

54034 The VVX 1500 phone no longer generates loud static when CNG packets are received.

54139 Consultative transfer uses the correct URI on REFER.

54262 The Ethernet status menu on the SoundPoint IP 320 and 321 displays the correct information.

54631 The Voice/Video call type prompt on the SoundStation IP 7000 defaults to Voice.

54765 The VVX 1500 phone fails to resend INVITE after 401 from server when second INVITE is roughly 1500 bytes.

54768 VVX 1500 phones can establish calls properly when booted without a network connection.

54886 The phones send re-Invite with SDP containing session attribute a=sendrecv upon resuming a call when the call is initiated with a=sendrecv offered.

54940 New REQUESTS sent directly to far end; route set ignored after a call is placed on MOH, resulting in a loss of audio.

55052 Additional parameter in the From header of INVITE no longer causes 1-way audio when it is not found in the ACK to a 200 OK.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP 3.2.2 configuration file parameters.

Software Version 3.2.2 Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
phone1	added	`call.autoOffHook.1.protocol`		
phone1	added	`call.autoOffHook.2.protocol`		
phone1	added	`call.autoOffHook.3.protocol`		
phone1	added	`call.autoOffHook.4.protocol`		
phone1	added	`call.autoOffHook.5.protocol`		
phone1	added	`call.autoOffHook.6.protocol`		
phone1	added	`reg.1.protocol.H323`		
phone1	added	`reg.1.protocol.SIP`		
phone1	added	`reg.1.server.H323.1.address`		
phone1	added	`reg.1.server.H323.1.expires`		
phone1	added	`reg.1.server.H323.1.port`		
phone1	added	`reg.2.protocol.H323`		
phone1	added	`reg.2.protocol.SIP`		
phone1	added	`reg.2.server.H323.1.address`		
phone1	added	`reg.2.server.H323.1.expires`		
phone1	added	`reg.2.server.H323.1.port`		
phone1	added	`reg.3.protocol.H323`		
phone1	added	`reg.3.protocol.SIP`		
phone1	added	`reg.3.server.H323.1.address`		
phone1	added	`reg.3.server.H323.1.expires`		
phone1	added	`reg.3.server.H323.1.port`		
phone1	added	`reg.4.protocol.H323`		
phone1	added	`reg.4.protocol.SIP`		
phone1	added	`reg.4.server.H323.1.address`		
phone1	added	`reg.4.server.H323.1.expires`		

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
phone1	added	`reg.4.server.H323.1.port`		
phone1	added	`reg.5.protocol.H323`		
phone1	added	`reg.5.protocol.SIP`		
phone1	added	`reg.5.server.H323.1.address`		
phone1	added	`reg.5.server.H323.1.expires`		
phone1	added	`reg.5.server.H323.1.port`		
phone1	added	`reg.6.protocol.H323`		
phone1	added	`reg.6.protocol.SIP`		
phone1	added	`reg.6.server.H323.1.address`		
phone1	added	`reg.6.server.H323.1.expires`		
phone1	added	`reg.6.server.H323.1.port`		
sip	added	`call.autoAnswer.H323`		0
sip	added	`call.autoAnswer.micMute`		1
sip	added	`call.autoAnswer.ringClass`		4
sip	added	`call.autoAnswer.SIP`		0
sip	added	`call.autoAnswer.videoMute`		0
sip	added	`call.autoRouting.preference`		line
sip	added	`call.autoRouting.preferredProtocol`		SIP
sip	removed	`httpd.lp.port`		
sip	removed	`httpd.ta.enabled`		
sip	added	`log.level.change.h323`		4
sip	added	`log.level.change.poll`		4
sip	added	`log.level.change.push`		4
sip	added	`log.level.change.wmgr`		4
sip	removed	`mb.launchpad.enabled`		
sip	removed	`mb.main.1.icon`		
sip	removed	`mb.main.1.text`		
sip	removed	`mb.main.1.url`		

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
sip	removed	mb.main.2.icon		
sip	removed	mb.main.2.text		
sip	removed	mb.main.2.url		
sip	removed	mb.main.3.icon		
sip	removed	mb.main.3.text		
sip	removed	mb.main.3.url		
sip	removed	mb.main.4.icon		
sip	removed	mb.main.4.text		
sip	removed	mb.main.4.url		
sip	removed	mb.main.5.icon		
sip	removed	mb.main.5.text		
sip	removed	mb.main.5.url		
sip	removed	mb.main.6.icon		
sip	removed	mb.main.6.text		
sip	removed	mb.main.6.url		
sip	added	sec.H235.mediaEncryption.enabled		1
sip	added	sec.H235.mediaEncryption.offer		0
sip	added	sec.H235.mediaEncryption.require		0
sip	added	up.callTypePromptPref		1
sip	added	up.enableCallTypePrompt		1
sip	changed	up.idleBrowser.enabled	0	
sip	added	up.manualProtocolRouting		1
sip	added	up.manualProtocolRouting.softKeys		1
sip	changed	video.autoStartVideoTx		1
sip	added	video.callRate		448
sip	added	video.codecPref.H261		4
sip	changed	video.enable		1
sip	added	video.forceRtcpVideoCodecControl		0

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
sip	changed	`video.maxCallRate`		512
sip	added	`video.profile.H261.annexD`		
sip	added	`video.profile.H261.CifMpi`		1
sip	added	`video.profile.H261.jitterBufferMax`		2000
sip	added	`video.profile.H261.jitterBufferMin`		150
sip	added	`video.profile.H261.jitterBufferShrink`		70
sip	added	`video.profile.H261.QcifMpi`		1
sip	changed	`video.screenMode`		normal
sip	changed	`video.screenModeFS`		normal
sip	added	`voice.audioProfile.G719.32Kbps.payloadType`		107
sip	added	`voice.audioProfile.G719.48Kbps.payloadType`		108
sip	added	`voice.audioProfile.G719.64Kbps.payloadType`		109
sip	added	`voice.audioProfile.G719.jitterBufferMax`		200
sip	added	`voice.audioProfile.G719.jitterBufferMin`		40
sip	added	`voice.audioProfile.G719.jitterBufferShrink`		1500
sip	added	`voice.audioProfile.G719.payloadSize`		20
sip	added	`voice.codecPref.VVX_1500.G719.32Kbps`		
sip	added	`voice.codecPref.VVX_1500.G719.48Kbps`		
sip	added	`voice.codecPref.VVX_1500.G719.64Kbps`		
sip	changed	`voice.gain.tx.digital.chassis.VVX_1500`	6	3
sip	added	`voIpProt.H323.autoGateKeeperDiscovery`		0
sip	added	`voIpProt.H323.dtmfViaSignaling.enabled`		1
sip	added	`voIpProt.H323.dtmfViaSignaling.H245alphanumericMode`		1
sip	added	`voIpProt.H323.dtmfViaSignaling.H245signalMode`		1

File	Change	Configuration Parameter	Old Value	New Value
sip	added	voIpProt.H323.enable		0
sip	added	voIpProt.H323.local.port		1720
sip	removed	voIpProt.local.port		
sip	added	voIpProt.server.H323.1.address		
sip	added	voIpProt.server.H323.1.expires		
sip	added	voIpProt.server.H323.1.port		
sip	added	voIpProt.SIP.dtmfViaSignaling.rfc2976		
sip	added	voIpProt.SIP.enable		1
sip	added	voIpProt.SIP.local.port		5060

Understanding Updates to SIP 3.2.1B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.1B beside their respective Polycom tracking ID number.

New or Enhanced Features

48947 Support for the SoundPoint IP 335 product.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP configuration file parameters.

Software Version 3.2.1B Parameter Enhancements

File	Change	Configuration Parameter	New Value	Description
sip	added	ind.anim.IP_335.42.frame.1.bitmap	Handset	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.42.frame.1.duration	1300	
sip	added	ind.anim.IP_335.42.frame.2.bitmap	PlumHd	
sip	added	ind.anim.IP_335.42.frame.2.duration	1300	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.1.bitmap	Headset	
sip	added	ind.anim.IP_335.43.frame.1.duration	1300	
sip	added	ind.anim.IP_335.43.frame.2.bitmap	PlumHd	

File	Change	Configuration Parameter	New Value	Description
sip	added	ind.anim.IP_335.43.frame.2.duration	1300	
sip	added	ind.anim.IP_335.44.frame.1.bitmap	Speaker	
sip	added	ind.anim.IP_335.44.frame.1.duration	1300	
sip	added	ind.anim.IP_335.44.frame.2.bitmap	PlumHd	
sip	added	ind.anim.IP_335.44.frame.2.duration	1300	

Understanding Updates to SIP 3.2.1

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.1 beside their respective Polycom tracking ID number.

Enhanced Capabilities

53322 Setting `voIpProt.local.port` to a non-standard port does not send from or advertise that port.

53611 User Language Selection is retained during an upgrade to SIP 3.2.0.

53685 The phones no longer ignore `nat.ip` parameters.

53852 DTMF duration on the SoundStation IP 7000 defaults to 300ms for HDX integration.

Understanding Updates to SIP 3.2.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.0 beside their respective Polycom tracking ID number.

New or Enhanced Features

22527 Implemented Scrolling Status Bar on the SoundPoint IP 320, 321, 330, 331, 550, 560, 650, 670, and SoundStation IP 6000 and 7000.

26754 Support for the iLBC codec on the SoundPoint IP 320, 321, 330, 331, 450, 550, 560, 650, and 670.

30079 Added support for mutual TLS authentication. See Technical Bulletin 52609 for more details on this feature.

32259 Microbrowser recognizes multiple mime types.

32753 Support for LLDP protocol. To take full advantage of this feature, you will need to use BootROM 4.2.0.

34782 Replaced libSRTP algorithms with OpenSSL versions.

35525 The DND icon contains text identifying that DND is active.

37118 Added the ability to take a screen capture.

39358 Added a Loud Ringer Ringtone selection. See Technical Bulletin 39358 for instructions on how this can be configured.

30855 Create a SoundStation IP 7000 Setup Guide.

41579 Met requirements of ETSI TS 102 027-2 v4.1.1 RFC 3261 compliance test for Anatel/Brazil.

43141 Support for Statically Configured BLF and Call Park and Retrieve enhancements.

43142 Support for single button Blind Transfer and Retrieve of a call designated as an automata in the dialog used for Statically Configured BLF.

43646 Improved boot speed in some situations where the boot server is incorrectly configured.

45057 Languages selection presented in appropriate language.

45174 Upgraded zlib to version 1.2.3.

45743 Upgraded curl library to version 7.19.2.

45787 Added instructions to the SoundPoint IP 450, 550, 560, 650, and 670 for changing label colors in the User Guides.

45791 Added a configuration option on the SoundStation IP 7000 to disable digit-map rules for Remote Dialing when connected to an HDX.

46093 Added the ability for User to enable/disable display of idle browser from menu.

46113 Added navigation button shortcuts in Idle Mode consistent with other phone models (*applies to SoundPoint IP 320, 321, 330, and 331*).

46248 Added an Admin menu option on the SoundStation IP 7000 to manually specify the value to be used as the extension displayed on the phone screen.

46424 Improved readability of Menu items when using Background images on the display.

46446 New menu option to view the status of feature licenses.

46683 Removed Background from scrolling Status Bar for improved readability.

47355 Scrolling Status Bar gives equal time to each status message.

47390 Added configuration parameters for select ETSI SIP compliance requirements.

47463 The phones allow for secure entry of passwords in the micro-browser API.

47487 Added the ability to enable/disable a Back soft key in the microbrowser

47689 Added support for SoundStation IP 7000/HDX6000 Integration. This feature requires a future update release to the HDX6000 software.

47749 Support Transmission of Join Header as per RFC 3911.

48004 Support for BLF call pick-up using Dialog-info within an INVITE with Replaces header.

48055 Improved user experience of the Enhanced BLF feature when an incoming call occurs whilst the user is viewing BLF monitored line call details.

48109 Included fmtp attribute specifying Mode=30 in the SDP when 13.33 Kbps iLBC is used.

48136 Removed platform specific TFTP code and instead used TFTP support in curl library 7.19.2.

48137 Support for BLF call pick-up using Dialog-info within an INVITE with Replaces header.

48205 Support for the iLBC Codec (*applies to SoundStation IP 6000 and 7000*).

48559 Consistent scrolling status line on various phones (*applies to SoundPoint IP 450, 550, 560, 650, and 670; SoundStation IP 6000 and 7000*).

48578 Reduced the local Contact Directory maximum to 99 on the SoundPoint IP 430.

48579 Reduced the maximum number of calls supported to 4 (from 8) on the SoundPoint IP 430.

48664 Added user accessible menu option to display whether a device certificate is installed.

48678 Media Statistics menu is more easily accessible. Accessed from Menu > Status > Diagnostics > Media Statistics.

48738 Added configurable behavior for Directed Call Pick-Up as used for Enhanced BLF.

48780 Added option to apply digit-map rules to tel:URI initiated calls.

48846 Added configuration option for whether the call appearance on a remotely monitored BLF line should be presented on the monitoring/attendant phone.

48861 Add configuration option `voIpProt.SIP.strictReplacesHeader` to control whether the phone requires call-id, to-tag, and from-tag to perform and INVITE with Replaces.

48984 The phone will populate the display-name field in the To header of responses that it generates.

48998 Added configuration option for the phone to send 486 Busy when a call is rejected.

49309 Combined the SoundPoint IP 550 and 560 user guides.

49465 Updated Destination of outbound call based on the display name in the SIP To header responses.

49660 During call forwarding *user=phone* should be included in `refer-to` parameter of Refer header.

49695 Allow for SDP offer or answer in provisional reliable response and PRACK request and response.

49839 RTP Rx detects and corrects for G.722, G.722.1, G.722.1C, and G.719 RTP timestamp increments based on different sample rates.

50769 Added support for Hook-Flash during POTS calls on the SoundStation IP 7000.

50927 Added Equifax Secure eBusiness CA-1 to the trusted CA list.

51419 RFC2543 Hold not working when video SDP present in certain scenarios.

Discontinued Features

48283 Removed support for SoundPoint IP 301, 501, 600, and 601 phones.

48698 Removed support for SoundStation IP 4000.

Enhanced Capabilities

Application load progress bar matches actual progress.

29148 The phone formats the file system when it notes an error on the screen while loading large configuration files.

29344 HTTP Digest Authentication works on IIS.

30219 Logs are uploaded when phone resets to factory default.

31858 When two phones with a shared line simultaneously resume a held call, the phone which did not retrieve the call shows call in progress on its shared line indicator.

34681 The parameters `stickyAutoLineSeize` and `call.enableOnNotRegistered=0` do not seize correctly if the 1st line is unregistered.

35288 The Web Configuration Utility uses less memory during initialization.

35991 The Roaming Buddy list with Office Communicator reports the proper status of all buddies.

36969 The SoundStation IP 6000 displays Japanese language correctly.

38348 The SRTP call displays proper line icons in a certain scenario on the SoundPoint IP 320, 321, 330, and 331.

38392 Performing a Blind Transfer from an encrypted phone to an unencrypted private line establishes the new call as encrypted.

38418 The phones no longer show SRTCP authentication failure at log level 0.

38824 After audio diagnostics such as Record and Play in handset, the 1st call is no longer established in handset mode even if the handset is ON-HOOK.

39013 Attaching a cell phone cable to the SoundStation IP 7000 no longer invokes the Cell phone UI until a physical cell phone is attached.

39143 The P-Asserted-Identity header in initial INVITE message is no longer used for caller ID.

39949 The navigation icon in the Corporate Directory correctly displays the available navigation options when using the keypad to navigate (*applies to SoundPoint IP 320, 321, 330, and 331*).

40679 Changing the status on the MyStatus menu of the SoundStation IP 6000 changes the OC client status when `roaming_buddies.reg= 1`.

40892 The Time/Date displays on the SoundStation IP 7000 when the first phone call is established.

41939 The user is not able to play the WAV file when it has a call on hold and also in remote busy state. Junk characters appear in audio player.

42092 Special Slovenian characters are included in the phone's fonts.

42213 The SIP: string displays on the SoundStation IP 7000 when using URL dialing.

42611 Recording no longer begins when a full USB drive is attached

42761 Pressing the Content soft key on the SoundStation IP 7000 no longer prompts the user to choose VGA input.

43910 The microbrowser can process an http response which contains an image/bmp.

43916 Configured sampled wave files can be downloaded onto the phone depending on sufficient RAM Disk size.

43990 Missing glyphs in the Katakana bit stream fonts on the SoundStation IP 7000.

44100 Call display names containing an @ symbol no longer truncate characters after the @ symbol.

44248 The microbrowser displays an error message when unsupported media is configured in the microbrowser URL.

44273 The phones can process all contacts in a SIP Contact header containing a comma separated list.

44278 The phone numbers are displayed correctly on line keys when the length of a phone number is more than 10 characters.

44301 The Date on the SoundStation IP 6000 and 7000 when the idle browser is enabled.

44377 The Redial key can be reassigned.

44443 The Menu exit via the Menu key is ignored while in Edit mode (*applies to SoundPoint IP 320, 321, 330, and 331*).

44635 The SoundStation IP 6000 phone uses the correct configuration parameters to download customizable fonts.

44783 The Cipher list is the same for different TLS transactions.

44844 USB Call Recording can be stopped using the Stop soft key.

44855 When using Call Lists, the Missed Calls are incremented on Call Forward on Busy.

44892 When using SCA Barge-In on the SoundStation IP 6000 and 7000 phones, the user no longer barges in to the wrong call in certain scenarios.

44962 The phone no longer displays 3-way animation icon in held screen when conference legs on hold.

45143 When the maximum conference size is reached when using Centralized Conference, the phone no longer displays a local conference UI.

45327 When the user establishes a call between two phones configured as shared lines, and presses the down arrow key, all soft keys no longer disappear.

45428 An unexpected re-INVITE no longer occurs before BYE when removing a leg from a conference call.

45650 In a double hold with music on hold and a non-Polycom SIP phone, – MOH no longer fails.

45658 The platform string in transmitted CDP packets is consistent across SoundPoint IP products.

45716 Text on the SoundPoint IP 450 is consistent as on other phones.

45835 Status Bar text on the SoundPoint IP 450 is easier to read on some backgrounds.

45943 Correct logic is used when picking line for outgoing call in a multiple registration scenario.

46068 Transfer On Proceeding is supported when using a proxy server.

46334 DTMF local rendering does not stop. If the far end holds while local digit key is pressed then the far end resumes.

46478 On the EFK feature, the phone sends invite when executing $Cwaitdialtone$.

46513 Dialog Event Package Content Guideline 6B (Local Identity).

46514 Dialog Event Package Content Guideline 6C (Local Target).

46547 Warning Header Text notification on the SoundStation IP 7000 displays on phone (when configured).

46550 Directed-Call-Pickup no longer fails when SIP server is a proxy.

46588 Info soft key on the SoundStation IP 7000 is no longer missing in the Contact Directory.

46738 The `attendant.ringType` parameter is removed from the override file when default (silent) attendant ring type is selected.

46741 Using enhanced BLF, when the watched line hangs up an outgoing call, the remote call appearance screen times out on the console phone.

46770 On the microbrowser, the * and # buttons work correctly when the text input mode is set to numeric on input fields.

46899 When using the electronic hookswitch, audio is heard during an active call if the user answers by pressing the hookswitch button immediately on a Jabra headset under a specific scenario.

47039 The line LED flashes instead of remaining a stable green when an active call is kept on hold during an incoming call.

47123 When using the USB Call Recording, the missed call notification no longer displays on the audio player screen if an incoming call is not answered during playback.

47207 When the MUTE is active on the SoundStation IP 7000, it no longer covers up the dialing fields.

47248 Hot dial works when lifting the handset for the second call when `call.stickyAutoLineSeize=1`.

47300 URL dial disabled message displays and successfully routes to voicemail from Message Center tab.

47336 The Received\Missed call list on the SoundStation IP 7000 no longer shows the IP address of the SIP server instead of the Extension number of a call received/Missed from a SIP extension.

47464 When two incoming calls are active on a phone, lifting the handset or pressing the handsfree key to answer the call no longer results in the most recent call being answered even though the ring tone is played according to the first incoming call (*applies to SoundPoint IP 320 and 330, and SoundStation IP 7000*).

47535 The soft keys no longer reset to the default layout on an inbound call in some multiple call handling scenarios.

47566 When an internal URI is executed with multiple VolUp and VolDown action URIs, the Ringer horizontal bar is seen and the Volume sound going UP and DOWN is heard.

47578 When using the Corporate Directory on the SoundPoint IP 320, 321, 330, 331, the sticky attributes are saved.

47612 When using BLF, cancelling a Transfer for a call that was initiated using Directed Call Pick-Up sequence results in the correct caller-id display to the user.

47641 The *Network Link Down* message on the SoundStation IP 7000 displays on the screen unless the phone reboots and comes up with Ethernet cable.

47695 When the phones have two registrations, the NewCall soft key no longer displays for alerting call appearance when there are max call appearances (*applies to SoundPoint IP 320, 321, 330, 331, 430, and 450*).

47699 When using XML API Internal URIs on the SoundStation IP 6000, the Tel URI is works properly if embedded within a couple of internal URI actions.

47712 A local contact directory search on the SoundPoint IP 320, 321, 330, and 331 works correctly.

47724 Mute icon and Call appearance counter on the SoundPoint IP 450 no longer conflict when DND is turned on and multiple call appearances are present on the phone.

47729 The on-hook dialing widget no longer uses multi-tap behavior but is not in multi-tap mode.

47746 The NewCall soft key is not displayed when phone holds max conference calls.

47798 The location of the Transfer and Conference soft keys on the SoundStation IP 7000 are more easily accessible during conference setup.

47847 When using BLF, the monitoring phone continues ringing if a shared line is seized while the monitored line has an incoming call.

47853 When the headset memory mode is active, the Headset key continues blinking during incoming calls after ending the first active call.

47862 The Time and Date on the SoundStation IP 6000 displays during a call.

47863 The phone's HTTP server is no longer sending some HTTP traffic in very small TCP segments.

47916 The Resume soft key on the SoundPoint IP 320, 321, 330, and 331 is available for second call appearance after splitting conf established through Join from different shared line registrations.

47921 The order of call appearances on the SoundPoint IP 320, 321, 330, and 331 is consistent with other phones after splitting a conference.

47929 Rendering special characters like no longer break the hyperlink style display.

47932 The Call widget counter (1/n) appears while in the dial tone state.

47951 Transfer has precedence over pickup of a ringing BLF line when pressing the line key during a call transfer.

47953 Call info display on the SoundStation IP 6000 displays properly when volume up/down key is pressed.

47958 More than one contact can be added when the SoundStation IP 7000 is configured with no Ethernet cable connected + HDX.

47962 An incorrect icon is no longer displayed when Redialing POTS call on the SoundStation IP 7000.

48003 The SoundStation IP 7000 phone no longer dials a POTS call as a video call when dialing from the idle state for a certain configuration.

48011 Use of the Idle Browser on the SoundStation IP 7000 no longer interferes with some display elements such as the Mute Icon, Video/Phone Call pop-up when connected to HDX.

48019 The pop-up message *Video or Phone Call?* is no longer overwritten by the idle browser on the SoundStation IP 7000.

48045 When using enhanced BLF, the phone holds the first call when pressing the Dial soft key to make the second call to the same called party.

48049 When using BLF, the attendant phone displays all remote calls on a BLF monitored line if the Monitored Phone has a call in the Ringing state.

48061 When using enhanced BLF, the attendant phone updates the 1/x widget when the BLF monitored line has one or multiple incoming calls being ended.

48069 When using the SCA Barge-In feature, extra soft keys are no longer displayed on remote shared phone while viewing call appearance list by long pressing line key.

48071 Key:Handsfree internal URI action is executed by the phone in a certain scenario.

48115 HDX no longer plays a ring sound after answering POTS call on the SoundStation IP 7000.

48131 Call Forwarding Status now shows multiple Call Forward Types are selected.

48149 SDP attribute is no longer truncated when first character of the value is a digit.

48162 The Boot Server status field no longer shows an incomplete or blank path if a / is included in the setting.

48174 A failed call no longer causes subsequent calls to skip URL/Number mode selection.

48179 A called Party number is no longer shown overlapped in incoming event notification when IP dialed calls are made between unregistered phones.

48209 Left-most character can be deleted before character selection timeout.

48213 Key:LineX is executed only if X is a supported line key for that platform.

48333 When using the USB Call Recording, the USB busy indicator appears on main screen when recording in progress.

48414 The phone no longer occasionally fails to act on the electronic hookswitch up/down signal from Plantronics and Hydra headsets.

48700 When using the USB Call Recording, playback can be stopped through a Stop soft key.

48745 LDAP Critical Extension Error 0x0c no longer causes the CD Server to not respond to messages from phone.

48981 SRTP no longer fails in 3.1.2 when the user presses Hold then Resume during a call. This happens on several different models of IP phone.

48996 The phone tags correct DSCP value to some packets (Trying, Ringing and OK).

49106 The entire dialed URL is saved in the phone’s call history

49251 The Polish XML Dictionary includes Polish characters.

49300 Ensure that the DTMF tones are being sent via the dtmf start/stop Clink2 API (*applies to SoundStation IP 7000*).

49417 The phone no longer reports MOH dialog if SUBSCRIBE received while on hold.

49459 Cancel works after entering hot dial digits.

49461 DND symbol(X) appears after the DND feature is disabled in a certain configuration.

49473 When using the Corporate Directory on the SoundPoint IP 320, 321, 330, 331, using the # key to change text entry mode it resets the Quick Search timeout timer.

49476 The scrolling indicators on the Corporate Directory work better.

49512 HTTP Refresh header response loads the specified URL on the phones after the specified amount of time has passed, in a certain situation.

49516 Hanging up the handset terminates calls in Audio or Display Diagnostics.

49523 Asian fonts are clearer on the SoundPoint IP 450 and SoundStation IP 7000.

49548 The Edit and Delete soft keys on the SoundPoint IP 320, 321, 330 and 331 disappear after deleting the last contact.

49572 When using the Corporate Directory on the SoundStation IP 7000, numeric characters can be entered in the Quick Search entry field.

49617 The phone plays a dial tone after a hold reminder is played in certain scenarios.

49619 The call waiting beep plays on phone when call hold reminder is set.

49620 Volume settings for Recording work in handsfree mode.

49639 The Handsfree dial tone is no longer interrupted by hold reminder and call waiting ringtones.

49641 Call info display on the SoundStation IP 6000 and 7000 displays properly while changing volume.

49677 The phone complies with RFC4475 3.1.2.3 Negative Content-Length.

49685 On SoundPoint IP 320, 321, 330, and 331, you can enter URLs with uppercase letters.

49692 The seconds colon in the time display blinks for every second on the SoundPoint IP 450.

49693 The ACD icon displays when the parameter `voIpProt.SIP.serverFeatureControl.cf=1` is enabled.

49696 After a long LAN outage while downloading a new application, when the phone re-connects to the network, it displays an error message.

49701 The SoundStation IP 7000 phone response with `reg.1.server.1.expires=5` setting is consistent.

49706 The SIP Extension display on the SoundStation IP 7000 is no longer disabled after disconnecting from HDX with HDX-Preference option.

49757 The SoundStation IP 7000 phone displays Network Link is Down after the cable is disconnected from a hub

49758 The SoundStation IP 7000 phone no longer gets into a bad state and can recover from temporarily unplugging network connection during an active call.

49776 If `dir.corp.user` is misconfigured, the phone displays Login Error.

49813 When using the Corporate Directory, the phones no longer display *Enter More Chars*... when submitting a string that returns no results in the Quick search mode.

49825 When using the Corporate Directory, the black background for the Search bar displays consistently on different platforms.

49829 NTP Time synchronization is reliable in a particular scenario.

49834 When using the Corporate Directory, if VLV indexing is configured and an Advanced Find yields more results than the configured page Size (Default is 64), scrolling through the entries works correctly.

49836 If the Corporate directory is down and the phone reboots, the phones displays a static Please try again message.

49911 Incoming ring tones are played on the phone in a certain enhanced BLF use case.

49926 The SoundPoint IP 320, 321, 330, and 331 phones no longer auto-increment the new contacts speed dial index to 100 even though the maximum amount of entries is 99.

49927 After an AdvFind search, exit and re-enter Corp Dir menu, phone displays search bar as Search: not Search (Filtered) (*applies to SoundPoint IP 320,321,330,331 and VVX 1500*).

49929 The SoundStation IP 7000 is displays HDX Extension, when voice call type is set to Auto and phone is not registered to SIP server.

49981 After rebooting the SoundStation IP 7000, the proper HDX extension displays.

49982 The SoundPoint IP 320, 321, 330, and 331 phones reconfigure when DHCP lease expires.

49989 The SoundStation IP 7000 phone is no longer adding contact directories from the call list with the existing speed dial number.

49977 The SoundPoint IP 320, 321, 330, and 331 phones display the selected status under MyStat menu.

50090 The SoundStation IP 7000 phone displays an Active Conference screen on joining a remotely held SLA call without first holding the local call.

50099 Consultative Transfer no longer fails if the second leg is forwarding and its 302 response is handled by proxy.

50109 Volume levels on the SoundStation IP 7000 are in Sync when Dialing a Video call

50110 An *Enter number* message displays for Video and audio calls once the Ethernet is removed on the SoundStation IP 7000.

50115 The DTMF tone of the first digit on the SoundStation IP 7000 plays at the SoundStation IP 7000 volume instead of the HDX volume.

50118 Dial tone volume and Hands Free volume are in sync on the SoundStation IP 7000.

50137 The volume no longer resets to default on the SoundStation IP 7000 after a POTS call is connected if `voice.volume.persists.handsfree=0`.

50153 When using the Corporate Directory, setting the Primary Attribute as sticky `dir.corp.attribute.1.sticky=1` gives a clearer user interface behavior.

50159 When using the Corporate Directory, a Quick search on a non-null sticky primary filter is no longer missing records.

50189 SIP responses are no longer missing the to-tag after the phone challenges INVITE.

50212 Scrolling upward for a while on the Corporate Directory sorts the phone entry list in order.

50253 When using the Corporate Directory on the SoundStation IP 7000and the edit phone number attribute in AdvFind menu, pressing on the 1/A/a soft key creates an Encoding soft key.

50254 The phone does honors SDP sent in PRACK.

50255 SIP Reliable Provisional responses are retransmitted.

50256 When not yet registered, phones will experience a random delay of 30-60 sec between registration attempts.

50264 Global prefix +present on calls made from Placed Calls list.

50299 When using the Corporate Directory on the SoundStation IP 7000, Quick search text input starts at the first multi tap character.

50381 Pressing the left navigation key on the SoundPoint IP 320, 321, 330, and 331 before the character selection timeout no longer moves cursor 2 spots.

50397 The SoundStation IP 7000 phone displays licenses correctly in the status screen.

50407 When the Corporate Directory server is down with phone connecting to LDAP server, a quick search results in the phone displaying a proper error message.

50523 When using the Corporate Directory on the SoundPoint IP 320, 321, 330, 331, the phone displays the Contact title in the View menu.

50546 With URL dialing disabled, a BLIND soft key appears in the third soft key slot after pressing TRNSFER.

50811 P-Asserted ID display name is a sticky on UI call appearance and in the placed call list.

50869 The phone will only offer SRTP when SRTP crypto suite is selected.

50891 The Resume soft key on the SoundStation IP 6000 and 7000 displays when the phone is put on hold on another shared line phone.

50989 Receiving a 603 Decline by a BLF monitored user plays a reorder tone.

51041 Regarding X-IdleBrowserSelectUrl, *http://url* is no longer remembered by the phone.

51245 BLF state is updated on receipt of the first full state NOTIFY after a reboot.

51320 The message Conference in Another Video or phone call? Is no longer displayed in a loop for each press on Conf hard key (*applies to SoundStation IP 7000*).

51432 The Conference Hard key Pop-up Message on the SoundStation IP 7000 does not display any message except directly allowing the user to make a video call.

51554 The phones no longer add an additional CRC to some 802.1X packets received on the PC port.

51567 Server based CFWD/DND sync no longer fails on 3.1.2.0392.

51605 API Push request will no longer be lost if it immediately follows another push request.

51631 The phone releases the first assigned IP address when VLAN is set via DHCP.

51633 The phone plays busy/reorder tone upon a refer-based transfer when it gets a 603 or 486 response.

51644 Certain Japanese strings now display correctly.

51690 The EFK feature is used for one touch Voicemail dialing. When using EFK with SIP 3.1.3, the phone honors the `stickyautolineseize`.

51718 The phone no longer continues to ring after a call has been answered with a certain call signaling sequence.

51763 When adding video to an existing call on a SoundStation IP 7000, pressing the Mute key successfully mutes the far end.

51838 Japanese characters are properly displayed.

52014/53597 In SIP 3.x.x, when an IP phone picks up a transferred call in a certain scenario, the call is connected instead of being placed on hold.

52017 The Web interface issue Password entry is masked when entered.

52108 The phone successfully restores destination to Asserted Identity or Remote ID after a transfer fails.

Configuration File Enhancements

Refer to the following table for a list of the parameters that have been added, changed, or deleted from the template **phone1.cfg** and **sip.cfg** files. You can find further descriptions of parameters in Administrators' Guide for the SIP 3.2.0 Release.

Note also that the template file **000000000000.cfg** has been modified in order to facilitate support for the Legacy phones and the VVX 1500 in this release.

Software Version 3.2.0 Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`call.directedCallPickupMethod`		native or legacy	See Administrators Guide for SIP 3.2.0 for details.
sip	added	`call.parkedCallRetrieveMethod`		native or legacy	
sip	added	`call.parkedCallRetrieveString`		Star code	See Administrators Guide for SIP 3.2.0 for details.
sip	added	`dialplan.applyToRemoteDialing`		0 or 1; Default is 0	A flag to determine if the dial plan applies to calls made through the Polycom HDX system.
sip	added	`dialplan.applyToTelUriDial`		0 or 1 Default is 1	A flag to determine if the dial plan applies to uses of the tel:// URI.
sip	added	`ind.class.2.state.35.index`		44	Changes Relating to screen layout modifications.
sip	added	`ind.class.2.state.36.index`		42	
sip	added	`ind.class.2.state.37.index`		43	
sip	changed	`ind.gi.IP _400.4.physX`	122	0	
sip	changed	`ind.gi.IP _400.5.physX`	112	10	
sip	changed	`ind.gi.IP _4000.6.physH`	12	0	
sip	changed	`ind.gi.IP _4000.6.physW`	14	0	
sip	changed	`ind.gi.IP _4000.6.physX`	16	0	
sip	changed	`ind.gi.IP _4000.6.physY`	2	0	

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	changed	ind.gi.IP _450.16.physX	176	196	
sip	changed	ind.gi.IP _450.17.physX	176	196	
sip	changed	ind.gi.IP _450.18.physX	176	196	
sip	changed	ind.gi.IP _450.19.physX	176	196	
sip	changed	ind.gi.IP _450.2.physX	40	20	
sip	changed	ind.gi.IP _450.3.physH	20	0	
sip	changed	ind.gi.IP _450.3.physW	20	0	
sip	changed	ind.gi.IP _450.3.physX	20	0	
sip	changed	ind.gi.IP _450.3.physY	2	0	
sip	changed	ind.gi.IP _600.13.physH	103	111	
sip	changed	ind.gi.IP _600.13.physY	0	25	
sip	changed	ind.gi.IP _600.4.physY	105	3	
sip	changed	ind.gi.IP _600.6.physH	20	0	
sip	changed	ind.gi.IP _600.6.physW	20	0	
sip	changed	ind.gi.IP _600.6.physX	113	0	
sip	changed	ind.gi.IP _600.6.physY	110	0	
sip	changed	ind.gi.IP _7000.3.physH	20	0	
sip	changed	ind.gi.IP _7000.3.physW	20	0	
sip	changed	ind.gi.IP _7000.3.physX	20	0	
sip	added	lcl.ml.lang.menu.1.label		简体中文 (zh-cn)	Language selection displayed in the appropriate language.
sip	added	lcl.ml.lang.menu.10.label		日本語 (ja-jp)	
sip	added	lcl.ml.lang.menu.11.label		한국어 (ko-kr)	
sip	added	lcl.ml.lang.menu.12.label		Norsk (no-no)	
sip	added	lcl.ml.lang.menu.13.label		Polski (pl-pl)	
sip	added	lcl.ml.lang.menu.14.label		Português (pt-br)	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`lcl.ml.lang.menu.15.label`		Сский (ru-ru)	
sip	added	`lcl.ml.lang.menu.16.label`		Slovenski (sl-si)	
sip	added	`lcl.ml.lang.menu.17.label`		Español (es-es)	
sip	added	`lcl.ml.lang.menu.18.label`		Svenska (sv-se)	
sip	added	`lcl.ml.lang.menu.2.label`		Dansk (da-dk)	
sip	added	`lcl.ml.lang.menu.3.label`		Nederlands (nl-nl)	
sip	added	`lcl.ml.lang.menu.4.label`		English (en-ca)	
sip	added	`lcl.ml.lang.menu.5.label`		English (en-gb)	
sip	added	`lcl.ml.lang.menu.6.label`		English (en-us)	
sip	added	`lcl.ml.lang.menu.7.label`		Français (fr-fr)	
sip	added	`lcl.ml.lang.menu.8.label`		Deutsch (de-de)	
sip	added	`lcl.ml.lang.menu.9.label`		Italiano (it-it)	
sip	added	`log.level.change.lldp`		4	Control the logging detail level for the LLDP feature.
sip	added	`mb.main.autoBackKey`		1	See Administrators Guide for SIP 3.2.0 for details.
sip	changed	`ramdisk.minfree`	3072	3150	Minimum amount of free space that must be left after the RAM disk has been created.
sip	changed	`se.pat.ringer.13.name`	Sampled 1		Customer ringer file names.

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	changed	`se.pat.ringer.14.name`	Sampled 2		
sip	changed	`se.pat.ringer.15.name`	Sampled 3		
sip	changed	`se.pat.ringer.16.name`	Sampled 4		
sip	changed	`se.pat.ringer.17.name`	Sampled 5		
sip	changed	`se.pat.ringer.18.name`	Sampled 6		
sip	changed	`se.pat.ringer.19.name`	Sampled 7		
sip	changed	`se.pat.ringer.20.name`	Sampled 8		
sip	changed	`se.pat.ringer.21.name`	Sampled 9		
sip	changed	`se.pat.ringer.22.name`	Sampled 10		
sip	added	`sec.srtp.requireMatchingTag`		0 or 1	A flag to determine whether or not to check the tag value in the crypto attribute in an SDP answer.
sip	changed	`tone.dtmf.rfc2833Payload`	101	127	The phone-event payload encoding in the dynamic range to be used in SDP offers.
sip	added	`up.idleBrowser.enabled`		0 or 1; default is 0	A flag to determine whether or not the background takes priority over the idle browser. Used in conjunction with up.prioritizeBackground.enable.

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	up.prioritizeBackgroundMenuItem.enabled		0 or 1; default is 1.	If set to 1, the Prioritize Background menu is available to the user. The user can then decide whether or not the background takes priority over the idle browser. Used in conjunction with up.idleBrowser.enabled.
sip	added	up.screenCapture.enabled		0 or 1; Default is 0	A flag to determine whether or not the user can get a screen capture of the current screen shown on a phone. The flag is cleared when the phone reboots.
sip	added	voice.audioProfile.iLBC.13_33Kbps.payloadSize		30	See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.audioProfile.iLBC.15_2Kbps.payloadSize		20	
sip	added	voice.audioProfile.iLBC.jitterBufferMax		160	
sip	added	voice.audioProfile.iLBC.jitterBufferMin		40	
sip	added	voice.audioProfile.iLBC.itterBufferShrink		500	
sip	added	voice.audioProfile.iLBC.payloadType		110	
sip	removed	voice.audioProfile.Lin16.44.1ksps.payloadType	120		Parameter renamed.
sip	added	voice.audioProfile.Lin16.44_1ksps.payloadType		120	See Administrators Guide for SIP 3.2.0 for details
sip	added	voice.audioProfile.Lin16.8ksps.payloadType		116	See Administrators Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.iLBC.13_33Kbps			

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	added	voice.codecPref.iLBC.15_2Kbps			
sip	added	voice.codecPref.IP_6000.iLBC.13_33Kbps			
sip	added	voice.codecPref.IP_6000.iLBC.15_2Kbps			
sip	added	voice.codecPref.IP_650.iLBC.13_33Kbps			
sip	added	voice.codecPref.IP_650.iLBC.15_2Kbps			
sip	added	voice.codecPref.IP_7000.iLBC.13_33Kbps			
sip	added	voice.codecPref.IP_7000.iLBC.15_2Kbps			
sip	added	voIpProt.SDP.early.answerOrOffer			If set to 1, an SDP offer or answer is generated in a provisional reliable response and PRACK request and response. If set to 0, an SDP offer or answer is not generated.
sip	added	voIpProt.SDP.offer.iLBC.13_33Kbps.includeMode			See Administrators Guide for SIP 3.2.0 for details.
sip	changed	voIpProt.server.1.port	5060		The port of a SIP server that accepts registration.
sip	added	voIpProt.server.2.address			
sip	added	voIpProt.server.2.expires			Minimum now 10
sip	added	voIpProt.server.2.expires.lineSeize		30	
sip	added	voIpProt.server.2.expires.overlap			
sip	added	voIpProt.server.2.lcs			
sip	added	voIpProt.server.2.port			

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`voIpProt.server.2.register`		1	
sip	added	`voIpProt.server.2.retryMaxCount`		0	
sip	added	`voIpProt.server.2.retryTimeOut`		0	
sip	added	`voIpProt.SIP.compliance.RFC3261.validate.contentLength`			If set to 1, validation of the SIP header content language is enabled.
sip	added	`voIpProt.SIP.compliance.RFC3261.validate.uriScheme`			If set to 1 or Null, validation of the SIP header URI scheme is enabled.
sip	added	`voIpProt.SIP.strictReplacesHeader`			This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.
sip	added	`voIpProt.SIP.use486forReject`			If set to 1 and the phone is indicating a ringing inbound call appearance, phone will transmit a 486 response to the received INVITE when the Reject soft key is pressed.
phone1	added	`attendant.behaviors.display.remoteCallerID.automata`		1	Flags to determine whether or not remote party caller ID information is presented to the attendant.
phone1	added	`attendant.behaviors.display.remoteCallerID.normal`		1	
phone1	added	`attendant.behaviors.display.spontaneousCallAppearances.automata`		0	Flags to determine whether or not a call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource
phone1	added	`attendant.behaviors.display.spontaneousCallAppearances.normal`		1	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
phone1	added	`attendant.resourceList.x.address`		The value of x depends on the phone. For IP 450 x=1-2; IP 550, 560 x=1-3; IP 650, 670 x=1-47	The user referenced by attendant.reg= will subscribe to this URI for dialog.
phone1	added	`attendant.resourceList.x.label`			Text label to appear on the display adjacent to the associated line key
phone1	added	`attendant.resourceList.x.type`		normal	Type of resource being monitored.
phone1	changed	`attendant.ringType`		1	
phone1	added	`dialplan.1.applyToTelUriDial`		1	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file.
phone1	added	`dialplan.2.applyToTelUriDial`		1	
phone1	added	`dialplan.3.applyToTelUriDial`		1	
phone1	added	`dialplan.4.applyToTelUriDial`		1	
phone1	added	`dialplan.5.applyToTelUriDial`		1	
phone1	added	`dialplan.6.applyToTelUriDial`		1	
phone1	changed	`divert.noanswer.1.timeout`	60	55	Modified No Answer Timeout
phone1	changed	`divert.noanswer.2.timeout`	60	55	
phone1	changed	`divert.noanswer.3.timeout`	60	55	
phone1	changed	`divert.noanswer.4.timeout`	60	55	
phone1	changed	`divert.noanswer.5.timeout`	60	55	
phone1	changed	`divert.noanswer.6.timeout`	60	55	
phone1	added	`reg.1.server.2.address`			
phone1	added	`reg.1.server.2.expires`			

File	Change	Configuration Parameter	Old Value	New Value	Description
phone1	added	reg.1.server.2.expires. lineSeize			See Administrators Guide for SIP 3.2.0 for details
phone1	added	reg.1.server.2.expires. overlap			
phone1	added	reg.1.server.2.lcs			
phone1	added	reg.1.server.2.port			
phone1	added	reg.1.server.2.register			
phone1	added	reg.1.server.2.retryMax Count			
phone1	added	reg.1.server.2.retryTimeOut			
phone1	added	reg.2.musicOnHold.uri			
phone1	added	reg.2.server.1.lcs			
phone1	added	reg.2.server.2.address			
phone1	added	reg.2.server.2.expires			
phone1	added	reg.2.server.2.expires.line Seize			
phone1	added	reg.2.server.2.expires.over lap			
phone1	added	reg.2.server.2.lcs			
phone1	added	reg.2.server.2.port			
phone1	added	reg.2.server.2.register			
phone1	added	reg.2.server.2.retryMax Count			
phone1	added	reg.2.server.2.retryTimeOut			
phone1	added	reg.2.tcpFastFailover			
phone1	added	reg.3.musicOnHold.uri			
phone1	added	reg.3.server.1.lcs			
phone1	added	reg.3.server.2.address			
phone1	added	reg.3.server.2.expires			
phone1	added	reg.3.server.2.expires.line eize			

File	Change	Configuration Parameter	Old Value	New Value	Description
phone1	added	reg.3.server.2.expires.overlap			
phone1	added	reg.3.server.2.lcs			
phone1	added	reg.3.server.2.port			
phone1	added	reg.3.server.2.register			
phone1	added	reg.3.server.2.retryMaxCount			
phone1	added	reg.3.server.2.retryTimeOut			
phone1	added	reg.3.tcpFastFailover			
phone1	added	reg.4.musicOnHold.uri			
phone1	added	reg.4.server.1.lcs			
phone1	added	reg.4.server.2.address			
phone1	added	reg.4.server.2.expires			
phone1	added	reg.4.server.2.expires.lineSeize			
phone1	added	reg.4.server.2.expires.overlap			
phone1	added	reg.4.server.2.lcs			
phone1	added	reg.4.server.2.port			
phone1	added	reg.4.server.2.register			
phone1	added	reg.4.server.2.retryMaxCount			
phone1	added	reg.4.server.2.retryTimeOut			
phone1	added	reg.4.tcpFastFailover			
phone1	added	reg.5.musicOnHold.uri			
phone1	added	reg.5.server.1.lcs			
phone1	added	reg.5.server.2.address			
phone1	added	reg.5.server.2.expires			
phone1	added	reg.5.server.2.expires.lineSeize			
phone1	added	reg.5.server.2.expires.overlap			

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
phone1	added	`reg.5.server.2.lcs`			
phone1	added	`reg.5.server.2.port`			
phone1	added	`reg.5.server.2.register`			
phone1	added	`reg.5.server.2.retryMaxCount`			
phone1	added	`reg.5.server.2.retryTimeOut`			
phone1	added	`reg.5.tcpFastFailover`			
phone1	added	`reg.6.musicOnHold.uri`			
phone1	added	`reg.6.server.1.lcs`			
phone1	added	`reg.6.server.2.address`			
phone1	added	`reg.6.server.2.expires`			
phone1	added	`reg.6.server.2.expires.lineSeize`			
phone1	added	`reg.6.server.2.expires.overlap`			
phone1	added	`reg.6.server.2.lcs`			
phone1	added	`reg.6.server.2.port`			
phone1	added	`reg.6.server.2.register`			
phone1	added	`reg.6.server.2.retryMaxCount`			
phone1	added	`reg.6.server.2.retryTimeOut`			
phone1	added	`reg.6.tcpFastFailover`			

Understanding Updates to SIP 3.1.7

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.7 beside their respective Polycom tracking ID number.

New or Enhanced Features

61028 Added support for SoundPoint IP 430.

61547 The phones now send a 486 (Busy) response to a received INVITE message when a call is rejected.

Enhanced Capabilities

51718 Under certain configurations, phone no longer continues to ring after the call has been answered.

52968 Deleted instant messages can be removed from the main screen.

53975 The phones send a SUBSCRIBE message in a certain scenario when using an SCA with barge-in enabled.

55884 The displays on a SoundPoint IP 650 with expansion modules no longer freeze during a consultative transfer.

58689 The phones no longer send a 486 if an INVITE is received after a NOTIFY for the alerting state and the configuration parameter `callsPerLineKey` is set to 1.

58728 The phone presents the New Call soft key and the End Call soft key to allow the user to release the call and place the phone into idle state after hanging up the call during a consultative transfer.

59789 On the SoundPoint IP 650, the user is able to properly resume a held call after answering a different call.

60051 On the SoundPoint IP 650 using a BLA, the display does shows the status of the remotely held call while there is an active call currently displayed. Pressing the Down Arrow key followed by the Up Arrow key refreshes the display to properly show the status of the held call.

60141 On the SoundPoint IP 650, on a Bridged Line Appearance BLA line, the display incorrectly indicates 2 call appearances when there should only be one for the active call. The 2nd call appearance is for the previously held remote call that is no longer on hold.

60145 On the SoundPoint IP 650 using a BLA, the display on the phone correctly presents 2 call appearances instead of only one.

60177 The display on the SoundPoint IP 5xx and 6xx presents hot-dialed digits when the idle display feature is enabled.

60264 During a call using a BLA line, when the display is showing the dialing screen, remote call appearances are no longer displayed when the remote phones BLA line resumes a call.

60340 The Join soft key no longer displays for phones with BLA lines when there is only one call active on the phone.

60480 A phone monitoring other BLA lines show the presence (LED goes out) of a BLA line when that monitored line joins two other calls.

60756 A phone monitoring a Shared Call Appearance line presents a correct presence indication of a BLA line when that monitored line joins two other calls in a centralized conference.

61264 Calls placed on hold using a shared BLA line timeout when a remote phone picks up the held call (on the BLA line).

61283 When a user attempts to place a conference call on hold and the phone receives a 400 Bad request. The phone no longer sends a NOTIFY with *<parampname=+sip.rendering pvalue=no />*.

61298 When 1.2Mbps of multicast traffic is passed through the PC port on the SoundPoint IP 601 phone, the data port no longer experiences a packet loss of 17%.

61299 When a phone has established a centralized conference call, the user is able to transfer a third incoming call.

61321 When a phone joins a centralized conference bridge, other monitoring phones correctly show the BLA line as being on hold instead of being in use.

61547 The phone sends a 486 Busy message when a call (INVITE) is rejected. A binary configuration parameter is added to **sip.cfg** called `voIpProt.SIP.use486forReject`. By default, (parameter is 0) the feature is disabled. If the parameter equals 1, the feature is enabled. If enabled and the phone is indicating a ringing inbound call appearance, then upon pressing the Reject soft key, the phone will transmit a 486 Response to the originator of the received INVITE message.

61725 Users can pick up a held call after multiple hold/resume interactions on the phone.

61950/62024 The phone honors a retry-after header in a 500 Glare message responding to a BLA re-SUBSCRIBE message.

62036 The SoundPoint IP 3xx phone continues sending DTMF RTP EVENTS when receiving a second incoming call while it is already active on a previously established call.

62050 The SoundPoint IP 650 phone properly updates the number of held calls after sending 200 OK messages as part of the notifications process.

62127 The Blind transfer soft key on the SoundPoint IP 650 is presented on the display when the Transfer soft key is pressed on the second call.

62223 The phone no longer crashes after resuming a held call using a BLA.

62226 The phone no longer proceeds to join a conference after receiving a 403 Forbidden from the switch.

62262 The phone no longer establishes a 1-way audio path after it has re-established a centralized conference call with the dropped 3rd party. This behavior is observed with Sylantro switches.

62279 The presence indicator on a Bridged Line Appearance displays correctly after the phone receives a 486 message.

62313 Using a BLA configuration, a dial tone is present when pressing the second line key followed by lifting handset after holding a call on first line appearance.

62361 The call status on a BLA Bridged Line Appearance (configured for 1 call per line appearance) of a monitoring phone is updated correctly when transfer/conference soft key is pressed.

62435 The SoundPoint IP 650 phone correctly displays a call appearance labeled Unknown Party if the remote party is held while reorder tone is played locally.

62511 In certain situations, the monitored Busy Lamp Field line invokes an incoming call notification (icon and tone).

62514 In certain situations, the status of the monitored Busy Lamp Field lines on the SoundPoint IP 670 is removed from the display even though the status has been updated by the switch.

62569 The phone no longer generates a redundant NOTIFY message when triggered by a 100 response during a re-INVITE.

62669 When multiple phones try to resume a held Bridge Line Appearance BLA line at the same time, the presence indicator on the BLA line is preserved on the trailing phone when the reorder tone is played.

62672 Either Directed Call Pickup DCP or Group Call Pickup feature (using soft keys instead of *53 and *54 feature access codes) no longer fail when the user enters an account code. The account code is appended to the user portion of the URI.

62704 The presence indicator of a Bridged Line Appearance BLA is updated correctly on monitoring phones when the phones LAN data cable is disconnected and then re-connected.

62926 The Resume soft key on the SoundPoint IP 3xx displays when the line key is pressed continuously while the line is in a remote held call state. This occurs when the line is configured as `callsPerLineKey=1`.

63099 The phones monitoring Bridged Line Appearance BLA line, configured for one call per line, can pick up the held call after the call on a BLA line has been put on hold using the Transfer/Conference key.

63286 The phone’s Part Number is listed correctly instead of YYYY-YYYYY-YYY.

64212 Invoking the Call Park feature with the soft key on the SoundPoint IP 3xx functions correctly when the soft key is configured as 1 line and 1 call per line.

64219 The SoundPoint IP 3xx phone sends a proper hold NOTIFY message after a consultative transfer is canceled when the configuration parameter `notifyTransferHoldAsActive` is disabled.

64271 In an attempt to answer an incoming call, the call is no longer unintentionally terminated. This occurs when the incoming calls line key is pressed simultaneously as the handset is lifted.

64274 In an attempt to resume a held call, the held call is no longer unintentionally terminated when the user inadvertently seizes two line keys simultaneously.

64327 In an attempt to answer an incoming call with the user inadvertently pressing 2 line keys, the user is no longer connected to both lines one with an incoming caller on one and a dial tone on the other.

64340 The indicator, on a Bridged Line Appearance BLA line that is monitoring other lines, blink after the monitored phone performs the following sequence: Transfer > Split > EndCall > Resume > Hold.

64356 The display on the SoundPoint IP 3xx showing a remote call appearance times out when the user presses continuously a BLA line key followed by pressing a down arrow key while there are multiple calls on hold on the remote BLA.

64822 When configuring the SoundPoint IP 3xx phones using **sip_att.cfg**, the phone no longer shows Service Unavailable when the speed dial key is pressed while the phone is off-hook.

64862 Joining an internal extension with an external PSTN call no longer causes one call to drop.

65119 When a Bridged Line Appearance BLA line is presented in a dialing screen, the remote call appearance is correctly displayed when the remote BLA line resumes a call.

65207 A slow memory leak due to the receipt of hunt group INVITE containing replaces no longer occurs in the SIP stack.

67186 All soft keys on the SoundPoint IP 301, 501, and IP 601 no longer disappear on the assistant phone when pressing down the arrow key after placing multiple calls on hold with the boss line appearance.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to software version 3.1.7 configuration file parameters.

Software Version 3.1.7 Parameter Enhancements

File	*Action*	*Parameter*	*Modification Description*
sip	added	`voIpProt.SIP.use486forReject`	Defaults to null
sip	added	`call.localConferenceEnabled=1`	Defaults to 1

Reference Documents

This section lists all documents referred to in these release notes as well as other relevant documents.

For information and support for all Polycom voice products and software, and for access to supporting documentation, see Polycom UC Software Support Center.

Polycom UC Software Administrators' Guide

- Polycom UC Software 4.1.0 Administrators' Guide

Technical Bulletins, Quick Tips, White Papers, and Engineering Advisories

- Feature Profile 84393: Using the Polycom BroadSoft UC-One Application on Polycom VVX Business Media Phones
- White Paper: Configuration File Management on Polycom® SoundPoint® IP, SoundStation® IP, and VVX® Phones
- Technical Bulletin 35311: Maintaining Older Polycom® Phones Beyond Their Last Supported Software Release
- Technical Bulletin 39358: Using Custom Ringtones on Polycom® SoundPoint® IP, SoundStation® IP, and Polycom VVX® 1500 Phones
- Technical Bulletin 52609: Mutual Transport Layer Security Provisioning Using Microsoft® Internet Information Services 6.0
- Technical Bulletin 56449: Polycom® SoundPoint® IP /SoundStation® IP /VVX® Software Changes in the Next Release
- Quick Tip 57215: Phone Lock Feature on Polycom® Phones Running Polycom UC Software
- Technical Bulletin 60519: Simplified Configuration Enhancements in Polycom® UC Software 3.3.0
- Engineering Advisory 64731: Polycom® UC Software 4.0.x: Upgrade and Downgrade Methods
- Technical Bulletin 66743: Security Advisory Relating to Denial of Service Vulnerability on Polycom® SoundPoint® IP and SoundStation® IP Phones
- Feature Profile 72430: Using Polycom Phones with Microsoft Lync Server 2010

User Guides

- SoundPoint IP Phones
- SoundStation IP Phones
- VVX Business Media Phones

Miscellaneous

- SIP/UC Software Downloads Matrix