Polycom® UC Software 3.3.3

Release Notes

Applies to SoundPoint® IP, SoundStation® IP, and VVX® Phones

UC Software 3.3.3 | November 2011 | 3804-11530-333RevB

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General

These release notes apply to UC Software (UCS) 3.3.3 and include previous versions of the Polycom UC Software. UC Software 3.3.3 will support SoundPoint IP, SoundStation IP, and VVX phones. For a complete guide to UC Software 3.3.3, refer to the *Administrator's Guide for Polycom UC Software 3.3.0.*

The UC Software 3.3.3 Release Notes contain the following sections:

- GeneralGeneral You will need to read this section in order to understand how the changes in UC Software 3.3.3 affect Polycom hardware and deployment and configuration of the software.
- What's New for Polycom UC Software 3.3.3? **Error! Reference source not found.**This section lists new, enhanced, and discontinued software features.
- Updates to Previous UC 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. Each item is linked for instant access.

Important Notes and Considerations in UC Software 3.3.3

This section contains important notes on Polycom hardware and software.

Configuration Files

Since UC Software 3.3.0, Polycom has simplified and extensively modified the configuration files, their respective parameters and defaults, and the provisioning methods. Some of the enhancements are not backward compatible with configuration parameters from previous software releases. To familiarize yourself with the enhancements, and to find out whether the enhancements are compatible with previous software releases, Polycom recommends reading *Technical Bulletin 60519: Simplified Configuration Enhancements in Polycom® UC Software 3.3.0* and the *Polycom UC Software Administrators' Guide – UC Software 3.3.0* before you install UC Software 3.3.3.

Upgrade and Downgrade Considerations for VVX 1500 Phones

Upgrading VVX 1500 phones to SIP 3.2.2 or later requires a specific procedure. Before you begin upgrading your VVX phones to SIP 3.2.6, refer to *Technical Bulletin 53522: Upgrading the Polycom VVX 1500 Phone to SIP 3.2.2*. VVX 1500 phones running SIP 3.2.2 or later cannot be downgraded to earlier SIP software or BootROM software versions.

Considerations for Legacy Phones

UC Software 3.3.3 does not include support for the SoundPoint IP 300, 301, 430, 500, 501, 600, 601, and SoundStation IP 4000 products. These products are termed *legacy products* and will be supported for critical issue fixes in the SIP 2.1.x release (IP 300, 500), SIP 3.2.x (IP 430), and SIP 3.1.x release (for the other legacy products). If you want to support these legacy phones using SIP 3.2.0 or later, see *Technical Bulletin 35311: Maintaining Older Polycom® Phones Beyond Their Last Supported Software Release*.

Managing SoundStation IP 7000 Phones with HDX Integration

If your phone deployment includes SoundStation IP 7000 phones with HDX integration, Polycom recommends using the following software versions: UC Software 3.3.1 with BootROM 4.3.1 and HDX version 2.6.1.3, 3.0, 3.0.0.1, 3.0.0.2, 3.0.1, 3.0.2, or 3.0.2.1.

Understanding Phone Platform Features and Licenses

UC Software 3.3.3 supports the Productivity Suite, which includes a Corporate Directory, Visual Conference Management, and USB Call Recording. The Voice Quality Monitoring (VQMon) feature will continue to be a licensed and paid product.

For customers using versions of Polycom software 3.3.3 and prior, Polycom will provide a site license for all features in the Productivity Suite, except VQMon. This license file is available on the Polycom portal and is free to download at http://www.polycom.com/products/voice/applications/index.html.

You can refer to Table 1: SoundPoint IP and VVX Series Features and Licenses and Table 2: SoundStation IP Series Features and Licenses to find out whether, in UC Software 3.3.3, a phone does not support a feature (No), a phone supports a feature without a license (Yes), or the phone requires a Productivity License to support a feature.

Table 1: SoundPoint IP and VVX Series Features and Licenses

Feature	*IP 320/330*	*IP 321/331/335*	*IP 450/550/560*	*IP 650/670*	*VVX 1500C/1500D*
VQ Mon	Productivity License	Productivity License	Productivity License	Productivity License	Yes
LDAP Directory	Productivity License*	Productivity License*	Productivity License*	Productivity License*	Yes
Call Recording	No	No	No	Productivity License*	Yes (Audio only)
Conference Management	No	No	Productivity License*	Productivity License*	Yes
4-way local conference	No	No	Yes	Yes	No

Feature	*IP 320/330*	*IP 321/331/335*	*IP 450/550/560*	*IP 650/670*	*VVX 1500C/1500D*
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes
Customizable UI Background	No	No	Yes	Yes	Yes
Local SRTP Conference	Yes	Yes	Yes	Yes	Yes, with limitations at high bandwidths
Asian Language	No	Yes	Yes	Yes	Yes
Configurable Soft Keys	Yes	Yes	Yes	Yes	Yes
XML API	Yes	Yes	Yes	Yes	Yes
Enhanced BLF	No	No	Yes	Yes	No
Warning Field Display	Yes	Yes	Yes	Yes	Yes
H.323 Video	No	No	No	No	License installed on the 1500D

Productivity License -> Free to download.*

Table 2: SoundStation IP Series Features and Licenses

Feature	*IP 5000*	*IP 6000*	*IP 7000*
VQMon	Productivity License	No	No
LDAP Directory	Productivity License*	Productivity License*	Productivity License*
Call Recording	No	No	No
Conference Management	No	No	Yes
4-way local conference	No	No	No
Electronic Hookswitch	No	No	No
Enhanced Feature Keys	No	No	No
Customizable UI Background	No	No	No
Local SRTP Conference	Yes	Yes	Yes

Feature	*IP 5000*	*IP 6000*	*IP 7000*
Asian Language	Yes	Yes	Yes
Configurable Soft Keys	No	No	No
XML API	Yes	Yes	Yes
Enhanced BLF	No	No	No
Warning Field Display	Yes	Yes	Yes
H.323 Video	No	No	No

System Requirements

Polycom recommends using UCS 3.3.3 with BootROM version 4.3.1. Otherwise, refer to Table 3: Recommended Platforms and BootROM Software to help you choose an appropriate BootROM version for your phone.

For more information about which phone models a particular software version supports, refer to the Polycom UC Software/ SIP Software Release Matrix.

Table 3: Recommended Platforms and BootROM Software

Platform	*BootROM Version*
SoundPoint IP 320/330	3.2.3RevB or later
SoundPoint IP 321/331	4.1.3 or later
SoundPoint IP 335	4.2.0RevB or later
SoundPoint IP 450	4.1.2 or later
SoundPoint IP 550	4.1.0 or later
SoundPoint IP 560	4.1.0 or later
SoundPoint IP 650	4.1.0 or later
SoundPoint IP 670	4.1.1 or later
SoundStation IP 5000	4.2.2 or later
SoundStation IP 6000	4.1.1 or later

Platform	*BootROM Version*
SoundStation IP 7000	4.1.1 or later
SoundStation IP 7000 **Used with Polycom HDX 4000, 6000, 7000, 8000, 9000 video systems running one of the following release:** **2.6.1.3, 3.0, 3.0.0.1, 3.0.0.2, 3.0.1, 3.0.2, 3.0.2.1**	Release UCS 3.3.1 with BootROM 4.3.1 is recommended.
VVX 1500	4.2.2 or later NOTE: As of 3.2.2, the SIP and BootROM are distributed as single package for the VVX 1500.

Downloading the Distribution Files

You can download UCS 3.3.3 using either the combined template file or the split template file, both in ZIP file format. For general use, Polycom recommends using the split file that corresponds to the phone model(s) for your deployment.

Refer to Table 4: Understanding the Split ZIP File to match the correct software file to your phone model. If you are provisioning your phones centrally using configuration files, download the corresponding file version and extract the configuration files to the provisioning server, maintaining the folder hierarchy in the ZIP file.

In some deployments, you may need to download both the combined and split file. For example, deployments running a BootROM release prior to 4.0.0 will require the combined file version. If you require both file versions, download both ZIP files, extract all the configuration files from the split version and extract the **sip.ld** file from the combined file version. The **sip.ld** from the combined version will be combined with the configuration files from the split version.

The current build ID for the **sip.ld** and resource files listed in Table 4: Understanding the Split ZIP File and Table 5: Understanding the Combined ZIP FileTable 5: Understanding the Combined ZIP File is 3.3.3.0069.

Downloading the Split ZIP File

Polycom recommends using the split ZIP file when possible for a shorter upgrade time. Use Table 4: Understanding the Split ZIP File for a brief description of each file in the split distribution. Downloading the split zip file requires that all phones be running BootROM release 4.0.0 or later.

Table 4: Understanding the Split ZIP File

Distributed Files	*File Purpose and Application*
2345-12200-002.sip.ld **2345-12200-005.sip.ld**	SIP application executables for SoundPoint IP 320
2345-12360-001.sip.ld	SIP application executables for SoundPoint IP 321
2345-12200-001.sip.ld **2345-12200-004.sip.ld**	SIP application executables for SoundPoint IP 330
2345-12365-001.sip.ld	SIP application executable for SoundPoint IP 331
2345-12375-001.sip.ld	SIP application executable for SoundPoint IP 335
2345-12450-001.sip.ld	SIP application executable for SoundPoint IP 450
2345-12500-001.sip.ld	SIP application executable for SoundPoint IP 550
2345-12600-001.sip.ld	SIP application executable for SoundPoint IP 650
2345-12670-001.sip.ld	SIP application executable for SoundPoint IP 670
3111-30900-001.sip.ld	SIP application executable for SoundStation IP 5000
3111-15600-001.sip.ld	SIP application executable for SoundStation IP 6000
3111-40000-001.sip.ld	SIP application executable for SoundStation IP 7000
2345-17960-001.sip.ld	SIP application executable for VVX 1500
sip.ver	Text file detailing build-id(s) for the release
000000000000.cfg	Example master configuration file
000000000000-directory~.xml	Example per-phone local contact directory XML file (edit and then remove '~' from name to seed phones which have no directory)
applications.cfg	Configuration parameters: micro-browser applications
features.cfg	Configuration parameters: telephony features
H323.cfg	Configuration parameters: H.323 signaling protocol

Distributed Files	*File Purpose and Application*
reg-advanced.cfg	Configuration parameters: Line/call registration – advanced feature settings-full
reg-basic.cfg	Configuration parameters: Line/call registration -– basic settings
region.cfg	Configuration parameters: Regional/localization settings (language, etc.)
sip-basic.cfg	Configuration parameters: VoIP server/softswitch registration. Basic settings
sip-interop.cfg	Configuration parameters: VoIP server/softswitch registration/inter-operability configuration settings/registration
site.cfg	Configuration parameters: parameters expected to be set on a per-site basis
video.cfg	Configuration parameters: Video connectivity
video-integration.cfg	Configuration parameters: For SoundStation IP7000/HDX Integration
SoundPointIP-dictionary.xml	Dictionary files for multilingual support include: • Chinese, China (for IP 450, 550, 560, 650 and IP 5000, 6000, 7000 only) • 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 and IP 5000, 6000, 7000 only) • Korean, Korea (for IP 450, 550, 560, 650, 670 and IP 5000, 6000, 7000 only) • Norwegian, Norway • Polish, Poland • Portuguese, Portugal • Russian, Russia • Slovenian, Slovenia • Spanish, Spain • Swedish, Sweden

Distributed Files	*File Purpose and Application*
SoundPointIPWelcome.wav	Start up welcome sound effect
LoudRing.wav	Loud ringer sound effect
Warble.wav	Loud ringer sound effect

Downloading the Combined ZIP File

Use Table 5: Understanding the Combined ZIP File as a reference guide to the files distributed in the combined ZIP file and a brief description of each file. The combined file is required for any phones running a BootROM release prior to 4.0.0, for example, BootROM 3.2.3 RevB.

Table 5: Understanding the Combined ZIP File

Distributed Files	*File Purpose and Application*
sip.ld	Concatenated SIP application executable
sip.ver	Text file detailing build-id(s) for the release
000000000000.cfg	Example master configuration file
000000000000-directory~.xml	Example per-phone local contact directory XML file (edit and then remove '~' from name to seed phones which have no directory)
applications.cfg	Configuration parameters: micro-browser applications
features.cfg	Configuration parameters: telephony features
H323.cfg	Configuration parameters: H.323 signaling protocol
reg-advanced.cfg	Configuration parameters: Line/call registration – advanced feature settings- full
reg-basic.cfg	Configuration parameters: Line/call registration -– basic settings
region.cfg	Configuration parameters: Regional/localization settings (language, etc.)
sip-basic.cfg	Configuration parameters: VoIP server/softswitch registration. Basic settings
sip-interop.cfg	Configuration parameters: VoIP server/softswitch registration/inter-operability configuration settings/registration

Distributed Files	*File Purpose and Application*
site.cfg	Configuration parameters: parameters expected to be set on a per-site basis
video.cfg	Configuration parameters: Video connectivity
video-integration.cfg	Configuration parameters: For SoundStation IP7000/HDX Integration
SoundPointIP-dictionary.xml	Dictionary files for multilingual support include: • Chinese, China (for IP 450, 550, 560, 650 and IP 5000, 6000, 7000 only) • 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 and IP 5000, 6000, 7000 only) • Korean, Korea (for IP 450, 550, 560, 650, 670 and IP 5000, 6000, 7000 only) • 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 3.3.3?

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to UC Software 3.3.3 beside their respective Polycom tracking identification number.

New or Enhanced Features

48734 In a server-based centralized conference, the phone now sends both REFERs without waiting for the 202 acknowledgment.

63780 Enhanced API for the VVX1500 browser Webkit.

72279 Enabled Call Back feature using an Enhanced Feature Keys (*applies to VVX 1500*).

Enhanced Capabilities

64742 Using a dial plan containing #, the phone sends out the proper invite message.

73247 The default entry mode in the phone quick setup menu is set to numeric.

73466 The phone updates the new provisioning server details when modified, instead of using the existing details.

73686 The phone can properly populate call list information, including missed, dialed, and received calls (*applies to VVX 1500*).

73689 The EFK configured for the Call Back feature no longer behaves like a redial key (*applies to VVX 1500*).

74074 On the phone Web UI, the phone line registration password can no longer be viewed using view source on the Web page.

74285 Restored the headset/handset audio changes to comply with IEC 60268-1 TUV safety test standards.

74431 When BLF feature is enabled, the monitoring phone receives a beep sound when the monitored phones go off hook.

74672 Phone now updates the centralized conference parameter `voIpProt.SIP.conference.address` when Update Configuration is performed on the phone menu.

74744 Phone now displays an error message for insufficient bandwidth during a central conference call.

74794 In a centralized conference call, when the maximum number of lines is reached, the phone no longer displays the local conference UI.

74849 Phone no longer displays wrong soft key option on the UI when the conference service is unavailable (*applies to SoundPoint IP 235,321,331*).

74895 Phones can properly handle a 403 response to first REFER sent to phone to a centralized conference server.

74912 Quick Setup: The Password entry field is now set to Numeric as default.

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

Configuration File Enhancements

Refer to Table 6: Software Version 3.3.3 - Configuration File Parameter Enhancements for a list of enhancements made to the UC Software 3.3.3 configuration file parameters.

Table 6: Software Version 3.3.3 - Configuration File Parameter Enhancements

Configuration File	*Modification*	*Parameter*	*Description of the Modification*
sip-interop	Changed	`voIpProt.SIP.failoverOn 503Response`	The default value has changed from 1 (Enabled) to 0 (Disabled).
sip-interop	Added	`voIpProt.SIP.conference. parallelRefer`	This parameter has been added to enable the phone to send parallel REFER messages without waiting for a 202 Accepted. 1: The phone sends parallel REFER messages without waiting for a 202 Accepted message. 0: The phone waits for a 202 Accepted message before sending the next REFER message.

Updates to Previous UC Software Releases

This section indicates updates to versions of the UC Software prior to 3.3.3.

Understanding Updates to UCS 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 Display enhancements: Provides simplified display option by removing protocol tag & host details (*applies to VVX 1500D*).

66624 Geographical redundancy enhancements.

67633 Added support for the Zero Touch Provisioning ZTP feature. Note that this feature should not be enabled unless you are using the ZTP service. To find out more about this service contact your Polycom sales representative.

68511 Added the functionality to set “NO Answer ringcount” to “1” via menu options / configuration files (*applies to VVX 1500D*).

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

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 popup 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 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 is displayed on the phone to enter more digits. After entering the other two digits it no longer append to the earlier two digits thereby resulting in a failed call (*applies to SoundPoint IP 3xx running UCS3.3.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 Phone displays the correct caller name in the user interface where there is a call from a group pickup number.

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 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 Table 7: Software Version 3.3.2 - Configuration File Parameter Enhancements for a list of enhancements made to the UC Software 3.3.2 configuration file parameters.

Table 7: Software Version 3.3.2 - Configuration File 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		DefaultCbRestart
		cfgParamType		param
		default		
		help		
		max		256
		min		0
		templates		device, site, new
		type		String
device.prov.ztpEnabled.set	added	callback		DefaultCbRestart
		cfgParamType		param
		default		0
		help		
		templates		device, site, new
		type		Bool
dialplan.x.routing.server.y.transport	changed	enum		transport
		max	256	
		min	0	
		type	String	Enum
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

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
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	
		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		

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		numReplace1Max		Const_NumLineReg
		templates		reg-advanced
		type		Bool
reg.x.outboundProxy.transport	changed	enum		transport
		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
		help		
		templates		new
		type		Bool
voIpProt.SIP.dialog.strictVersionValidation	added	accessType		Admin
		callback		CbSipStack

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		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.only SignalWithRegistered	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
voIpProt.SIP.serverFeatureControl.missed Calls	added	accessType		Admin
		callback		DefaultCbRestart
		cfgParamType		param
		default		0
		help		
		templates		sip-interop
		type		Bool
voIpProt.server.x.failOver.onlySignal WithRegistered	added	callback		CbReg
		cfgParamType		paramArray
		defaultAll		1
		help		
		numReplace1Max		Const_NumServers
		templates		sip-interop

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		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
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

Configuration Parameter	*Action*	*Property*	*Old Value*	*New Value*
		min		-1000
		templates		new
		type		SInt

Understanding Updates to UCS 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 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 Security Advisory Relating to Denial of Service Vulnerability on Polycom® SoundPoint® IP and SoundStation® IP Phones.*

Understanding Updates to UCS 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 for CMA presence (*applies to VVX 1500*).

52476 Added support for Premium extensions to server synchronized ACD feature.

55059 Added support for Feature Key Synchronization using FAC/NOTIFY message combination. Hosted IP solutions are implementing Synchronization of Feature key Functions (e.g. DND/CFWD) using a Feature Access Code (FAC) to set the Feature, and a SIP NOTIFY message to inform the phone of the feature state.

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 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 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 - 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 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 sipX server, there is no music on resume from a double Music On Hold (MOH) between two phones.

60814 The Login soft key now displays when `feature.acdLoginLogout.enabled` is set (*applies to SoundStation IP 6000*).

60831 Ringback tone continues to play for 30 seconds after the phone sends a BYE message (*applies to SoundStation IP 6000*).

60848 After invoking the Update Configuration menu option, the phone now returns to the idle screen.

60897 The “Custom Ringer Types” menu uses the proper configured name.

61030 The Buddy Watch presence no longer fails on the phone after it boots initially with `voIpProt.H323.enable`=1 (*applies to VVX 1500*).

61031 Active call does not have 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 VVX 1500*).

61042 The Directed call pick-up feature does not work when the SUBSCRIBE message has “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; SoundStation IP 5000, 7000*).

61088 The phone no longer freezes and reboots after making a call with `tcpIpApp.port.rtp.forceSend`=1024 (*applies to VVX 1500*).

61090 The configuration parameter `voIpProt.SIP.musicOnHold.uri` is now updated upon a configuration change.

61095 While dialing a URL using the on-screen keyboard, the first entered character is unexpectedly deleted (*applies to VVX 1500*).

61102 The Ringer icon properly displays when you adjust the ringer volume while the phone is idle (*applies to SoundStation IP 5000, 6000*).

61104 With a shared line configured on the phone, activity on the remote shared line will cause the idle browser content to cycle off then on (*applies to VVX 1500*).

61114 Phone can successfully boot up with the DHCP VLAN 256 DVD option. The user interface no longer halts at the BootROM count-down screen (*applies to VVX 1500*).

61115 You can now answer calls after a configuration update is invoked.

61242 The configuration parameter `voIpProt.SIP.useCompleteUriForRetrieve` now updates upon a configuration change.

61246 The `voIpProt.SIP.allowTransferOnProceeding` XML schema has the proper value type as specified in the administrator's guide. The values are: 0, 1, & 2.

61273 Joining calls into local conference when 1 leg is a remotely held BLA line no longer results in no audio between both remote users.

61314 The number of characters for custom names is limited to 12. The number has been extended to 127.

61367 When dialing a number with a '+' sign, e.g. +492101099210, "user=phone" is not added to the "To" header.

61677 The phone escapes the '%' character as '%25' when it is present in the destination of a call (*applies to VVX 1500*).

61723 The phone is missing the first string "<?xml version="1.0" encoding="utf-8" ?>" in FAST UPDATE request which causes an integrated RMX to reject the INFO method (*applies to VVX 1500*).

61779 Under certain conditions, the phone no longer reboots spontaneously from idle state or in-use state.

61904 A call is placed with the correct signaling protocol when the line is configured as "dual line" protocol (*applies to VVX 1500*).

62036 Phone now continues sending DTMF RTP EVENTS when receiving a second incoming call during an active primary call (*applies to SoundPoint IP 320,330*).

62114 Users can now unlock the phone after the phone is locked with a password containing letters (*applies to VVX 1500*).

62325 VVX 1500: Chinese characters cause the phone to become unresponsive to user requests.

62333 VVX 1500: Incorrect Chinese characters are displayed in the "reboot" menu.

62417 When an off-hook event is received from the headset base station, the phone sends 3 events to the base station. This results in unusual audio effects at the headset. This affects all DHSG headsets and platforms.

62453 Phone displays ghost call appearance labeled 'Unknown Party' if remote party is held while 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 plays dial tone and RTP media when resuming on a call held at another phone.

62704 BLA presence recovers properly on the monitoring phone when the LAN cable is disconnected and then re-connected.

62906 Phones correctly provision using the HTTPS protocol option when using a server certificate with an older MD2 digest message algorithm.

63076 Phones with BLA lines are able to establish more than 10 outgoing calls.

63214 Phone will no longer reboot if it receives more REFERs that `reg.x.callsPerLineKey` is configured for.

Configuration File Enhancements

Refer to Table 8: Software Version 3.3.1 – Configuration File Parameter Enhancements for a list of enhancements made to the UC Software 3.3.1 configuration file parameters. The list applies only to the enhancements made since UCS 3.3.0.

Table 8: Software Version 3.3.1 – Configuration File Parameter Enhancements

Configuration Parameter	*Action*
`attendant.resourceList.x.callAddress`	added
`attendant.resourceList.x.proceedingIsRecipient`	added
`device.cma.disableTlsForDebug`	added
`device.cma.disableTlsForDebug.set`	added
`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 UCS 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.lineKeyCallTerminate`. Type=Bool. Default=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 TB60519 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.

52459 Website uses the new configuration system.

52493 Added support for MD4 encryption key (OpenSSL).

52532 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. The associated configuration parameter is `call.transferOnConferenceEnd`.

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 is displayed 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 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 schemeä, ö, ü / Ä, Ö, Ü ß.

55745 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 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 display 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 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 call.

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 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 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.

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 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 on occasion from 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*).

58177 Blind transfer: in certain scenarios, when two phones receive a PSTN call and two people attempt to blind transfer to an internal extension, they will hear a series of “beeps” after pressing the "Send" soft key indicating that the transfer was not successful. If they cancel the transfer and try again, the transfer will complete properly.

58197 After upgrading from 3.0.0 to 3.1.3 RevC, 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 through 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

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, Polycom recommends 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*.

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.6 beside their respective Polycom tracking identification number.

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 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 Section 0.

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 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 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 pickup 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.

Configuration File Enhancements

Refer to Table 10: Software Version 3.2.5 - Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.2.5 configuration file parameters.

Table 10: Software Version 3.2.5 - Configuration File Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
phone1	added	`reg.n.server.m.failOver.onlySignal WithRegistered`	N/A	Null
phone1	added	`reg.n.outboundProxy.failOver.onlySignal WithRegistered`	N/A	Null
phone1	added	`reg.n.filterReflectedBlaDialogs`	N/A	Null
sip	added	`voIpProt.server.n.failOver.onlySignal WithRegistered`	N/A	Null
sip	added	`voIpProt.SIP.CID.sourcePreference`	N/A	Null
sip	added	`voIpProt.SIP.failoverOn503Response`	N/A	1

File	Change	Configuration Parameter	Old Value	New Value
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 version 3.2.4B beside their respective Polycom tracking ID number.

Enhanced Capabilities

66743 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: Security Advisory Relating to Denial of Service Vulnerability on Polycom® SoundPoint® IP and SoundStation® IP Phones*

Understanding Updates to SIP 3.2.4

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP version 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 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, 335*).

55884 On occasion, the display freezes and both BLF Expansion Modules' display may become blank during a consultative transfer. The phone does not recover and has to be rebooted (*applies to SoundPoint IP 650 with attached expansion modules*).

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 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.

Configuration File Enhancements

Refer to Table 11: Software Version 3.2.3 - Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.2.3 configuration file parameters.

Table 11: Software Version 3.2.3 - Configuration File Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
phone1	added	`reg.n.server.1.failOver.reRegisterOn`	N/A	
phone1	added	`reg.n.server.1.failOver.failBack.mode`	N/A	
phone1	added	`reg.n.server.1.failOver.failBack.timeout`	N/A	
phone1	added	`reg.n.server.2.failOver.reRegisterOn`	N/A	
phone1	added	`reg.n.server.2.failOver.failRegistrationOn`	N/A	
phone1	added	`reg.n.server.2.failOver.failBack.mode`	N/A	
phone1	added	`reg.n.server.2.failOver.failBack.timeout`	N/A	
phone1	added	`reg.n.outboundProxy.failOver.reRegisterOn`	N/A	
phone1	added	`reg.n.outboundProxy.failOver.failRegistrationOn`	N/A	

File	Change	Configuration Parameter	Old Value	New Value
phone1	added	reg.n.outboundProxy.failOver.failBack.mode	N/A	
phone1	added	reg.n.outboundProxy.failOver.failBack.timeout	N/A	
phone1	added	reg.n.useCompleteUriForRetrieve	N/A	1
sip	added	voIpProt.server.1.failOver.reRegisterOn	N/A	
sip	added	voIpProt.server.1.failOver.failRegistrationOn	N/A	
sip	added	voIpProt.server.1.failOver.failBack.mode	N/A	
sip	added	voIpProt.server.1.failOver.failBack.timeout	N/A	
sip	added	voIpProt.server.2.failOver.reRegisterOn	N/A	
sip	added	voIpProt.server.2.failOver.failRegistrationOn	N/A	
sip	added	voIpProt.server.2.failOver.failBack.mode	N/A	
sip	added	voIpProt.server.2.failOver.failBack.timeout	N/A	
sip	added	voipPort.SIP .useCompleteUriForRetrieve	N/A	1
sip	added	voIpProt.SIP.outboundProxy.failOver. reRegisterOn	N/A	
sip	added	voIpProt.SIP.outboundProxy.failOver.fail RegistrationOn	N/A	
sip	added	voIpProt.SIP.outboundProxy.failOver.failBack. mode	N/A	
sip	added	voIpProt.SIP.outboundProxy.failOver.failBack. timeout	N/A	
sip	added	voIpProt.H323.blockFacilityOnStartH245	N/A	0
sip	added	se.destination	N/A	chassis
sip	added	voice.codecPref.IP _5000.G711Mu	N/A	2
sip	added	voice.codecPref.IP _5000.G711A	N/A	3
sip	added	voice.codecPref.IP _5000.G729AB	N/A	4
sip	added	voice.codecPref.IP _5000.G722	N/A	1
sip	added	voice.codecPref.IP _5000.iLBC.13_33kbps	N/A	

File	Change	Configuration Parameter	Old Value	New Value
sip	added	voice.codecPref.IP _5000.iLBC.15_2kbps	N/A	
sip	added	voice.gain.rx.analog.chassis.IP _5000	N/A	0
sip	added	voice.gain.rx.analog.ringer.IP _5000	N/A	0
sip	added	voice.gain.rx.digital.chassis.IP _5000	N/A	11
sip	added	voice.gain.rx.digital.ringer.IP _5000	N/A	-12
sip	added	voice.gain.tx.analog.chassis.IP _5000	N/A	0
sip	added	voice.gain.tx.digital.chassis.IP _5000	N/A	15
sip	added	voice.aes.hf.duplexBalance.IP _5000.0	N/A	10
sip	added	voice.aes.hf.duplexBalance.IP _5000.1	N/A	9
sip	added	voice.aes.hf.duplexBalance.IP _5000.2	N/A	8
sip	added	voice.aes.hf.duplexBalance.IP _5000.3	N/A	7
sip	added	voice.aes.hf.duplexBalance.IP _5000.4	N/A	6
sip	added	voice.aes.hf.duplexBalance.IP _5000.5	N/A	5
sip	added	voice.aes.hf.duplexBalance.IP _5000.6	N/A	4
sip	added	voice.aes.hf.duplexBalance.IP _5000.7	N/A	3
sip	added	voice.aes.hf.duplexBalance.IP _5000.8	N/A	2
sip	added	voice.ns.hf.IP _5000.enable	N/A	1
sip	added	voice.ns.hf.IP _5000.signalAttn	N/A	-6
sip	added	voice.ns.hf.IP _5000.silenceAttn	N/A	-9
sip	added	voice.rxEq.hf.IP _5000.preFilter.enable	N/A	1
sip	added	voice.rxEq.hf.IP _5000.postFilter.enable	N/A	0
sip	added	voice.txEq.hf.IP _5000.preFilter.enable	N/A	0
sip	added	voice.txEq.hf.IP _5000.postFilter.enable	N/A	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 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 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 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 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 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 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 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 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 Table 12: Software Version 3.2.2 - Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.2.2 configuration file parameters.

Table 12: Software Version 3.2.2 - Configuration File 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`		

File	Change	Configuration Parameter	Old Value	New Value
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		
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		

File	Change	Configuration Parameter	Old Value	New Value
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	removedd	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		
sip	removed	mb.main.2.icon		
sip	removed	mb.main.2.text		

File	Change	Configuration Parameter	Old Value	New Value
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.H245 alphanumericMode		1
sip	added	voIpProt.H323.dtmfViaSignaling.H245 signalMode		1
sip	added	voIpProt.H323.enable		0

File	Change	Configuration Parameter	Old Value	New Value
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 Table 13: Software Version 3.2.1B - Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.2.1B configuration file parameters.

Table 13: Software Version 3.2.1B - Configuration File Parameter Enhancements

File	Change	Configuration Parameter	Old Value	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	See *Administrators Guide for SIP 3.2.2 for details*
sip	added	ind.anim.IP_335.42.frame .2.bitmap		PlumHd	See *Administrators Guide for SIP 3.2.2 for details*
sip	added	ind.anim.IP_335.42.frame .2.duration		1300	See *Administrators Guide for SIP 3.2.2 for details*

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	added	ind.anim.IP_335.43.frame.1.bitmap		Headset	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.1.duration		1300	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.2.bitmap		PlumHd	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.2.duration		1300	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.1.bitmap		Speaker	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.1.duration		1300	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.2.bitmap		PlumHd	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.2.duration		1300	See Administrators Guide for SIP 3.2.2 for details

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 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 Add support for mutual TLS authentication. See *Technical Bulletin 52609: Mutual Transport Layer Security Provisioning using Microsoft® Internet Information Services 6.0* 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: Using Custom Ringtones on Polycom® SoundPoint® IP, SoundStation® IP, and VVX® 1500 Phones* 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 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 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

27048 Application load progress bar doesn't match actual progress

29148 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 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 is displayed 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 display 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 Phones can process all contacts in a SIP Contact header containing a comma separated list.

44278 Phone numbers are displayed correctly on line keys when the length of a phone number is more than 10 characters.

44301 The Date is displayed 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 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 2nd 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 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 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 is displayed 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 is displayed.

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 is displayed 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 Popup Message on the SoundStation IP 7000 does not display any message except directly allowing the user to make a video call..

51554 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 Some Japanese strings display correctly.

51690 The EFK feature is used for one touch Voicemail dialing. When using EFK with 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 Table 14: Software Version 3.2.0 - Configuration File Parameter Enhancements 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 the 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.

Table 14: Software Version 3.2.0 - Configuration File Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`call.directedCallPickup Method`		native or legacy	See Administrators Guide for SIP 3.2.0 for details.
sip	added	`call.parkedCallRetrieve Method`		native or legacy	See Administrators Guide for SIP 3.2.0 for details.
sip	added	`call.parkedCallRetrieve String`		Star code	See Administrators Guide for SIP 3.2.0 for details.
sip	added	`dialplan.applyToRemote Dialing`		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	
sip	changed	`ind.gi.IP _450.16.physX`	176	196	

File	Change	Configuration Parameter	Old Value	New Value	Description
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)	

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	added	lcl.ml.lang.menu.13.label		Polski (pl-pl)	
sip	added	lcl.ml.lang.menu.14.label		Português (pt-br)	
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.

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
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.
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.

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
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`.
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.

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	added	voice.audioProfile.iLBC.15_2kbps.payloadSize		20	See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.audioProfile.iLBC.jitterBufferMax		160	See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.audioProfile.iLBC.jitterBufferMin		40	See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.audioProfile.iLBC.jitterBufferShrink		500	See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.audioProfile.iLBC.payloadType		110	See Administrators Guide for SIP 3.2.0 for details.
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			See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.codecPref.iLBC.15_2kbps			See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.codecPref.IP_6000.iLBC.13_33kbps			See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.codecPref.IP_6000.iLBC.15_2kbps			See Administrators Guide for SIP 3.2.0 for details.

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	added	voice.codecPref.IP_650.iLBC.13_33kbps			See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.codecPref.IP_650.iLBC.15_2kbps			See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.codecPref.IP_7000.iLBC.13_33kbps			See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.codecPref.IP_7000.iLBC.15_2kbps			See Administrators Guide for SIP 3.2.0 for details.
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			

File	Change	Configuration Parameter	Old Value	New Value	Description
sip	added	voIpProt.server.2.port			
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.strictRepliesHeader			This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.
sip	added	voIpProt.SIP.use486forReject			If set to1 and the phone is indicating a ringing inbound call appearance, phone will transmit a 486 response to the received INVITE when the Reject soft key is pressed.
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	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
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	
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, 670x=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	

File	Change	Configuration Parameter	Old Value	New Value	Description
phone1	added	dialplan.5.applyToTel UriDial		1	
phone1	added	dialplan.6.applyToTel UriDial		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			See Administrators Guide for SIP 3.2.0 for details
phone1	added	reg.1.server.2.expires			
phone1	added	reg.1.server.2.expires.lineSeize			
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.retryMaxCount			
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.lineSeize			

File	Change	Configuration Parameter	Old Value	New Value	Description
phone1	added	reg.2.server.2.expires.overlap			
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.retryMaxCount			
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.lineSeize			
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			

File	Change	Configuration Parameter	Old Value	New Value	Description
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			
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			

File	Change	Configuration Parameter	Old Value	New Value	Description
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 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 NewCall soft key and the EndCall 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 is displayed 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 Table 15: Software Version 3.1.7 - Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.1.7 configuration file parameters.

Table 15: Software Version 3.1.7 - Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP .use486forReject`	Defaults to null
sip	added	`call.localConferenceEnabled`	Defaults to 1

Understanding Updates to SIP 3.1.6

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.6 beside their respective Polycom tracking ID number.

Enhanced Capabilities

54423 Phone no longer reboots under heavy SIP traffic while using Buddy Watch as a BLF (*applies to SoundPoint IP 601*).

54479 After upgrading from 2.1.2 to 3.1.3RevB, users can transfer calls using the Transfer key with no delay (*applies to SoundPoint IP 601 + 32 member BLF*).

Understanding Updates to SIP 3.1.5 (Limited Distribution)

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.5 beside their respective Polycom tracking ID number.

Enhanced Capabilities

54165 A phone can pick up a call on hold after it receives a NOTIFY message with dialog state="full" in response to its BLA re-subscribe message

Understanding Updates to SIP 3.1.4

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.4 beside their respective Polycom tracking ID number.

Discontinued Features

Removed support for the SoundPoint IP 320, 321, 330, 331, 430, 450, 550, 560, 650, 670 products.

Removed support for the SoundStation IP 6000, 7000 products

Removed support for the VVX 1500 product.

Enhanced Capabilities

50189 SIP responses contain a To tag after a phone challenges an INVITE message

51031 Russian is supported on the phones.

52237/52017 Web interface Password entry is masked when entered.

53826/50546 If URL dialing is disabled and you press the Transfer soft key, the Blind soft key displays in the proper position.

53827/51690 If EFK feature is used for one touch voicemail dialing, the phone adheres to the configuration set by `stickyAutoLineSeize`.

53828/52014 When an IP phone picks up a transferred call in a certain scenario, the call properly connects.

53829/50254 Phones honour SDP sent in PRACK.

54214/50869 Phones no longer only offer SRTP when SRTP crypto suite is selected.

Understanding Updates to SIP 3.1.3 C

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.3 C beside their respective Polycom tracking ID number.

New or Enhanced Features

Added support for the SoundPoint IP 321 and 331 products.

Understanding Updates to SIP 3.1.3 B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.3 B beside their respective Polycom tracking ID number.

Enhanced Capabilities

50103 Volume changes are maintained after a POTS call is established (*applies to SoundStation IP 7000 with HDX*).

50104 Performing an Advanced Find search on a corporate directory with `ViewPersistency` enabled maintains the attribute filters even after exiting and re-entering the search results menu.

50117 Incoming POTS call no longer resets the Ringer volume (*applies to SoundStation IP 7000 with HDX*).

Understanding Updates to SIP 3.1.3.0336 (Limited Distribution)

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.3.0336 beside their respective Polycom tracking ID number.

New or Enhanced Features

45869 Added support for LDAP directory queries using VLV Indexing in the corporate directory.

47179 Extended fast-failover mechanism to transactions initiated over TCP transport

47493 Improved the User Interface for the corporate directory. Refer to *Technical Bulletin 41137: Best Practices When Using Corporate Directory on SoundPoint IP, SoundStation IP and Polycom VVX Phones* for more details.

47495 Screen idle timeout resets while a corporate directory search is in process

48183 Add network jitter computation and reporting for video packet channels (*applies to VVX 1500*).

48467 Touching the LCD screen at any location wakes the LCD from the dim state to full brightness (*applies to VVX 1500*).

48484 Users can control the dial tone sound level when adding a POTS call to an existing video call (*applies to SoundStation IP 7000 with HDX*).

48854 Default value for the configuration parameter `mb.main.idleTimeout` increased from 20 to 40 seconds.

48567 When Do Not Disturb/call forwarding sync is enabled, phones do not forward or deny any calls that they receive.

Discontinued Features

48567 Removed License Requirement on uaCSTA feature

Enhanced Capabilities

23634 Computing packet stats jitter is done as explained in RFC3550 (*applies to SoundPoint IP 320/330, 430, 450, 550, 560, 650, 670; SoundStation IP 4000, and VVX 1500*). Issue remains on SoundPoint IP 301, 501, 600, 601, and SoundStation IP 6000, 7000 phones.

43517 REFER-based click-to-dial no longer causes errors and a phone reboot.

44973 Line label no longer disappears after SCA phone views remote shared line's call appearance list and the view screen times out (*applies to SoundPoint IP 301*).

46795 Colon in time display blinks correctly (*applies to SoundPoint IP 450*).

46480 Loud static ‘pop’ and ‘hiss’ are no longer heard when receiving audio using G.729AB as the codec with VAD enabled (*applies to SoundPoint IP 301, 501, 600, 601*).

46613 Audio not transmitted or routed via default gateway when phone’s subnet mask does not match phone’s IP address network class.

47303 URL BLF speed dial calls use the correct "@domain" in certain signaling scenarios.

47492 Message LED no longer flashes continuously after receiving blind transfer from a ‘centralized conference’ leg (*applies to SoundPoint IP 501*).

47609 Phone is able to display more than two status notifications if server controlled ACD is enabled (*applies to SoundPoint IP 450*).

47878 Phone is no longer generating malformed XML with ACD Login/Logout for some parameters.

47911 Forked INVITE back to caller successfully connects to voicemail on call timeout

47915 Phone no longer ignores 401 challenge after responding to 407 in a certain call scenario.

47960 Redialing POTS call from placed call list dials as video call if the call was dialed from contact directory (*applies to SoundStation IP 7000 with HDX*).

47964 Phone displays correct icon when conferencing and adding a POTS call (*applies to SoundStation IP 7000 with HDX*).

48002 Speaker volume no longer drops to two bars after making a video call (*applies to SoundStation IP 7000 with HDX*)

48039 Phone plays the proper ring tone if a remote line and local phone are both ringing and the remote line is answered and then put on hold.

48046 On G.729AB gateway calls, speaker phone volume is loud enough for low level signals.

48076 If `call.stickyAutoLineSeize`=”1”, BLF attendant phone is automatically placed on hold if a BLF or speed dial key is used to dial while an active call is in process on the attendant phone.

48123 If the idle browser is enabled, clock time increments properly while a call is active (*applies to SoundStation IP 4000/6000/7000*)

48171 De-registration attempts successfully authenticate and de-register some lines.

48280 When using TFTP or FTP as the provisioning server type, phone saves directory entries locally when TFTP or FTP server is not available (*applies to SoundStation IP 6000/7000*).

48385 SSRC header field correct for RFC2833 packets (*applies to VVX 1500*).

48462 Ring LED indicator no longer continues flashing when a call is answered if an INVITE with "sendonly" SDP is received by the phone (*applies to SoundStation IP 6000/7000*).

48485 Audio call recording during video calls no longer fails with certain USB drives (*applies to VVX 1500*).

48577 Default headset gains have been changed to correct values to ensure good audio quality with certain headsets (*applies to SoundPoint IP 430*).

48591 Click-to-hold works correctly (*applies to VVX 1500*).

48605 The behaviour set in `call.stickyAutoLineSeize` is applied correctly when a line is ringing and SilentRing is selected

48615 If `call.StickyAutoLineSeize`="1", transfer no longer fails if it's attempted while a second call is alerting.

48667 If there is an incoming call while there is an existing outgoing call in the proceeding state, the phone will audibly alert the user for the incoming call.

48668 401 Authentication challenge to a VQMon PUBLISH no longer causes the phone to reboot.

48672 Received volume on the handset is lower than desired for low input signal levels. Addressed by adding 4dB gain at low input levels on the handset. Gain at high input levels is unchanged.

48685 The MWI NOTIFY contains the message summary for the MWI LED to be lit.

48697 An incoming call without a caller ID name but with caller ID number is no longer matched with the first local contact that has a blank name.

48699 TelURI can process "tel://*50"

48756 Using a shared line, if there is an incoming call with only a number, the phone displays a blank in the caller ID instead of Unknown Party.

48778 Motion detection now begins when a video conference call begins (*applies to VVX 1500*).

48858 BLF attendants monitoring both initiator and recipient obtain the proper state even when initiator and recipient use the same dialog ID.

48912 REFER transaction timeout set too high due to subscription state expires from a NOTIFY with sipfrag on a successful blind transfer

48920 When placing a video conference call with 8 legs, the UI also shows the two last call appearances (*applies to SoundStation IP 7000 with HDX*).

48959 After upgrading to SIP 3.1.2, the time portion of the date and time display are no longer cut off when using a custom idle display (*applies to SoundPoint IP 430*).

48985 The phone no longer reboots if you receive or miss a call while looking at information about a previously received or missed call.

49013 The DND icon (X) updates next to a line key when BroadWorks ACD is enabled.

49068 Receiving an OPTIONS message no longer causes the phone to send a false dialog Notification

49129 User interface properly updates when soft keys and physical keys are pressed (*applies to VVX 1500*).

49181 When using the idle microbrowser, the phone display no longer randomly freezes (*applies to VVX 1500*).

49201 Receiving updates with confirmed SDP before 200 OK no longer cause the phone to drop the outgoing call .

49233 Incoming call line key animation is shown even after ending the call at far end when the phone is initiating conference or transfer.

49237 When `callWaiting.ring` = *ring*, changing the termination mode during a call waiting no longer results in one-way audio

49256 The microbrowser can access URLs longer than 54 characters without the phone rebooting (*applies to VVX 1500*).

49281 Adjusting the SoundStation IP 7000 volume no longer causes an integrated HDX's volume to decrease to 0 (*applies to SoundStation IP 7000 with HDX*).

49287 SUBSCRIBE terminate no longer causes BLF labels to disappear for 2 - 4 seconds

49323 While browsing an empty call list, the phone no longer reboots after lifting the handset (*applies to VVX 1500*).

49402 Seizing one SCA line and then resuming a held call on another SCA line before the line seize completes no longer causes a race condition.

49533 Correct UDP checksum in DHCP Decline message.

49599 BLF attendant phone updates 1/x widget when BLF monitored line has 1 or multiple incoming calls ended.

49810 Phone seizes the correct line when `call.stickyAutoLineSeize`=*1* and the speed dial key is used to place an outgoing call (*applies to VVX 1500*).

Configuration File Enhancements

Refer to Table 16: Software Version 3.1.3.0336 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.1.3.0336 configuration file parameters.

Table 16: Software Version 3.1.3.0336 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.serverFeature Control.localProcessing.dnd`	If set to 0 and voIpProt.SIP.serverFeature Control.dnd ="1", the phone will not perform local DND call behavior. If set to 1 or Null, the phone will perform local DND call behavior on all calls received.
sip	added	`voIpProt.SIP.serverFeature Control.localProcessing.cf`	If set to 0 and voIpProt.SIP.serverFeature Control.cf="1", the phone will not perform local Call Forward behavior. If set to 1 or Null, the phone will perform local Call Forward behavior on all calls received.
sip	added	`voIpProt.SIP.tcpFastFailover`	If set to 1, failover occurs based on the values of reg.x.server.y.retryMaxCount voIpProt.server.x.retryTimeOut. If set to 0, use old behavior. If reg.x.tcpFastFailover is Null, this attribute is checked. If voIpProt.SIP.tcpFast Failover is Null, then this feature is disabled. If both attributes are set, the value of reg.x.tcpFastFailover takes precedence.
sip	changed	`voice.gain.tx.digital.headset .IP_430`	Changed from 10 to 6
sip	changed	`voice.headset.txag.adjust.IP_ 430`	Changed from 39 to 21
sip	changed	`dir.corp.pageSize`	Changed from 16 to 32
sip	changed	`dir.corp.cacheSize`	Changed from 64 to 128
sip	added	`dir.corp.leg.pageSize`	pageSize applied to LDAP queries on SoundPoint IP 301, 501, 600 and 601 phones. Range is 8 to 64. Default value is 8
sip	added	`dir.corp.leg.cacheSize`	cacheSize applied to LDAP queries on SoundPoint IP 301, 501, 600 and 601 phones. Range is 32 to 256 Default value is 32.

File	*Action*	*Parameter*	*Description*
sip	added	`dir.corp.sortControl`	Controls how client makes queries and does it sort entries locally. It should not be used by customers. If set to 0 or Null, leave sorting as negotiated between client and server. If set to 1, force "non-sorting" Queries (Not recommended due to possible performance issues)
sip	added	`dir.corp.autoquerySubmit Timeout`	To control if there is a timeout after the user stops entering characters in the quick search and, if there is, how long the timeout is. If set to 0, there is not (disabled).
sip	added	`dir.corp.vlv.allow`	A flag to determine whether or not VLV queries can be made if the LDAP server supports VLV. If set to 0, VLV queries are disabled. If set to 1 or Null, VLV queries are enabled.
sip	added	`dir.corp.vlv.sortOrder`	The list of attributes (in the exact order) to be used by the LDAP server when indexing.
sip	added	`dir.corp.attribute.x. searchable`	A flag to determine if the attribute is searchable through quick search. This flag applies for x = 2 or greater. If set to 0 or Null, quick search on this attribute is disabled. If set to 1, quick search on this attribute is enabled.
sip	changed	`ind.gi.IP_400.6.physW`	Changed from 10 to 0
sip	changed	`ind.gi.IP_400.6.physH`	Changed from 10 to 0
sip	added	`pnet.remoteCall.localDialtone`	0=no DialTone played when IP 7000 makes an outgoing POTS call on HDX 1=Play DialTone when IP 7000 makes an outgoing POTS call on HDX Default=0
sip	added	`pnet.remoteCall.callProgAtten`	Attenuation (in dB) applied to tones played by the IP 7000 for POTS calls on HDX when HDX is the active speaker. Range -60 to 0; default=-15

Understanding Updates to SIP 3.1.2 B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.2 B beside their respective Polycom tracking identification number.

New or Enhanced Features

Added Support for the VVX 1500 product.

Configuration File Enhancements

Several parameters added for the VVX 1500 product. See Addendum to SIP 3.1 Administrator's Guide for VVX 1500 for details.

Understanding Updates to SIP Version 3.1.2

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.2 beside their respective Polycom tracking identification number.

Added or Changed Features

34787 Add Support for ACD Call Center Agent functionality using the 'Feature Synchronization' method. See *Technical Bulletin 34787: Using Feature Synchronized Automatic Call Distribution with Polycom SoundPoint IP Phones* for details.

38442 Add support for multiple NTP servers via DHCP Options 42 or 4 or DNS SRV or A records.

44612 License file should be provisioned along with configuration files at application startup.

45233 Implement a 'scrolling status bar' on phones to match the capability on the SoundPoint IP 450. This feature applies to all phones except SoundPoint IP 301.

45460 Add "Quick Set-Up" option. See *Technical Bulletin 45460: Using Quick Setup with SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones* for details.

45795 Change "Browse Files" to "Browse Recordings" in USB Device menu

46270 Remove DHCP timeout menu option from UI

46631 XML API: Softkeys don't allow for having multiple submit buttons on the page containing items list

46758 Modify 000000000000.cfg to reference the Configuration File White Paper

47128 Lifting the handset whilst a BLF monitored line is ringing should seize a line not answer the remote call. *Quick Tip 37381: Understanding Enhanced BLF on SoundPoint IP Phones* has been updated with to reflect this change.

47309 BLF indicator for a monitored phone should flash when the monitoring phone calls the monitored phone.

Enhanced Capabilities

25666 1/A/a not visible when editing some items on SoundPoint IP301.

42425 XML API: Two browser links highlighted after scrolling up a page in a certain scenario.

43484 CMR/P: Recording does not happen if started while call was on hold and then resumed.

44271 200 Response to Cancel is not matched, such that retransmission of Cancel continues.

44681 SIP 3.0.0 – 3.1.1 Releases: An internal line registration error could occur if the phone was unable to reach its provisioning server on boot up. This could result in the phone displaying “Service Unavailable” when the associated line key was selected.

44727 Microbrowser may display overlapped text if multiple spaces are included in the page.

45080 Line-seize behavior incorrect for speed-dial when call.stickyAutoLineSeize.onHookDialing = "0"

45102 SoundStation IP 7000: 1/A/a soft key is missing in Corp Dir search screen.

45169 When using sampled audio as local hold notification Local hold notification may play inaudibly or muffled.

45273 SoundStation IP4000 will not register when qos.ip.callControl.dscp = "24"

45422 Adding speed dial entry using Expansion Module may place new entry in an unexpected place

45479 SoundStation IP7000: Time&Date setting returns to the default when the phone is rebooted.

45715 Ringing stops when users goes on-hook after lifting handset during incoming call when up.offHookAction.none = 1

45799 XML API: Internal URIs: softkey:Exit, softkey:Submit and softkey:Reset do not work when called from hyperlink anchor tags

46051 Manage N-way conference menu has overlapping items if long caller-ids are present.

46144 JPEG decoder fails on some files

46242 XML API: If an account supports 2 line keys, API notifications of call events are sent for only 1 of them

46293 Phones may lock up if a CHECK-SYNC is received while a CHECK-SYNC is in progress

46422 Five to six second delay in UI when using the SPLIT soft key to cancel a transfer

46488 Phone plays continuous Reorder tone if a BLA line is successfully seized with a new line ID after a previous GLARE response.

46539 Centralized Conferencing: Conference call is terminated if the phone tries to join a conference that has reached its maximum number of participants.

46553 When `call.stickyAutoLineSeize`=”1”, an active call is not put on hold when 2nd call is made via speed dial or from calls list menu

46569 No ACK sent after receiving VM 200 OK w/ SDP, CANCEL sent 60 seconds later.

46610 Errors in Polish language dictionary

46737 BLF: Softkeys & Call appearance disappears on the console phone in a certain scenario using a shared line.

46757 XML API: Issue with order of call appearances on a single line registration and single line key

46763 XML API: URI softkey:exit does not work when executed from soft key or hyperlink anchor XHTML tags

46767 Configuration parameters `bg.gray.selection` are repeated in sip.cfg

46807 XML API: Ringer volume adjust tone is repeated every 5s in certain play URI scenarios

46808 BLF: The 2nd and 3rd Expansion Modules may not work when IP601 monitors 47 BLF lines

46812 XML API: SoundStation IP4000 and IP6000 reboot when attempting to execute the URI key:line2

46831 Phone locked up with "Reboot initiated" on the display, when it received corrupted JPEG data.

46843 Using TCP as the transport and BLF line monitoring: An attendant in an active call cannot perform a directed call pick-up on a remote ringing line.

46858 SoundStation IP 7000 may reboot/freeze if the TRANSFER and CANCEL soft-keys are pressed in rapid succession.

46861 Call appearance is sometimes missing when a conference is split during ringback on shared line

46939 Digest Authentication fails on first file in the CONFIG_FILES list with a certain configuration.

46968 SIP "auth-int" digest authentication mode does not work.

46978 EFK: Configurable soft keys cannot call functions unless at least one valid efklist entry is present

47083 SoundStation IP 4000: Phone does not send a register request when parameters qos.ip.rtp.dscp and qos.ip.callControl.dscp are set to a different value between 0 and 60

47110 SoundStation IP 7000: Enter user password in Advanced menu, phone goes to Admin menu instead of User menu

47163 603 Decline sent instead of 486 on DND

47185 In some scenarios, Directed Call-Pickup via BLF drops call and leaves phone UI in a strange state.

47262 Microbrowser URL in configuration file is not recognized if it is preceded by spaces

47310 Going on-hook on the handset of the BLF attendant during incoming call to a BLF monitored line initiates a BLF Call-Pickup.

47345 If `call.stickyAutoLineSeize`=”1”; In some scenarios, initiating a call whilst a BLF monitored phone is in the Alerting state may cause the phone to lock-up.

47450 Port 17185 is open, presenting a security risk

47500 If `call.stickyAutoLineSeize`=”1”; Active call is not placed on hold when another call is initiated by a BLF/Speed-dial key.

47530 Using a BLF or Speed Dial key for a Transfer operation does not work.

47531 Using a BLF or Speed Dial key for a Conference operation does not work.

47537 If `call.stickyAutoLineSeize`=”1”, initiating a second call whilst a first call is in the “Outgoing Proceeding” State will result in two calls in the Proceeding state

47681 BLF: Attendant may not be able to perform directed call pick up on a monitored line if using a shared line.

47705 When a phone holds a call, press headset button->EndCall sk->NewCall sk, the phone does not switch back to hands free mode

47716 `call.stickyAutoLineSeize`="1", phone does not seize correct line key when dialing from Call List or Contact Directory

47728 SoundPoint IP 601: Attendant does not display incoming call appearance and does not hear attendant ringing tone when a monitored line is on the 2nd or 3rd Expansion Module

47741 When using 1, 3, 7, 5 key combo to reset flash settings, the UI has some errors.

47866 SoundPoint IP 320/330/430/450/550/560/650/670: The phone may reboot when hold reminder tone is enabled and a call is active on the speakerphone.

47537 If `call.stickyAutoLineSeize`=”1”, initiating a second call whilst a first call is in the “Outgoing Proceeding” State will result in two calls in the Proceeding state

47538 On-hook entered digits on a BLF attendant phone are erased if a remote BLF phone in ringing state is answered on the remote BLF phone.

47559 In some scenarios a BLF attendant phone incorrectly plays the attendant ringing tone.Configuration File Parameter Changes

Configuration File Enhancements

Refer to Table 17: Software Version 3.1.2 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.1.2 configuration file parameters.

Table 17: Software Version 3.1.2 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
phone1	added	`acd.reg`	See Technical Bulletin 34787 for details
phone1	added	`acd.stateAtSignIn`	
sip	added	`voIpProt.SIP.acd.signalingMethod`	
sip	added	`voIpProt.SIP.compliance.RFC3261.validate.contentLanguage`	If set to 1, validation of the SIP header content language is enabled. If set to 0 or Null, validation is disabled.

File	Action	Parameter	Description
sip	removed	bg.gray.selection	Modified the method in which the background settings are managed across multiple phone models
sip	added	bg.hiRes.gray.selection	
sip	removed	bg.color.selection	
sip	added	bg.hiRes.color.selection	
sip	added	bg.medRes.gray.selection	
sip	changed	ind.gi.IP_600.13.physH	Changed from 109 to 103
sip	changed	ind.gi.IP_7000.7.physH	Changed from 60 to 76
sip	added	log.level.change.cmr	Control the logging detail level for individual components: call media recording, call media playback, USB I/O respectively.
sip	added	log.level.change.cmp	
sip	added	log.level.change.usbio	
sip	added	prov.quickSetup.enabled	See Technical Bulletin 45460 for details
sip	added	pnet.hdx.ext	HDX Extension Number. For HDX/IP 7000 integration

Understanding Updates to SIP 3.1.1 B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.1 B beside their respective Polycom tracking identification number.

Enhanced Capabilities

47034 SoundStation IP 7000 connected to HDX: Cannot make POTS call when Ethernet is connected and Call preference configured to Auto.

47082 SoundStation IP 7000 connected to HDX: Phone does not Mute on Auto-Answer.

47251 SoundStation IP 7000 connected to HDX: When participants in a multi-point call are disconnected the phone unmutes the local phone incorrectly.

47432 SoundStation IP 7000 connected to HDX: In a certain scenario the phone sends audio to the far end even though it shows that the call is muted.

Understanding Updates to SIP 3.1.1

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.1 beside their respective Polycom tracking identification number.

New or Enhanced Features

Add Support for SoundStation IP 7000 integration with HDX Video systems. This feature requires BootROM 4.1.2.

41705 Revise error message, when USB drive is plugged into an IP650/670 and is not supported, to direct phone user to Polycom support web-site.

45411 Change hands-free volume control to give user improved volume level adjustment capability.

45736 Reset Device Settings menu option will clear log files on the phone.

45969 Add a menu option to enable/disable headset echo cancellation.

46131 SoundPoint IP 450: Phone does not flash Time and Date when time server is not configured.

Enhanced Capabilities

27694 Interdigit interval of DTMF signal is less than `tone.dtmf.offTime` setting.

30380 In some situations the MWI state is not cleared when all voice msgs on the phone are deleted.

34586 Phone redials incorrect number after cancelling transfer or conference.

41615 Idle display animation will not appear unless phone is used in some way if the .bmp image only completes downloading after the phone has booted to the idle screen.

42233 Phone does not attempt Digest Authentication after redirect.

43408 BLA line status not updated correctly with a particular signaling timing scenario.

44099 If attempting to perform a Barge-In on an SCA and the INVITE gets a 403 Forbidden the call no longer shows as active on the phone that tried to Barge-In.

44319 SoundStation IP 6000 and 7000 phones do not use exponential back-off for TCP retransmissions.

44728 Call is not automatically resumed when pressing Cancel soft key after pressing URL.

44784 The To-Tag should not be included in an INVITE after a 401 challenge.

45039 Unnecessary Refer is sent by phone as it is being blind transferred to a conference focus.

45073 Phones do renew their DHCP Lease when they have been operational for longer than 99 days.

45187 Voice streams are not resumed automatically after a play uri.

45316 Phones can re-boot when they are sent a check-sync while under some load.

45364 In a certain scenario, when SCA phone views remote shared line's call appearance list, the UI does not return back to its previous state.

45380 XML API: Phone may reboot when accessing XHTML pages containing <softkey> tag.

45386 When remote shared line is on hold, press NewCall >Cancel/EndCall sk, both shared line displays hold screen.

45410 Phone's micro-browser is not honoring DNS TTL.

45657 BLF Console Phone does not behave correctly when List URI is removed from the server configuration.

45750 Rapidly pressing a new speed dial key after it has just been entered may cause the phone to re-boot.

45602 Early dialog state not reported by NOTIFY if the far end does not support (100rel) or send PRACK.

45713 dialog-info document is empty in NOTIFY to subscription 2,3,n when dialog state is terminated.

45827 Entered number cannot be edited by pressing left arrow key to move cursor to the left in some scenarios.

45870 When bitmap is loaded as background for idle display and either the plus or minus volume key is pressed, the volume indicator graphic does not clear automatically.

45895 Phone will not dial from contact directory when separators are part of the contact e.g. 604-450-1234.

45954 SUBSCRIBE to phone with expires less than 2 seconds will never receive a NOTIFY.

46047 BLF lamps remain on when no explicit terminated state sent for BLF but it has been omitted in the Full list.

46407 Soft keys do not show up after a call is taken off hold quickly - one-way audio issue.

46412 BLF: Memory Fragmentation and leak with receipt of BLF messaging.

46500 BLF: DisplayName is not included in Remote Identity of Dialog when phone receives REQUEST.

46543 BLA: phone should NOT send dialog NOTIFY with terminated after receiving a cancel.

46486 Enabling Idle Browser on IP330 may cause dialed digits to not display.

46888 The phone erroneously sends G.711 mu-law audio with zero SSRC field regardless of negotiated codec after a conference leg is resumed, a call held by the far end is resumed, or a remotely held call on a shared/bridged line is resumed.

Configuration File Enhancements

Refer to Table 18: Software Version 3.1.1 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.1.1 configuration file parameters.

Table 18: Software Version 3.1.1 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	changed	`voice.gain.rx.digital.chassis.IP_330`	Changed from 6 to 5
sip	changed	`voice.gain.rx.digital.chassis.IP_430`	Changed from 6 to 5
sip	changed	`voice.gain.rx.digital.chassis.IP_450`	Changed from 6 to 5
sip	changed	`voice.gain.rx.digital.chassis.IP_650`	Changed from 6 to 5
sip	changed	`voice.gain.rx.digital.chassis.IP_7000`	Changed from 6 to 5
sip	changed	`voice.gain.rx.digital.chassis.IP_6000`	Changed from 6 to 5

Understanding Updates to SIP 3.1.0 C

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.0 C beside their respective Polycom tracking identification number.

New or Enhanced Features

Add Support for the SoundPoint IP 450 product.

Configuration File Enhancements

Refer to Table 19: Software Version 3.1.0 C – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.1.0 C configuration file parameters.

Table 19: Software Version 3.1.0 C – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*

sip	added	`voice.gain.rx.analog.chassis.IP_450`
		`voice.gain.rx.analog.ringer.IP_450`
		`voice.gain.rx.digital.chassis.IP_450`
		`voice.gain.rx.digital.ringer.IP_450`
			Add DSP parameters for IP 450 platform.

sip	added	`voice.gain.tx.analog.chassis.IP_450`
		`voice.gain.tx.digital.handset.IP_450`
		`voice.gain.tx.digital.headset.IP_450`
		`voice.gain.tx.digital.chassis.IP_450`
			Add DSP parameters for IP 450 platform.

File	Action	Parameter	Description
sip	added	voice.rxEq.hs.IP_450.preFilter.enable voice.rxEq.hs.IP_450.postFilter.enable voice.rxEq.hd.IP_450.preFilter.enable voice.rxEq.hd.IP_450.postFilter.enable voice.rxEq.hf.IP_450.preFilter.enable voice.rxEq.hf.IP_450.postFilter.enable voice.txEq.hs.IP_450.preFilter.enable voice.txEq.hs.IP_450.postFilter.enable voice.txEq.hd.IP_450.preFilter.enable voice.txEq.hd.IP_450.postFilter.enable voice.txEq.hf.IP_450.preFilter.enable voice.txEq.hf.IP_450.postFilter.enable	Add DSP parameters for IP 450 platform.
sip	added	voice.handset.rxag.adjust.IP_450 voice.handset.txag.adjust.IP_450 voice.handset.sidetone.adjust.IP_450 voice.headset.rxag.adjust.IP_450 voice.headset.txag.adjust.IP_450 voice.headset.sidetone.adjust.IP_450	Add DSP parameters for IP 450 platform.
sip	added	bitmap.IP_450.* ind.anim.IP_450.* ind.gi.IP_450.*	Add UI parameters for IP 450 platform.

Understanding Updates to SIP 3.1.0 B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.0 B beside their respective Polycom tracking identification number.

Enhanced Capabilities

45605 Missing closing XML tag in a configuration file causes a phone reboot.

Understanding Updates to SIP 3.1.0.0073(Limited Distribution)

This version should be replaced by 3.1.0RevB. This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.1.0.0073 beside their respective Polycom tracking identification number.

New or Enhanced Features

22971 Phone should re-register after changing auth parameters.

26010 Add support for Music On Hold (per IETF draft-worley-service-example-01).

26765 Phone does not handle forked INVITE properly.

29788 Ensure transfer and call termination behavior is robust against predictable failure modes.

30210 Phone should be able to upload a 'tech-support' information dump.

31171 Provide New Call soft key when alerting call appearance is in focus.

31556 EFK: Add ability to configure Telephony Soft-Keys.

32534 Allow on-hook dialing during the alerting state.

32757 XML API: Make Micro-browser soft-keys configurable from Server.

33428 Exit should exit, Back should take you back.

33479 When entering 0 and 00 as speed dial number and saving, phone should display error message saying invalid Speed Dial number.

33481 Phone should warn if user tries to enter duplicate Speed Dial.

34248 Location of Transfer and Conference soft key should not change during Transfer and Conference process.

34364 Add GeoTrust to the built in trusted CA list.

37592 Add configuration to give 'dead air' when phone goes off-hook.

37644 Limit the number of conference groups to one on SoundStation IP 7000.

38022 XML API: Support for asynchronous HTTP URL Push and HTTP POST to the microbrowser.

38032 XML API extensions for application support of telephony functions and telephony integration.

38286 Add support for Plantronics electronic hook switch. This feature requires BootROM 4.1.0 or newer to operate.

38585 EFK: Add support for enhanced soft key (ESK) capability.

38741 EFK: Add the ability to specify a HTTP or HTTPS URL to be loaded by the microbrowser.

38882 Update default list of trusted CAs on the phone.

39145 Include Diversion Header Information in the caller-id display.

39146 Add ability for the phone to display contents of the SIP warning field to the user.

39647 On registration failure (TCPOnly) phone waits 30-60 seconds for retry.

39666 Improve directory configuration parameters – see Administrator’s Guide for details.

39821 Add label field to local contact directory.

40000 EFK: Add ability to invoke internal key functions via the macro engine.

40265 Hide SAS-VP Provisioning Option from the User Interface.

40278 SIP stack Tx support of Accept-Language.

40341 XML API: Play API - audio file to be downloaded from the HTTP server and played using the phones speaker.

40431 CMR/P: Add support for USB flash drives larger than 2GB on SoundPoint IP 650/670 phones.

40543 DTMF dialing will process, character as 2 sec. pause.

40559 When phone is rebooted, it should first deregister before starting reboot process.

40978 EFK: Ensure that all soft key functions can be mapped to hard keys.

41016 Add Slovenian to the list of languages supported by certain SoundPoint/SoundStation IP Phones.

41017 Add Polish to the list of languages supported by certain SoundPoint/SoundStation IP Phones.

41050 Enhanced BLF: Add indication of remote phone ringing to Dialog Package BLF implementation.

41161 Add decode support for JPEG image format on SoundStation IP 6000 and 7000 phones.

41177 Add configuration to control whether name or number comes first in caller-id.

41217 Show Diversion Header Information in the caller-id display.

41264 Associate key colors with background bitmaps.

41366 Update phone UI and Administrator Documents to properly reference 'CDP'.

41622 Enhanced BLF: BLF Dialog Handling in SIP Stack.

41629 Enhanced BLF: BLF call appearance UI changes.

41928 EFK: Remove License requirement from EFK feature.

42812 Add EFK support to SoundPoint IP 670.

42979 CMR/P: Increase recording buffer size to accommodate flash drives larger than 2GB.

42980 CMR/P: Reject user attempts to perform USB operations while another operation is still in progress, to support large flash drives.

42982 CMR/P: Add UI icon to show when USB drive is busy, to help user avoid accidentally removing the drive before an operation finishes.

43144 Remove CFS restriction on SSAWC.

44546 Set Handset AEC and AES to ‘on’ in default configuration files to avoid handset echo issues.

44740 SoundStation IP 7000: Call lists do not display sip: prefix for URL dialed calls.

45222 Reduce the default maximum memory size for tones from 600kbytes to 300kbytes to avoid memory issues on SoundPoint IP 320, 330, and 430 products. See *Technical Bulletin 35704: Allocating Adequate Memory for Resources on SoundPoint IP and SoundStation IP Phones* for details on managing the memory usage on phones.

Enhanced Capabilities

24740 Not all SIP header compact form supported.

29946 Log files are not uploaded if an Apache 2.0.X boot server requires authentication.

34586 Phone redials incorrect number after cancelling transfer or conference in a certain scenario.

35315 URL dialing fails, when shared line is in unregistered state.

35766 Phone locks up after receiving MWI due to extra space in config.

36060 nonVolatile.maxSize does not set the contact limit.

36728 MWI Caching across re-boots does not work as expected.

36770 In ring type menu, ring gets played twice if the wav file is of more than 300kb.

36782 Pressing any digit key should close the pop-up volume control widget.

36933 Menu should not time out when custom certificate fingerprint is being displayed and user input is expected.

37173 Charge-For-Software: Features not immediately deactivated upon license key expiration, post license.polling.time.

37233 SoundPoint IP330, IP430, IP650, IP550 and IP4000 phones malfunction if you enter > 40 digit contact number in directory.xml file.

37449 The phone may re-boot when the user tries to end a local conference if the call server does not respond to the REFER message.

37580 DoS: Multicast rate limiting is not enabled on IP601.

37848 LED indication functionality is not consistent among platforms when IMs are exchanged between phones while on Instant messages screen.

37924 Peer-to-peer presence: More soft key appears in Buddy Status menu when there are no more soft keys to display.

38284 Volume adjust text labels along with volume bar are incorrect in some scenarios.

38403 RFC2543 Hold cannot be correctly set using phone's menu and web Configuration.

38452 Press and hold line key, assigning the in-focus entry to that speed dial key does not work correctly.

38548 Typing some value in the Send message to: field and exiting causes problem when Instant Messages is re-selected.

38610 Burst of ring tone happens before ring back when call is placed for the 2nd time after the 1st call is dropped.

38631 Go to Directory menu, down scrolling icon does not display until down arrow key is pressed if contact does not have last/first name.

38633 When there are no records in Corporate Directory menu, Search soft key should not display.

38636 CMR/P: Wav file cannot be opened when consultation call (of Conference) is on hold.

38798 Operation of menus using the 'Back' soft key is confusing.

39022 Transfer and Conference soft keys are still available on IP650/IP550/IP301/IP4000 after the maximum number of outgoing calls is reached on these phones.

39208 Content Type Header field not handled properly in Microbrowser.

39317 Call cannot be resumed when reINVITE is given a 404 error.

39533 Malicious connection to TCP port 5060 may cause phone to reboot.

39546 Phone should not send Presence SUBSCRIBE signaling when `pres.reg`=invalid line number.

39553 Corporate Directory: when DNS record timeouts, Corp Dir does not honour TTL and sends a new DNS query.

39598 VQMon: use of partition byte count (magic number) to detect SID/CNG is too small - use buffer flags instead.

39623 Headset: Headset icon (active path icon) disappears during call in a certain scenario on the SoundPoint IP 430 phone.

39642 SoundStation IP 6000 and 7000 products reply to IP packets of unknown protocol with ICMP messages.

39788 SoundPoint IP 501, 601: Phone should not play incoming rtp when offered recvonly stream.

39935 Users of the IP650 hands free complain that sometimes, the phone goes dead silent and they wonder if the far-end is still on the line.

39987 Corporate Directory: In phone CD status menu the port displayed is wrong, though internally the functionality is fine.

39988 DNS NAPTR mis-configuration can cause phone to reset.

39996 Only one of the two calls appears on the UI when transferring a conference between shared lines.

40005 Phone does not remove BLFs from the U/I if all monitored users are removed at once.

40057 Volume Control not visible when adjusting volume while in Manage Conference menu.

40066 N-way conf: In manage menu, Animations icon disappear from the screen when user selects the participant by pressing its corresponding number (digit) on dial pad.

40101 USB: Backlight does not get turned on when USB memory stick is attached/removed.

40117 Corporate Directory: Modify algorithms for displaying CD status and entry details.

40125 CMR/P: In Browse Files menu the file name gets appended with ellipses (...) when exit from the Delete screen.

40126 CMR/P: File name is partially truncated at the beginning in audio player screen in a certain scenario.

40197 CMR/P: The menu title for Browse Files... option is USB Device which is a duplicate of the parent menu screen.

40328 Phone hanging on HTTP PUT with authentication.

40399 Phones generate multiple SOA queries and eventually lock up if the DNS domain is incorrectly configured.

40400 Phone issuing DHCP Inform packet when it doesn't need to.

40416 Backlight does not go to Dim mode (medium) under these scenarios (when On intensity=High, Idle intensity = Medium).

40436 Backlight intensity should not change from medium to low under these scenarios when configured (On=medium & Idle = Off).

40445 Place an incoming call to a phone that enables call forward, screen flickers incoming caller id for 1 time if the phone is in dial tone state.

40503 The scroll down bar is still available even if corporate directory list is accessed to the end.

40561 Backspace or << soft key is not available on Add Buddy Page for IP 4000 and IP 6000 phones.

40562 The first option in the Mystat list gets highlighted even if option other than the first option is selected.

40586 SoundStation IP 7000: Phone's UI does not display ''date and time'' in the call appearance screen during multiple calls.

40660 + being ‘escaped’ as %2B in INVITE URI.

40664 To establish a 2nd call using speaker key while the first call is on hold, one has to press the speaker key twice.

40716 CMR/P: Renaming the new wav file to an already existing old wav file should be prohibited. Currently, this failure replaces the new file completely (content, length, size) with old file.

40718 CMR/P: Rename screen: (1) Title is incomplete. (2) Encoding soft key appears after second press of 1/A/a sk.

40804 CMR/P: When new call arrives while user is in the audio player screen but not playing audio, incorrect soft keys are displayed.

40831 Corporate Directory: Page and Cache size parameters should be configurable.

40862 Wrong soft key displayed while transferring a URL call and selecting blind.

40898 Usage bar shows behind customer bitmap display.

40945 Pressing DND feature during hot dial creates problem with new call establishment.

41002 When entering contact directory entry, there is no soft key (1/A/a) to change number/lower case/upper case.

41034 CMR/P: No audio in Jabra 9350 headset when wav file is played through headset mode, though the visual indicators show it in Playing state.

41173 Japanese XML dictionary needs a review.

41184 SoundStation IP 7000: Wrong Date Time format when you select Japanese language.

41186 SoundStation IP 7000: Date Time format is wrong on the Placed/Received Calls info when Japanese Language is selected.

41364 Phones does not honor MIME type for telephone event in SDP Answer.

41448 Phone stops sending DTMF in a certain scenario.

41700 RTP does not go to correct destination following reINVITE.

42252 Configuring VLAN discovery does not incur a restart.

42261 Phone will not search sub containers in the corporate directory.

42749 Phone connects to LDAP server, but does not return records.

42792 Media Attribute missing in Hold ReINVITE when SRTP is enabled.

42841 Echo is experienced when calling IP 650 to IP 650 using G.722 HD at full volume.

43014 `call.stickyAutoLineSeize` is not working correctly when a second call is initiated from a speed dial.

43121 safeReconfig on SoundStation IP 4000 results in the phone rebooting.

43360 Phone sends a ‘terminated’ notify with two different dialogs for the same call.

43513 SoundPoint IP 650 experiencing Echo at full volume on handset.

43745 French XML Dictionary needs updating.

44066 Ringer diminishes on some phones over time and stops working.

44164 SoundPoint IP 320 does not respond to UPDATE when sent more than 9 seconds after INVITE.

44223 SoundStation IP 7000: # key behaves as if pressing the 1/A/a soft key.

44324 Feature key remapping does not always work.

44029 When ANALOG HEADSET MODE is set to JABRA mode, there is no audio call waiting tone.

44413 Speed dial labels on line keys are switched from first, last to last first.

44423 Speed dial entries on 650s are coming up URL Call Disabled.

44509 SoundPoint IP 600/601: Transferring and originating calls generates URL Call Disabled message.

44520 Phone is generating an unexpected NOTIFY on an incoming call which puts the BLA status out of sync.

44763 Phones ignoring DNS SRV records response from Session Border Controller in certain scenario.

45093 SoundStation IP4000 and 6000 have no way to delete or backspace on the Password entry screen.

45118 Digest authentication for SIP transactions fail when digest token is in lower-case characters.

45198 Dialing EFK macros from speed dial key does not work if URL dialing is disabled.

Configuration File Enhancements

Refer to Table 20: Software Version 3.1.0.0073 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.1.0.0073 configuration file parameters.

Table 20: Software Version 3.1.0.0073 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.strictLine Seize`	If set to 1, forces the phone to wait for 200 OK response when receiving a TRYING notify. If set to 0 or Null, this is old behavior.
sip	added	`voIpProt.SIP. strictUserValidation`	If set to 1, forces the phone to match user portion of signaling exactly. If set to 0 or Null, phone will use first registration if the user part does not match any registration.
sip	added	`voIpProt.SIP.lineSeize. retries`	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA).
sip	added	`voIpProt.SIP.header. diversion.enable`	If set to 1, the diversion header is displayed if received. If set to 0 or Null, the diversion header is not displayed.
sip	added	`voIpProt.SIP.header.list. useFirst`	If set to 1 or Null, the first diversion header is displayed. If set to 0, the last diversion header is displayed.

File	Action	Parameter	Description
sip	added	voIpProt.SIP.header.warning.codes.accept	A list of accepted warning codes. If set to Null, all codes are accepted. Only codes between 300 and 399 are supported.
sip	added	voIpProt.SIP.header.warning.enable	If set to 1, the warning header is displayed if received. If set to 0 or Null, the warning header is not displayed.
sip	added	voIpProt.SIP.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. If reg.x.musicOnHold is set to Null, this attribute is checked.
sip	added	lcl.ml.lang.tags.x	The format is: • The first two letters are the ISO-639 language abbreviation. • The next two letters are the ISO-3166 country code. • The next two letters are the ISO-639 language abbreviation. • The remainder of the string is the preference level for the display of the language, or English if the language is not available
sip	added	up.numberFirst CID	If set to 0 or Null, caller ID display will show caller's name first. If set to 1, caller ID display will show caller's number first.
sip	changed	saf.1	The default value is Null. To allow the SoundPoint IP welcome sound to be played on reboots and restarts, set to SoundPointIPWelcome.wav
sip	changed	voice.aec.hs.enable	The default value is enabled (1).
sip	changed	voice.aes.hs.enable	The default value is enabled (1).
sip	added	call.directedCallPickup String	The star code to initiate a directed call pickup.

File	*Action*	*Parameter*	*Description*
sip	added	`dir.corp.pageSize`	The maximum number of entries requested from the corporate directory server with each query.
sip	added	`dir.corp.cacheSize`	The maximum number of entries that can be cached locally on the phone.
sip	added	`dir.corp.scope`	Type of search. If set to one, a search of the level one below the baseDN is performed. If set to sub or Null, a recursive search (of all levels below the baseDN) is performed. If set to base, a search at the baseDN level is performed.
sip	changed	`voice.ns.hs.enable`	The default value is enabled (1).
sip	changed	`res.quotas.1.value`	The default value is 300KB for tones.
sip	added	`apps.telNotification.URL`	The URL to which the phone sends notifications of specified events. The protocol used can be either HTTP or HTTPS.
sip	added	`apps.telNotification.incomingEvent`	If set to 0, incoming call notification is disabled. If set to 1, incoming call notification is enabled.
sip	added	`apps.telNotification.outgoingEvent`	If set to 0, outgoing call notification is disabled. If set to 1, outgoing call notification is enabled.
sip	added	`apps.telNotification.offhookEvent`	If set to 0, offhook notification is disabled. If set to 1, offhook notification is enabled
sip	added	`apps.telNotification.onhookEvent`	If set to 0, onhook notification is disabled. If set to 1, onhook notification is enabled
sip	added	`apps.statePolling.URL`	The URL to which the phone sends call processing state/device/network information. The protocol used can be either HTTP or HTTPS
sip	added	`apps.statePolling.username`	The user name to access the state polling URL.

File	*Action*	*Parameter*	*Description*
sip	added	`apps.statePolling.password`	The password to access the state polling URL.
sip	added	`apps.push.messageType`	Select the allowable push priority messages on phone.
sip sip	added	`apps.push.serverRootURL`	The relative URL (received from HTTP URL Push message) is appended to the application server root URL and the resultant URL is sent to the Microbrowser.
sip	added	`apps.push.username`	The user name to access the push server URL.
sip	added	`apps.push.password`	The password to access the push server URL.
	added	`softkey.x.label`	This is the text displayed with the soft key. If set to Null, the label to display is determined as follows: • If the soft key is mapped to a enhanced feature key macro, the label of the enhanced feature key macro will be used. • If the soft key is mapped to a speed dial, the label of the corresponding directory entry will be used. If this label does not exist as well and the directory entry is an enhanced feature key macro, then the label of the enhanced feature key macro will be used. • If the soft key is mapped to chained actions, only the first one is considered for label, using the rules above. • If no labels are found after the above steps, the soft key label will be blank.
sip	added	`softkey.x.action`	The same syntax as the enhanced feature key action.
sip	added	`softkey.x.enable`	If set to 0 or Null, the soft key is disabled. If set to 1, the soft key is enabled.
sip	added	`softkey.x.precede`	If set to 0 or Null, the soft key replaces any empty space from the leftmost position. If set to 1, the soft key is displayed before the first standard soft key.

File	*Action*	*Parameter*	*Description*
sip	added	`softkey.x.use.idle`	If set to 0 or Null, the soft key is not displayed in the idle state. If set to 1, the soft key is displayed in the idle state.
sip	added	`softkey.x.use.active`	If set to 0 or Null, the soft key is not displayed in the active call state. If set to 1, the soft key is displayed in the active call state.
sip	added	`softkey.x.use.alerting`	If set to 0 or Null, the soft key is not displayed in the alerting state. If set to 1, the soft key is displayed in the alerting state.
sip	added	`softkey.x.use.dialtone`	If set to 0 or Null, the soft key is not displayed in the dialtone state. If set to 1, the soft key is displayed in the dialtone state.
sip	added	`softkey.x.use.proceeding`	If set to 0 or Null, the soft key is not displayed in the proceeding state. If set to 1, the soft key is displayed in the proceeding state.
sip	added	`softkey.x.use.setup`	If set to 0 or Null, the soft key is not displayed in the setup state. If set to 1, the soft key is displayed in the setup state.
sip	added	`softkey.x.use.hold`	If set to 0 or Null, the soft key is not displayed in the hold state. If set to 1, the soft key is displayed in the hold state.
sip	added	`softkey.feature.newcall`	If set to 0, the New Call soft key is not displayed when there is another way to place a call. If set to 1 or Null, the New Call soft key is displayed.
sip	added	`softkey.feature.endcall`	If set to 0, the End Call soft key is not displayed. If set to 1 or Null, the EndCall soft key is displayed.

File	Action	Parameter	Description
sip	added	softkey.feature.split	If set to 0, the Split soft key is not displayed. If set to 1 or Null, the Split soft key is displayed.
sip	added	softkey.feature.join	If set to 0, the Join soft key is not displayed. If set to 1 or Null, the Join soft key is displayed.
sip	added	softkey.feature.forward	If set to 0, the Forward soft key is not displayed. If set to 1 or Null, the Forward soft key is displayed.
sip	added	softkey.feature.directories	If set to Null, the Dir soft key is displayed on the SoundPoint IP 320/330 phone, but not on any other phone. If set to 0, the Dir soft key is not displayed on any phone. If set to 1, the Dir soft key is displayed on all phones as follows: • In the idle state, it is displayed after the New Call and Callers soft keys. • In the dialtone state, it is displayed after the End Call and Callers soft keys. • During a conference or transfer, it is displayed after the Callers and Cancel soft keys.
sip	added	softkey.feature.callers	If set to Null, the Callers soft key is displayed on the SoundPoint IP 320/330 phone, but not on any other phone. If set to 0, the Callers soft key is not displayed on any phone. If set to 1, the Callers soft key is displayed on all phones as follows: • In the idle state, it is displayed after the New Call soft key and before the Dir soft key. • In the dialtone state, it is displayed after the End Call soft key and before the Dir soft key. • During a conference or transfer, it is displayed before the Cancel soft key.
sip	added	softkey.feature.mystatus	If set to 0, the MyStatus soft key is not displayed. If set to 1 or Null, the MyStatus soft key is displayed.

File	Action	Parameter	Description
sip	added	`softkey.feature.buddies`	If set to 0, the Buddies soft key is not displayed. If set to 1 or Null, the Buddies soft key is displayed.
sip	added	`softkey.feature.basicCall Management.redundant`	If set to 0 and the phone has hard keys mapped for Hold, Transfer, and Conference functions (all must be mapped), all of these soft keys are not displayed. If set to 1 or Null, all of these soft keys are displayed.
phone1	added	`reg.x.strictLineSeize`	If set to 1, forces phone to wait for 200 OK on registration x when receiving a TRYING notify. If set to 0 or Null, this is old behavior. If this parameter is Null, `voIpProt.SIP.strictLineSeize` is checked. If both parameters are set, this parameter takes precedence.
phone1	added	`reg.x.musicOnHold.uri`	A URI that provides the media stream to play for the remote party on hold. When present, and if `reg.x.musicOnHold` is not Null, this attribute overrides the global Music on Hold defined in the sip.cfg configuration file.
phone1	added	`attendant.ringType`	The ring tone to play when a BLF dialog is in the offering state. Permitted values are 1 to 22. The default is Null.

Understanding Updates to SIP 3.0.4

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.4 beside their respective Polycom tracking identification number.

Note that SIP 3.0.4 was released after SIP 3.1.0, so it should not be assumed that the changes in SIP 3.0.4 also apply to SIP 3.1.0.

New or Enhanced Features

44546 Set Handset AEC and AES to ‘on’ in default configuration files to avoid handset echo issues.

45411 Adjust Speaker phone (Hands Free) volume control for better user experience.

Enhanced Capabilities

43264 Phone is not able to answer calls due to duplicate INVITEs with same details and new BRANCH ID.

43513 SoundPoint IP 650 to 650 calls experiencing Echo at full volume on the handset.

44029 When ANALOG HEADSET MODE is set to JABRA, there is no audio call waiting tone.

44066 Ringer (including call waiting tone) diminishes on some phones over time and stops being audible.

44413 Speed dial labels on line leys are labeled switched from first, last to last, first.

44423 Speed dial entries on 650s are coming up URL Call Disabled.

44509 SoundPoint IP 600/601: Transferring and originating calls causing URL Call Disabled due to unnecessary attempt to provision CFS license file via HTTPS.

44520 Phone generating an unexpected NOTIFY on incoming call, putting BLA status out of sync.

44763 Phones ignoring DNS SRV records response from Session Border Controller in certain scenario.

44818 Danish dictionary is Chinese.

45073 Phones do not renew their DHCP Lease when they have been operational for longer than 99 days.

45118 Digest Authentication for SIP transactions fail when Digest token is all lower-case.

45221 Oneway voice in handset/headset mode during call waiting when call.callWaiting.ring= ring is set.

45719 Corporate Directory: Phone not sending correct details when connecting to SUNldap Server.

45761 DND Sync feature failing across reSUBSCRIBE.

Configuration File Enhancements

Refer to Table 21: Software Version 3.0.4 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.0.4 configuration file parameters.

Table 21: Software Version 3.0.4 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*

sip	changed	`voice.aec.hs.enable`
		`voice.aes.hs.enable`
		`voice.ns.hs.enable`
			Changed default value from ‘0’ to ‘1’

sip	changed	`voice.gain.rx.digital.chassis.IP_330`
		`voice.gain.rx.digital.chassis.IP_430`
		`voice.gain.rx.digital.chassis.IP_650`
			Changed default value from ‘6’ to ‘5’

Understanding Updates to SIP 3.0.3 B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.3 B beside their respective Polycom tracking identification number.

Changes made apply only to the SoundStation IP 7000 product.

Enhanced Capabilities

41974 SoundStation IP 7000 no longer randomly reboots when the idle browser is enabled

Understanding Updates to SIP 3.0.3

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.3 beside their respective Polycom tracking identification number.

New or Enhanced Features

39423 Change default boot config and packaged sip.cfg value for parameter `voice.vad.signalAnnexB`.

40385 Add config parameters `voIpProt.SIP.strictLineSeize`, `reg.x.strictLineSeize` and `voIpProt.SIP.lineSeize.retries`.

40387 SIP stack will use config parameter `voIpProt.SIP.strictLineSeize` and `voIpProt.SIP.lineSeize.retries` to make fault-tolerant behavior optional.

40447 Add a User Option to restart the phone.

Enhanced Capabilities

39635 Phones configured for a bridged line appearance reboot when they receive an improperly forked duplicate packet.

39792 The phone is requesting a SIP URI on transfer instead of a number with some call servers.

40175 Digitmap problem with IP330 and IP320s not processing single digit map entry correctly

40287 Phone is not returning fast busy on a timeout when sending TRYING state; it continues to send call EARLY causing BLA sync issues

40318 Buddy Status indicator not working when a function key is mapped to a speed dial.

40632 Phones hang at the welcome screen when DHCP server specifies a subnet mask of 255.255.254.0.

40673 Phone does not handle NOTIFY message correctly in Glare (race condition).

40709 Phone responding to subscribe that does not match its configuration.

40766 Phone must match To: header with third-party subscribe.

41203 Phones not responding to DHCP offer using an option other than 160 if Option 160 also has an entry (*applies to SoundPoint IP 320, 330, 430, 550, 560, 650*).

41351 Call lists may show SIP URI on SoundPoint IP 330/320 phones.

41403 CMR/P: Wrong popup appears when usb is removed after exiting from the playback to the browse files menu.

41475 After upgrade to SIP 3.0 The SIP Config option `msg.bypassInstantMessage=1` does not work correctly.

41614 Phone repeating USER AGENT string in HTTP request.

41645 Transfer of Held call causes party on Hold to automatically resume in certain call server interactions.

41654 CMR/P: Call gets answered in speaker mode when off-hook if an incoming call happens while in audio player screen.

41657 CMR/P: Headset memory persistence status goes wrong if an incoming call happens while in audio player screen.

41666 CMR/P: While in audio player screen, ringing for an incoming call happens in wrong termination mode. It should always happen on speaker.

41789 AsFeature doesn't reSUBSCRIBE after losing the TLS connection.

41808 Idle logo does not display correctly in certain configurations.

41903 Corporate Directory searches may not return complete results if results contain Unicode character values > 127 (server supports sorting).

41926 Navigation select button does not get call details.

41983 SCA Caller ID displays wrong direction as From: when remote shared line places an outgoing call.

42605 Speed dial shortcut should not apply if contact directory is disabled on SoundPoint IP 330/320 phones.

Configuration File Enhancements

Refer to Table 22: Software Version 3.0.3 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.0.3 configuration file parameters.

Table 22: Software Version 3.0.3 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.strictUserValidation`	If set to 1, forces phone to match user portion of signaling exactly. If set to 0, phone will use first registration if the user part does not match any registration

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.strictLineSeize`	If set to 1, forces phone to wait for 200 OK when receiving a TRYING notify.
sip	added	`voIPProt.SIP.lineSeize.retries`	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA). Valid values are 3 to 10. Note that in this release, a value of 3 results in 10. A value of 2 can be used to get 3 retries.
phone1	added	`reg.n.strictLineSeize`	If set to 1, forces phone to wait for 200 OK on registration n when receiving a TRYING notify. If this parameter is Null, voIpProt.SIP.strictLineSeize is checked. This parameter takes precedence.

Understanding Updates to SIP 3.0.2 C

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.2 C beside their respective Polycom tracking identification number.

Enhanced Capabilities

42034 Phone freezes when booting from TFTP server in certain scenarios.

42060 When an IP601 with Expansion Modules attached is configured with many speed-dials with long names. Removing or adding a speed-dial during a period of high activity (e.g. call in progress) may result in sluggish UI response or in extreme cases re-boot.

Understanding Updates to SIP 3.0.2.0917 B (Limited Distribution)

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.2.0917 B beside their respective Polycom tracking identification number.

New or Enhanced Features

Added support for the SoundPoint IP 670 product.

Added support for the SoundStation IP 6000 product.

Added support for the SoundStation IP 7000 product.

39292 Add dynamic test for un-recognized USB devices.

39532 After a 500 Glare response, phone should retry call attempt on a different line ID.

39585 Add support for JPEG images (in addition to BMP format).

40351 Add additional USB flash drives to the internal list of supported drives.

40591 Add background preference configuration to the phone's configuration web server.

41025 Set default LDAP Corporate Directory background re-sync period to 24 hours.

41045 Make initial background LDAP Contact Directory synchronization optional.

41363 Add additional graphic backgrounds to the IP 550, 560, 650 phones.

41517 Add JPEG support to the micro-browser.

Enhanced Capabilities

38539 Microbrowser does not display Asian fonts on IP 550, 560 and 650 phones.

39603 Rapid hold-resume with SRTP can cause one-way audio.

39608 Phone does not play ring tone when conference put on hold in certain scenarios.

39610 Idle display not fully cleared when making new call.

39657 Phone may reboot if user removes USB flash drive while recording is in progress.

39678 Authorization response changes during multi-stage dialing.

39716 Speed dial from up arrow shortcut using speed dial index does not work correctly when BLF is configured.

39932 Unicode text entry does not work correctly.

39979 SoundPoint IP 301, 501, 601 phones with SRTP disabled reject calls offering both SRTP and non-SRTP media.

40115 CMR/P: File browser continues to display file in file list after user has deleted file.

40266 Voice Quality Metrics incorrectly reports packet losses when VAD is enabled.

40346 Corporate Directory: Improve message when connection is lost after CD server initialized successfully.

40427 Phone will send a 486 (Busy Here) SIP response if the reject soft key is used after DND is enabled and disabled.

40574 Phone ignores 'Require: 100rel' header in INVITE.

40593 2-way audio (call made from Shared line) gets lost after cancelling transfer once the far end has performed hold/resume (or cancelled transfer/conf).

40598 Original call does not get resumed when transfer attempt is cancelled by pressing the active termination key in certain call scenarios.

40669 Caller ID using `up.useDirectoryNames=1` stops working when sip and so logs set at 0.

40686 DTMF tones are transmitted in band when RFC 2833 is negotiated on a SoundStation IP 4000.

40694 When call is put on hold at shared line the soft keys New Call, Transfer, Conf, More don't appear.

40724 SoundStation IP 4000: Call Waiting Tone echoed to far end caller.

40804 When new call arrives while user is in the USB Recording 'play' screen but not playing audio, incorrect soft keys are displayed.

41199 802.1x packets do not get forwarded by SoundPoint IP 320, 330, 430, 550, 560, 650 phones.

41355 Phone responds with 501 to UPDATE request, which it should not do.

41364 Phone does not honor MIME Type for Telephone-Event in SDP Answer.

Configuration File Enhancements

Refer to Table 23: Software Version 3.0.2.0917 B – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.0.2.0917 B configuration file parameters.

Table 23: Software Version 3.0.2.0917 B – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voice.codecPref.IP_(6|7)000.*`	Codec support for IP 6000 and IP 7000.
sip	added	`voice.gain.(r|t)x.analog.*.IP_(6|7)000`	Gain levels for IP 6000 and IP 7000.
sip	added	`voice.(r|t)xEq.hf.IP_(6|7)000.(pre|post)Filter.enable`	Prefilter and postfilter enable for IP 6000 and IP 7000.
sip	added	`dir.corp.backGroundSync`	Changed from 1 to 0, disabling background sync.
sip	changed	`dir.corp.backGroundSync.period`	Changed value from 43200 (12 hours) to 86400 (24 hours).
sip	changed	`bg.ranges`	
sip	removed	`bg.color.selection`	Defines which background is used. Default is 1,1. First (left) index is the type of background. Second is the index into the table of that type. **Index** / **Type** 1 – Predefined backgrounds 2 – Solid patterns 3 – User-defined bitmaps

File	*Action*	*Parameter*	*Description*
sip	changed	`bg.hiRes.color.pat.solid.*.name|red|green|blue)`	Defines the name and colour of solid backgrounds.
sip	added	`bg.hiRes.color.bm.*.(em.)?name`	Defines colour backgrounds for the phone's display and the expansion modules' displays (em).
sip	added	`button.color.selection.*.*.modify`	Defines the transform applied to the button image used for line keys and soft keys. The two indexes operate as defined above in bg.color.selection. The value comprises a transform method, and parameters for the transform. Two transforms are supported – rbgHiLo and none. The rgbHiLo has six parameters. The first two apply to the red channel, the next two to the green and the last to the blue. The first parameter of a pair defines the value to use for the brightest pixels of the button graphic. The second parameter of a pair defines the value to use for the darkest pixels. Intermediate values are scaled between the pair.
sip	added	`bg.hiRes.gray.(pr|bm).*.adj`	Defines the adjustment applied to backgrounds when displayed on a gray hiRes phone. pr in the parameter name refers to the predefined background table. bm refers to the user-defined bitmaps table. The index is the index into the respective table. The value is the number of steps to brighten the image (negative values darken the image). Each step is 1/16th of full scale.
sip	added	`bg.hiRes.gray.bm.*.name`	Defines gray-scale backgrounds for the phone's display and the expansion modules' displays (em).
sip	added	`button.gray.selection.*.*.modify`	See button.color.selection.*.*.modify above.

File	Action	Parameter	Description
sip	added	`bitmap.IP_7000.*.name`	Defines the bitmaps used in the user interface of the IP 7000 phone. This is the same format as used with other SPIP phones.
sip	added	`ind.anim.IP_7000.*.frame.*. (bitmap| duration)`	Defines the animations used by the IP 7000 phone. This is the same format as used with other SPIP phones.
sip	added	`ind.gi.IP_7000.*.(index|class|physX |physY|physW|physH)`	Defines the graphical indications used by the IP 7000 phone. This is the same format as used with other SPIP phones.
sip	added	`log.level.change.(clink|pnetm|peer)`	Three new logging types have been added. clink logs low-level Clink2 activity in the IP 7000. pnetm logs mid-level Clink2 activity. peer logs high-level activity.
sip	added	`ramdisk.nBlocks.IP_650`	This controls the number of blocks of memory devoted to the ramdisk in the IP 650 phone.

Understanding Updates to SIP 3.0.1RevB

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.1 RevB beside their respective Polycom tracking identification number.

Enhanced Capabilities

42034 Phone freezes when booting from TFTP server in certain scenarios.

42121 SoundPoint IP 550 and 650 phones will not provision using the ‘large’ sip.ld software image. Phone reports Application does not support self provisioning.

Understanding Updates to SIP 3.0.1.0032 (Limited Distribution)

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.1.0032 beside their respective Polycom tracking identification number.

New or Enhanced Features

40475 Set VLAN Filtering to 'Off' by default.

41025 Set default Corporate Directory background re-sync period to 12 hours.

Discontinued Features

35285 Add check for user part of check-sync. This was causing issues with the use of Check-Sync for remote re-boot of phones.

Enhanced Capabilities

36320 Echo is heard on handset to handset call during single talk setting hsAec to 1 on IP650/550/430/330.

38960 Enhance packet loss handling on IP 650 to match performance of IP 601 in large packet loss situations.

39330 DHCPINFORM should apply if boot server address is Null or 0.0.0.0. (0.0.0.0 checking was not working correctly).

39430 Port component in refer-to target URI is needed in a certain situation.

40121 VLAN tag not added to frame that is an IP fragment with between 1 and 3 octets of payload.

Configuration File Enhancements

Refer to Table 24: Software Version 3.0.1.0032 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.0.1.0032 configuration file parameters.

Table 24: Software Version 3.0.1.0032 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	change	`dir.corp.backGroundSync.period`	Changed value from 300 (5 minutes) to 43200 (12 hours)

Understanding Updates to SIP 3.0.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.0.0 beside their respective Polycom tracking identification number.

An asterisk indicates that the feature requires a license-key to be enabled.

New or Enhanced Features

26088* Add RTCP reporting via SIP protocol according to RFC draft draft-ietf-sipping-rtcp-summary – all supported phone models except SoundPoint IP 301.

29851* Support Statistics gathering and reporting for QOS monitoring according to RFC3611 (RTCP-XR) – all supported phone models except SoundPoint IP 301.

30091* Add a Conference Management User Interface for conferences hosted locally on the phone (SoundPoint IP 550, 560, 650 phones).

30099* Add uaCSTA support.

30134 Allow speakerphone to be disabled by configuration file.

30993 Submit from Web Browser should not initiate a reconfig/restart when no changes have been made on the phone.

31442 Make automatic resume on centralized conference optional. Implemented for uaCSTA purposes; configured using *call.disableAutoResumeCentralConference.*

31576* Add 4-way local conferencing on SoundPoint IP 550, 560, 650 phones.

32054* Integrate with corporate directories using LDAP and Active Directory.

32058 Added configurable behavior to support Single Keypress Conference Set-up. Uses `call.singleKeyPressConference` parameter.

32223 Added sound effects to accompany USB device insertion and removal.

32848* Added call recording and playback on USB flash drive. Refer to Technical Bulletin 38084 for details on supported USB devices.

33230 Added SCA Bridging for BroadWorks. Refer to Technical Bulletin 33230 for more details.

34949 Added support for min-expires header.

35150 Added electronic hookswitch capability using Jabra DHSG protocol on SoundPoint IP 320, 330, 430, 550, 560, 650 phones. This feature requires BootROM 4.1.0 to operate. Refer to *Technical Bulletin 35150: Using an Electronic Hookswitch (EHS) with Polycom SoundPoint IP and VVX 1500 Phones* for more details.

37159 Handle MIME type application/vq-rtcpxr in SIP stack.

37256 Jabra Jx10 electronic hook switch support on SoundPoint IP 320, 330, 430, 550, 560, 650 phones. Requires an Interface Cable from the headset base to the phone for use. Refer to technical bulletin 35150 for more details.

37551* Add enhanced speed dial capability.

38443 Support full complement of BLF parties on SoundPoint IP 650 plus 3 EMs using UDP.

38847 Line-Key and Soft-Key Labels changed to white text with 3-D appearance on SoundPoint IP 550, 560, 650 phones.

38979 Make UI background bitmap configurable – SoundPoint IP 550, 560 and 650 phones.

39071 DHCPINFORM should apply if boot server address is null.

39072 Reduce DHCPINFORM retry timeouts.

39305 Increase Handset transmit loudness by 3dB to better meet standards AS/NZS 60950 and AS/ACIF S004, as directed by Category C33 of the Telecommunications Labeling Notice (TLN) (for Australia).

39330 DHCPINFORM should apply if boot server address is 0.0.0.0.

39344 Update XML Dictionaries for SIP 3.0.0.

39695 Lower minimum `syslog.renderLevel` to 0 from 1.

Discontinued Features

37321 Remove support for Asian languages from IP 600 and IP 601 phones (due to memory limitations)

Enhanced Capabilities

30170 Icon Frame is missing when pressing menu key.

30814 Phone sends INVITE with an incomplete SDP section in a certain call sequence.

30903 Packet Loss statistics 'jump' if calls are transferred.

30990 LED does not blink for incoming call on IP 301 when DND enabled and `call.rejectBusyOnDnd=0`.

32668 When a call on shared line is put on hold, pressing and holding line key of a remote shared line causes incorrect soft keys to appear.

34445 Do Not Disturb feature fails on cancellation of second incoming call when `call.rejectBusyOnDnd=0`.

35459 On configuring Identification - Auth Password in web interface for configuration, the parameter value is entered in override mac-phone.cfg.

35937 SoundPoint IP 550,560,650 phones do not support setting Tx Digital gain in config file.

35963 Large XHTML document can trigger reboot on phones with more than 16MB RAM.

36063 HD-Voice Handsets are marginal with respect to hearing aid compatibility (HAC).

36296 Dialing from directory or hot-dialing bypasses automatic off-hook call placement.

36490 Display Diagnostics has some areas that do not work correctly.

36583 IP 301 logs ssps errors during bootup and when establishing a handsfree call.

36677 IP320/330 does not update its Presence status when a roaming buddy changes their status.

36680 Dial tone can become momentarily very loud when cancelling conf call.

36751 EM display diagnostics fails during hot plug-in.

37071 Internal per-line call limit can be overridden on platforms that do not allow 24 calls per line.

37111 Using default certs log message appears when configuring for Custom cert only.

37116 Date and time disappear from the phone's idle screen when browsing menu during call.

37184 Digest Authentication Password used for downloading configuration files is displayed in log files.

37227 The registration icon disappears when IP301 establishes a conference call.

37391 Phone does not start correctly if the contact directory XML syntax is not correct.

37420 SIP Server Fall-back, line Information screen does not show the server info when 3rd SIP server becomes the working server (*applies to SoundPoint IP 320,330*).

37426 Cannot change selection in Clock Time menu more than once without exiting.

37428 Selecting another language forces exit from language menu.

37603 Key remapping does not show correct values in diagnostics menu on IP 320, IP 330 and IP 4000.

37679 File TX Tries setting in flash could be set to invalid value 0.

37690 Phone does not retry ACK when receiving duplicate 200 OK.

37709 SoundPoint IP 320 and IP 330 phones may re-boot after several days when the idle micro-browser is configured and active.

37711 Brief audio ‘noise’ due to SRTP encryption key change.

37719 Pressing Resume soft key on phone after sending an unresolvable hostname during a blind transfer reboots or freezes the far end phone.

37726 DNS SRV queries can incorrectly append search domain when it is already present.

37851 SRTP phone doesn't include crypto suite in group pickup signaling.

37855 Join soft-key is not available when maximum call appearances are used.

37906 IP301 does not show watch buddy icon when peer-to-peer watch buddy is enabled.

37915 Peer-to-Peer Presence: Blocking contact in Watcher List creates extra contact SPIP in directory menu.

38021 Ringer type 12 is not playing correctly.

38219 While receiving multiple NOTIFY messages, the phone may not send an invite to initiate a call.

38279 If a 403 response is received by the phone when attempting to complete a call transfer in the proceeding state the phone may re-boot.

38308 Packet Loss count does not increment correctly when a Held call is resumed and the SSRC value changes.

38334 MKI format in RTP and RTCP packets is incorrect.

38540 Packet channel statistics computation not resetting properly when SSRC changes.

38732 Line status icon does not change back on line 2 after being on speaker or handset – SoundPoint IP 330/320.

38902 UI malfunctions when remote shared line is in hold status and local phone attempts a new call.

39041 Icon may indicate phone is unregistered after successful re-registration if voIpProt.SIP.serverFeatureControl.cf=1 or `voIpProt.SIP.serverFeatureControl.dnd=1`.

39074 Microbrowser: clicking a link to non-responsive server takes a long time to timeout.

39184 Read-only directory can be edited on IP 320 and IP 330 if phone is in digit collection state when contact directory is opened.

39338 Some of the SRTP session parameters are incorrectly spelled in the SDP (e.g. UNENCRYPTED_SRTCP is represented as UNENCRYPTED_RTCP).

39362 Phone does not play incoming RTP when offered send-only stream.

39419 Maximum Backlight Intensity setting has very little effect on SoundPoint IP 560 phones.

39431 Display Diagnostics shows very minimal changes on the display on IP 550 and IP 650.

39438 Backlight does not update immediately after pressing cancel on the maximum intensity screen.

39490 In some call scenarios the phone may not display the SRTP secure line icon even though the call is encrypted.

39502 DigitMap: The + character does not get matched in a dial plan.

39601 In IP 320 and IP 330 phone's local contact edit menu, cursor flashes on the character just entered instead of after the character.

39618 font500Prop_16_U0000_U00FF.fnt has anomalously wide K.

39629 When reg.1.callsPerLineKey=1 is set, and a conference is established while transferring the call, the phone hangs and reboots.

39631 Idle browser cuts volume icon.

39652 Some layered windows are incorrectly clipped.

Configuration File Enhancements

Refer to Table 25: Software Version 3.0.0 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 3.0.0 configuration file parameters.

Table 25: Software Version 3.0.0 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SDP.useLegacyPayloadTypeNegotiation`	
sip	added	`voIpProt.SIP.csta`	Enables uaCSTA.
sip	added	`up.handsfreeMode`	Enables or disables hands-free speakerphone.

File	*Action*	*Parameter*	*Description*
phone1	added	`up.analogHeadsetOption`	Selects optional external hardware for use with a headset attached to the phone's analog headset jack.
sip	changed	`tone.chord.callProg.6.offDur`	Changed from 0 to 10000.
sip	changed	`tone.chord.callProg.6.repeat`	Changed from 1 to 2.
sip	changed	`se.pat.ringer.12.name=Ringback-style`	Added 100ms of silence to start of pattern.

sip	removed	`voice.gain.rx.analog.handset.wideband`
		`voice.gain.rx.analog.handset.sidetone.wideband`
		`voice.gain.tx.analog.handset.wideband`
		`voice.handset.wideband`
		`voice.handset.wideband.rxdg.adjust`
			Controlled gain for wideband handset. This control is now performed through the parameters that do not include .wideband.

sip	added	`voice.qualityMonitoring`	The voice.qualityMonitoring section controls the Voice Quality Monitoring feature.

sip	added	`tcpIpApp.keepalive.tcp.idleTransmitInterval`
		`tcpIpApp.keepalive.tcp.noResponseTrasmitInterval`
		`tcpIpApp.keepalive.tcp.sip.tls.enable`
			Controls TCP keep-alive on SIP TLS connections.

sip	added	`call.singleKeyPressConference` `call.localConferenceCallHold`	Enables new conference behaviors.
sip	added	`call.disableAutoResumeCentralConference`	For use with uaCSTA feature for centralized confrerencing.

sip	added	`bg.hiRes.gray.pat.solid.x.name`
		`bg.hiRes.gray.pat.solid.x.red`
		`bg.hiRes.gray.pat.solid.x.green`
		`bg.hiRes.gray.pat.solid.x.blue`
		`bg.hiRes.gray.bm.x.name`
			Sets up color (gray-scale) and graphical backgrounds for IP 550, IP560 and IP 650 phones.

sip	added	`feature.x.name`	Added new features nway-conference, call-recording and corporate-directory

File	*Action*	*Parameter*	*Description*
phone1	added	`reg.x.bargeInEnabled`	Enables barge in feature for SCAs.
sip	added	`dir.corp`	The dir.corp section controls the Corporate Directory feature.
sip	added	`usb.set1.device.1.vendor` `usb.set1.device.1.product`	Identifies supported USB devices. This list should be populated only with devices that are known to work with the phones. See Technical Bulletin 38084 for details.

Understanding Updates to SIP 2.2.2

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.2.2 beside their respective Polycom tracking identification number.

New or Enhanced Features

35534 De-couple Presence Signaling from Idle Screen Soft-key UI.

36931 Add support for SoundPoint IP 560 product.

37053 Add ability to make local contact directory read-only from the phone.

38328 Add check for local contact directory changes during configuration change checks.

38357 Add ability to adjust the maximum brightness of the SoundPoint IP 550 and 650 phones.

38371 Allow for TCP keep-alive on SIP signaling TLS connections.

38654 Add support for SoundPoint IP 320 Part Number 2345-12200-005 and SoundPoint IP 330 Part Number 2345-12200-004 for China market.

38888 Add ability to adjust the maximum brightness of SoundPoint IP Backlit Expansion Modules.

Discontinued Features

38813 Remove 1000 half duplex as a valid ethernet configuration.

Enhanced Capabilities

34800 MWI Notify: Message Waiting Counts are ignored if Messages-Waiting is set to no

35692 Functionality breaks down on pressing conference>>cancel soft keys after transfer try is rejected. Phone reboots.

36566 Microbrowser: Left arrow when on first field in a form makes cursor turn invisible

36786 Changing audio modes (e.g. handsfree to handset) during call set-up mode may not work correctly in some circumstances.

37284/37661 During a Blind Transfer the phone should terminate the call on receipt of a 180 Ringing Response.

37313 RTP packet size incorrect when SRTP authentication turned off.

37316 Authentication failing when phones have different payload size.

37334 Disabling CDP from the phone menu causes an unnecessary reboot.

37709 SoundPoint IP 330/320 phones using the idle micro-browser may re-boot after several days due to low memory.

38112 Logging message indicates that default cert bundle in use when custom only has been selected.

38344 If URL-dialing is disabled in the configuration file, the phone shows Number@ServerIP for caller ID (This issue occurs on SIP 2.2.0 and SIP 2.2.1 releases only).

38430 In a BLA configuration attempting to make a call on a remotely busy shared line may cause the phone to re-boot instead of displaying Service Unavailable. Occurs on SoundPoint IP 330/320, 430, 550, 650 phones.

38435 When the phone's local directory is writable, unable to add a new contact by selecting new entry on SoundPoint IP 330/320 phones.

38666 If a call is initiated in hands-free mode and the Ringback Tone is server generated the far-end user may experience echo when they answer the call. If the originating phone is switched to handset mode and back to hands-free mode the echo goes away. Occurs on SoundPoint IP 330/320, 430, 550, 650 phones.

38678 In a particular network configuration when using BLA the bridged line indication does not light up properly due to a missing NOTIFY from the phone.

Configuration File Enhancements

Refer to Table 26: Software Version 2.2.2 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP version 2.2.2 configuration file parameters.

Table 26: Software Version 2.2.2 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`tcpIpApp.keepalive.tcp.idleTransmitInterval`	Sets the interval of the TCP keep-alive packets.

File	Action	Parameter	Description
sip	added	`tcpIpApp.keepalive.tcp.noResponse TransmitInterval`	Set the retransmission interval when the server fails to acknowledge the TCP keep-alive.
sip	added	`tcpIpApp.keepalive.tcp.sip.tls.enable`	Enables sending a TCP keep-alive packet from the phone to the server. The server is expected to respond with a TCP keep-alive ack. This is only used with TLS sessions.
sip	added	`dir.local.readonly`	When set to 1, the contact directory cannot be changed and [MACADDRESS]-directory.xml is not uploaded.
sip	added	`pres.idleSoftKeys`	If set to 0, appearance of presence idle soft keys is disabled.

Understanding Updates to SIP 2.2.1 (Limited Distribution)

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.2.1 beside their respective Polycom tracking identification number.

New or Enhanced Features

38371 When SIP over TLS is configured the phone will send TCP Keep-Alive messages to the SIP server every 30 seconds, and will retry 3 times (at 20 seconds) before resetting (RST) the connection if no response is received.

Enhanced Capabilities

36557 When SRTP is enabled and so logging level is set to 1, the RTCP sender report displays encrypted values in the log file.

37651 RTP Timestamp not updated correctly for silence packets.

37690 Phone does not retry ACK when receiving duplicate 200 OK.

37708 Phones fail SIP TLS registration when SNTP server is not configured.

37851 SRTP phone doesn't include Crypto Suite in Group Pickup signaling.

37873 Crypto line in answer does not have correct tag field.

37878 Multiple crypto suites not handled when there is a re-INVITE.

37879 SRTCP packets have invalid authentication tags.

37968 Phone with multiple lines using TLS not re-registering on loss of connection.

38110 Far end hears noise when an SRTP call is taken off hold with some SIP servers.

38249 SRTP lifetime value cannot be parsed correctly by the called party.

38384 During a local SRTP conference, a far end holding then resuming may result in one-way audio or noise with some SIP servers.

Configuration File Enhancements

Refer to Table 27: Software Version 2.2.1 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 2.2.1 configuration file parameters.

Table 27: Software Version 2.2.1 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`sec.srtp.offer.HMAC_SHA1_80`	If set to 1 or Null, a crypto line with the AES_CM_128_HMAC_SHA1_80 crypto-suite will be included in offered SDP. If set to 0, the crypto line is not included.
sip	added	`sec.srtp.offer.HMAC_SHA1_32`	If set to 1, a crypto line with the AES_CM_128_HMAC_SHA1_32 crypto-suite will be included in offered SDP. If set to 0 or Null, the crypto line is not included.

Understanding Updates to SIP 2.2.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.2.0 beside their respective Polycom tracking identification number.

New or Enhanced Features

22532 When there has been no activity in a menu for a configurable period of time, the phone returns to the idle display. This does not happen if the user is entering data using a menu.

25274 Added sending vendor identifier information through DHCP.

25702 Added microbrowser support for accepting and displaying a URL that points directly to a BMP image (previously it was necessary to embed BMP images in an XHTML document).

27040 Added new configurable ring-while-busy options.

28029 Added microbrowser support for two-dimensional table navigation using all four arrow keys.

28747 Added a general flash file system caching mechanism so that downloaded resources can be stored in non-volatile memory.

29030 Added automatic provisioning support for individual image files.

29854 Added support for tracking of missed calls to be configurable on a per-line basis.

31558 Added synchronization of local DND/CF features with server-based DND/CF features.

31840 Set transfer time-out for image file download to worst case scenario.

32259 Added microbrowser support for recognizing mime types.

32648 Reformatted call list entries.

33616 Added configuration option for default transfer type for SoundPoint IP 320 and 330 phones.

33748 Improved resistance to denial of service attacks aimed at phone's web server.

34131 Changed URL dialing terminology from Name to URL.

34434 Implemented 300Hz high pass transmit filter to reduce low frequency noise (noise creates problems in some network line echo cancellers). This can be enabled or disabled.

34573 Added support for re-establishing a TLS connection if the connection closes.

34625 Added ability to discover provisioning server address using DHCPINFORM.

34651 Added phone serial number (MAC address) to user-agent string HTTP Gets.

34685 Renamed Services menu entry to Applications.

34705 Added support in microbrowser for form functionality when embedded in tbody or out of tbody.

34707 Added low-delay handset acoustic echo canceller for SoundPoint IP 320, 330, 430, 550 and 650 phones. This can be enabled or disabled.

34874 If all DNS servers are found to be unreachable, the phone suppresses DNS queries for 5 minutes (as per RFC 2308 Sec 7.1).

34998 Increased maximum number of registrations on SoundPoint IP 650 phones to 34.

35039 Pressing Exit soft key when using the microbrowser should return user to telephony application.

35040 Added configurable timeout parameter to allow microbrowser to return to telephony application after a period of inactivity in the microbrowser.

35043 Added configurable option to display or hide browser status messages in microbrowser.

35087 Changed boot-up behaviour so that idle browser only starts about 2 minutes after the phone has booted up (this is to optimize memory use).

35099 Added support for TLS transport to Syslog.

35199 Improved some translations in Norwegian XML dictionary file.

35285 Add check for user part of check-sync.

35296 Added support for managing TLS custom certificates via the configuration file system.

35311 Added support for specifying different versions of the application executable and configuration files in the <*Ethernet address*>.cfg file on the boot server.

35372 Pressing the Exit function key on the SoundStation IP 4000 phone when using the microbrowser should return user to telephony application.

35373 Changed appearance of soft keys when running microbrowser so that they look the same as when running the telephony application.

35419 Added user interface for configuring no-answer and busy forwarding behavior.

35481 Added support for Backlit Expansion Module.

35507 Adding configuration parameter to control the timeout back to the idle display after a period of inactivity in a menu.

36030 Implemented Ethernet ingress filtering for DoS suppression and VLAN filtering.

36277 Added ability to delete the contact number entered in the Forward menu.

36531 Updated all translation dictionary files to rename Services menu entry to Applications.

Discontinued Features

36079 Removed support for the SoundPoint IP 300 and 500 phones.

Enhanced Capabilities

24021 Call display gets corrupted in IP-dialed call if caller presses a digit then puts call on hold.

25744 Spaces go missing in text in microbrowser occasionally.

26110 Volume level cannot be changed in audio diagnostics mode.

26231 ACD login failure should cause busy tone to be played.

26389 Forward contact which has been disabled is not displayed after a reboot.

26935 ACD icon not suppressed if feature is disabled in sip.cfg but activated in phone1.cfg.

27105 The idle browser occasionally displays when the menu is being updated.

27958 Phone hears busy tone for 2 seconds after far end hangs up and another call is already in the incoming state and has triggered the call waiting alert.

28419 Divert settings for lines 7 to 12 are not used.

28503 When in the held state, a shared line hears ring tone instead of call waiting tone when another call comes in.

28570 Stuttered dial tone (indicating voice mail waiting) does not work on shared line.

28622 Some UNICODE ranges are not properly mapped.

28681 Forward is not removed from menu when function disabled.

29014 Cannot edit the local directory on the phone if the file is corrupt on the server.

29358 Phone may malfunction/reboot if the specified DNS server is down and an invalid SNTP address is configured.

29470 Cursor is in wrong position when performing a factory reset on the SoundPoint IP 301 phone.

29573 Phone may freeze if a DNS server address is all zeroes.

29966 Phone may reboot if incorrect information is entered in the menu for custom CA certificate.

30880 Phone may malfunction/reboot when editing a server address which is 255 characters long.

30902 Auto reject or divert settings changed in a contact after entering contact directory by pressing and holding a speed dial line key are not correctly displayed when next pressing and holding that speed dial line key.

31019 There is no confirmation pop-up message after choosing to reset the local security key.

31326 Transferring a call to windows messenger or office communicator may leave the phone in a frozen state.

31886 Remote resume does not work on BLA line when call between two other phones sharing the same line has been put on hold.

31994 Trying to delete a null unicode character in the contact list causes the phone to lock-up/reboot.

32179 When SAS-VP provisioning is used, the boot server password is visible in the application log file.

32816 Phone may lock-up on subsequent call if using NTLM and received transfer from a non-NTLM phone.

32476 IP601 does not work correctly when Presence feature is enabled with LCS server without using Roaming Buddies.

33105 Hold does not work if selected just before a Conference is completed.

33748 Web server has vulnerability to DOS attacks.

33931 Not all keys on phone can be remapped to Null.

34089 SoundPoint IP 430 phone keeps rebooting if a function key is remapped to null in the configuration files.

34196 Phone keeps rebooting when SIP server address is not a fully qualified domain name and primary DNS server replies to queries with ICMP destination unreachable packets (due to service being turned off) and secondary DNS server is not configured with NAPTR and SRV entries for the SIP server.

34237 Default directory file (000000000000-directory.xml) is not downloaded by the phone when the <*Ethernet-address*>-directory.xml file does not exist on the boot server.

34258 Log file is deleted when it reaches the configured size limit even though `log.render.file.upload.append.limitMode` is set to stop.

34271 SoundPoint IP 430/550/650 phones may reboot when microbrowser XHTML page contains combined FORM and TABLE elements.

34460 Local directory file larger than 10kB is downloaded by phone once but on subsequent reboots the phone freezes.

34578 Phones may lock-up when downloading a directory file which contains an empty contact field.

34636 Call on a shared line may lose audio when cancelling a transfer after the far end has already cancelled a transfer or conference.

34641 Emergency Call Routing does not work correctly if multiple numbers are configured in a single entry in the configuration file e.g. dialplan.1.routing.emergency.1.value=911,9911.

34649 First call after a reboot may demonstrate one-way audio if phones have different codec preferences and voIpProt.SDP.answer.useLocalPreferences parameter is set to default.

34891 SoundStation IP 4000 loudness does not decrease for bottom six volume settings.

35320 If two function keys are remapped to dial specific speed dial numbers, only the first one will work.

35480 SoundPoint IP 320 and 330 phones allow watching only 7 buddies instead of 8 and may lock-up when an 8th watched buddy is added.

35490 SoundPoint IP 320 and 330 phones do not display SAS-VP failure messages during boot-up.

35879 Nonce counter not incremented in PRACK.

36031 If a phone is configured to use TLS for the 2nd line and TCP for the 1st, the 2nd line does not register.

36107 SoundStation IP 4000 phone drops maximum size packets when VLAN is enabled.

36477 Configuring the nat.signalPort parameter may cause the phone to lock-up.

36775 Route-Set susceptible to change mid-dialog in certain situations.

36882 Selecting a speed dial number using the ‘nn#’ key sequence does not work on SoundPoint IP 320 and 330 phones when the phone is unregistered or is using URL dialing mode.

36905 CDP packet always advertises LAN duplex mode as Duplex: Full.

36948 On SoundPoint IP 320 and 330 phones, if the Dial and Menu keys are pressed at the same time after entering digits from the idle display, incorrect soft keys are displayed.

36967 If the phone receives an INVITE with SDP which contains video information, it returns a malformed response.

37086 Phone ignores expiration date of CA certificate if SNTP is only set via DHCP.

37632 Out of order SCA signaling can lead to improper handling of Shared Lines in some situations.

37646 DNS SRV querying after a record cache makes registration fail.

Configuration File Enhancements

Refer to Table 28: Software Version 2.2.0 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 2.2.0 configuration file parameters.

Table 28: Software Version 2.2.0 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.csta`	Not currently used, will be used in a future release.
sip	added	`voIpProt.SIP.serverFeatureControl.dnd`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
sip	added	`voIpProt.SIP.serverFeatureControl.cf`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
sip	added	`up.toneControl.bass`	Not currently used, will be used in a future release.
sip	added	`up.toneControl.treble`	
sip	added	`up.audioSetup.auxInput`	
sip	added	`up.audioSetup.auxOutput`	
sip	added	`up.idleTimeout`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
sip	added	`se.pat.ringer.12.inst.5.type=branch` `se.pat.ringer.12.inst.5.value=-4`	
sip	added	`voice.txPacketFilter`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
sip	added	`voice.codecPref.IP_7000.xxx`	Not currently used, will be used in a future release.

File	*Action*	*Parameter*	*Description*

sip	added	`voice.audioProfile.Lin16.frequency`
		`voice.audioProfile.G7221.xxx`
		`voice.audioProfile.G7221C.xxx`
		`voice.audioProfile.Siren14.xxx`
		`voice.audioProfile.Siren22.xxx`
			Not currently used, will be used in a future release.

sip	added	`Several gain and other voice parameters have been added.`	The entire gain section in sip.cfg must be updated. Failure to do this will affect the audio performance of the phone.
sip	added	`voice.rxEq.hf.IP_7000.xxx` `voice.txEq.hf.IP_7000`	Not currently used, will be used in a future release.
sip	added	`call.dialtoneTimeOut`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
sip	added	`call.disableAutoResumeCentralConference`	Not currently used, will be used in a future release.
sip	added	`call.singleKeyPressConference`	
sip	added	`call.transfer.blindPreferred`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
sip	added	`call.cellPhoneAutoBridging`	Not currently used, will be used in a future release.
sip	added	`bitmap.IP_7000.xxx`	
sip	added	`log.level.change.srtp`	See *Administrator's Guide* for SIP 2.2.0 for details[1]

sip	added	`log.level.change.clink`
		`log.level.change.pnetm`
		`log.level.change.peer`
			Not currently used, will be used in a future release.

File	Action	Parameter	Description
sip	added	sec.srtp.enable sec.srtp.leg.enable sec.srtp.offer sec.srtp.require sec.srtp.key.lifetime sec.srtp.mki.enabled sec.srtp.sessionParams.noAuth.offer sec.srtp.sessionParams.noAuth.require sec.srtp.sessionParams.noEncrypRTP.offer sec.srtp.sessionParams.noEncrypRTP.require sec.srtp.sessionParams.noEncrypRTCP.offer sec.srtp.sessionParams.noEncrypRTCP.require sec.srtp.sessionParams.leg.noAuth.offer sec.srtp.sessionParams.leg.noAuth.require sec.srtp.sessionParams.leg.noEncrypRTP.offer sec.srtp.sessionParams.leg.noEncrypRTP.require sec.srtp.sessionParams.leg.noEncrypRTCP.offer sec.srtp.sessionParams.leg.noEncrypRTCP.require sec.srtp.sessionParams.IP_4000.noAuth.offer sec.srtp.sessionParams.IP_4000.noAuth.require sec.srtp.sessionParams.IP_4000.noEncrypRTP.offer sec.srtp.sessionParams.IP_4000.noEncrypRTP.require	
sip	added	license.polling.time	
sip	added	feature.16.name feature.16.enabled	
sip	added	mb.main.idleTimeout	
sip	added	mb.main.statusbar	
sip	added	pnet.role	
sip	changed	tone.chord.ringer.46.offDur=200 to 0 tone.chord.ringer.46.repeat=2 to 1	
sip	changed	se.pat.ringer.12.inst.1.type=silence to chord se.pat.ringer.12.inst.1.value=100 to 46 se.pat.ringer.12.inst.2.type=chord to silence se.pat.ringer.12.inst.2.value=46 to 200 se.pat.ringer.12.inst.3.type=silence to chord se.pat.ringer.12.inst.3.value=2000 to 46 se.pat.ringer.12.inst.4.type=branch to silence se.pat.ringer.12.inst.4.value=-2 to 2000	Note: also added se.pat.ringer.12.inst.5.type=branch and se.pat.ringer.12.inst.5.value=-4

File	*Action*	*Parameter*	*Description*
sip	changed	voice.audioProfile.G722.jitterBufferShrink=500 to 1500 voice.audioProfile.G722.jitterBufferMax=160 to 200	Audio performance tuning.
sip	changed	Several gain and other voice parameters have been changed.	The entire gain section in sip.cfg must be updated. Failure to do this will affect the audio performance of the phone.
sip	changed	voice.rxEq.hd.IP_650.preFilter.enable=1 to 0 voice.txEq.hs.IP_650.preFilter.enable=1 to 0 voice.txEq.hd.IP_650.preFilter.enable=1 to 0 voice.txEq.hf.IP_650.preFilter.enable=1 to 0	Audio performance tuning.
sip	changed	voice.handset.txag.adjust.IP_430=24 to 9 voice.handset.sidetone.adjust.IP_430=-13 to 0	Audio performance tuning.
sip	changed	Multiple parameters in the ind.anim.xxx, ind.class.xxx and ind.gi.xxx sections.	The entire indicator section in sip.cfg must be updated. Failure to do this will affect the appearance of the display.
sip	changed	res.finder.minFree=1200 to 600	
sip	removed	ind.anim.xxx parameters from CTX_CUSTOM1 to CTX_CUSTOM8 and CTX_UNASSIGNED for all platforms	These parameters were not used.
sip	removed	usb.enable usb.bulkDrive.enable usb.bulkDrive.name	These parameters were not used.
phone1	added	reg.x.csta	Not currently used, will be used in a future release.
phone1	added	reg.x.serverFeatureControl.dnd reg.x.serverFeatureControl.cf	See *Administrator's Guide* for SIP 2.2.0 for details[1]

File	Action	Parameter	Description
phone1	added	`call.missedCallTracking.x.enabled`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
phone1	added	`call.callWaiting.ring`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
000000 000000	added	`LICENSE_DIRECTORY`	See *Administrator's Guide* for SIP 2.2.0 for details[1]
000000 000000	added	`APP_FILE_PATH_SPIP300=sip_212.ld` `CONFIG_FILES_SPIP300=phone1_212.cfg, sip_212.cfg`	These are samples of the new fields which can specify application images and configuration files for specific hardware platforms, in this case the SoundPoint IP 300. See *Administrator's Guide* for SIP 2.2.0 for details[1]
000000 000000	added	`APP_FILE_PATH_SPIP500=sip_212.ld` `CONFIG_FILES_SPIP500=phone1_212.cfg, sip_212.cfg`	These are samples of the new fields which can specify application images and configuration files for specific hardware platforms, in this case the SoundPoint IP 500. See *Administrator's Guide* for SIP 2.2.0 for details[1]

Understanding Updates to SIP 2.1.2

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.1.2 beside their respective Polycom tracking identification number.

New or Enhanced Features

35361 Added ability for parameters in <Ethernet address>.cfg to be overridden by model- or platform-specific versions.

35969 Changed behavior of the select button or right arrow button in call lists and contact directory on SoundPoint IP 320 and 330 to give contact information instead of acting the same as the dial key.

36538 Added configurable failover behavior for authentication signaling to specify that the phone first retries a SIP transaction with the server that has just sent a 401 or 407 response. Uses new parameters `voIpProt.SIP.authOptimizedInFailover` and/or `reg.x.auth.optimizedInFailover`.

36647 Added configurable option allowing message waiting indicator to be displayed although voicemail cannot be accessed. Uses new parameter `up.mwiVisible`.

36681 Added logging of version information for configuration files.

Enhanced Capabilities

34899 Phone may continuously reboot if a configuration change is made then power is disconnected and the provisioning server is unavailable.

35873 Registration expiry period is limited to 65535 seconds.

35914 Scheduled logging stops after 99 days.

35961 Cannot use call/group/directed pickup on SoundPoint IP 320 and 330 phones while a call is incoming or the phone is off hook.

35974 SoundPoint IP 320 and 330 phones do not show status for watched contacts until after the next reboot.

35979 SoundPoint IP 320 and 330 phones reboot while trying to use call pickup on a remote hold BLA call.

36011 After changing termination while in a local conference, the first time the volume is adjusted the volume slider shows minimum.

36044 Downloadable character sets are not working correctly in certain scenarios.

36053 On SoundPoint IP 320 and 330 phones, Add and Delete soft keys should not be available in buddy list if roaming buddy feature is disabled.

36072 On SoundPoint IP 320 and 330 phones, the digit map is not applied to numbers selected from a call list when in the dial-tone state.

36074 On SoundPoint IP 320 and 330 phones, the digit map is not correctly applied when using hot dialing from the second line key.

36225 Phone may reboot if several voicemail NOTIFY messages are received from the server in a short interval.

36233 Specially crafted Via: header in an INVITE can lock-up the phone.

36504 A call is dropped if a blind transfer to an invalid number is attempted.

36581 SoundPoint IP 320 and 330 phones cannot send #nn codes.

36753 One phone drops the call when 2nd party attempts another blind transfer to an invalid number.

36877 All microbrowser text, regardless of which tag is used (except for href), is dim on SoundPoint IP 550 and 650 phones.

Configuration File Enhancements

Refer to Table 29: Software Version 2.1.2 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 2.1.2 configuration file parameters.

Table 29: Software Version 2.1.2 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.authOptimized InFailover`	This parameter controls failover behavior during authentication signaling. 0 = default behavior which obeys the RFC 1 = optimization enabled, phone first retries a SIP transaction with the server that has just sent a 401 or 407 response
sip	added	`up.mwiVisible`	0 = same behavior as SIP 2.1.1, this is the default behavior 1 = if `msg.mwi.x.callBackMode` parameter is set to disabled, message waiting indicator is displayed but voicemail cannot be accessed
sip	changed	`Changed file header from $Revision: $ $Date: $ to $RCSfile: sip.cfg,v $ $Revision: $`	This is required to support the new feature 36681 described above.

File	Action	Parameter	Description
phone1	added	`reg.x.auth.optimizedInFailover`	If this parameter is set, it overrides the global `voIpProt.SIP.authOptimizedInFailover` parameter. x is the registration index. See the description for `voIpProt.SIP.authOptimizedInFailover`
phone1	changed	`Changed file header from $Revision: $ $Date: $ to $RCSfile: phone1.cfg,v $ $Revision: $`	This is required to support the new feature 36681 described above.
0000000 00000	changed	`Changed file header from $Revision: $ $Date: $ to $RCSfile: 000000000000.cfg,v $ $Revision: $`	This is required to support the new feature 36681 described above.
0000000 00000-directory ~.xml	changed	`Changed file header from $Revision: $ $Date: $ to $RCSfile: 000000000000-directory~.xml,v $ $Revision: $`	This is required to support the new feature 36681 described above.

Understanding Updates to SIP 2.1.1 C

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.1.1 C beside their respective Polycom tracking identification number.

New or Enhanced Features

32146 Added support for SoundPoint IP 330.

33391 Added support for SoundPoint IP 320.

35415 Added translations for new phrases needed for SoundPoint IP 320 and 330 phones.

Enhanced Capabilities

35913 SoundPoint IP430, 550, 650 phones may reboot while in a call under certain network conditions.

Understanding Updates to SIP 2.1.1

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.1.1 beside their respective Polycom tracking identification number.

New or Enhanced Features

33263 Added support for G.729 Annex B SDP signaling per RFC 3555

Note: New parameter `voice.vad.signalAnnexB` has been added to support this.

35268 Added support for 16 levels of gray on the LCD of SoundPoint IP 550 and 650 phones.

35643 Added support for new SoundPoint IP 320 and 330 phones in the configuration files to allow easier addition of these phones in a future software release.

Enhanced Capabilities

The following issues have been resolved with this release:

32273 Failure of call park action results in a dropped call.

32609 Heavy call volume may cause phone to reject calls due to resource depletion.

33390/35392/35482 Voice activity detection (VAD) comfort noise generation (CNG) packets can be discarded by the jitter buffer or interpreted as out-of-order packets which may result in delayed receive audio when the G.729B codec is in use.

33586 The To URI is used in a refer-to header instead of the contact URI
Note: New parameter `voIpProt.SIP.useContactInReferTo` has been added to **sip.cfg** to control the source of the URI used in the refer-to header.

33647 The phone may reboot because it detects a suspended task even though that task may have been suspended intentionally.

33967 An error message is logged if a daylight savings time (DST) start or stop time of 0 (12am) is selected (although the selection is correctly used).

34325 Microbrowser display is closed when shared line is opened on other phone.

34431 When changing the configuration of a phone via the web interface, the phone may lock up.

34443 A remote-on-hold call on a line is not picked up by the first press of the line key with some SIP servers.

34508 In a G.729 call, SoundPoint IP 50X and 60X phones may reboot with a DSP assertion failure. This problem is more likely in conference calls and can be reliably reproduced within 20 minutes of the call start.

34723 RTCP transmission interval is not consistent with industry norms.

34772 The value of the DLSR field in RTCP sent by the phone can be wrong by up to about one second.

34827 There are two places to configure the microbrowser from the phone web server.

34882 The configuration page on the phone web server has two Event 2 entries in the Global Log Level Limit drop-down list.

34906 NOTIFY request without dialog content (an 'empty' NOTIFY request, such as you would get with a subscription renewal when the line is idle) does not extinguish LED's lit as a result of previous active dialogs.

35049 DSP load graph on SoundPoint IP 550 shows slightly incorrect value.

35228 Phone may have one-way audio when SDP is received with c line below m line.

35293 Soft keys have some missing pixels on the SoundPoint IP 430 when the microbrowser is accessed.

35308 A known problem in the SoundPoint IP 430 processor may cause the phone to reboot with a DSP assertion failure instead of restarting the affected driver.

35477 When handset AEC is enabled on SoundPoint IP 50X and 60X phones, echo may occur on speaker phone when switching between handset and speaker phone.

35533 The phone's web server shows the DST start and stop days as Monday by default instead of Sunday.

35537 A saturated transmit signal may cause SoundPoint IP 430 phone to reboot.

35573 After selecting the Russian language and accessing the microbrowser, the phone may freeze.

36012 Conference host may indicate phone is muted but audio is heard by far end after one leg ends call.

Configuration File Enhancements

Refer to Table 30: Software Version 2.1.1 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 2.1.1 configuration file parameters.

Table 30: Software Version 2.1.1 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.useContactInReferTo`	0 = default behavior which is the same as previous behavior, use URI from initial call's To header in REFER's refer-to header 1 = use URI from initial call's Contact header in REFER's refer-to header when setting up a transfer

File	*Action*	*Parameter*	*Description*
sip	added	voice.gain.rx.analog.chassis.IP_330 voice.gain.rx.analog.ringer.IP_330 voice.gain.rx.digital.chassis.IP_330 voice.gain.rx.digital.ringer.IP_330 voice.gain.tx.analog.chassis.IP_330 voice.gain.tx.digital.chassis.IP_330 voice.rxEq.hs.IP_330.preFilter.enable voice.rxEq.hs.IP_330.postFilter.enable voice.rxEq.hd.IP_330.preFilter.enable voice.rxEq.hd.IP_330.postFilter.enable voice.rxEq.hf.IP_330.preFilter.enable voice.rxEq.hf.IP_330.postFilter.enable voice.txEq.hs.IP_330.preFilter.enable voice.txEq.hs.IP_330.postFilter.enable voice.txEq.hd.IP_330.preFilter.enable voice.txEq.hd.IP_330.postFilter.enable voice.txEq.hf.IP_330.preFilter.enable voice.txEq.hf.IP_330.postFilter.enable	New parameters to support SoundPoint IP 320 and 330 platforms which will be supported in a future software release. Do not change these values.
sip	added	voice.vad.signalAnnexB	A new line can be added to SDP depending on the setting of this parameter and the voice.vadEnable parameter. Default behavior is the same as voice.vad.signalAnnexB = 0: No change to the SDP voice.vad.signalAnnexB = 1: If voice.vadEnable=1, add attribute line a=fmtp:18 annexb=yes below a=rtpmap… attribute line (where ‘18’ could be replaced by another payload) If voice.vadEnable=0, add attribute line a=fmtp:18 annexb=no below a=rtpmap… attribute line (where ‘18’ could be replaced by another payload)

File	Action	Parameter	Description
sip	added	voice.handset.rxag.adjust.IP_330 voice.handset.txag.adjust.IP_330 voice.handset.sidetone.adjust.IP_330 voice.headset.rxag.adjust.IP_330 voice.headset.txag.adjust.IP_330 voice.headset.sidetone.adjust.IP_330 dir.search.field font.IP_330.1.name bitmap.IP_330.1.name tobitmap.IP_330.66.name ind.idleDisplay.mode ind.anim.IP_330.38.frame.1.bitmap ind.anim.IP_330.38.frame.1.duration ind.gi.IP_330.1.index to ind.gi.IP_330.10.index ind.gi.IP_330.1.class to ind.gi.IP_330.10.class ind.gi.IP_330.1.physX to ind.gi.IP_330.10.physX ind.gi.IP_330.1.physY to ind.gi.IP_330.10.physY ind.gi.IP_330.1.physW to ind.gi.IP_330.10.physW ind.gi.IP_330.1.physH to ind.gi.IP_330.10.physH	New parameters to support SoundPoint IP 320 and 330 platforms which will be supported in a future software release. Do not change these values.

Understanding Updates to SIP 2.1.0

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 2.1.0 beside their respective Polycom tracking identification number.

New or Enhanced Features

5844 Enhanced support for server fall-back configurations.

7275 Microbrowser should auto-navigate to first selectable item.

7444 Added table support to microbrowser.

8452 Added microbrowser support to the SoundStation IP 4000.

9268 Added unique prompt for billing code entry.

9649 Enhanced '+' global prefix character for E.164 user parts in sip: URIs.

11572 Added ability to strip or insert leading digits for outgoing calls.

13497 Updated default daylight savings time rules.

13818 Added ability to disable message waiting indication on a line by line basis.

13882 Added support for setting RTP streams to inactive when on hold.

14485 Increased maximum number of digit map segments to 30.

14733 Improved text entry efficiency in the microbrowser.

14740 Improved visibility of cursor in text entry fields of microbrowser.

14759 Added microbrowser support to the SoundPoint IP 501 platform.

14760 Added microbrowser support to the SoundPoint IP 430 platform.

14900 Changed line-seize subscription failure handling to be biased towards providing dial tone.

15934 Added more low-end dynamic range to volume control.

16110 Added support for SoundPoint IP 550 platform.

16515 Improved aresDnsLookup: time out on socket select log message.

16527 Added a debugging command to display cached DNS NAPTR records.

17124 Added support for SYSLOG reporting of system status and errors.

18434 Changed call timer clock display to have no leading colon.

18966 Added support for adding phone serial number (*Ethernet address*) to user agent string in HTTP GET's used by microbrowser, and modified format of user agent string used during provisioning process and used by microbrowser.

19111 Added TCPOnly as a transport option.

19425 Added microbrowser support for form input elements with checked = true attribute.

19443 Added microbrowser support for forms within tables.

19572 Added configurable sticky line seize behavior only for on-hook dialing.

Enhanced Capabilities

The following issues have been resolved with this release:

7301 Phone doesn't ring if one line has Do Not Disturb enabled.

16354 Inconsistent error message given when attempting to make a call on an unregistered line using different methods when `call.enableOnNotRegistered` is set to '0'.

16477 When phone is configured for NAPTR transport but server does not contain NAPTR and SRV, the phone may do SRV lookups for A records or A lookups for SRV records.

16899 Phone can send a malformed target URI in some NOTIFY messages in certain scenario.

17179 Transfer may fail in some scenarios if the Transfer soft key is pressed before the second party answers.

17318 Phone does not update presence status (e.g. to offline) when reboot initiated.

17422 When using a bridged line, if a call is transferred to an invalid number it cannot be retrieved.

17614 Setting the phone's own status through MyStat does not work properly.

17868 Boot server password is displayed in Configuration menu if boot server is specified as a full URL including user name and password.

17911 Per-registration DND does not work on SoundPoint IP 430.

17918 call.enableOnNotRegistered parameter is not working correctly.

17920 Incorrect icon displayed for offline status when using peer-to-peer presence.

18078 When using an LCS server, contacts cannot be added on the phone when the contact list is empty.

18147 Expansion modules may display solid background if SoundPoint IP 601 or 650 has maximum number of registrations configured and maximum number of roaming buddies enabled.

18198 Value of reg.x.callsPerLineKey parameter is not taken into account when additional calls are placed using hot (static) dialing.

18297 VAD/CNG Rx synthesis not working on SoundPoint IP 650.

18333 Received data on any socket resets timeout of all sockets.

18393 DTMF levels 3dB lower than configured level when RFC 2833 disabled.

18501 Incoming call is sent to wrong line in some scenarios when the phone has an active call and reg.x.lineKeys > 1.

18688 Value of reg.1.callsPerLineKey parameter is not taken into account when two lines are configured and reg.2.callsPerLineKey is set to default and there is a call on hold on both lines.

18772 SoundPoint IP 650 phone does not show 'HD' animation when a wide-band call is transferred to it.

18773 After a transfer, a SoundPoint IP 650 phone may incorrectly display the 'HD' animation when the call is no longer a wide-band call.

18785 After receiving a transferred call which is not a wide-band call, a SoundPoint IP 650 phone may incorrectly display the 'HD' animation.

18985 The log render level for the sip module cannot be changed.

19113 Phone sends incorrect authorization header in some hold scenarios.

19124 Setting codec preferences using web interface does not work correctly for SoundPoint IP 650.

19252 Phone does not send a final NOTIFY to initiator of transfer if the phone cancels the transfer before it completes.

19292 SoundPoint IP 650 phone may freeze after restarting after configuration changed using one of the menus.

19427 Phone can display Cache bounced error message when submitting forms from the microbrowser.

19524 Problems resuming a call which is on hold on a remote bridged line for a specific SIP server.

19605 Phone may continue to send INVITE's in specific scenario if a call is initiated then ended but the SIP servers are not reachable.

19664 Phone may reboot in some scenarios with log file showing a Null pointer in a specific task.

19702 Receipt of a re-transmitted invalid SIP ACK message may cause phone to reboot.

19754 Do Not Disturb key cannot be remapped to Null.

19827 Phone using Bridged Line Appearance can send corrupt message header in SUBSCRIBE message.

19875 Phone should use NTP time to check validity of SSL server certificate.

19876 Phone will lose some memory if microbrowser displays Cache bounced error message due to unresponsive server.

19883 Handset sidetone level is 3dB too hot on SoundPoint IP 430.

35063 Power levels reported via CDP for SoundPoint IP 650 are too low.

35068 Power levels reported via CDP for SoundPoint IP 601 with EM Power option enabled are too high.

Configuration File Enhancements

Refer to Table 31: Software Version 2.1.0 – Configuration File Parameter Enhancements for a list of enhancements made to the SIP 2.1.0 configuration file parameters.

Table 31: Software Version 2.1.0 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
phone1	added	`reg.x.server.y.lcs`	Refer to Technical Bulletin 5844.

phone1	added	`dialplan.x.applyToUserSend=1`
		`dialplan.x.applyToUserDial=1`
		`dialplan.x.applyToCallListDial=0`
		`dialplan.x.applyToDirectoryDial=0`
			Refer to Technical Bulletin 11572.

phone1	added	`reg.x.server.y.transport` and `reg.x.outboundProxy.transport`	Added TCPOnly as a possible value for these existing parameters.

File	*Action*	*Parameter*	*Description*
phone1	changed	`msg.mwi.x.callBackMode=disabled to msg.mwi.x.callBackMode=registration (for x = 2, 3, 4, 5, 6) [changed for bug 13818]`	
sip	added	`voIpProt.server.1.lcs`	Refer to Technical Bulletin 5844.
sip	added	`voIpProt.SIP.useSendonlyHold`	Can be set to 0 or 1. Null default is 0. Default in sip.cfg is 1. If set to 1, the phone will send a reinvite with a stream mode attribute of sendonly when a call is put on hold. This is the same as the previous behavior. If set to 0, the phone will send a reinvite with a stream mode attribute of inactive when a call is put on hold. Note: The phone will ignore the value of this parameter if set to 1 when the parameter `voIpProt.SIP.useRFC2543hold` is also set to 1 (default is 0).

sip	added	`dialplan.applyToUserSend=1`
		`dialplan.applyToUserDial=1`
		`dialplan.applyToCallListDial=0`
		`dialplan.applyToDirectoryDial=0`
			Refer to Technical Bulletin 11572.

sip	changed	`dialplan.digitmap.timeOut=3 to 3|3|3|3|3|3`	Refer to Technical Bulletin 11572.
sip	changed	`tcpIpApp.sntp.daylightSavings.start.month=4 to 3`	Changes to support new daylight savings time rules.
sip	changed	`tcpIpApp.sntp.daylightSavings.start.date=1 to 8`	
sip	changed	`tcpIpApp.sntp.daylightSavings.stop.month=10 to 11`	
sip	changed	`tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth=1 to 0`	
sip	added	`call.stickyAutoLineSeize.onHookDialing`	Refer to *Administrator's Guide Addendum* for SIP 2.1
sip	changed	`voice.gain.rx.digital.chassis.IP_650=-9 to 6`	Gain changes required to match new

File	*Action*	*Parameter*	*Description*
sip	changed	`voice.gain.rx.digital.ringer.IP_650=-21 to -12`	software load.
sip	changed	`voice.handset.sidetone.adjust.IP_430=-12 to -13`	
sip	added	`voIpProt.server.x.transport` and `voIpProt.SIP.outboundProxy.transport`	Added TCPOnly as a possible value for these existing parameters.

Known Issues and Suggested Workarounds

The following issues will be fixed in a subsequent release.

24805 Cannot answer an incoming call while directory is being saved. No workaround is currently available.

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.

Workaround: Use the correct subnet mask.

26920 Centralized conference fails due to RTP port being slow to open in some cases. No workaround is currently available.

27469 Local Conferencing on IP4000 phones is disabled if G.729 is in the Codec preference list.

Workaround: Disable G.729 as a Codec option on the phone by setting `voice.codecPref.IP_4000.G729AB=""`

30086 Boot servers running explicit FTPS are not supported.

Workaround: Use implicit FTPS or HTTPS.

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

Workaround: Start the pattern with a short period of silence then 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.

33593 Shared line does not show remote active for the second incoming call if `callsPerLineKey` parameter is set to 1.

Workaround: Set `callsPerLineKey` parameter to a value greater than 1.

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.

37437 When SRTP is used with both Authentication and Encryption enabled on SoundPoint IP 301, 501, 600 and 601 platforms and three-way conferencing is enabled, the phone will reboot when a local conference is attempted.

Workaround: Disable local conferencing by setting `sec.srtp.leg.allowLocalConf`="0" or disable SRTP Authentication.

37984 Enabling the Idle bit-map on SoundPoint IP330/320 phones causes the line key labels and dialed digits to be invisible.

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

39001 Difficulties with phone operation due to memory limitations may be experienced if phone directories larger than 50Kbytes are used with SoundPoint IP 330, 330, 430 phones.

Workaround: Keep the local contact directory to less than 50kbytes in size.

38347 The video call appearance disappears from the SoundStation IP 7000 UI and the far end audio is routed to both the IP7000 and the HDX when the external microphone is plugged into the IP7000 during a video call (*applies to SoundStation IP 7000 with HDX Integration*). No workaround is currently available.

41706 Regarding USB Call Recording, a phone does not detect the USB if re-attached quickly after removal before the popup "USB device removed" disappears.

Workaround: Wait until the USB device removed message has disappeared before re-inserting the USB device.

41993 Scrolling through the Corporate Directory may not return complete results if results contain Unicode character values greater than 127.

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

42027 In certain scenarios, the log file time stamps of a SoundStation IP 7000 used as a secondary/slave device are 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 (*applies to VVX 1500*).

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

46997 Camera brightness adjustment does not work between levels 3 to 6 (*applies to VVX 1500*). No workaround is currently available.

47651 URL Dialing must be enabled in order to place calls (*applies to SoundStation IP 7000 with HDX Integration*). No workaround is currently available.

48905 Packet statistics jitter parameter is not correctly computed as per RFC3550 (*applies to SoundStation IP 6000, 7000*). No workaround is currently available.

52141 Daisy chained phone sometimes gets 'stuck' during software upgrade (*applies to SoundStation IP 7000*).

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 such 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. 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* 512kbps to avoid the issue.

54976 H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway with 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. No workaround is currently available.

59773 Phone locks and resets when answering 13th incoming call (*applies to SoundPoint IP 550, 650*).

Workaround: Set the number of calls per line to less than 13.

59812 During an active call, a blind transfer by URL does not work and the URL soft key disappears after some time (*applies to SoundStation IP 7000*).

Workaround: End the call to restore normal state.

61091 The configuration parameter `tcpIpApp.port.rtp.forceSend` = 1024 works only for the SoundStation IP 6000, 7000 and VVX 1500. No workaround is currently available.

61143 Server controlled DND should not work on shared lines.

Workaround: Disable server based DND feature.

62387 When a new registration is added to the phone (update is done during runtime) and the BLF keys are bumped down, incoming calls and notifications to those BLF lines are still showing their indications for the previous line keys. No workaround is currently available.

62482 Server certificate Serial Number SN should not be checked against the host name if the outbound proxy is configured. No workaround is currently available.

63123 Instead of initiating a new call, attendant phone plays reorder tone when BLF line key is pressed for the second time. No workaround is currently available.

65133 Redial does not work after making a call with an account code (*applies to SIP 3.1.7*). No workaround is currently available.

66086 During an active call, the phone does not insert numbers using left arrow key.

Workaround: You can cancel the transfer/conference to restore normal state

67394 Eliminating the BLF line monitoring on the server does not clear the BLF line icons and the indicators on the phone.

Workaround: Need to reboot the phone to restore normal state.

67551 ACD Conference by Emergency Escalation causes all phones to show 'Connected' State. No workaround is currently available.

69469 Phones send a 400 Bad Request response to messages containing "<" or ">" in a display name. No workaround is currently available.

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.

70396 Second call name does not display from click to dial feature. No workaround is currently available.

71248 After the maximum number of calls are established on the phone, the phone displays the Transfer and Conference soft key (*applies to SoundStation IP 5000, 6000, 7000, Duo*). No workaround is currently available.

71694 When Update Configuration is selected on the phone menu, the User ID and Password are truncated to 128 characters. No workaround is currently available.

72299 When phone is registered with BLA lines, phone continues to display remote hold appearance even after remote BLA resumes the call (*applies to SoundPoint IP 450, 560, 650*). No workaround is currently available.

72315 On SoundPoint IP 650 with 2 expansion modules attached, the phone does not show more than 24 lines when the parameter `reg.x.lineKeys` is set to a value greater than 24.

Workaround: Do not set `reg.x.lineKeys` to a value greater than 24.

72387 After pressing transfer soft key remote BLA line doesn't show remote hold status when `call.shared.exposeAutoHolds` is set to 1. No workaround is currently available.

72449 When the phone is registered to a shared line, the Forward option is displayed in the phone menu (*applies to VVX 1500*). No workaround is currently available.

72453 Phone displays only the last characters of a long line label (*applies to SoundStation IP 5000/6000/7000*).

Workaround: Use short line labels

72512 Browsing phone menus using EFK action strings do not work. No workaround is currently available.

72601 Phone fails to dial authorized call (*applies to SoundPoint IP 235T, 335*). 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*). No workaround is currently available.

73089 When joining two parties for a conference call, phone displays the local conference UI if the centralized conference server is not available.

Workaround: End the conference to restore normal state

73617 Phone displays *Split* soft key for a fraction of second while transferring a call (*applies to VVX 1500*). No workaround is currently available.

73721 Phone does not display the caller ID properly for an incoming call with a URL (*applies to VVX 1500*). No workaround is currently available.

73926 Phone displays incorrect caller ID on an active centralized conference call (*applies to SoundStation IP 7000*).

Workaround: End the conference to restore normal state

74003 The phone restarts automatically when the set lease time is expired after enabling/disabling the DHCP server. No workaround is currently available.

74109 A phone number entered on the phone's idle screen is erased when there is an incoming call (*applies to SoundStation IP5000/6000*).

Workaround: Need to press Arrow up/down/left key to restore normal state

74121 Using shared lines with barge in enabled, a SoundStation IP 7000 / VVX1500 cannot barge in to an active call on the shared line. No workaround is currently available.

74126 A calling phone does not get a call waiting ring when the receiving phone is busy placing another call. No workaround is currently available.

74199 Phone displays incorrect caller ID while calling the last dialed number using the Last Call Return (LCR) soft key (*applies to SoundStation IP 7000*). No workaround is currently available.

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

74535 Phone displays incorrect soft key on UI for hold call appearance (*applies to SoundPoint IP 450/560/ 670, and VVX 1500*). No workaround is currently available.

74797 The phone does not display a message on the UI when a conference transfer (blind/attended) is not possible with the call server (*applies to SoundPoint IP 321, 331; VVX 1500)*.

74898 Regarding ZTP, an unregistered phone configured with the 000000000000_profile.cfg file does not reboot when ZTP is set to Enabled. No workaround is currently available.

74956 Phone will go into an INVITE loop and reboot if it receives a 503 response to its initial INVITE. No workaround is currently available.

75156 Regarding instant conferencing, the phone which initiated the conference call experiences different behaviour during a Blind and Consultative transfer (*applies to all SoundPoint IP phones*). No workaround is currently available.

75157 Phones using a Sylantro call server display incorrect soft keys after a Conference service unavailable error is displayed. No workaround is currently available.

75166 Phone displays Enter Number on the status bar instead of Transfer to, Blind Transfer to, or Conference With when the user initiates a transfer/conference (*applies to all SoundStation IP phones*). No workaround is currently available.

75229 Phones using a Sylantro call server display the local conference UI when establishing a centralized conference using the Join soft key. No workaround is currently available.

75263 Phone reboots when attempting a conference using an SCA line (*applies to SoundPoint IP 321*). No workaround is currently available.

Reference Documents

This section lists all documents referred to in these release notes as well as other relevant documents.

Polycom UC Software Administrators' Guide

Version 3.3.0

Version 3.2.0

Technical Bulletins, Quick Tips, and White Papers

White Paper: Configuration File Management on Polycom® SoundPoint® IP , SoundStation® IP , and VVX® Phones

Technical Bulletin 5844: SIP Server Fallback Enhancements on SoundPoint IP, SoundStation IP, and VVX 1500 Phones

Technical Bulletin 11572: Changes to Local Digit Maps on SoundPoint® IP, SoundStation® IP, and Polycom® VVX® 1500 Phones

Technical Bulletin 34787: Using Feature Synchronized Automatic Call Distribution with Polycom SoundPoint IP Phones

Technical Bulletin 35150: Using an Electronic Hookswitch (EHS) with Polycom SoundPoint IP and VVX 1500 Phones

Technical Bulletin 35704: Managing Memory Allocation on SoundPoint IP and SoundStation IP Phones

Quick Tip 37381: Understanding Enhanced BLF on SoundPoint IP Phones

Technical Bulletin 39358: Using Custom Ringtones on Polycom® SoundPoint® IP, SoundStation® IP, and VVX® 1500 Phones

Technical Bulletin 41137: Best Practices When Using Corporate Directory on SoundPoint IP, SoundStation IP and Polycom VVX Phones

Technical Bulletin 45460: Using Quick Setup with 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 Improvements in Polycom® UC Software 3.3.0.

Technical Bulletin 66743: Security Advisory Relating to Denial of Service Vulnerability on Polycom® SoundPoint® IP and SoundStation® IP Phones

User Guides

SoundPoint IP Phones

SoundStation IP Phones

VVX Business Media Phones

SoundStation IP 7000 and HDX Integration Information

SoundStation IP 7000 Integration with Polycom HDX Overview

SoundStation IP 7000 Integration with Polycom HDX FAQs

User Guide for the Polycom® SoundStation® IP 7000 Phone Connected to a Polycom HDX™ System in Unsupported VoIP Environments

Integration Guide for the Polycom® SoundStation® IP 7000 Conference Phone Connected to a Polycom® HDX™ System in Unsupported VoIP Environments

Miscellaneous

SIP/UCS Downloads Matrix