POLYCOM®

Polycom SIP 3.2.6

Release Notes

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

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Trademarks

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General

These release notes apply to Polycom SIP 3.2.6 and include previous versions of the SIP software. SIP 3.2.6 will support SoundPoint IP, SoundStation IP, and VVX phones. For a complete guide to SIP 3.2.6, refer to the Administrator's Guide for Polycom SoundPoint IP/SoundStation IP/VVX Family.

The SIP 3.2.6 Release Notes contain the following sections:

- General You will need to read this section in order to understand how the changes in SIP 3.2.6 affect Polycom hardware and deployment and configuration of the software.
- What's New for SIP 3.2.6? This section lists new, enhanced, and discontinued software features.
- 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 SIP 3.2.6

This section contains important notes on Polycom hardware and software.

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

SIP 3.2.6 does not include support for the SoundPoint IP 300, 301, 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) 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: SIP 3.2.5 for the SoundStation IP 7000 and HDX 2.6.0, 2.6.0.2, 2.6.1 and 2.6.1.3 for the HDX.

Using Compatible Programs to Edit XML Files

The **sip.cfg** file included with the SIP 3.2.6 software contains language selections in the native font for that language. To use these language files, your XML editor needs to support Unicode. If you edit the **sip.cfg** using an XML editor that is not compatible with Unicode, the language selections on your phone will not display correctly.

Understanding Phone Platform Features and Licenses

SIP 3.2.6 supports the Productivity Suite, which includes features such as a Corporate Directory, Visual Conference Management, and USB Call Recording. Upgrading to SIP 3.2.6 automatically enables the Productivity Suite; no license is required. The Voice Quality Monitoring (VQMon) feature will continue to be licensed as a charged product.

For customers using versions of Polycom software prior to 3.2.6, Polycom will provide a site license for all features in the Productivity Suite, except for the VQMon feature. This license file is available on the Polycom portal and is free to download at
http://www.polycom.com/products/voice/applications/index.html.

SIP 3.2.6 supports features that are available on the SoundPoint IP, SoundStation IP, and VVX 1500 phones. Refer to ***Table 1****: SoundPoint IP Series Features and Licenses* and ***Table 2****: SoundStation IP and VVX 1500 Series Features and Licenses* to find out whether, in SIP 3.2.6, 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 Series Features and Licenses

Feature	*IP 320/330*	*IP 321/331/335*	*IP 430*	*IP 450/550/560*	*IP 650/670*
VQMon	Productivity License	Productivity License	Productivity License	Productivity License	Productivity License
LDAP Directory	Productivity License	Productivity License	Productivity License	Productivity License	Productivity License
Call Recording	No	Yes	No	No	Productivity License
Conference Management	No	No	No	Productivity License	Productivity License
4-way local conference	No	No	No	Productivity License	Productivity License
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes
Customizable UI Background	No	No	No	Yes	Yes
Local SRTP Conference	Yes	Yes	Yes	Yes	Yes
Asian Language	No	No	No	Yes	Yes
Configurable Soft keys	Yes	Yes	Yes	Yes	Yes

Feature	*IP 320/330*	*IP 321/331/335*	*IP 430*	*IP 450/550/560*	*IP 650/670*
XML API	Yes	Yes	Yes	Yes	Yes
Enhanced BLF	No	No	No	Yes	Yes
Warning Field Display	Yes	Yes	Yes	Yes	Yes
H.323 Video	No	No	No	No	No

Table 2: SoundStation IP and VVX 1500 Series Features and Licenses

Feature	*IP 5000*	*IP 6000*	*IP 7000*	*VVX 1500*
VQMon	Yes	Yes	No	Yes (audio only)
LDAP Directory	Productivity License	Productivity License	Yes	Yes
Call Recording	No	No	No	Yes (audio only)
Conference Management	No	No	Yes	Yes
4-way local conference	No	No	No	No
Electronic Hookswitch	No	No	No	Yes
Enhanced Feature Keys	No	No	No	Yes
Customizable UI Background	No	No	No	Yes
Local SRTP Conference	Yes	Yes	Yes	Yes (limitations at high video bandwidths)
Asian Language	Yes	Yes	Yes	Yes
Configurable Soft keys	No	No	No	Yes
XML API	Yes	Yes	Yes	Yes
Enhanced BLF	No	No	No	No
Warning Field Display	Yes	Yes	Yes	Yes
H.323 Video	No	No	No	License installed on 1500D

System Requirements

Polycom recommends using ***Table 3***: *Recommended Platforms and BootROM Software* to help you choose an appropriate BootROM version for your phone. For SIP 3.2.6, Polycom recommends using BootROM 4.3.1.

For more information about which Polycom phones support which software versions, 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 430	3.1.3 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
SoundStation IP 7000	4.1.1 or later
SoundStation IP 7000 (used with Polycom HDX 4000, 7000, 8000, 9000 video systems)	4.2.2 or later
SoundStation IP 7000 used with HDX 6000 video systems. SIP 3.2.1 Cannot integrate with HDX until HDX version supporting integration is announced.	4.2.2 or later
VVX 1500	4.2.2 (NOTE: As of 3.2.2, the SIP and BootROM are distributed as single package for VVX1500)

Downloading the Distribution Files

You can download SIP 3.2.6 using either the combined file or the split 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* 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 SIP 3.3.0 may 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. All files other than the **sip.ld** files will be duplicated across the two ZIP file versions.

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 File* is 3.2.6.0314.

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 as a reference guide to the files distributed in the split ZIP file and a brief description of each file.

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 executables for SoundPoint IP 331
2345-12375-001.sip.ld	SIP application executables for SoundPoint IP 335
2345-11402-001.sip.ld	SIP application executable for SoundPoint IP 430
2345-12450-001.sip.ld	SIP application executable for SoundPoint IP 450
2345-12500-001.sip.ld	SIP application executable for SoundPoint IP 550

Distributed Files	*File Purpose and Application*
2345-12560-001.sip.ld	SIP application executable for SoundPoint IP 560
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.cfg	main core and SIP configuration file
phone1.cfg	example per-phone SIP configuration
sip.ver	Text file detailing the build-identification(s) for the release
000000000000.cfg	Master configuration template file
000000000000-directory~.xml	Local contact directory template file. To apply on a per-phone basis, replace the 0s with the MAC address of the phone, and remove '~' from the file name.

Distributed Files	*File Purpose and Application*
SoundPointIP-dictionary.xml	Includes native support for the following languages: • Chinese, China (for IP 450, 550, 560, 650 and IP 5000, 6000, 7000) • 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). • Korean, Korea (for IP 450, 550, 560, 650, 670, and IP 5000, 6000, 7000). • 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

Downloading the Combined ZIP File

Use ***Table 5****: Understanding the Combined ZIP File* as a reference guide to each of the files distributed in the combined ZIP file and a brief description of each file. The combined file is required for any phones running a BootROM version prior to SIP 3.3.0, 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

Distributed Files	*File Purpose and Application*
sip.cfg	main core and SIP configuration file
phone1.cfg	example per-phone SIP configuration
sip.ver	Text file detailing build-identification(s) for the release
000000000000.cfg	Master configuration template file
000000000000-directory~.xml	Local contact directory template file. To apply on a per-phone basis, replace the 0s with the MAC address of the phone, and remove '~' from the file name.
SoundPointIP-dictionary.xml	Includes native support for the following languages: • Chinese, China (for IP 450, 550, 560, 650 and IP 6000, 7000). • 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 6000, 7000) • Korean, Korea (for IP 450, 550, 560, 650, 670 and IP 6000, 7000) • 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

What’s New for SIP 3.2.6?

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.

New or Enhanced Features

59207 Added VeriSign intermediate CA certificates.

62800 Updated VeriSign 2048-bit Trusted CA Root Certificate.

64378 Added RSA 2048 V3 Root certificate.

69573 Added event notification version checking configuration parameter.

71997 Added full support to RFC 2782.

Enhanced Capabilities

53509 Adding an existing contact in the buddies list no longer changes the contact display and presence status.

62036 During an active call, if there is an incoming call, phones can send DTMF tones while the incoming call is alerting (*applies to SoundPoint IP 330*).

67622 Phone displays correct caller ID from a group pickup number.

67753 Phone no longer continues to play ring back tone even after the call is timed out.

68063 Phones no longer reboot when DHCP fails (*applies to SoundPoint IP 560*).

68184 Phone no longer sends double confirmation on auto answer.

68267 A ringing tone is no longer heard while the phone is on an active call (*applies to SoundPoint IP 650*).

68344 Request for registration on the second line is accepted by the phone.

69022 Repeated call transfer between phones no longer causes a hold tone.

69166 Music on hold now terminates on resuming a held call.

70154 Phone prevents users using Putty from establishing an SSH connection to the phone (*applies to VVX 1500*).

70228 Conference calls using a Shared Call Appearance (SCA) line no longer cause the phone to reboot (*applies to SoundPoint IP* 321).

70633/72089 Phone properly handles the 403 response to a REFER message for a central conference.

70988/71595 A phone set at maximum volume and powered through AC no longer reboots during certain calls.

71071 During an active call, if there is an incoming call, the caller ID displays properly on the phone (*applies SoundPoint IP* 331).

71616 Phones can properly handle a 480 response to a BroadSoft SCA line seize.

71757 During a conference call, holding/resuming the call no longer causes the Presence feature to fail.

71763 During a conference call, the phone no longer sends a Re-INVITE message to the conference server after sending a REFER message (*applies to SoundPoint IP 321/331*).

71874 Phone audio quality is improved when DND and call forwarding features are enabled on the call server (*applies to SoundPoint IP 501/601*).

71987 The RTP timestamp gets updated on a video call (*applies to VVX 1500*).

72179 Phones no longer display local conference user interface when the centralized conference service is not available.

72298 When maximum call appearances are configured on the phones the join soft key now displays (*applies to SoundPoint IP 650 and VVX 1500*).

72333 Phones can sort DNS SRV records based on priority and weight and can sort NAPTR records based on order (lowest to highest).

72337 A user configured with Shared Call Appearance no longer loses audio while transferring a conference call (*applies to SoundPoint IP 560*).

72752 Video no longer freezes while holding/resuming a call (*applies to VVX 1500*).

70641 Phone properly sends 1 off-hook event to the base station.

72949 A Split soft key no longer displays in place of a Transfer soft key (*applies to SoundPoint 650*)

72822/72996 Conference call between three parties now successfully connects all parties.

73075 Exit key enables you to return to the previous menu (*applies to SoundStation 6000*).

73219 Adding an existing contact properly displays a Duplicate contact value message (*applies to SoundPoint IP 320/321/330/331*).

73288 Using shared call appearance, an attempt to setup a second conference call from the same line is now successful.

73377 Phone can fragment packets when instructed by an ICMP message.

73411 A call transferred to a busy user will update the presence information on a monitoring phone.

73695 Phone pointing to the DNS no longer fails to send the SRV.

73718 The caller name and number are now displayed while placing a call (*applies to SoundPoint IP 430*).

73763 Request line now displays the @ symbol.

73776 Phone displays the proper software version (*applies to VVX 1500*).

73824 During a blind call transfer, a phone will display a ring icon and animation for an incoming call (*applies to SoundPoint IP 330/335*).

73888 During an n-way conference call, audio is no longer lost between calling phones.

73930 Phones pointed to DNS servers evenly distribute outgoing calls.

73932 Dialed digits no longer overlap when a call is held (*applies to SoundPoint IP 550/650*).

73937 Phones automatically recover when a directed pick up fails.

73994 When the last call return (LCR) soft key is pressed, it properly returns the call to the last received number (*applies to VVX 1500*).

73997 The phone user interface displays the extension when a transfer/conference call is cancelled (*applies to SoundPoint IP 450, 550, 670*).

74003 Phone acquires IP address and recovers functionality without reboot even after enabling/disabling the DHCP server lease time.

Configuration File Enhancements

Refer to ***Table 6****: Software Version 3.2.6 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.2.6 configuration file parameters.

Table 6: Software Version 3.2.6 – Configuration File Parameter Enhancements

Configuration File	*Action*	*Parameter*	*Description*
sip.cfg	added	voIpProt.SIP.dialog.strictVersionValidation	Enables the phone to disregard the version number of an event notification and manage state based on the contents of the NOTIFY most recently received

Updates to Previous SIP Releases

This section indicates updates to SIP versions prior to SIP 3.2.6.

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 7****: Software Version 3.2.5 - Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.2.5 configuration file parameters.

Table 7: Software Version 3.2.5 - Configuration File Parameter Enhancements

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

Understanding Updates to SIP 3.2.4B

This section lists additions and changes, removals, enhancements, and configuration file parameter changes to SIP 3.2.4B beside their respective Polycom tracking ID number.

Enhanced Capabilities

66743 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 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 Phones no longer reboot during G.729 packet loss concealment such as when the remote phone is placed on hold (*applies to VVX 1500*).

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, and 335).

55884 The displays on a SoundPoint IP 650 with expansion modules no longer freeze during a consultative transfer.

56032 SoundPoint IP 650 phones with two expansion modules no longer reboot while monitoring continuous BLF traffic.

56488 In packets sent from the client, the Parameter Request List option no longer contains two duplicate requests for the options Router (3) and Domain Name (15) (*applies to SoundStation IP 6000 and 7000*).

56641 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 8****: Software Version 3.2.3 - Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.2.3 configuration file parameters.

Table 8: 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	
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.failRegistrationOn	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 9****: Software Version 3.2.2 - Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.2.2 configuration file parameters.

Table 9: 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`		
phone1	added	`call.autoOffHook.4.protocol`		
phone1	added	`call.autoOffHook.5.protocol`		
phone1	added	`call.autoOffHook.6.protocol`		
phone1	added	`reg.1.protocol.H323`		

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
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		
phone1	added	reg.5.server.H323.1.expires		
phone1	added	reg.5.server.H323.1.port		
phone1	added	reg.6.protocol.H323		

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
phone1	added	`reg.6.protocol.SIP`		
phone1	added	`reg.6.server.H323.1.address`		
phone1	added	`reg.6.server.H323.1.expires`		
phone1	added	`reg.6.server.H323.1.port`		
sip	added	`call.autoAnswer.H323`		0
sip	added	`call.autoAnswer.micMute`		1
sip	added	`call.autoAnswer.ringClass`		4
sip	added	`call.autoAnswer.SIP`		0
sip	added	`call.autoAnswer.videoMute`		0
sip	added	`call.autoRouting.preference`		line
sip	added	`call.autoRouting.preferredProtocol`		SIP
sip	removed	`httpd.lp.port`		
sip	removed	`httpd.ta.enabled`		
sip	added	`log.level.change.h323`		4
sip	added	`log.level.change.poll`		4
sip	added	`log.level.change.push`		4
sip	added	`log.level.change.wmgr`		4
sip	removed	`mb.launchpad.enabled`		
sip	removed	`mb.main.1.icon`		
sip	removed	`mb.main.1.text`		
sip	removed	`mb.main.1.url`		
sip	removed	`mb.main.2.icon`		
sip	removed	`mb.main.2.text`		
sip	removed	`mb.main.2.url`		
sip	removed	`mb.main.3.icon`		

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
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
sip	changed	`video.maxCallRate`		512

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
sip	added	`video.profile.H261.annexD`		
sip	added	`video.profile.H261.CifMpi`		1
sip	added	`video.profile.H261.jitterBufferMax`		2000
sip	added	`video.profile.H261.jitterBufferMin`		150
sip	added	`video.profile.H261.jitterBufferShrink`		70
sip	added	`video.profile.H261.QcifMpi`		1
sip	changed	`video.screenMode`		normal
sip	changed	`video.screenModeFS`		normal
sip	added	`voice.audioProfile.G719.32kbps.payloadType`		107
sip	added	`voice.audioProfile.G719.48kbps.payloadType`		108
sip	added	`voice.audioProfile.G719.64kbps.payloadType`		109
sip	added	`voice.audioProfile.G719.jitterBufferMax`		200
sip	added	`voice.audioProfile.G719.jitterBufferMin`		40
sip	added	`voice.audioProfile.G719.jitterBufferShrink`		1500
sip	added	`voice.audioProfile.G719.payloadSize`		20
sip	added	`voice.codecPref.VVX_1500.G719.32kbps`		
sip	added	`voice.codecPref.VVX_1500.G719.48kbps`		
sip	added	`voice.codecPref.VVX_1500.G719.64kbps`		
sip	changed	`voice.gain.tx.digital.chassis.VVX_1500`	6	3
sip	added	`voIpProt.H323.autoGateKeeperDiscovery`		0
sip	added	`voIpProt.H323.dtmfViaSignaling.enabled`		1
sip	added	`voIpProt.H323.dtmfViaSignaling.H245alphanumericMode`		1
sip	added	`voIpProt.H323.dtmfViaSignaling.H245signalMode`		1
sip	added	`voIpProt.H323.enable`		0
sip	added	`voIpProt.H323.local.port`		1720

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*
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 Added support for the SoundPoint IP 335 product.

Configuration File Enhancements

Refer to ***Table 10****: Software Version 3.2.1B - Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.2.1B configuration file parameters.

Table 10: 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 *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.42.frame.1.duration`		1300	See *Administrator's Guide* for SIP 3.2.2 for details[1]

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`ind.anim.IP _335.42.frame.2.bitmap`		PlumHd	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.42.frame.2.duration`		1300	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.43.frame.1.bitmap`		Headset	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.43.frame.1.duration`		1300	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.43.frame.2.bitmap`		PlumHd	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.43.frame.2.duration`		1300	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.44.frame.1.bitmap`		Speaker	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.44.frame.1.duration`		1300	See *Administrator's Guide* for SIP 3.2.2 for details[1]
sip	added	`ind.anim.IP _335.44.frame.2.bitmap`		PlumHd	See *Administrator's Guide* for SIP 3.2.2 for details[1]

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`ind.anim.IP _335.44.frame.2.duration`		1300	See *Administrator's Guide* for SIP 3.2.2 for details[1]

[1] See the Administrator's Guide for SIP 3.2.

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.

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 Created 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

Application load progress bar matches 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 now displays on the SoundStation IP 7000 when using URL dialing.

42611 Recording no longer begins when a full USB drive is attached

42761 Pressing the Content soft key on the SoundStation IP 7000 no longer prompts the user to choose VGA input.

43910 The microbrowser can process an http response which contains an image/bmp.

43916 Configured sampled wave files can be downloaded onto the phone depending on sufficient RAM Disk size.

43990 Missing glyphs in the Japanese 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 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 correctly uses multi-tap behavior 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 603 or 486 responses.

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 11****: 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 Administrator's 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 11: Software Version 3.2.0 - Configuration File Parameter Enhancements

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`call.directedCallPickupMethod`		native or legacy	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	`call.parkedCallRetrieveMethod`		native or legacy	See *Administrator's Guide* for SIP 3.2.0 for details[1]

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`call.parkedCallRetrieveString`		Star code	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	`dialplan.applyToRemoteDialing`		0 or 1; Default is 0	A flag to determine if the dial plan *applies to* calls made through the Polycom HDX system.
sip	added	`dialplan.applyToTelUriDial`		0 or 1 Default is 1	A flag to determine if the dial plan *applies to* uses of the *tel:// URI*.
sip	added	`ind.class.2.state.35.index`		44	Changes Relating to screen layout modifications.
sip	added	`ind.class.2.state.36.index`		42	
sip	added	`ind.class.2.state.37.index`		43	
sip	changed	`ind.gi.IP _400.4.physX`	122	0	
sip	changed	`ind.gi.IP _400.5.physX`	112	10	
sip	changed	`ind.gi.IP _4000.6.physH`	12	0	
sip	changed	`ind.gi.IP _4000.6.physW`	14	0	
sip	changed	`ind.gi.IP _4000.6.physX`	16	0	
sip	changed	`ind.gi.IP _4000.6.physY`	2	0	
sip	changed	`ind.gi.IP _450.16.physX`	176	196	
sip	changed	`ind.gi.IP _450.17.physX`	176	196	
sip	changed	`ind.gi.IP _450.18.physX`	176	196	
sip	changed	`ind.gi.IP _450.19.physX`	176	196	
sip	changed	`ind.gi.IP _450.2.physX`	40	20	
sip	changed	`ind.gi.IP _450.3.physH`	20	0	
sip	changed	`ind.gi.IP _450.3.physW`	20	0	
sip	changed	`ind.gi.IP _450.3.physX`	20	0	
sip	changed	`ind.gi.IP _450.3.physY`	2	0	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	changed	ind.gi.IP _600.13.physH	103	111	
sip	changed	ind.gi.IP _600.13.physY	0	25	
sip	changed	ind.gi.IP _600.4.physY	105	3	
sip	changed	ind.gi.IP _600.6.physH	20	0	
sip	changed	ind.gi.IP _600.6.physW	20	0	
sip	changed	ind.gi.IP _600.6.physX	113	0	
sip	changed	ind.gi.IP _600.6.physY	110	0	
sip	changed	ind.gi.IP _7000.3.physH	20	0	
sip	changed	ind.gi.IP _7000.3.physW	20	0	
sip	changed	ind.gi.IP _7000.3.physX	20	0	
sip	added	lcl.ml.lang.menu.1.label		简体中文 (zh-cn)	Language selection displayed in the appropriate language.
sip	added	lcl.ml.lang.menu.10.label		日本語 (ja-jp)	
sip	added	lcl.ml.lang.menu.11.label		한국어 (ko-kr)	
sip	added	lcl.ml.lang.menu.12.label		Norsk (no-no)	
sip	added	lcl.ml.lang.menu.13.label		Polski (pl-pl)	
sip	added	lcl.ml.lang.menu.14.label		Português (pt-br)	
sip	added	lcl.ml.lang.menu.15.label		Cский (ru-ru)	
sip	added	lcl.ml.lang.menu.16.label		Slovenski (sl-si)	
sip	added	lcl.ml.lang.menu.17.label		Español (es-es)	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`lcl.ml.lang.menu.18.label`		Svenska (sv-se)	
sip	added	`lcl.ml.lang.menu.2.label`		Dansk (da-dk)	
sip	added	`lcl.ml.lang.menu.3.label`		Nederlands (nl-nl)	
sip	added	`lcl.ml.lang.menu.4.label`		English (en-ca)	
sip	added	`lcl.ml.lang.menu.5.label`		English (en-gb)	
sip	added	`lcl.ml.lang.menu.6.label`		English (en-us)	
sip	added	`lcl.ml.lang.menu.7.label`		Français (fr-fr)	
sip	added	`lcl.ml.lang.menu.8.label`		Deutsch (de-de)	
sip	added	`lcl.ml.lang.menu.9.label`		Italiano (it-it)	
sip	added	`log.level.change.lldp`		4	Control the logging detail level for the LLDP feature.
sip	added	`mb.main.autoBackKey`		1	See *Administrator's Guide* for SIP 3.2.0 for details[1]
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		

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	changed	`se.pat.ringer.15.name`	Sampled 3		
sip	changed	`se.pat.ringer.16.name`	Sampled 4		
sip	changed	`se.pat.ringer.17.name`	Sampled 5		
sip	changed	`se.pat.ringer.18.name`	Sampled 6		
sip	changed	`se.pat.ringer.19.name`	Sampled 7		
sip	changed	`se.pat.ringer.20.name`	Sampled 8		
sip	changed	`se.pat.ringer.21.name`	Sampled 9		
sip	changed	`se.pat.ringer.22.name`	Sampled 10		
sip	added	`sec.srtp.requireMatchingTag`		0 or 1	A flag to determine whether or not to check the tag value in the crypto attribute in an SDP answer.
sip	changed	`tone.dtmf.rfc2833Payload`	101	127	The phone-event payload encoding in the dynamic range to be used in SDP offers.
sip	added	`up.idleBrowser.enabled`		0 or 1; default is 0	A flag to determine whether or not the background takes priority over the idle browser. Used in conjunction with `up.prioritizeBackground.enable`.

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`up.prioritizeBackgroundMenuItem.enabled`		0 or 1; default is 1	If set to 1, the Prioritize Background menu is available to the user. The user can then decide whether or not the background takes priority over the idle browser. Used in conjunction with `up.idleBrowser.enabled`.
sip	added	`up.screenCapture.enabled`		0 or 1; Default is 0	A flag to determine whether or not the user can get a screen capture of the current screen shown on a phone. The flag is cleared when the phone reboots.
sip	added	`voice.audioProfile.iLBC.13_33kbps.payloadSize`		30	See *Administrator's Guide* for SIP 3.2.0 for details.
sip	added	`voice.audioProfile.iLBC.15_2kbps.payloadSize`		20	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	`voice.audioProfile.iLBC.jitterBufferMax`		160	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	`voice.audioProfile.iLBC.jitterBufferMin`		40	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	`voice.audioProfile.iLBC.jitterBufferShrink`		500	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	`voice.audioProfile.iLBC.payloadType`		110	See *Administrator's Guide* for SIP 3.2.0 for details[1]

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	removed	voice.audioProfile.Lin16.44.1ksps.payloadType	120		Parameter renamed.
sip	added	voice.audioProfile.Lin16.44_1ksps.payloadType		120	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.audioProfile.Lin16.8ksps.payloadType		116	See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.iLBC.13_33kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.iLBC.15_2kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.IP_6000.iLBC.13_33kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.IP_6000.iLBC.15_2kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.IP_650.iLBC.13_33kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.IP_650.iLBC.15_2kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.IP_7000.iLBC.13_33kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	added	voice.codecPref.IP_7000.iLBC.15_2kbps			See *Administrator's Guide* for SIP 3.2.0 for details[1]

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
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 *Administrator's Guide* for SIP 3.2.0 for details[1]
sip	changed	voIpProt.server.1.port	5060		The port of a SIP server that accepts registration.
sip	added	voIpProt.server.2.address			
sip	added	voIpProt.server.2.expires			Minimum now 10
sip	added	voIpProt.server.2.expires.lineSeize		30	
sip	added	voIpProt.server.2.expires.overlap			
sip	added	voIpProt.server.2.lcs			
sip	added	voIpProt.server.2.port			
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.

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
sip	added	`voIpProt.SIP .strictReplacesHeader`			This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.
sip	added	`voIpProt.SIP .use486forReject`			If set 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	
phone1	added	`attendant.behaviors.display .spontaneousCallAppearances .automata`		0	Flags to determine whether or not a call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource
phone1	added	`attendant.behaviors.display .spontaneousCallAppearances .normal`		1	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
phone1	added	`attendant.resourceList.x.address`		The value of x depends on the phone For IP 450 x=1-2; IP 550, 560 X=1-3; IP 650, 670 x=1-47	The user referenced by `attendant.reg=` will subscribe to this URI for dialog.
phone1	added	`attendant.resourceList.x.label`			Text label to appear on the display. adjacent to the associated line key
phone1	added	`attendant.resourceList.x.type`		normal	Type of resource being monitored.
phone1	changed	`attendant.ringType`		1	
phone1	added	`dialplan.1.applyToTelUriDial`		1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file.
phone1	added	`dialplan.2.applyToTelUriDial`		1	
phone1	added	`dialplan.3.applyToTelUriDial`		1	
phone1	added	`dialplan.4.applyToTelUriDial`		1	
phone1	added	`dialplan.5.applyToTelUriDial`		1	
phone1	added	`dialplan.6.applyToTelUriDial`		1	
phone1	changed	`divert.noanswer.1.timeout`	60	55	Modified No Answer Timeout
phone1	changed	`divert.noanswer.2.timeout`	60	55	
phone1	changed	`divert.noanswer.3.timeout`	60	55	
phone1	changed	`divert.noanswer.4.timeout`	60	55	

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
phone1	changed	divert.noanswer.5.timeout	60	55	
phone1	changed	divert.noanswer.6.timeout	60	55	
phone1	added	reg.1.server.2.address			See *Administrator's Guide* for SIP 3.2.0 for details[1]
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			
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			

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

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
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.line Seize			
phone1	added	reg.5.server.2.expires.over lap			
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.retryMaxCoun t			
phone1	added	reg.5.server.2.retryTimeOut			
phone1	added	reg.5.tcpFastFailover			
phone1	added	reg.6.musicOnHold.uri			
phone1	added	reg.6.server.1.lcs			
phone1	added	reg.6.server.2.address			
phone1	added	reg.6.server.2.expires			
phone1	added	reg.6.server.2.expires.line Seize			
phone1	added	reg.6.server.2.expires.over lap			
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.retryMaxCoun t			

File	*Change*	*Configuration Parameter*	*Old Value*	*New Value*	*Description*
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 third 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 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 Phone correctly displays a call appearance labeled Unknown Party if the remote party is held while reorder tone is played locally (*applies to SoundPoint IP 650*).

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 12****: Software Version 3.1.7 - Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.1.7 configuration file parameters.

Table 12: 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=1`	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

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

Remove support for the SoundStation IP 6000, 7000 products

Remove 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 a 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 13****: Software Version 3.1.3.0336 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.1.3.0336 configuration file parameters.

Table 13: Software Version 3.1.3.0336 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	voIpProt.SIP.serverFeatureControl.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.serverFeatureControl.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.

File	*Action*	*Parameter*	*Description*
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.
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.autoquerySubmitTimeout	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.

File	Action	Parameter	Description
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: Soft keys & 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 Config `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 Enhancements

Refer to ***Table 14****: Software Version 3.1.2 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.1.2 configuration file parameters.

Table 14: Software Version 3.1.2 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
phone1	added	`acd.reg`	See Technical Bulletin34787 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.
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
sip	added	`log.level.change.cmp`	

File	Action	Parameter	Description
sip	added	`log.level.change.usbio`	recording, call media playback, USB I/O respectively.
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 15***: *Software Version 3.1.1 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.1.1 configuration file parameters.

Table 15: 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 16***: *Software Version 3.1.0 C – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.1.0 C configuration file parameters.

Table 16: 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.
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.

File	Action	Parameter	Description
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 3.2.6 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.

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

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 17***: *Software Version 3.1.0.0073 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.1.0.0073 configuration file parameters.

Table 17: Software Version 3.1.0.0073 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.strictLineSeize`	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.
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.

File	*Action*	*Parameter*	*Description*
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.directedCallPickupString`	The star code to initiate a directed call pickup.
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.

File	*Action*	*Parameter*	*Description*
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.
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.

File	*Action*	*Parameter*	*Description*
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.
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.

File	*Action*	*Parameter*	*Description*
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.
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.

File	*Action*	*Parameter*	*Description*
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.
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.

File	*Action*	*Parameter*	*Description*
sip	added	`softkey.feature.basicCallManagement.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 3.2.6 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 18**: Software Version 3.0.4 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.0.4 configuration file parameters.

Table 18: 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.

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. Affects SoundPoint IP 320, 330, 430, 550, 560, 650 phones.

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 19**: Software Version 3.0.3 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.0.3 configuration file parameters.

Table 19: 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
sip	added	`voIpProt.SIP.strictLineSeize`	If set to 1, forces phone to wait for 200 OK when receiving a TRYING notify.

File	Action	Parameter	Description
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

Add Support for the SoundPoint IP 670 product.

Add Support for the SoundStation IP 6000 product.

Add 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 Micro-Browser 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 20***: *Software Version 3.0.2.0917 B – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.0.2.0917 B configuration file parameters.

Table 20: Software Version 3.0.2.0917 B – Configuration File Parameter Enhancements

<table>
<tr><th>File</th><th>Action</th><th>Parameter</th><th>Description</th></tr>
<tr><td>sip</td><td>added</td><td><code>voice.codecPref.IP_(6|7)000.*</code></td><td>Codec support for IP 6000 and IP 7000.</td></tr>
<tr><td>sip</td><td>added</td><td><code>voice.gain.(r|t)x.analog.*.IP_(6|7)000</code></td><td>Gain levels for IP 6000 and IP 7000.</td></tr>
<tr><td>sip</td><td>added</td><td><code>voice.(r|t)xEq.hf.IP_(6|7)000.(pre|post)Filter.enable</code></td><td>Prefilter and postfilter enable for IP 6000 and IP 7000.</td></tr>
<tr><td>sip</td><td>added</td><td><code>dir.corp.backGroundSync</code></td><td>Changed from 1 to 0, disabling background sync.</td></tr>
<tr><td>sip</td><td>changed</td><td><code>dir.corp.backGroundSync.period</code></td><td>Changed value from 43200 (12 hours) to 86400 (24 hours).</td></tr>
<tr><td>sip</td><td>changed</td><td><code>bg.ranges</code></td><td></td></tr>
<tr><td>sip</td><td>removed</td><td><code>bg.color.selection</code></td><td>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.
<table><tr><th>Index</th><th>Type</th></tr><tr><td>1</td><td>Predefined backgrounds</td></tr><tr><td>2</td><td>Solid patterns</td></tr><tr><td>3</td><td>User-defined bitmaps</td></tr></table></td></tr>
<tr><td>sip</td><td>changed</td><td><code>bg.hiRes.color.pat.solid.*.name|red|green|blue)</code></td><td>Defines the name and colour of solid backgrounds.</td></tr>
</table>

File	*Action*	*Parameter*	*Description*
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 21****: Software Version 3.0.1.0032 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 3.0.1.0032 configuration file parameters.

Table 21: 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 3.2.6 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 Add configurable behavior to support Single Keypress Conference Set-up. Uses *call.singleKeyPressConference* parameter.

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

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

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

34949 Add support for min-expires header.

35150 Add electronic hook-switch 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 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 --- IP 320 and IP 330 -- Line Information screen does not show the server info when 3rd SIP server becomes the working server.

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 22**: Software Version 3.0.0 – Configuration File Parameter Enhancements for a list of enhancements made to software version 3.0.0 configuration file parameters.

Table 22: 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.

File	*Action*	*Parameter*	*Description*
sip	added	`up.handsfreeMode`	Enables or disables hands-free speakerphone.
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 23****: Software Version 2.2.2 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 2.2.2 configuration file parameters.

Table 23: 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 24***: *Software Version 2.2.1 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 2.2.1 configuration file parameters.

Table 24: 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 25****: Software Version 2.2.0 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 2.2.0 configuration file parameters.

Table 25: 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	See *Technical Bulletin* 25751 for details
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	

File	*Action*	*Parameter*	*Description*
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
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 <u>must be updated</u>. 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 <u>must be updated</u>. 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.

File	*Action*	*Parameter*	*Description*

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]
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]

[1] Administrator's Guide for SIP 2.2.0

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 26****: Software Version 2.1.2 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 2.1.2 configuration file parameters.

Table 26: Software Version 2.1.2 – Configuration File Parameter Enhancements

File	*Action*	*Parameter*	*Description*
sip	added	`voIpProt.SIP.authOptimizedInFailover`	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.
000000000000	changed	`Changed file header from $Revision: $  $Date: $ to $RCSfile: 000000000000.cfg,v $  $Revision: $`	This is required to support the new feature 36681 described above.
000000000000-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

The following issues have been resolved with this release:

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 signalling 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 27**: Software Version 2.1.1 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 2.1.1 configuration file parameters.

Table 27: 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 to bitmap.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 28***: *Software Version 2.1.0 – Configuration File Parameter Enhancements* for a list of enhancements made to software version 2.1.0 configuration file parameters.

Table 28: 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.

File	*Action*	*Parameter*	*Description*
phone1	added	reg.x.server.y.transport and reg.x.outboundProxy.transport	Added TCPOnly as a possible value for these existing parameters.
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	

File	*Action*	*Parameter*	*Description*
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[1]
sip	changed	`voice.gain.rx.digital.chassis.IP_650=-9 to 6`	Gain changes required to match new software load.
sip	changed	`voice.gain.rx.digital.ringer.IP_650=-21 to -12`	
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.

[1] Administrator's Guide Addendum for SIP 2.1

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 IP 4000 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=` .

27777 SoundStation IP 4000 does not play a local hold reminder tone No workaround is current available.

29144 When parking a call to an empty destination, the phone does not display an error message. No workaround is currently available.

30086 Boot servers running explicit FTPS are not supported.

*Workaround:*Use implicit FTPS or HTTPS.

30371 Pattern generator for tones does not work well 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.

34454 If the microbrowser is enabled, page refreshes are very frequent, and pages contain large images, the phone may lock-up (*applies most frequently to SoundPoint IP 601*).

Workaround: Do not refresh Microbrowser too frequently in configuration settings or by rapidly pressing the Refresh soft key. Design the pages so that the content is within reasonable limits.

34743 A phone may freeze when it receives a check-sync if the resources on the phone are heavily used by downloaded .wav files or large/complex microbrowser pages

Workaround: Reduce the RAM disk size configured in sip.cfg (this will reduce the amount of space available for downloaded wave files and other resources) by setting `ramdisk.nBlocks` to `3072`. Design web pages used by the Microbrowser carefully.

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` (this is the default setting) or disable SRTP Authentication.

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

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

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.

39630 Blocking a roaming buddy from the Privacy list also prevents the user from viewing the blocked buddy's status (*applies to SoundPoint IP 330/320 using LCS 2005*).

Workaround: Do not block users from viewing your status if you wish to view their status.

41706 USB call Recording 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 > 127 (server does not support sorting).

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

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

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

43295 During a call with SRTP enabled, if one user holds the call, the receiving user's phone does not display the held call UI. No workaround is currently available.

44478 Configurable soft key feature does not work. No workaround is currently available (*applies to VVX 1500*).

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

*Workaround:*If SRTP is being used limit the video bit rate to 384kbps.

45247 Browsing microbrowser pages while other functions requiring internal memory are heavily used may cause the phone to reboot (*applies to SoundPoint IP 430*).

Workaround: See *Technical Bulletin 35704: Managing Memory Allocation on SoundPoint IP and SoundStation IP Phones* for information on managing the memory resources on SoundPoint IP/SoundStation IP phones.

46997 Camera brightness adjustment does not work between levels 3 to 6 on the VVX 1500. No workaround is currently available.

47606 Phone configured with BLA and using a Sylantro call server displays the second incoming call on the 1st line even if it's configured with `reg.1.lineKeys=2`.

47651 URL Dialing must be enabled in order to place calls on the SoundStation IP 7000. No workaround is currently available.

47827 Phone uses incorrect units for Jitter in SIP PUBLISH VQSession Report (*applies to SoundPoint IP*). No workaround is currently available.

48463 Cannot view JPEG images with .jpe or .jfif extensions (*applies to VVX 1500*).

Workaround: Ensure that JPEG images use the .jpg extension.

48905 Jitter parameter is not correctly computed on the SoundStation IP 6000/7000 as per RFC3550. No workaround is currently available.

49189 The parameter `up.numberFirstCID` does not apply to call lists. No workaround is currently available.

51904 If an HDX (release 2.5.x and possibly other releases) is configured for SIP using UDP, it does not make a video connection with a VVX 1500.

Workaround: Configure HDX for Auto protocol instead of UDP.

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

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

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

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

52592 Phone fails to provision if using the combined sip.ld file and a TFTP provisioning server that does not support the *bulksize* option (*applies to SoundStation IP 6000*).

Workaround: Either use the *split* image for the SoundStation IP 6000 or use a TFTP server that supports the *bulksize* option.

52782 Video issues are experienced when VVX 1500 phones are bridged on HDX and VSX MCUs

Workaround: Issue appears to be less evident at higher video bit rates.

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.

54292 Status menu displays that the phone is registered to the primary gatekeeper even though it has registered with the alternate gatekeeper (*applies to VVX 1500*). No workaround is current available, but note that this does not affect the phone's operation.

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.

54656 Phone does not display x/y indicator when multiple calls are active if the Time and Date display is disabled.

Workaround: Enable the Time and Date display.

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.

54834 VVX 1500 connects with audio only when an MGC IVR 'Video Welcome Slide' is used.

Workaround: Disable the video welcome slide on the IVR.

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.

55287 Phone drops the incorrect call if the user selects a held call and then attempts to terminate the active call.

Workaround: Ensure that the active call is in focus when terminating the call.

55392 Centralized conferencing is not compatible with an Avaya CS2100 call server. No workaround is currently available.

55477 SRTP key renewal does not occur during local conference calls. No workaround is currently available.

55910 Phone stops operating after appearing to boot up (*applies to SoundPoint IP 430*).

Workaround: The phone must be power-cycled (occasionally more than once) in order for it to operate correctly. See *Technical Bulletin 35704: Managing Memory Allocation on SoundPoint IP and SoundStation IP Phones* for additional details.

58574 Phone does not invalidate an existing registration when it is registered with a BroadSoft server (*applies to SoundPoint IP 650*). No workaround is currently available.

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.

60086 Phone does not generate the event notification when Auto-Answer is enabled (*applies to SoundPoint IP 650*). No workaround is currently available.

62450 When the value in the configuration parameter `mb.idleDisplay.home` is set to point to a URL containing an image, the idle display shows a break in the border located at the bottom left corner. No workaround is currently available.

63123 Instead of initiating a new call, a BLF attendant phone plays a reorder tone when the BLF line key is pressed for the second time. No workaround is currently available.

63262 When dialing a call using the Out of Dialog REFER based method, the user needs to press the speakerphone key twice in order to terminate the call (*applies to SoundPoint IP 650*). No workaround is currently available.

64859 One-way audio will result after resuming a held call when using SRTP and TLS. This only occurs when calls are held after the SRTP packet sequence counter rolls over to zero.

Workaround: Terminate the existing call and establish a new one.

65133 Cannot invoke the Redial feature after making a call and entering an account code (*applies to SoundPoint IP 3xx*). No workaround is currently available.

65650 During a BLA call, you may notice a flicker on your phone UI (*applies to SoundPoint IP 450*).

Workaround: Wait for 5 seconds for the flicker to disappear.

65758 A space character is added to each side of an umlauted character in the microbrowser idle display. For example, G Ä rtner instead of GÄrtner.

Workaround: If the umlaut character is encoded into the word using UTF-8 format (supported by the browser), then the characters will render properly.

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.

66106 The phone will incorrectly select line 2 initiate a call when dialing from the dial pad and pressing the speakerphone key. No workaround is currently available.

67178 Centralized conference, on occasion, will not be established when `reg.1.lineKeys` is set to 5 or greater. No workaround is currently available.

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.

69209 Deleting a voice call recording from the USB flash drive on a VVX1500 phone does not display the icon/busy indicator. No workaround is currently available.

69469 Phone responds with a bad request when a message contains special characters < or > in the display names. 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.

72453 Phone displays only the last characters of a long line label (*applies to SoundStation IP 5000/6000/7000*).

Workaround: Use short line labels.

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.

73996 The phone user interface displays black lines when a transfer call is cancelled. No workaround is currently available.

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.

74392 On an active call between two SoundPoint IP 331 phones, a user with a headset on the receiving phone hears low quality audio. 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.

Reference Documents

This section lists all documents referred to in these release notes as well as other relevant documents.

Polycom SIP Software Administrators' Guide

Version 3.2.2

Technical Bulletins, Quick Tips, and White Papers

White Paper: Configuration File Management on Polycom® SoundPoint® IP , SoundStation® IP , and VVX® Phones

Technical Bulletin 34787: Using Feature Synchronized Automatic Call Distribution with Polycom SoundPoint IP Phones

Technical Bulletin 35704: Managing Memory Allocation on SoundPoint IP and SoundStation IP 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 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