



Release Notes
SIP Application
SoundPoint® IP, SoundStation® IP, and VVX™

Version 3.2.2
3 December 2009

Part Number 3804-11530-322

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1. General

These release notes apply to version 3.2.2 of the SoundPoint IP, SoundStation IP and VVX SIP application. This release includes significant changes for the Polycom VVX 1500 product. The prior SIP 3.2.0 and 3.2.1 releases included the 3.1.3 image for the VVX 1500 product.

For all other products this release is a patch release that includes corrections to issues that have been uncovered in the SIP 3.2.1 release.

For more information, refer to the documents listed in Section 4.

1.1 Important Notes

- **VVX 1500 products running release SIP 3.2.2 CANNOT BE DOWNGRADED TO EARLIER SIP SOFTWARE OR BOOTROM SOFTWARE.**
- Upgrading VVX1500 products to release SIP 3.2.2 requires a more complex procedure than is typical. This procedure is documented in technical bulletin [TB53522](#) Please consult this document before starting the upgrade.
- This release *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 on the SIP 2.1.x release (IP 300, 500) and SIP 3.1.x release (other Legacy models). Technical Bulletin TB35311 describes how to support these Legacy models in an environment where SIP 3.2.0 is deployed for other phones. This bulletin may be downloaded from: http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_publications.html. The template 000000000000.cfg file included with this release is set up to facilitate this type of deployment.
- **SoundStation IP 7000/HDX Integration:** This release is compatible for deployments where the SoundStation IP 7000 is integrated with a Polycom HDX 4000/7000/8000/9000 video system running release HDX 2.5.0.2_29-3382 (hot fix 29). Integration with the Polycom HDX 6000 (in addition to the 4000/7000/8000/9000) systems will be supported by HDX version 2.5.0.7 due for release in December 2009.
- The sip.cfg template file included with this release contains language selections in the 'native' font for that language. These include fonts that are not supported in certain **XML editors**. If the sip.cfg is edited using such an editor the language selections shown in the Languages menu on the phone may not display correctly. To confirm whether your editor properly supports these characters, view the language parameter for languages such as Chinese, Japanese, Korean, Russian – e.g. lcl.ml.lang.menu.1.label
- Documentation on how to enable and use the new features in the SIP 3.2.x release is included in the **Administrator's Guide for SIP 3.2.2** (See Section 4 for details on how to access the document). There is a specific section in this document that references the major new features in the SIP 3.2 Release.

1.2 Feature License and Platform limitations

The following table summarizes several features that require a particular hardware platform and/or a license key for activation.

SoundPoint IP and Polycom VVX Family of Products (Desktop Phones)

Feature	IP 320/330	IP 321/331/335	IP 430	IP 450/550/560	IP 650/670	VVX 1500/-C/-D
VQMon	Productivity License	Productivity License	Productivity License	Productivity License	Productivity License	Yes (Audio only)
LDAP Directory	Productivity License	Productivity License	Productivity License	Productivity License	Productivity License	Yes
Call Recording	No	No	No	No	Productivity License	Yes (Audio only)
Conference Management	No	No	No	Productivity License	Productivity License	Yes
4-way local conference	No	No	No	Productivity License	Productivity License	No
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes	Yes
Customizable UI Background	No	No	No	Yes	Yes	Yes
Local SRTP Conference	Yes	Yes	Yes	Yes	Yes	Yes, with limitations at high video bandwidths
Asian Language	No	No	No	Yes	Yes	Yes
Configurable Soft-Keys	Yes	Yes	Yes	Yes	Yes	Yes
XML API	Yes	Yes	Yes	Yes	Yes	Yes
Enhanced BLF	No	No	No	Yes	Yes	No
Warning Field Display	Yes	Yes	Yes	Yes	Yes	Yes
H.323 Video	No	No	No	No	No	License (pre-installed on 1500D)

Productivity License – licensed as part of the Productivity Suite

SoundStation IP Product Family (Conference Phones)

Feature	IP 6000	IP 7000
VQMon	No	No
LDAP Directory	Productivity License	Yes
Call Recording	No	No
Conference Management	Yes	Yes
4-way local conference	No	No
Electronic Hookswitch	No	No
Enhanced Feature Keys	No	No
Customizable UI Background	No	No
Local SRTP Conference	Yes	Yes
Asian Language	Yes	Yes
Configurable Soft-Keys	No	No
XML API	Yes	Yes
Enhanced BLF	No	No
Warning Field Display	Yes	Yes
H.323 Video	No	No

1.3 System Requirements

Platform	BootROM version
SoundPoint IP 320/330	3.2.3RevB or greater
SoundPoint IP 321/331	4.1.3 or greater
SoundPoint IP 335	4.2.0RevB or greater
SoundPoint IP 430	3.1.3 or greater
SoundPoint IP 450	4.1.2 or greater
SoundPoint IP 550	4.1.0 or greater
SoundPoint IP 560	4.1.0 or greater
SoundPoint IP 650	4.1.0 or greater
SoundPoint IP 670	4.1.1 or greater
SoundStation IP 6000	4.1.1 or greater
SoundStation IP 7000	4.1.1 or greater
SoundStation IP 7000 used with Polycom HDX 4000, 7000, 8000, 9000 video systems.	4.1.2 or greater
SoundStation IP 7000 used with HDX 6000 video systems. SIP 3.2.1 Cannot integrate with HDX until HDX version supporting integration is announced.	4.2.0 or greater
VVX 1500	4.2.1 (NOTE: As of this release the SIP and BootROM are distributed as single package for VVX1500)

For details on historical software version support by platform please refer to the “SIP Downloads Matrix” table accessible from the Polycom Support site at http://downloads.polycom.com/voice/voip/sip_sw_releases_matrix.html

1.4 Distribution Files

The SIP 3.2.2 distribution of the SoundPoint / SoundStation IP/VVX SIP application is done using two methods. Select the downloadable zip file(s) appropriate for your deployment model.

In some cases it may be beneficial to download both release files. If this is necessary, download both zip files, extract all the files from the ‘individual’ release and then extract the

sip.ld file from the 'combined' release file. All files other than ".ld" files are duplicated between the two release zip files.

For centrally provisioned systems, download the appropriate file and extract the files to the provisioning/boot server, maintaining the folder hierarchy present in the zip file.

Some of the configuration files must be modified. Refer to the documents listed in Section 4 for details.

1.4.1 Release using individual (split) files

Use of 'individual files' is recommended as it will result in a faster upgrade time for the phone.

This method requires that all phones be running BootROM release 4.0.0 or newer.

Files	Description
2345-12200-002.sip.ld 2345-12200-005.sip.ld	SIP application executables for SoundPoint IP 320 – Version 3.2.2.0477
2345-12360-001.sip.ld	SIP application executables for SoundPoint IP 321 – Version 3.2.2.0477
2345-12200-001.sip.ld 2345-12200-004.sip.ld	SIP application executables for SoundPoint IP 330 – Version 3.2.2.0477
2345-12365-001.sip.ld	SIP application executables for SoundPoint IP 331 – Version 3.2.2.0477
2345-12375-001.sip.ld	SIP application executables for SoundPoint IP 335 – Version 3.2.2.0477
2345-11402-001.sip.ld	SIP application executable for SoundPoint IP 430 – Version 3.2.2.0477
2345-12450-001.sip.ld	SIP application executable for SoundPoint IP 450 – Version 3.2.2.0477
2345-12500-001.sip.ld	SIP application executable for SoundPoint IP 550 – Version 3.2.2.0477
2345-12560-001.sip.ld	SIP application executable for SoundPoint IP 560 – Version 3.2.2.0477
2345-12600-001.sip.ld	SIP application executable for SoundPoint IP 650 – Version 3.2.2.0477
2345-12670-001.sip.ld	SIP application executable for SoundPoint IP 670 – Version 3.2.2.0477
3111-15600-001.sip.ld	SIP application executable for SoundStation IP 6000 – Version 3.2.2.0477
3111-40000-001.sip.ld	SIP application executable for SoundStation IP 7000 – Version 3.2.2.0477
2345-17960-001.sip.ld	SIP application executable for VVX 1500 – Version 3.2.2.0481
sip.cfg	main core and SIP configuration file
phone1.cfg	example per-phone SIP configuration
sip.ver	Text file detailing build-id(s) for the release.
000000000000.cfg	example master configuration file
000000000000-directory~.xml	example per-phone local contact directory XML file (edit and then remove '~' from name to seed phones which have no directory)

Files	Description
SoundPointIP-dictionary.xml	dictionary files for multilingual support include: 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

1.4.2 Release using Combined Image

The 'combined' sip.ld file contains images for all members of the SoundPoint IP/SoundStation IP/VVX products. This file is required for any phones that may be running a BootROM release older than SIP 4.0.0 (e.g. SIP 3.2.3RevB).

Files	Description
sip.ld	Concatenated SIP application executable, Version 3.2.1.0078 for IP 335, Version 3.2.2.0481 for VVX 1500, 3.2.2.0477 for all other supported models.
sip.cfg	main core and SIP configuration file
phone1.cfg	example per-phone SIP configuration
sip.ver	Text file detailing build-id(s) for the release.
000000000000.cfg	example master configuration file
000000000000-directory~.xml	example per-phone local contact directory XML file (edit and then remove '~' from name to seed phones which have no directory)
SoundPointIP-dictionary.xml	dictionary files for multilingual support include: Chinese, China (for IP 450, 550, 560, 650 and IP 6000, 7000 only) Danish, Denmark Dutch, Netherlands English, Canada English, United Kingdom English, United States French, France German, Germany Italian, Italy Japanese, Japan (for IP 450, 550, 560, 650, 670 and IP 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

Files	Description
SoundPointIPWelcome.wav	start up welcome sound effect

2. Changes

2.1 Version 3.2.2

2.1.1 Added or Changed Features

- 41450: VVX 1500: Change of the real time operating system.
- 43760: VVX 1500: H.323 signaling protocol support for video.
- 43862: VVX 1500: Add support for Webkit browser to replace the XHTML browser.
- 45172: VVX 1500: Add support for iLBC audio codec.
- 47173: VVX 1500: Add support for H.261 video codec
- 48557: VVX1500: Set Default max video bit rate to 384 kbps
- 48743: VVX 1500: Upgrade curl library to version 7.19.
- 48961: VVX 1500: Add support for H.235 security
- 49069: SoundStation IP 6000, 7000: Add support for iLBC audio codec
- 49079: VVX1500: Add support for mutual TLS authentication.
- 49277: VVX1500: Add support for LLDP protocol.
- 49430: VVX 1500: Add ITU-T G.719 vocoder
- 50125: VVX 1500: Outgoing calls support dual (SIP/H.323) protocols
- 51084: VVX 1500: Add support for video fast update request via RTCP, RFC 5104
- 52944: VVX 1500: Add menu support applicable to H.323 usage.
- 53849: Formalize support for “DTMF via SIP INFO” (initially supported in SIP 3.2.0)
- 54025: Increase maximum size of contact directory to 128 to facilitate complex dialing scenarios.
- 54239: VVX 1500: Add user accessible menu option to select the video call rate. Default configured using video.callRate.

2.1.2 Removed Features

- 52522: VVX 1500: Remove “Launchpad” Feature.

2.1.3 Corrections

- 44782: **VVX 1500: Improve phone UI response when a local conference is active.**
- 44980: **VVX 1500: Fall back to configured video codec configuration for Tx video when incoming signalling lacks codec modifiers**
- 47023: **VVX 1500: Occasionally the text font changes.**
- 47476: **XML API: When the user is inside an XHTML Form Field the Submit soft-key does not show up**
- 47768: **SoundPoint IP 450: CDP power usage advertisement is low for peak power conditions.**
- 48175: **VVX 1500: Conference not established using EFK feature.**
- 48784: **VVX-1500: Softkeys not restored after rejecting a call from within the 'Applications' UI context.**
- 48857: **VVX 1500: Recording (R) 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**
- 48921: **VVX 1500 Digit key presses may be missed in certain scenarios**
- 50152: **VVX 1500; Corporate Directory: Change non-null sticky primary filter, search (filtered) bar remains on old data**
- 50192: **VVX 1500: Media Statistics menu is not displayed correctly for several languages**
- 50286: **VVX 1500; Corporate Directory: Pressing page down key "#" does not move entry list after pressing page up key "*" in quick search menu**
- 50531: **SoundStation IP7000: Phone will not startup without network connection when using the PIC cable**
- 50624: **Inbound call is rejected due to timeout but no 603 is ever sent because TCP stream has already been reset.**
- 51141: **Remove the small number on the left side of the scrolling status bar**
- 51449: **VVX 1500: Out of Dialog Refer based dialing is failing. SDP on INVITE from VVX is missing media attributes, generating a 606 response.**
- 51533: **Backlight intensity change is not updated appropriately in Overrides config file.**
- 51605: **VVX 1500: Push request will get lost if it follows another push request immediately.**
- 51643: **SoundStation IP 6000, VVX 1500: Japanese Language is not properly displayed.**
- 51753: **SoundPoint IP 450. Display text look fuzzy especially when using Asian fonts**

- 51959: **Handling of Hold re-Invites is incorrect after one-touch blind transfer to full park orbit.**
- 51965: **HTTP request messages are not directed to proxy**
- 52164: **VVX 1500: Hot-dial does not work in headset mode.**
- 52360: **'Auth Password' field' can be viewed in web configuration page.**
- 52365: **Phones don't transition very well from LLDP to CDP.**
- 52370: **SoundStation IP 7000/HDX Integration: Removing Ethernet cable, unmutes the Muted phone.**
- 52376: **SoundStation IP 6000, 7000: Unable to disable daylight Savings Time. Introduced in SIP 3.2.0**
- 52381: **On some phones; "Retrieve", "Directed" and "Group" soft keys disappear after entering some digits. This occurs when using the call-park/pick-up feature using SIP signaling. Introduced in SIP 3.2.0.**
- 52415: **Enhanced BLF: Ringtones are suppressed when a user is parked**
- 52568: **SoundStation IP 7000/HDX Integration Onyx VI: Phone does not play DTMF tone with default configuration**
- 52580: **SoundStation IP 7000/HDX Integration: Delayed DTMF audio feedback is heard when conferencing third POTS end while using the IP 7000 User Interface.**
- 52656: **VVX 1500: Phone does not support transcoding of video codecs that are not included in the far-end's capability set**
- 52678: **Corporate Directory: When quick/AdvFind search on full last name, some entries are missing.**
- 52709: **License menu reports expiry date of 31-Dec-1969 for license with no expiry date.**
- 52770: **Message-summary SUBSCRIBE is not sent when reg.x.type=shared**
- 52836: **Phone allows the user to enter more than maximum allowed (32) characters in hot dial and contact directory operations. Introduced in SIP 3.2.0.**
- 52860: **Split sk should not be available for a Transfer consultation call if the call per line limit is reached.**
- 52883: **In a particular signaling scenario; When a call is placed to a shared line, the ringer for an IP650 stutters when the call is picked up at another station.**
- 52943: **LLDP reported power usage in logs indicates inappropriate power consumption.**
- 52950: **VQMon: Packet Loss and Burst Gap Loss metrics too high when calling IVR, caused by valid gap in audio sent from IVR**

- **52963: SoundPoint IP 320, 321, 330, 331: Phone re-boots when user press NN# from idle screen to invoke Contact Directory entry screen for NN speed dial index. Occurs if "Presence" feature is enabled. Introduced in SIP 3.2.0.**
- **52971: EFK: Phone re-boots when efkprompt label is longer than 32 characters.**
- **52977: VVX 1500: "Directory" soft key unexpectedly disappears after selecting "Blind" transfer mode**
- **53007: VVX 1500; VQMon: Phone does not compute RFactor and MOS quality scores for the G7221C codec**
- **53034: SUBSCRIBE for BLA with expires:0 received from server is not recognized as terminating the subscription**
- **53254: VVX 1500: It is not possible to change Auth Password for SIP Lines via on-phone Admin Settings**
- **53598: Side-tone still present after call hangup on headset. Resets only after headset button is pressed. Using GN9350e with EHS.**
- **53656: Part number in Phone Status menu is displaying as YYYY-YYYYY-YYY. Introduced in SIP 3.2.1.**
- **53855: When a phone's extension has an underscore in the name, followed only by numbers, the underscore is removed in SIP signaling and the device is not found**
- **53917: Phone Reboots in a certain scenario when using the 'Join' key**
- **53944: SoundPoint IP 320, 330, 321, 331; SoundStation IP 7000: Phone does not display Dir soft-key in Korean and Slovenian languages**
- **53946: SoundPoint IP 550, 560, 650, 670: Sometimes the phone displays the time and date behind a custom idle display.**
- **53975: Phones will not send a SUBSCRIBE message in a certain scenario when using SCA with barge in enabled.**
- **54034: VVX 1500: Phone generates loud static when CNG packets are received.**
- **54139: Consultative Transfer uses wrong URI on REFER. Issue introduced in SIP 3.2.0.**
- **54262: SoundPoint IP 320, 321: Ethernet status menu displays incorrect information**
- **54631: SoundStation IP7000/HDX Integration: The Voice/Video call type prompt needs to be removed when hot dialing and pressing the Hook hard key. The call type should default to Voice by default.**
- **54765: VVX 1500: Phone fails to resend INVITE after 401 from server when second INVITE is roughly 1500 bytes.**
- **54768: VVX 1500: Phone cannot establish calls properly when booted without a network connection.**

- 54886: **Phone does not 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. Loss of audio results.**
- 55052: **Additional parameter in From header of INVITE causes 1-way audio when it is not found in the ACK to a 200 OK**

2.1.4 Configuration File Parameter Changes

File	Change	Attribute	Old value	New value
phone1	added	call.autoOffHook.1.protocol		
phone1	added	call.autoOffHook.2.protocol		
phone1	added	call.autoOffHook.3.protocol		
phone1	added	call.autoOffHook.4.protocol		
phone1	added	call.autoOffHook.5.protocol		
phone1	added	call.autoOffHook.6.protocol		
phone1	added	reg.1.protocol.H323		
phone1	added	reg.1.protocol.SIP		
phone1	added	reg.1.server.H323.1.address		
phone1	added	reg.1.server.H323.1.expires		
phone1	added	reg.1.server.H323.1.port		
phone1	added	reg.2.protocol.H323		
phone1	added	reg.2.protocol.SIP		
phone1	added	reg.2.server.H323.1.address		
phone1	added	reg.2.server.H323.1.expires		
phone1	added	reg.2.server.H323.1.port		
phone1	added	reg.3.protocol.H323		
phone1	added	reg.3.protocol.SIP		
phone1	added	reg.3.server.H323.1.address		
phone1	added	reg.3.server.H323.1.expires		
phone1	added	reg.3.server.H323.1.port		
phone1	added	reg.4.protocol.H323		
phone1	added	reg.4.protocol.SIP		
phone1	added	reg.4.server.H323.1.address		
phone1	added	reg.4.server.H323.1.expires		
phone1	added	reg.4.server.H323.1.port		
phone1	added	reg.5.protocol.H323		
phone1	added	reg.5.protocol.SIP		
phone1	added	reg.5.server.H323.1.address		
phone1	added	reg.5.server.H323.1.expires		
phone1	added	reg.5.server.H323.1.port		
phone1	added	reg.6.protocol.H323		
phone1	added	reg.6.protocol.SIP		
phone1	added	reg.6.server.H323.1.address		
phone1	added	reg.6.server.H323.1.expires		
phone1	added	reg.6.server.H323.1.port		
sip	added	call.autoAnswer.H323		0

sip	added	call.autoAnswer.micMute		1
sip	added	call.autoAnswer.ringClass		4
sip	added	call.autoAnswer.SIP		0
sip	added	call.autoAnswer.videoMute		0
sip	added	call.autoRouting.preference		line
sip	added	call.autoRouting.preferredProtocol		SIP
sip	removed	httpd.lp.port		
sip	removed	httpd.ta.enabled		
sip	added	log.level.change.h323		4
sip	added	log.level.change.poll		4
sip	added	log.level.change.push		4
sip	added	log.level.change.wmgr		4
sip	removed	mb.launchpad.enabled		
sip	removed	mb.main.1.icon		
sip	removed	mb.main.1.text		
sip	removed	mb.main.1.url		
sip	removed	mb.main.2.icon		
sip	removed	mb.main.2.text		
sip	removed	mb.main.2.url		
sip	removed	mb.main.3.icon		
sip	removed	mb.main.3.text		
sip	removed	mb.main.3.url		
sip	removed	mb.main.4.icon		
sip	removed	mb.main.4.text		
sip	removed	mb.main.4.url		
sip	removed	mb.main.5.icon		
sip	removed	mb.main.5.text		
sip	removed	mb.main.5.url		
sip	removed	mb.main.6.icon		
sip	removed	mb.main.6.text		
sip	removed	mb.main.6.url		
sip	added	sec.H235.mediaEncryption.enabled		1
sip	added	sec.H235.mediaEncryption.offer		0
sip	added	sec.H235.mediaEncryption.require		0
sip	added	up.callTypePromptPref		1
sip	added	up.enableCallTypePrompt		1
sip	changed	up.idleBrowser.enabled	0	
sip	added	up.manualProtocolRouting		1
sip	added	up.manualProtocolRouting.softKeys		1
sip	changed	video.autoStartVideoTx		1
sip	added	video.callRate		448
sip	added	video.codecPref.H261		4
sip	changed	video.enable		1
sip	added	video.forceRtcpVideoCodecControl		0
sip	changed	video.maxCallRate		512
sip	added	video.profile.H261.annexD		
sip	added	video.profile.H261.CifMpi		1
sip	added	video.profile.H261.jitterBufferMax		2000

sip	added	video.profile.H261.jitterBufferMin		150
sip	added	video.profile.H261.jitterBufferShrink		70
sip	added	video.profile.H261.QcifMpi		1
sip	changed	video.screenMode		normal
sip	changed	video.screenModeFS		normal
sip	added	voice.audioProfile.G719.32kbps.payloadType		107
sip	added	voice.audioProfile.G719.48kbps.payloadType		108
sip	added	voice.audioProfile.G719.64kbps.payloadType		109
sip	added	voice.audioProfile.G719.jitterBufferMax		200
sip	added	voice.audioProfile.G719.jitterBufferMin		40
sip	added	voice.audioProfile.G719.jitterBufferShrink		1500
sip	added	voice.audioProfile.G719.payloadSize		20
sip	added	voice.codecPref.VVX_1500.G719.32kbps		
sip	added	voice.codecPref.VVX_1500.G719.48kbps		
sip	added	voice.codecPref.VVX_1500.G719.64kbps		
sip	changed	voice.gain.tx.digital.chassis.VVX_1500	6	3
sip	added	volpProt.H323.autoGateKeeperDiscovery		0
sip	added	volpProt.H323.dtmfViaSignaling.enabled		1
sip	added	volpProt.H323.dtmfViaSignaling.H245alpha numericMode		1
sip	added	volpProt.H323.dtmfViaSignaling.H245signal Mode		1
sip	added	volpProt.H323.enable		0
sip	added	volpProt.H323.local.port		1720
sip	removed	volpProt.local.port		
sip	added	volpProt.server.H323.1.address		
sip	added	volpProt.server.H323.1.expires		
sip	added	volpProt.server.H323.1.port		
sip	added	volpProt.SIP.dtmfViaSignaling.rfc2976		
sip	added	volpProt.SIP.enable		1
sip	added	volpProt.SIP.local.port		5060

2.2 Version 3.2.1 B

2.2.1 Added or Changed Features

- 48947: Add Support for the SoundPoint IP 335 product.

2.2.2 Removed Features

None.

2.2.3 Corrections

None

2.2.4 Configuration File Parameter Changes

File	Change	Attribute	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
sip	added	ind.anim.IP_335.42.frame.1.duration		1300	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.42.frame.2.bitmap		PlumHd	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.42.frame.2.duration		1300	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.1.bitmap		Headset	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.1.duration		1300	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.2.bitmap		PlumHd	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.2.duration		1300	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.1.bitmap		Speaker	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.1.duration		1300	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.2.bitmap		PlumHd	See Administrator's Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.44.frame.2.duration		1300	See Administrator's Guide for SIP 3.2.2 for details

2.3 Version 3.2.1

2.3.1 Added or Changed Features

None

2.3.2 Removed Features

None.

2.3.3 Corrections

- 53322: **Setting volpProt.local.port to a non standard port does not send from or advertise that port**
- 53611: **User Language Selection is lost on Upgrade to SIP 3.2.0**
Note that the fix for this issue will guarantee retention of language setting when upgrading from releases prior to 3.2.0 (e.g. 3.1.3) but WILL NOT preserve language changes made when the phone was running SIP 3.2.0 .
- 53685: **Phones ignoring nat.ip parameters.**
- 53852: **SoundStation IP 7000/HDX Integration: DTMF duration should be set to 300ms. for HDX integration.**

2.3.4 Configuration File Parameter Changes

None.

2.4 Version 3.2.0

2.4.1 Added or Changed Features

- 22527: **SoundPoint IP320, 321, 330, 331, 550, 560, 650, 670; SoundStation IP 6000, 7000: Implement 'Scrolling Status Bar'.**
- 26754: **SoundPoint IP 320,321,330,331,450, 550, 560, 650, 670: Add support for the iLBC codec**
- 30079: **Add support for mutual TLS authentication. See technical bulletin TB52609 and the Administrator's Guide for more details on this feature.**
- 32259: **Recognize multiple mime types in the microbrowser.**
- 32753: **Add support for LLDP protocol. To take full advantage of this feature BootROM 4.2.0 should be used.**
- 34782: **Replace libSRTP algorithms with OpenSSL versions**
- 35525: **Modify DND Status Message.**
- 37118: **Add ability to invoke a 'screen capture'**
- 39358: **Add a 'Loud Ringer' Ring-Tone selection. See technical Bulletin 39358 for instructions on how this can be configured.**
- 30855: **SoundStation IP 7000: Create a SoundStation IP 7000 Setup Guide.**
- 41579: **Meet requirements of ETSI TS 102 027-2 v4.1.1 RFC 3261 compliance test for Anatel/Brazil**
- 43141: **Add support for 'Statically Configured' BLF and Call park and retrieve enhancements**
- 43142: **Add support for single button Blind Transfer and Retrieve of a call designated as an 'automata' in the Dialog used for 'Statically Configured' BLF.**
- 43646: **Improve boot-speed in some situations where the boot server is incorrectly configured.**
- 45057: **Languages selection presented in appropriate language**
- 45174: **Upgrade zlib to version 1.2.3**
- 45743: **Upgrade curl library to version 7.19.2**
- 45787: **SoundPoint IP 450, 550, 560, 650, 670: Add instructions for changing label colors in the User Guides.**
- 45791: **SoundStation IP 7000/HDX Integration: Add a configuration option to disable Digit-map rules for 'Remote Dialing' when connected to an HDX.**
- 46093: **Add ability for User to enable/disable display of idle browser from menu**

- 46113: **SoundPoint IP 320, 321, 330, 331: Add navigation button 'shortcuts' in 'Idle Mode' consistent with other phone models.**
- 46248: **SoundStation IP 7000/HDX Integration: Add Admin menu option to manually specify the value to be used as the 'extension' displayed on the phone screen.**
- 46424: **Improve readability of Menu items when using Background images on the display.**
- 46446: **Provide a menu option to view the status of feature licenses.**
- 46683: **Remove Background from scrolling Status Bar for improved readability.**
- 47355: **Scrolling Status Bar should give equal time to each status message**
- 47390: **Add configuration parameters for select ETSI SIP compliance requirements**
- 47463: **Allow for secure entry of passwords in the micro-browser API**
- 47487: **Forcing a 'Back' soft-key in the micro-browser soft-keys is cumbersome**
- 47689: **Add 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: **Add support for BLF call pick-up using Dialog-info within an INVITE with Replaces header**
- 48055: **Enhanced BLF: Improve user experience when an incoming call occurs whilst the user is viewing BLF monitored line call details.**
- 48109: **Include "fmtp" attribute specifying Mode=30 in the SDP when 13.33 kbps iLBC is used.**
- 48136: **Remove platform specific TFTP code and instead use tftp support in curl library 7.19.2**
- 48137: **Add support for BLF call pick-up using Dialog-info within an INVITE with Replaces header**
- 48205: **SoundStation IP 6000, 7000: Add support for the iLBC Codec.**
- 48559: **Scrolling status line should have similar look on various phones.**
- 48578: **SoundPoint IP 430: Reduce the local Contact Directory maximum to 99.**
- 48579: **SoundPoint IP 430: Reduce the maximum number of call supported to 4 (from 8).**
- 48664: **Add User accessible menu option to display whether a device certificate is installed.**
- 48678: **During local conferencing it is now call diagnostics for each call leg. Accessed from Menu->Status->Diagnostics->Media Statistics.**

- 48738: **Add configurable behavior for Directed Call Pick-Up as used for Enhanced BLF.**
- 48780: **Add option to apply digit-map rules to tel:URI initiated calls**
- 48846: **Add 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 volpProt.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: **Add configuration option for the phone to send 486 Busy when call is rejected.**
- 49309: **Combine SoundPoint IP 550 and 560 User Guides.**
- 49465: **Update Destination of outbound call based on display-name in SIP To header responses**
- 49660: **Call Forward: "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 must detect and correct for G.722, G.722.1, G.722.1C, and G.719 RTP timestamp increments based on different sample rates**
- 50769: **SoundStation IP 7000/HDX Integration: Add support for Hook-Flash during POTS calls.**
- 50927: **Add Equifax Secure eBusiness CA-1 to the trusted CA list.**
- 51419: **RFC2543 hold not working when video SDP present in certain scenarios**

2.4.2 Removed Features

- 48283: **Remove support for SoundPoint IP 301, 501, 600, 601 phones.**
- 48698: **Remove support for SoundStation IP 4000**

2.4.3 Corrections

- 27048: **Application load progress bar doesn't match actual progress**
- 29148: **Phone doesn't format the file system when it notes error on screen while loading large configuration files.**
- 29344: **HTTP Digest Authentication does not work on IIS.**
- 30219: **Logs are not uploaded when phone resets to factory default**
- 31858: **Shared line indicator led turns off when 2 phones resume simultaneously**

- 34681: **stickyAutoLineSeize** and **call.enableOnNotRegistered="0"** seize wrong line if 1st is unregistered
- 35288: **Config web-site** takes too much memory during initialization
- 35991: **Roaming Buddy list** with Office Communicator reports all buddies as offline
- 36969: **SoundStation IP 6000** doesn't display Japanese language correctly
- 38348: **SoundPoint IP 320, 321, 330, 331**: SRTP call displays incorrect line icons in a certain scenario.
- 38392: **Blind Transfer** from encrypted phone to an unencrypted private line does not establish the new call as encrypted
- 38418: **Phones** sometimes show SRTCP authentication failure at log level 0
- 38824: **After audio diagnostics** (i.e Record and Play in handset), 1st call gets established in handset mode even if the handset is ON-HOOK.
- 39013: **SoundStation IP 7000** should not recognize cell phone cable without physical cell phone attached
- 39143: **P-Asserted-Identity** header in initial INVITE message not used for caller ID
- 39949: **SoundPoint IP 320, 321, 330, 331**; **Corporate Directory**: Navigation icon is incorrect when using keypad to navigate
- 40679: **SoundStation IP 6000**: Changing the status on "MyStatus" menu does not change the OC client status when **roaming_buddies.reg = 1**.
- 40892: **SoundStation IP 7000**: There is no Time/Date displayed as first phone call established.
- 41939: **Call Recording**: 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** not included in phone's fonts
- 42213: **SoundStation IP 7000**: There is no "SIP:" string displayed whe using URL dialing.
- 42611: **USB Call Recording**: When full USB drive is attached recording should not begin and no new file should be created
- 42761: **SoundStation IP 7000/HDX Integration**: Pressing Content soft key on SoundStation IP 7000 prompts the user to choose VGA input
- 43910: **Microbrowser** fails to process http response with image/bmp directly in a certain situation.
- 43916: **Some of the configured sampled wave files** are not downloaded onto phone becuase of insufficient RAM Disk size.
- 43990: **SoundStation IP 7000**: Missing glyphs in the Katakana bitstream fonts.

- 44100: **If a Call display name includes an @ then the display is truncated after "@" character.**
- 44248: **Micro Browser not displaying any error message when an unsupported media configured in the microbrowser URL.**
- 44273: **When SIP Contact header is a comma separated list only the first contact is processed**
- 44278: **Phone number is not displayed correctly on line key when the length of phone number is more than 10 characters.**
- 44301: **SoundStation IP 6000,7000: Date is not displayed when idle browser is enabled**
- 44377: **Redial key cannot be reassigned**
- 44443: **SoundPoint IP 320,321,330,331: Menu exit via Menu key is not ignored while in Edit mode.**
- 44635: **SoundStation IP 6000: Phone uses incorrect configuration parameters to download customizable fonts**
- 44783: **Cipher list displays different items for different TLS transactions**
- 44844: **USB Call Recording: Stopping Playback through "Back" key not intuitive**
- 44855: **Call Lists: Missed Calls not incremented on Call Forward on Busy**
- 44892: **SoundStation IP 6000, 7000; SCA Barge-In: Phone barges in to the wrong call in a certain scenario.**
- 44962: **Phone displays 3-way animation icon in held screen when conference legs on hold**
- 45143: **Centralized Conference: When max conference size is reached phone displays local conference UI**
- 45327: **Establish a call between two phones configured as shared lines, press down arrow key, all soft keys disappear**
- 45428: **Unexpected re-INVITE occurs before BYE, when removing a leg from a conference call**
- 45650: **Double hold w/ MOH and a non-Polycom SIP phone: one way audio - MOH fails**
- 45658: **Platform string in transmitted CDP packets is not consistent across SoundPoint IP products.**
- 45716: **SoundPoint IP 450: Text is not as clear as on other phones.**
- 45835: **SoundPoint IP 450: Status Bar text is difficult to read on some backgrounds.**
- 45943: **Incorrect logic used when picking line for outgoing call in a multiple registration scenario.**
- 46068: **"Transfer On Proceeding" is not supported when server is a proxy**

- **46334: DTMF local rendering does not stop if far end holds while local digit key is pressed then far end resumes**
- **46478: EFK: Phone does not send invite when executing \$Cwaitdialtone\$**
- **46513: Dialog Event Package Content Guideline 6B (Local Identity)**
- **46514: Dialog Event Package Content Guideline 6C (Local Target)**
- **46547: SoundStation IP 7000: Warning Header Text notification does not display on phone (when configured)**
- **46550: Directed-Call-Pickup fails when SIP server is a proxy.**
- **46588: SoundStation IP 7000/HDX Integration: Info Soft key is missing in Contact Directory**
- **46738: Enhanced BLF: attendant.ringType parameter is not removed from the override file when default (silent) attendant ring type is selected**
- **46741: Enhanced BLF: The remote call appearance screen does not time out on console phone until the watched line hangs up an outgoing call**
- **46770: Microbrowser: * and # buttons do not work correctly when text input mode is set to numeric on input fields**
- **46899: Electronic hook switch: No audio during active call if answer by pressing hook switch button immediately on Jabra headset under specific scenario.**
- **47039: The line LED does not flash instead remains stable green, when an active call is kept on hold during an incoming call.**
- **47123: USB Call Recording: Missed call notification is getting displayed on the audio player screen if an incoming call is not answered during playback**
- **47207: SoundStation IP 7000/HDX Integration: When the MUTE is active it covers up the dialing fields so I cannot see what I am dialing**
- **47248: Hot dial doesn't work when lifting the handset for the second call when call.stickyAutoLineSeize="1"**
- **47300: URL dial disabled message never displayed - Failed to route to voicemail from "Message Center" tab**
- **47336: SoundStation IP 7000/HDX Integration: Received\Missed call list is showing IP address of SIP server instead of the Extension number of a call received/Missed from a SIP extension.**
- **47464: SoundPoint IP 320/330; SoundStation IP 7000: When two incoming calls are active on a phone lifting the handset or pressing the hands free key to answer the call results in the most recent call being answered even though the ring-tone is played according to the first incoming call.**
- **47535: Soft keys reset to default layout on an inbound call in some multiple call handling scenarios**

- 47566: XML API; Internal URIs: When a internal URI is executed with multiple VoUp and VoDown action uri's, the Ringer horizontal bar is not seen, only the Volume sound going UP and Down is heard.
- 47578: SoundPoint IP 320, 321, 330, 331; Corporate Directory: The 'sticky' attributes are not saved.
- 47612: BLF: Cancelling a Transfer for a call that was initiated using Directed Call Pick-Up sequence will result in incorrect caller-id display to the user.
- 47641: SoundStation IP 7000/HDX Integration: Network Link down message should stay unless phone reboot and comes up with Ethernet cable.
- 47695: SoundPoint IP 320, 321, 330, 331, 430, 450: When phone has 2 registrations, NewCall soft key is still displayed for alerting call appearance when there are max call appearances
- 47699: SoundStation IP6000; XML API; Internal URIs: Tel URI is not working properly if embedded within a couple of internal URI actions.
- 47712: SoundPoint IP 320,321,330,331: Local contact directory search does not always work correctly.
- 47724: SoundPoint IP 450: Mute icon and Call appearance counter conflict when DND is turned on and multiple call appearances are present on the phone
- 47729: On-hook dialing widget uses multi-tap behavior but is not in multi-tap mode
- 47746: NewCall soft key should not be displayed when phone holds max conference calls
- 47798: SoundStation IP 7000: Improve location of Transfer and Conference soft keys during conference setup.
- 47847: BLF: Monitoring phone stops ringing if shared line is seized while monitored line has an incoming call
- 47853: Headset memory mode active: Headset key stops blinking during incoming call after ending 1st active call.
- 47862: SoundStation IP6000 : Time and Date doesn't display during call
- 47863: Phone's HTTP server is sending some HTTP traffic in very small TCP segments
- 47916: SoundPoint IP 320, 321, 330, 331: Resume soft key is not available for 2nd call appearance after splitting conf established through Join from different shared line registrations.
- 47921: SoundPoint IP 320, 321, 330, 331: The order of call appearances is different compared with other phones after splitting conf. This discrepancy results in bringing focus to 2/3 (or 2/4) when split conf.
- 47929: Rendering special characters like " ' " will break the hyperlink style display.

- 47932: **Call widget counter (1/n) does not appear while in dial tone state. It flashes for a fraction of sec and then disappears.**
- 47951: **Transfer should have precedence over pickup of a ringing BLF line when pressing the linekey during a call transfer**
- 47953: **SoundStation IP 6000: Call info display not displayed properly when volume up/down key press.**
- 47958: **SoundStation IP 7000/HDX Integration: Unable to add more than one contact dir when Onyx is configured with no Ethernet cable connected + HDX**
- 47962: **SoundStation IP 7000/HDX Integration: Incorrect icon displayed when Redialing POTS call but there is nothing in the buffer to redial. Phone should not attempt to dial when redial buffer is empty for the call type selected.**
- 48003: **SoundStation IP 7000/HDX Integration: Phone dials POTS call as video call when dialing from idle state for a certain configuration.**
- 48011: **SoundStation IP 7000/HDX Integration: Use of the Idle Browser interferes with some display elements e.g. Mute Icon, Video/Phone Call Pop-up when connected to HDX.**
- 48019: **SoundStation IP 7000/HDX Integration: The pop-up message "Video or Phone Call?" is overwritten by idle browser**
- 48045: **Enhanced BLF: Phone does not hold the 1st call when press Dial soft key to make the 2nd call to the same called party**
- 48049: **BLF: Attendant phone does not display all remote calls on a BLF monitored line if the Monitored Phone has a call in the 'Ringing' state.**
- 48061: **Enhanced BLF: Attendant phone does not update 1/x widget when BLF monitored line has 1 or multiple incoming calls being ended**
- 48069: **U/I : SCA Barge-In: Extra softkeys are displayed on remote shared phone while viewing call appearance list by long pressing line key**
- 48071: **XML Push API: Key:Handsfree internal URI action is not executed by phone in a certain scenario.**
- 48115: **SoundStation IP 7000/HDX Integration: HDX plays ring sound after answering POTS call**
- 48131: **Call Forwarding Status Not Always Shown if multiple Call Forward Types are selected.**
- 48149: **SDP attribute truncated when first character of the value is a digit**
- 48162: **"Boot Server" status field shows incomplete or blank path if a "/" is included in the setting.**
- 48174: **Failed call may cause subsequent calls to skip URL/Number mode selection**
- 48179: **XML API; Telephony Notifications: Called Party number is shown overlapped in incoming event notification in case of IP dialed calls between unregistered phones.**

- 48209: **Cannot delete left-most character before character selection timeout**
- 48213: **XML API; Internal URI: Key:LineX should be executed only if "X" is a supported line key for that platform.**
- 48333: **USB Call Recording: USB busy indicator does not appear on main screen when recording in progress.**
- 48414: **Phone occasionally fails to act on electronic hookswitch up/down signal from Plantronics and Hydra headsets.**
- 48700: **USB call Recording: Stopping Playback through "Back" key not intuitive**
- 48745: **Corporate Directory: LDAP "Critical Extension Error 0x0c" causes CD Server not responding message from phone.**
- 48981: **SRTP 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 not tagging correct DSCP value to some packets (Trying, Ringing and OK)**
- 49106: **Entire dialed URL is not always saved in call history**
- 49251: **Update Polish XML Dictionary to include Polish characters**
- 49300: **SoundStation IP 7000/HDX Integration: Insure that DTMF tone are being sent via the dtmf start/stop Clink2 API**
- 49417: **Phone reports MOH dialog if SUBSCRIBE received while on hold**
- 49459: **Cancel doesn't work after entering hotdial digits.**
- 49461: **DND symbol(X) does not disappears after DND feature is disabled in a certain configuration.**
- 49473: **SoundPoint IP 320,321,330,331;Corporate Directory: If I use the # key to change text entry mode it should reset the Quick Search timeout timer**
- 49476: **Corporate Directory: Scrolling indicators work poorly**
- 49512: **XML: HTTP Refresh header response is not loading the specified URL on the phones after the specified amount of time has passed, in a certain situation.**
- 49516: **Hanging up handset does not terminate call in Audio or Display Diagnostics**
- 49523: **SoundPoint IP 450, SoundStation IP 7000: Asian fonts appear 'fuzzy'**
- 49548: **SoundPoint IP 320, 321, 330, 331: Edit and Delete softkeys remain after deleting last contact**
- 49572: **SoundStation IP7000; Corporate Directory: Numeric characters cannot be entered in the Quick Search entry field.**
- 49617: **Phone does not play dial tone after a hold reminder is played in certain scenarios.**

- 49619: **Call waiting beep does not play on phone when call hold reminder is set.**
- 49620: **Volume settings for Recording do not work in handsfree mode.**
- 49639: **Handsfree dial tone is interrupted by hold reminder and call waiting ring tones**
- 49641: **SoundStation IP 6000, 7000: Call info display does not display properly while changing volume.**
- 49677: **Phone does not comply with rfc4475 3.1.2.3. Negative Content-Length**
- 49685: **SoundPoint IP 320, 321, 330, 331: Cannot enter URLs with uppercase letters**
- 49692: **SoundPoint IP 450 : Seconds Colon in time does not blink for every second.**
- 49693: **ACD icon not displayed when parameter(volpProt.SIP.serverFeatureControl.cf=1) is enabled.**
- 49696: **After a long LAN outage during "Downloading new application" the phone is re-connected to the network. It gets back an IP but it does not reboot and it does not display any error message**
- 49701: **SoundStation IP 7000/HDX Integration: Phone response with "reg.1.server.1.expires = "5" setting is inconsistent**
- 49706: **SoundStation IP 7000/HDX Integration: SIP Extension display disabled after dis-connecting from HDX with HDX-Preference option**
- 49757: **SoundStation IP 7000: Phone does not display "Network Link is Down" after the cable is disconnected from a hub**
- 49758: **SoundStation IP7000 : Phone gets into a bad state and does not recover from temporarily unplugging network connection during an active call.**
- 49776: **If dir.corp.user is mis-configured, the phone does not display "Login Error"**
- 49813: **Corporate Directory: Phone displays 'Enter More Chars...' when submitting a string that returns no results in the Quick search mode.**
- 49825: **Corporate Directory: Black background for Search bar displays inconsistently on different platforms**
- 49829: **NTP Time synchronization unreliable in a particular scenario.**
- 49834: **Corporate Directory: If VLV indexing is configured and an Advanced Find yields more results than the configured 'pageSize' (Default is 64) scrolling through the entries may not work correctly.**
- 49836: **Corporate Directory: Phone flashes "Please try again" msg for 1 time if Corp Dir server is down->phone reboots up->Open Corp Dir menu**
- 49911: **Incoming ring tone not played on the phone in a certain enhanced BLF use case.**

- 49926: **SoundPoint IP 320,321,330,331: Phone auto-increments new contact's speeddial index to 100 even though the maximum entries is 99.**
- 49927: **SoundPoint IP 320,321,330,331 and VVX 1500: After an AdvFind search, exit and re-enter Corp Dir menu, phone should displays search bar as "Search:" not "Search (Filtered):"**
- 49929: **SoundStation IP 7000/HDX Integration: SoundStation IP 7000 is not displaying HDX Extension , when voice call type is set to Auto and phone is not registered to SIP server**
- 49981: **SoundStation IP 7000/HDX Integration: After reboot, 2 digits HDX extension replaces the last two digits of SIP extension and displays 4 digits(2Digits of sip+2Digits of HDX) with call type HDX.**
- 49982: **SoundPoint IP 320, 321, 330, 331: Phone doesn't reconfigure when DHCP lease expires**
- 49989: **SoundStation IP 7000: Phone is adding contact directories from call list with the existing speed dial number.**
- 49977: **SoundPoint IP 320, 321, 330, 331: Phone does not display the selected status under "MyStat" menu**
- 50090: **SoundStation IP 7000: Phone does not display Active Conference screen on Joining a remotely held SLA call without first holding the local call**
- 50099: **Consultative Transfer fails if 2nd leg is forwarding and its 302 response is handled by proxy**
- 50109: **SoundStation IP 7000/HDX Integration: Volume levels are not in Sync when Dialing a Video call**
- 50110: **SoundStation IP 7000/HDX Integration:There is no Enter number message for Video and audio calls, once the Ethernet is removed.**
- 50115: **SoundStation IP 7000/HDX Integration: The DTMF tone of the first digit is played at HDX volume instead of SoundStation IP 7000 volume.**
- 50118: **SoundStation IP 7000/HDX Integration: Dial tone volume and Hands Free volume are not in Sync.**
- 50137: **SoundStation IP 7000/HDX Integration: The volume is reset to default after the POTS call is connected if voice.volume.persists.handsfree=0**
- 50153: **Corporate Directory: Setting the Primary Attribute as 'sticky' (dir.corp.attribute.1.sticky="1") can give confusing user interface behavior.**
- 50159: **Corporate Directory: Quick search on non-null sticky primary filter missing records**
- 50189: **SIP responses missing to-tag after Phone challenges INVITE**
- 50212: **Corporate Directory: Scrolling upward for a while, phone displays entry list not sorted in order**
- 50253: **SoundStation IP 7000, Corporate Directory: When edit phone number attribute in AdvFind menu, pressing on 1/A/a sk creates Encoding sk**

- 50254: **Phone does not honor SDP sent in PRACK.**
- 50255: **SIP Reliable Provisional responses are not retransmitted.**
- 50256: **When not yet registered, random delay of 30-60 sec between registration attempts is not observed**
- 50264: **Global prefix + not present on calls made from Placed Calls list.**
- 50299: **SoundStation IP 7000, Corporate Directory: Quick search text input starts at the second multitap character instead of the first (e.g. B instead of A or E instead of D)**
- 50381: **SoundPoint IP 320, 321, 330, 331: Pressing left navigation key before character selection timeout moves cursor 2 spots**
- 50397: **SoundStation IP 7000: Phone not displaying licenses correctly in status screen**
- 50407: **Corporate Directory: When server is down with phone connecting to ldap server, do a quick search, phone displays "No entries found"**
- 50523: **SoundPoint IP 320, 321, 330, 331; Corporate Directory: Phone should display "Contact" title in View menu but it displays quick search bar with a flashing cursor**
- 50546: **When URL dialing disabled, BLIND soft key appears in the 4th soft key slot, as opposed to the 3rd slot, after pressing TRANSFER.**
- 50811: **P-Asserted ID display name should be sticky on UI call appearance and in placed call list**
- 50869: **Phone will only offer SRTP when SRTP crypto suite is selected**
- 50891: **SoundStation IP 6000,7000: Resume soft key is not displayed when the phone is put on hold on another shared line phone.**
- 50989: **Receiving a 603 Decline by a BLF monitored user does not play a reorder tone**
- 51041: **X-IdleBrowserSelectUrl: http://url is remembered by the phone even though idle page doesn't specify it.**
- 51245: **BLF state is not updated on receipt of 1st full state NOTIFY after a reboot**
- 51320: **SoundStation IP 7000/HDX Integration: "Conference in Another Video or phone call?" message is displayed in a loop for each press on "Conf" hard key.**
- 51432: **SoundStation IP 7000/HDX Integration: Conference Hard key Popup Message need to be altered or displayed appropriately**
- 51554: **Phones add an additional CRC to some 802.1X packets received on the PC port. This causes the 802.1X authentication to fail in some situations.**
- 51567: **Server based CFWD/DND sync fails on 3.1.2.0392 [NOTIFY no longer refreshing target of dialog]**

- 51605: **API: Push request will get lost if it follows another push request immediately.**
- 51631: **Phone not releasing first assigned IP address when VLAN is set via DHCP.**
- 51633: **Phone fails to play busy/reorder tone upon a refer based transfer when it gets a 603 or 486 response**
- 51644: **Some Japanese strings do not display correctly.**
- 51690: **EFK feature is used for onetouch Voicemail dialling. When using on 3.1.3 the phone appears not to honour the stickyautolinesize**
- 51718: **Phone continues to ring after the call has been answered with a certain call signaling sequence.**
- 51763: **SoundStation IP 7000/HDX Integration: When Adding video to an existing call. IP7000 shows as on Mute but Far end can hear them.**
- 51838: **Some Japanese characters are not properly displayed.**
- 52014: **In SIP 3.x.x when an IP phone picks up a transferred call in a certain scenario, the call is immediately placed on Hold instead of being connected.**
- 52017: **Web interface issue Password entry is not masked when entered (since SIP 3.0.0)**
- 52108: **Phone fails to restore destination to Asserted Identity or Remote ID after a transfer fails**

2.4.4 Configuration File Parameter Changes

This section lists the parameters that have been added/changed or deleted from the template phone1.cfg and sip.cfg files. For further description of parameters please refer to the Administrator's Guide for the SIP 3.2 Release.

Note also that the template 000000000000.cfg file has been modified in order to facilitate support for the Legacy phones and the VVX 1500 in this release.

File	Change	Attribute	Old value	New value	Description
sip	added	call.directedCallPickupMethod		"native" or "legacy"	See Administrator's Guide for SIP 3.2.0 for details
sip	added	call.parkedCallRetrieveMethod		"native" or "legacy"	See Administrator's Guide for SIP 3.2.0 for details
sip	added	call.parkedCallRetrieveString		Star code	See Administrator's Guide for SIP 3.2.0 for details
sip	added	dialplan.applyToRemoteDialing		0 or 1; Default is 0	A flag to determine if the dial plan applies to for 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
sip	added	ind.class.2.state.36.index		42	

sip	added	ind.class.2.state.37.index		43	screen layout modifications
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	
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)	
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		русский (ru-ru)	
sip	added	lcl.ml.lang.menu.16.label		Slovenski (sl-si)	
sip	added	lcl.ml.lang.menu.17.label		Español (es-es)	
sip	added	lcl.ml.lang.menu.18.label		Svenska (sv-se)	
sip	added	lcl.ml.lang.menu.2.label		Dansk (da-dk)	
sip	added	lcl.ml.lang.menu.3.label		Nederlands (nl-nl)	
sip	added	lcl.ml.lang.menu.4.label		English (en-ca)	

sip	added	lcl.ml.lang.menu.5.label		English (en-gb)	
sip	added	lcl.ml.lang.menu.6.label		English (en-us)	
sip	added	lcl.ml.lang.menu.7.label		Français (fr-fr)	
sip	added	lcl.ml.lang.menu.8.label		Deutsch (de-de)	
sip	added	lcl.ml.lang.menu.9.label		Italiano (it-it)	
sip	added	log.level.change.lldp		4	Control the logging detail level for the LLDP feature.
sip	added	mb.main.autoBackKey		1	See Administrator's Guide for SIP 3.2.0 for details
sip	changed	ramdisk.minfree	3072	3150	Minimum amount of free space that must be left after the RAM disk has been created
sip	changed	se.pat.ringer.13.name	Sample d 1		Customer ringer file names.
sip	changed	se.pat.ringer.14.name	Sample d 2		
sip	changed	se.pat.ringer.15.name	Sample d 3		
sip	changed	se.pat.ringer.16.name	Sample d 4		
sip	changed	se.pat.ringer.17.name	Sample d 5		
sip	changed	se.pat.ringer.18.name	Sample d 6		
sip	changed	se.pat.ringer.19.name	Sample d 7		
sip	changed	se.pat.ringer.20.name	Sample d 8		
sip	changed	se.pat.ringer.21.name	Sample d 9		
sip	changed	se.pat.ringer.22.name	Sample d 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 .

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
sip	added	voice.audioProfile.iLBC.jitterBufferMax		160	See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.audioProfile.iLBC.jitterBufferMin		40	See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.audioProfile.iLBC.jitterBufferShrink		500	See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.audioProfile.iLBC.payloadType		110	See Administrator’s Guide for SIP 3.2.0 for details
sip	removed	voice.audioProfile.Lin16.44.1ksps.payloadType	120		Parameter renamed.
sip	added	voice.audioProfile.Lin16.44_1ksps.payloadType		120	See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.audioProfile.Lin16.8ksps.payloadType		116	See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.iLBC.13_33kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.iLBC.15_2kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.IP_6000.iLBC.13_33kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.IP_6000.iLBC.15_2kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.IP_650.iLBC.13_33kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.IP_650.iLBC.15_2kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.IP_7000.iLBC.13_33kbps			See Administrator’s Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.IP_7000.iLBC.15_2kbps			See Administrator’s Guide for SIP 3.2.0 for details

sip	added	volpProt.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	volpProt.SDP.offer.iLBC.13_33kbps.includeMode			See Administrator's Guide for SIP 3.2.0 for details
sip	changed	volpProt.server.1.port	5060		port of a SIP server that accepts registration
sip	added	volpProt.server.2.address			
sip	added	volpProt.server.2.expires			Minimum now 10
sip	added	volpProt.server.2.expires.lineSeize		30	
sip	added	volpProt.server.2.expires.overlap			
sip	added	volpProt.server.2.lcs			
sip	added	volpProt.server.2.port			
sip	added	volpProt.server.2.register		1	
sip	added	volpProt.server.2.retryMaxCount		0	
sip	added	volpProt.server.2.retryTimeOut		0	
sip	added	volpProt.SIP.compliance.RFC3261.validate.contentLength			If set to 1, validation of the SIP header content language is enabled.
sip	added	volpProt.SIP.compliance.RFC3261.validate.uriScheme			If set to 1 or Null, validation of the SIP header URI scheme is enabled.
sip	added	volpProt.SIP.strictReplacesHeader			This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.
sip	added	volpProt.SIP.use486forReject			If set to 1 and the phone is indicating a ringing inbound call appearance, phone will transmit a 486 response to the received INVITE when the Reject soft key is pressed.
phone1	added	attendant.behaviors.display.remoteCallerID.automata		1	Flags to determine whether or
phone1	added	attendant.behaviors.display.remoteCallerID.normal		1	not remote party caller ID information is presented to the attendant.
phone1	added	attendant.behaviors.display.spontaneousCallAppearances.automata		0	Flags to determine whether or

phone1	added	attendant.behaviors.display.spontaneousCallAppearances.normal		1	not a call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource
phone1	added	attendant.resourceList.x.address		The value of x depends on the phone For IP 450: x=1-2; IP 550, IP 560: X=1-3; IP 650, IP 670: x=1-47	The user referenced by attendant.reg="" will subscribe to this URI for dialog.
phone1	added	attendant.resourceList.x.label			Text label to appear on the display adjacent to the associated line key
phone1	added	attendant.resourceList.x.type		"normal"	Type of resource being monitored.
phone1	changed	attendant.ringType		1	
phone1	added	dialplan.1.applyToTelUriDial		1	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file.
phone1	added	dialplan.2.applyToTelUriDial		1	
phone1	added	dialplan.3.applyToTelUriDial		1	
phone1	added	dialplan.4.applyToTelUriDial		1	
phone1	added	dialplan.5.applyToTelUriDial		1	
phone1	added	dialplan.6.applyToTelUriDial		1	
phone1	changed	divert.noanswer.1.timeout	60	55	Modified No Answer Timeout
phone1	changed	divert.noanswer.2.timeout	60	55	
phone1	changed	divert.noanswer.3.timeout	60	55	
phone1	changed	divert.noanswer.4.timeout	60	55	
phone1	changed	divert.noanswer.5.timeout	60	55	
phone1	changed	divert.noanswer.6.timeout	60	55	
phone1	added	reg.1.server.2.address			See Administrator's Guide for SIP 3.2.0 for details
phone1	added	reg.1.server.2.expires			
phone1	added	reg.1.server.2.expires.lineSeize			
phone1	added	reg.1.server.2.expires.overlap			
phone1	added	reg.1.server.2.lcs			
phone1	added	reg.1.server.2.port			
phone1	added	reg.1.server.2.register			
phone1	added	reg.1.server.2.retryMaxCount			
phone1	added	reg.1.server.2.retryTimeOut			
phone1	added	reg.2.musicOnHold.uri			
phone1	added	reg.2.server.1.lcs			
phone1	added	reg.2.server.2.address			
phone1	added	reg.2.server.2.expires			
phone1	added	reg.2.server.2.expires.lineSeize			
phone1	added	reg.2.server.2.expires.overlap			
phone1	added	reg.2.server.2.lcs			
phone1	added	reg.2.server.2.port			

phone1	added	reg.2.server.2.register		
phone1	added	reg.2.server.2.retryMaxCount		
phone1	added	reg.2.server.2.retryTimeOut		
phone1	added	reg.2.tcpFastFailover		
phone1	added	reg.3.musicOnHold.uri		
phone1	added	reg.3.server.1.lcs		
phone1	added	reg.3.server.2.address		
phone1	added	reg.3.server.2.expires		
phone1	added	reg.3.server.2.expires.lineSeize		
phone1	added	reg.3.server.2.expires.overlap		
phone1	added	reg.3.server.2.lcs		
phone1	added	reg.3.server.2.port		
phone1	added	reg.3.server.2.register		
phone1	added	reg.3.server.2.retryMaxCount		
phone1	added	reg.3.server.2.retryTimeOut		
phone1	added	reg.3.tcpFastFailover		
phone1	added	reg.4.musicOnHold.uri		
phone1	added	reg.4.server.1.lcs		
phone1	added	reg.4.server.2.address		
phone1	added	reg.4.server.2.expires		
phone1	added	reg.4.server.2.expires.lineSeize		
phone1	added	reg.4.server.2.expires.overlap		
phone1	added	reg.4.server.2.lcs		
phone1	added	reg.4.server.2.port		
phone1	added	reg.4.server.2.register		
phone1	added	reg.4.server.2.retryMaxCount		
phone1	added	reg.4.server.2.retryTimeOut		
phone1	added	reg.4.tcpFastFailover		
phone1	added	reg.5.musicOnHold.uri		
phone1	added	reg.5.server.1.lcs		
phone1	added	reg.5.server.2.address		
phone1	added	reg.5.server.2.expires		
phone1	added	reg.5.server.2.expires.lineSeize		
phone1	added	reg.5.server.2.expires.overlap		
phone1	added	reg.5.server.2.lcs		
phone1	added	reg.5.server.2.port		
phone1	added	reg.5.server.2.register		
phone1	added	reg.5.server.2.retryMaxCount		
phone1	added	reg.5.server.2.retryTimeOut		
phone1	added	reg.5.tcpFastFailover		
phone1	added	reg.6.musicOnHold.uri		
phone1	added	reg.6.server.1.lcs		
phone1	added	reg.6.server.2.address		
phone1	added	reg.6.server.2.expires		
phone1	added	reg.6.server.2.expires.lineSeize		
phone1	added	reg.6.server.2.expires.overlap		
phone1	added	reg.6.server.2.lcs		
phone1	added	reg.6.server.2.port		
phone1	added	reg.6.server.2.register		

phone1	added	reg.6.server.2.retryMaxCount			
phone1	added	reg.6.server.2.retryTimeOut			
phone1	added	reg.6.tcpFastFailover			

2.5 Version 3.1.3 C

2.5.1 Added or Changed Features

- Add Support for the SoundPoint IP 321 and 331 products.

2.5.2 Removed Features

None.

2.5.3 Corrections

None.

2.5.4 Configuration File Parameter Changes

None.

2.6 Version 3.1.3 B

2.6.1 Added or Changed Features

None.

2.6.2 Removed Features

None.

2.6.3 Corrections

- 50103 **SoundStation IP 7000/HDX: Volume change before dialing is discarded after the POTS call is established**
- 50104: **Corporate Directory: If ViewPersistency is enabled, Scrolling down the list of results from an Advanced Find query, after exit ->re-enter->scroll up, attribute filter in previous AdvFind is not maintained**
- 50117: **SoundStation IP 7000/HDX: Incoming POTS call resets the Ringer volume.**

2.6.4 Configuration File Parameter Changes

None.

2.7 Version 3.1.3 (Limited Release – Version 3.1.3.0336)

2.7.1 Added or Changed Features

- 45869: **Corporate Directory: Add support for LDAP directory queries using VLV Indexing.**
- 47179: **Extend fast-fail over mechanism to transactions initiated over TCP transport**
- 47493: **Corporate Directory: Improvements to User Interface. See Technical Bulletin TB 41137 for details.**
- 47495: **Corporate Directory: Screen Idle Timeout needs to be reset whilst a Corporate Directory search is in process**
- 48183: **VVX 1500: Add network jitter computation and reporting for video packet channels**
- 48467: **VVX 1500: Touching the LCD screen at any location should wake the LCD from the "dim" state to full brightness.**
- 48484: **IP7000/HDX: Allow Configuration control of the Dialtone sound level when adding a POTS call to an existing Video call.**
- 48854: **Change default for parameter mb.main.idleTimeout from 20 to 40 seconds.**
- 48567: **When DND/CF Sync is enabled the phone should not Forward or deny any calls that it receives**

2.7.2 Removed Features

- 47376: **Remove License Requirement on uaCSTA feature**

2.7.3 Corrections

- 23634: **SoundPoint IP 320/330, 430, 450, 550, 560, 650, 670, SoundStation IP 4000, VVX 1500: Packet stats jitter should be computed exactly as shown in RFC3550. Issue remains on SoundPoint IP 301, 501, 600, 601 and SoundStation IP 6000, 7000 phones.**
- 43517: **REFER-based 'click-to-dial' causes errors and may cause a phone reboot.**
- 44973 **SoundPoint IP 301: Line label disappears after SCA phone views remote shared line's call appearance list and the view screen times out**
- 46795: **SoundPoint IP 450: Colon in time display does not blink**
- 46480: **SoundPoint IP 301, 501, 600, 601: Loud static 'pop' and 'hiss' may be heard when receiving audio using G.729AB as the codec with VAD enabled.**
- 46613: **SoundPoint IP 301, 501, 600, 601; SoundStation IP 4000: 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 are using the incorrect "@domain" in Signalling in certain scenarios.**
- 47492: **SoundPoint IP501: Message LED flashes continuously after receiving blind transfer from a 'centralized conference' leg**
- 47609: **SoundPoint IP 450: Phone is not able to display more than two status notifications if server controlled ACD is enabled**
- 47878: **CLONE -Phone generating malformed XML with ACD Login/Logout for some parameters.**
- 47911: **Forked INVITE back to caller fails to connect to voicemail on call timeout**
- 47915: **Phone ignores 401 challenge after responding to 407 in a certain call scenario.**
- 47960: **SoundStation IP 7000/HDX: Redialing POTS call from placed call list dials as video call if the call was dialed from contact directory.**
- 47964: **SoundStation IP 7000/HDX: Phone displays wrong icon when conferencing and adding a POTS call**
- 48002: **SoundStation IP 7000/HDX: Speaker volume drops to two bars after making a video call**
- 48039: **BLF: Phone plays the 'Attendant Ring-Tone' instead of the 'Regular Ring-Tone' if the 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 not loud enough for low level signals**
- 48076: **BLF: Attendant phone does not automatically get placed on Hold if a BLF or speed dial key is used to dial whilst an active call is in process on the attendant phone. Only occurs if call.stickyAutoLineSeize="1".**
- 48123: **SoundStation IP 4000/6000/7000: Clock time does not increment while a call is active if the idle browser is enabled.**
- 48171: **De-registration attempts do not authenticate and so fail to de-register some lines.**
- 48280: **SoundStation IP 6000, 7000: When using TFTP or FTP as the provisioning Server Type, phone does not save directory entries locally when TFTP or FTP server is not available.**
- 48385: **VVX 1500: SSRC header field is not correct for RFC2833 packets.**
- 48462: **SoundPoint IP 501: Ring LED indicator continues flashing even when the call is answered if an INVITE with "sendonly" SDP is received by the phone.**
- 48485: **VVX 1500: Audio call recording during video calls may fail with certain USB drives.**

- 48577: **SoundPoint IP 430: Default headset gains not correctly set which may result in poor audio quality with certain headsets.**
- 48591: **VVX 1500: Click-to-Hold does not work correctly.**
- 48605: **call.stickyAutoLineSeize is not applied correctly when a line is ringing and SilentRing is selected**
- 48615: **If call.StickyAutoLineSeize="1": Transfer fails if attempted whilst 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 not audibly alert the user for the incoming call**
- 48668: **401 Authentication challenge to a VQMon PUBLISH may cause 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 **In SIP 3.1.2 the MWI NOTIFY must have the message summary for the MWI LED to be lit.**
- 48697: **An incoming call without Caller ID Name but with Caller ID Number is matched with the first local contact that has Name blank.**
- 48699: **TelURI doesn't process "tel://*50"**
- 48756: **Unknown Party displayed on caller ID when using a shared line and only number is provided, no name.**
- 48778: **VVX 1500: Motion detection is not starting after a video conference call.**
- 48858: **BLF attendants monitoring both initiator and recipient get confused about state 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: **IP7000/HDX: When placing a Video conference call with 8 legs, the UI does not show the two last call appearances.**
- 48959: **SoundPoint IP 430: After upgrading to SIP 3.1.2, the time portion of date and time cut off when using a custom Idle Display.**
- 48985: **The phone may reboot if you receive or miss a call while looking at information about a previously received or missed call.**
- 49013: **DND X icon does not update next to line key when BroadWorks ACD is enabled.**
- 49068: **Receiving an OPTIONS message results in a spurious dialog Notification being sent**
- 49129: **VVX1500 U/I not showing updates while soft keys, physical buttons do work.**

- **49181: VVX 1500: When using the idle micro-browser the phone display sometimes /freezes'.**
- **49201: Receiving Update with confirmed SDP before 200 ok caused 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 SoundPoint IP601: One-way audio when changing termination mode during call waiting when callWaiting.ring="ring" is set.**
- **49256: VVX 1500: If the micro-browser tries to access a URL longer than 54 characters the phone may re-boot or lock-up.**
- **49281: IP7000/HDX integration: When the IP7000 is used to adjust the volume this may cause the HDX volume level to be reduced to 0.**
- **49287: SUBSCRIBE terminate causes BLF labels to disappear for 2~4 seconds**
- **49323: VVX 1500 reboots after lifting handset while in an empty call list**
- **49402: Race condition when you seize one SCA line and then resume a held call on another SCA before the line seize completes**
- **49533: Incorrect UDP checksum in DHCP Decline message**
- **49599: BLF: Attendant phone does not update 1/x widget when BLF monitored line has 1 or multiple incoming calls being ended**
- **49810: VVX 1500 seizes line key 1 when "call.stickyAutoLineSeize=1" and the speed dial key is used to dial.**

2.7.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.SIP.serverFeatureControl.localProcessing.dnd	If set to 0 and voIpProt.SIP.serverFeatureControl.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	volpProt.SIP.serverFeatureControl.localProcessing.cf	If set to 0 and voIpProt.SIP.serverFeatureControl.cf = "1", the phone will not perform local Call Forward behavior. If set to 1 or Null, the phone will perform local Call Forward behavior on all calls received.
sip	added	volpProt.SIP.tcpFastFailover	If set to 1, failover occurs based on the values of reg.x.server.y.retryMaxCount and voIpProt.server.x.retryTimeOut. If set to 0, use old behavior. If reg.x.tcpFastFailover is Null, this attribute is checked. If voIpProt.SIP.tcpFastFailover 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.
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	aded	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

2.8 Version 3.1.2 B

2.8.1 Added or Changed Features

- Add Support for the VVX 1500 product.

2.8.2 Removed Features

None.

2.8.3 Corrections

None.

2.8.4 Configuration File Parameter Changes

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

2.9 Version 3.1.2

2.9.1 Added or Changed Features

- 34787: Add Support for ACD Call Center Agent functionality using the 'Feature Synchronization' method. See *Technical Bulletin 34787* 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* 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 (see Section 4) 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.

2.9.2 Removed Features

N/A

2.9.3 Corrections

- 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: **Ringling 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 softkey 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 secs later.**
- 46610: **Errors in Polish language dictionary**
- 46737: **BLF: Softkeys & Call appearance disappears on the console phone in a certain scenario using a shared line.**
- 46757: **XML API: Issue with order of call appearances on a single line registration and single line key**
- 46763: **XML API: URI softkey:exit does not work when executed from softkey 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.**

2.9.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
phone1	added	acd.reg	See Technical Bulletin34787 for details
phone1	added	acd.stateAtSignIn	
sip	added	volpProt.SIP.acd.signalingMethod	
sip	added	volpProt.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 recording, call media playback, USB I/O respectively.
sip	added	log.level.change.cmp	
sip	added	log.level.change.usbio	
sip	added	prov.quickSetup.enabled	See Technical Bulletin 45460 for details
sip	added	pnet.hdx.ext	HDX Extension Number. For HDX/IP 7000 integration

2.10 Version 3.1.1 B

2.10.1 Added or Changed Features

None.

2.10.2 Removed Features

None.

2.10.3 Corrections

- 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.**

2.10.4 Configuration File Parameter Changes

2.11 Version 3.1.1

2.11.1 Added or Changed 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**

2.11.2 Removed Features

N/A

2.11.3 Corrections

- **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 a 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.**

2.11.4 Configuration File Parameter Changes

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

2.12 Version 3.1.0 C

2.12.1 Added or Changed Features

- Add Support for the SoundPoint IP 450 product.

2.12.2 Removed Features

None.

2.12.3 Corrections

None.

2.12.4 Configuration File Parameter Changes

.cfg 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 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 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 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.

2.13 Version 3.1.0 B

2.13.1 Added or Changed Features

None.

2.13.2 Removed Features

None.

2.13.3 Corrections

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

2.13.4 Configuration File Parameter Changes

None.

2.14 Version 3.1.0 (Limited Distribution; build-id 3.1.0.0073)

This version should be replaced by 3.1.0RevB

2.14.1 Added or Changed 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 Tech Bulletin TB35704 for details on managing the memory usage on phones.**

2.14.2 Removed Features

N/A

2.14.3 Corrections

- 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 are crashing if we 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: **[Corporate Directory] 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' softkey are confusing**
- 39022: **Transfer and Conference softkeys are still available on IP650/IP550/IP301/IP4000 after maximum number of outgoing calls are made from 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: **[Presence]: 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 duplicate of parent menu screen.**
- 40328: **Phone hanging on HTTP PUT with authentication**
- 40399: **Phones generates multiple SOA queries and eventually locks 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: **[Corporate Directory] The scroll down bar is still available even if corporate directory list is accessed to the end.**
- 40561: **[Presence] Backspace or "<<" softkey is not available on Add Buddy Page for IP 4000 and IP 6000 phones.**

- 40562: **[Presence]** 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 softkeys 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 crash**
- 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.**

2.14.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.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	volpProt.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	volpProt.SIP.lineSeize.retries	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA).
sip	added	volpProt.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	volpProt.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	volpProt.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	volpProt.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	volpProt.SIP.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. If reg.x.musicOnHold is set to Null, this attribute is checked.
sip	added	lcl.ml.lang.tags.x	The format is: <ul style="list-style-type: none"> • 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.
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.
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.

sip	added	softkey.x.label	<p>This is the text displayed with the soft key.</p> <p>If set to Null, the label to display is determined as follows:</p> <ul style="list-style-type: none"> • 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	<p>If set to 0 or Null, the soft key is disabled.</p> <p>If set to 1, the soft key is enabled.</p>
sip	added	softkey.x.precede	<p>If set to 0 or Null, the soft key replaces any empty space from the leftmost position.</p> <p>If set to 1, the soft key is displayed before the first standard soft key.</p>
sip	added	softkey.x.use.idle	<p>If set to 0 or Null, the soft key is not displayed in the idle state.</p> <p>If set to 1, the soft key is displayed in the idle state.</p>
sip	added	softkey.x.use.active	<p>If set to 0 or Null, the soft key is not displayed in the active call state.</p> <p>If set to 1, the soft key is displayed in the active call state.</p>
sip	added	softkey.x.use.alerting	<p>If set to 0 or Null, the soft key is not displayed in the alerting state.</p> <p>If set to 1, the soft key is displayed in the alerting state.</p>
sip	added	softkey.x.use.dialtone	<p>If set to 0 or Null, the soft key is not displayed in the dialtone state.</p> <p>If set to 1, the soft key is displayed in the dialtone state.</p>
sip	added	softkey.x.use.proceeding	<p>If set to 0 or Null, the soft key is not displayed in the proceeding state.</p> <p>If set to 1, the soft key is displayed in the proceeding state.</p>
sip	added	softkey.x.use.setup	<p>If set to 0 or Null, the soft key is not displayed in the setup state.</p> <p>If set to 1, the soft key is displayed in the setup state.</p>

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.
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: <ul style="list-style-type: none"> • 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: <ul style="list-style-type: none"> • 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.
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, volpProt.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.

2.15 Version 3.0.4

Note that Versio 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.

2.15.1 Added or Changed 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.**

2.15.2 Removed Features

N/A

2.15.3 Corrections

- 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**

2.15.4 Configuration File Parameter Changes

.cfg 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'

2.16 Version 3.0.3 B

Change made applies to the SoundStation IP 7000 product only.

2.16.1 Added or Changed Features

None.

2.16.2 Removed Features

None.

2.16.3 Corrections

- 41974: **SoundStation IP 7000 occasionally reboots when the idle browser is enabled**

2.16.4 Configuration File Parameter Changes

None.

2.17 Version 3.0.3

2.17.1 Added or Changed Features

- 39423: **Change default boot config and packaged sip.cfg value for parameter voice.vad.signalAnnexB**
- 40385: **Add config parameters volpProt.SIP.strictLineSeize, reg.x.strictLineSeize and volpProt.SIP.lineSeize.retries**
- 40387: **SIP stack will use config parameter volpProt.SIP.strictLineSeize and volpProt.SIP.lineSeize.retries to make fault-tolerant behavior optional**
- 40447: **Add a User Option to Restart the phone**

2.17.2 Removed Features

None

2.17.3 Corrections

- 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**

2.17.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.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	volpProt.SIP.strictLineSeize	If set to "1", forces phone to wait for 200 OK when receiving a TRYING notify.
sip	added	volIPProt.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, volpProt.SIP.strictLineSeize is checked. This parameter takes precedence.

2.18 Version 3.0.2 C

2.18.1 Added or Changed Features

None.

2.18.2 Removed Features

None.

2.18.3 Corrections

- 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.**

2.18.4 Configuration File Parameter Changes

None.

2.19 Version 3.0.2 B (Limited Release – build-id 3.0.2.0917)

2.19.1 Added or Changed 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 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**

2.19.2 Removed Features

None.

2.19.3 Corrections

- **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 echo'd to far end caller.**
- **40804: When new call arrives while user is in the USB Recording 'play' screen but not playing audio, incorrect softkeys 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**

2.19.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description								
sip	added	voice.codecPref.IP_(6 7)000.*	Codec support for IP 6000 and IP 7000.								
sip	added	voice.gain.(r t)x.analog.*.IP_(6 7)000	Gain levels for IP 6000 and IP 7000.								
sip	added	voice.gain.(r t)x.analog.*.IP_6000	Gain levels for IP 6000.								
sip	added	voice.(r t)xEq.hf.IP_(6 7)000.(pre post)Filter.enable	Prefilter and postfilter enable for IP 6000 and IP 7000.								
sip	changed	dir.corp.backGroundSync	Changed from 1 to 0, disabling background sync.								
sip	changed	dir.corp.backGroundSync.period	Changed value from 43200 (12 hours) to 86400 (24 hours).								
sip	removed	bg.ranges									
sip	changed	bg.color.selection	<p>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.</p> <table border="1"> <thead> <tr> <th>Index</th> <th>Type</th> </tr> </thead> <tbody> <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> </tbody> </table>	Index	Type	1	Predefined backgrounds	2	Solid patterns	3	User-defined bitmaps
Index	Type										
1	Predefined backgrounds										
2	Solid patterns										
3	User-defined bitmaps										
sip	added	bg.hiRes.color.pat.solid.*(name red green blue)	Defines the name and colour of solid backgrounds.								
sip	added	bg.hiRes.color.bm.*(em.)?name	Defines colour backgrounds for the phone's display and the expansion modules' displays (em).								
sip	added	button.color.selection.*.*.modify	<p>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.</p> <p>The value comprises a transform method, and parameters for the transform. Two transforms are supported – rgbHiLo 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.</p>								

sip	added	bg.hiRes.gray.(pr bm).*.adj	<p>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.</p> <p>The value is the number of steps to brighten the image (negative values darken the image). Each step is 1/16th of full scale.</p>
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.
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.

2.20 Version 3.0.1RevB

2.20.1 Added or Changed Features

None

2.20.2 Removed Features

None

2.20.3 Corrections

- 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”.**

2.21 Version 3.0.1 (Limited Distribution – build-id 3.0.1.0032)

2.21.1 Added or Changed Features

- 40475: Set VLAN Filtering to 'Off' by default
- 41025: Set default Corporate Directory background re-sync period to 12 hours

2.21.2 Removed 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.

2.21.3 Corrections

- 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

2.21.4 Configuration File Parameter Changes

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

Table 2-1

2.22 Version 3.0.0

** Indicates a feature that requires a license-key to be enabled.

2.22.1 Added or Changed 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-rtcpvxr 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)**

2.22.2 Removed Features

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

2.22.3 Corrections

- 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 volpProt.SIP.serverFeatureControl.cf=1 or volpProt.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**

2.22.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.SDP. useLegacyPayloadTypeNegotiation	
sip	added	volpProt.SIP.csta	Enables uaCSTA.
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 conferencing.
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"
phone1	added	reg.x.bargelnEnabled	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.

Table 2-2

2.23 Version 2.2.2

2.23.1 Added or Changed 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.**

2.23.2 Removed Features

- **38813: Remove 1000 half duplex as a valid ethernet configuration.**

2.23.3 Corrections

- **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.**

2.23.4 Configuration File Parameter Changes

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

2.24 Version 2.2.1 (Limited Release)

2.24.1 Added or Changed 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**

2.24.2 Removed Features

None.

2.24.3 Corrections

- **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**

2.24.4 Configuration File Parameter Changes

.cfg 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.

2.25 Version 2.2.0

2.25.1 Added or Changed 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"**

2.25.2 Removed Features

- 36079: **Removed support for the SoundPoint IP 300 and 500 phones**

2.25.3 Corrections

- 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 crash 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 crash 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 crash**
- 32179: **When SAS-VP provisioning is used, the boot server password is visible in the application log file**
- 32816: **Phone may crash 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 crash 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 volpProt.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 crash 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 crash**
- 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

2.25.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.SIP.csta	Not currently used, will be used in a future release.
sip	added	volpProt.SIP.serverFeatureControl.dnd	See Administrator's Guide for SIP 2.2.0 for details
sip	added	volpProt.SIP.serverFeatureControl.conf	See Administrator's Guide for SIP 2.2.0 for details
sip	added	up.toneControl.bass	Not currently used, will be used in a future release.
sip	added	up.toneControl.treble	Not currently used, will be used in a future release.
sip	added	up.audioSetup.auxInput	Not currently used, will be used in a future release.
sip	added	up.audioSetup.auxOutput	Not currently used, will be used in a future release.
sip	added	up.idleTimeout	See Administrator's Guide for SIP 2.2.0 for details
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
sip	added	voice.codecPref.IP_7000.xxx	Not currently used, will be used in a future release.
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 <u>must be updated</u> . 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
sip	added	call.disableAutoResumeCentralConference	Not currently used, will be used in a future release.
sip	added	call.singleKeyPressConference	Not currently used, will be used in a future release.

.cfg File	Action	Parameter	Description
sip	added	call.transfer.blindPreferred	See Administrator's Guide for SIP 2.2.0 for details
Sip	added	call.cellPhoneAutoBridging	Not currently used, will be used in a future release.
Sip	added	bitmap.IP_7000.xxx	Not currently used, will be used in a future release.
Sip	added	log.level.change.srtp	See Administrator's Guide for SIP 2.2.0 for details
Sip	added	log.level.change.clink log.level.change.pnetm log.level.change.peer	Not currently used, will be used in a future release.
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 sec.srtp.sessionParams.IP_4000.noEncrypRTCP.offer sec.srtp.sessionParams.IP_4000.noEncrypRTCP.require sec.srtp.leg.allowLocalConf	See Technical Bulletin 25751 for details.

.cfg File	Action	Parameter	Description
sip	added	license.polling.time	See Administrator's Guide for SIP 2.2.0 for details
sip	added	feature.16.name feature.16.enabled	Not currently used, will be used in a future release.
sip	added	mb.main.idleTimeout	See Administrator's Guide for SIP 2.2.0 for details
sip	added	mb.main.statusbar	See Administrator's Guide for SIP 2.2.0 for details
sip	added	pnet.role	Not currently used, will be used in a future release.
sip	changed	tone.chord.ringer.46.offDur="200" to "0" tone.chord.ringer.46.repeat="2" to "1"	
sip	changed	se.pat.ringer.12.inst.1.type="silence" to "chord" se.pat.ringer.12.inst.1.value="100" to "46" se.pat.ringer.12.inst.2.type="chord" to "silence" se.pat.ringer.12.inst.2.value="46" to "200" se.pat.ringer.12.inst.3.type="silence" to "chord" se.pat.ringer.12.inst.3.value="2000" to "46" se.pat.ringer.12.inst.4.type="branch" to "silence" se.pat.ringer.12.inst.4.value="-2" to "2000"	Note: also added se.pat.ringer.12.inst.5.type="branch" and se.pat.ringer.12.inst.5.value="-4"
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"	

.cfg File	Action	Parameter	Description
sip	removed	ind.anim.xxx parameters from CTX_CUSTOM1 to CTX_CUSTOM8 and CTX_UNASSIGNED for all platforms	These parameters were not used.
sip	removed	usb.enable usb.bulkDrive.enable usb.bulkDrive.name	These parameters were not used.
phone1	added	reg.x.csta	Not currently used, will be used in a future release.
phone1	added	reg.x.serverFeatureControl.dnd reg.x.serverFeatureControl.cf	See Administrator's Guide for SIP 2.2.0 for details
phone1	added	call.missedCallTracking.x.enabled	See Administrator's Guide for SIP 2.2.0 for details
phone1	added	call.callWaiting.ring	See Administrator's Guide for SIP 2.2.0 for details
000000000000	added	LICENSE_DIRECTORY	See Administrator's Guide for SIP 2.2.0 for details
000000000000	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
000000000000	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

2.26 Version 2.1.2

2.26.1 Added or Changed 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 volpProt.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**

2.26.2 Removed Features

None.

2.26.3 Corrections

- 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 phone 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 crash 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**

2.26.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.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.
phone1	added	reg.x.auth.optimizedInFailover	If this parameter is set, it overrides the global volpProt.SIP.authOptimizedInFailover parameter. x is the registration index. See the description for volpProt.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.

2.27 Version 2.1.1 C

2.27.1 Added or Changed 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**

2.27.2 Removed Features

None.

2.27.3 Corrections

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

2.27.4 Configuration File Parameter Changes

None.

2.28 Version 2.1.1

2.28.1 Added or Changed 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**

2.28.2 Removed Features

None.

2.28.3 Corrections

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 `volpProt.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**

2.28.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	volpProt.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
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)

.cfg 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.

2.29 Version 2.1.0

2.29.1 Added or Changed 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**
Example showing format of user agent in HTTP GET's previously:
User-Agent: Polycom-Microbrowser/1.0 (SIP/2.0.2.0060; SoundPoint IP PolycomSoundPointIP-SPIP_650) libcurl/7.12.1\r\n
Example showing format of user agent in HTTP GET's now (with security sec.tagSerialNo set to 1):
User-Agent: Microbrowser/1.1 PolycomSoundPointIP-SPIP_430-UA/2.1.0.2643 (SN:0004f210013a)
- 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**

2.29.2 Removed Features

None.

2.29.3 Corrections

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 softkey 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

2.29.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
phone1	added	reg.x.server.y.lcs	Refer to Technical Bulletin 5844.
phone1	added	dialplan.x.applyToUserSend="1" dialplan.x.applyToUserDial="1" dialplan.x.applyToCallListDial="0" dialplan.x.applyToDirectoryDial="0"	Refer to Technical Bulletin 11572.
phone1	added	reg.x.server.y.transport and reg.x.outboundProxy.transport	Added “TCPOnly” as a possible value for these existing parameters.
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	volpProt.server.1.lcs	Refer to Technical Bulletin 5844.

.cfg File	Action	Parameter	Description
sip	added	volpProt.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 volpProt.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"	Refer to Technical Bulletin 11572.
sip	changed	tcplpApp.snmp.daylightSavings.start.month="4" to "3"	Changes to support new daylight savings time rules.
sip	changed	tcplpApp.snmp.daylightSavings.start.date="1" to "8"	
sip	changed	tcplpApp.snmp.daylightSavings.stop.month="10" to "11"	
sip	changed	tcplpApp.snmp.daylightSavings.stop.dayOfWeek.lastInMonth="1" to "0"	
sip	added	call.stickyAutoLineSeize.onHookDialing	Refer to Administrator's Guide Addendum for SIP 2.1.
sip	changed	voice.gain.rx.digital.chassis.IP_650="-9" to "6"	Gain changes required to match new 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	volpProt.server.x.transport and volpProt.SIP.outboundProxy.transport	Added "TCPOnly" as a possible value for these existing parameters.

3. Outstanding Issues

The following issues will be fixed in a subsequent release.

- **24805: Cannot answer an incoming call while directory is being saved**
Workaround: None.
- **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**
Workaround: None.
- **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.
- **34743: A phone may freeze when it receives a check-sync if the resources on the phone are heavily used by downloaded wave files or large or 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.
- **37984: Enabling the Idle bit-map on SoundPoint IP330/320 phones causes the Line Key labels and 'dialed digits' to be invisible.**
Workaround: Do not use the idle bit-map on 330/320 phones; i.e. set ind.idleDisplay.enabled="0" for 330/320 phones
- **39001: Difficulties with phone operation due to memory limitations may be experienced if phone directories larger than 50Kbytes are used with SoundPoint IP 330, 330, 430 phones**
Workaround: Keep the local contact directory to less than 50kbytes in size.
- **39630: Using SoundPoint IP 330/320 phone with LCS2005; Blocking a roaming buddy from the Privacy list also prevents the user from viewing the 'Blocked' buddy's status**
Workaround: Do not block user's from viewing your status if you wish to view their's

- **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.
- **44478: VVX 1500: Configurable soft key feature does not work.**
Workaround: None.
- **44764: VVX 1500: SRTP processing may cause performance degradation with certain video/audio codec combinations.**
Workaround: If SRTP is being used limit the video bit rate to 384kbps.
- **45188: SoundPoint IP 320, 330 and 430 phones are low on memory. This affects the maximum size of functions such as ring-tones that may be used on these phones.**
Workaround: See Technical Bulletin TB 35704 for information on managing the memory resources on SoundPoint IP/SoundStation IP phones.
- **45247: SoundPoint IP 430 may re-boot when browsing Microbrowser pages if other functions requiring internal memory are heavily used.**
Workaround: See Technical Bulletin TB 35704 for information on managing the memory resources on SoundPoint IP/SoundStation IP phones.
- **47651: SoundStation IP7000/HDX: URL Dialing must be enabled in order to place calls.**
Workaround: None
- **47827: VQM: Soundpoint IP uses incorrect units for Jitter in SIP PUBLISH VQSession Report**
Workaround: None
- **48463: VVX 1500: Cannot view JPEG images with .jpe or .jfif extensions are used.**
Workaround: Ensure that JPEG images use .jpg extension for the name.
- **48905: SoundStation IP 6000/7000: Packet Statistics, Jitter parameter is not correctly computed as per RFC3550.**
Workaround: None.
- **51904: VVX 1500; Video Interop.; If an HDX (Release 2.5.x and maybe 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: SoundStation IP 7000: daisy Chained phone sometimes gets ‘stuck’ during software upgrade.**
Workaround: Pressing any key on the phone will continue the upgrade.
- **52142: VVX 1500; Video Interop.; Video connections with CounterPath Eyebeam client do not work if H.263-1998 codec is selected. Experienced with Eyebeam version 1.5.19.5 build 52345.**
Workaround: use a different codec. Try other versions of Eyebeam client (some do work okay).
- **52592: SoundStation IP6000: Phone fails to provision if using the combined sip.ld file and a TFTP provisioning server that does not support the ‘bulksize’ option.**
Workaround: Either use the ‘split’ image for the SoundStation IP 6000 or use a TFTP server that supports the ‘bulksize’ option.
- **52782: VVX 1500; Video Interop.; Video issues 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: VVX 1500; Video Interop.; MGC50: H.264 calls to an HDX9002 device using an MGC 50 Gateway using H.320 result in lip sync. issues.**
Workaround: Set the call for transcoding on the MGC.
- **54027: SRTP: 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: SRTP: Key Changes do not function correctly when multiple crypto suites are enabled.**
Workaround: Configure a single crypto suite on the phone.
- **54219: SoundPoint IP 560/670: Phones do not establish link when connected to some switches when both phone and switch are configured for 100Mbps/Full Duplex.**
Workaround: Configure the phone or switch for Auto negotiation.
- **54292: VVX 1500; Video Interop.: Status menu displays that phone is registered to the primary gatekeeper even though it has registered with the alternate gatekeeper.**
Workaround: None – does not affect unit operation.
- **54321: VVX 1500; Video Interop.: VVX 1500 does not receive video (does receive audio) when calls are initiated from a Tandberg C20 (running 2.0.0.191232) device using SIP.**
Workaround: None.
- **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: VVX 1500; Video Interop.: VVX 1500 transmits H.264 QCIF video to Tandberg MXP in H.323 calls.**
Workaround: Set the video bit rate on the VVX 1500 to 512kbps to avoid the issue.

- **54834: VVX 1500; Video Interop.: 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: VVX 1500; Video Interop.; Tandberg: H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway using encrypted media (offered but not required) results in distorted audio and no video on the VVX 1500.**
Workaround: Configure system for encryption required.
- **54977: VVX 1500; Video Interop.; Tandberg: H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway result in lip sync. issues.**
Workaround: None.
- **55287: Phone drops the incorrect call if the user selects (on the phone UI) a ‘held call’ and then attempts to terminate the active call (e.g. by placing the handset on hook).**
Workaround: Ensure that the active call is ‘in focus’ when terminating the call.
- **55415: Phone allows the user to enter more characters than it is capable of saving in Contact Directory fields. Introduced in SIP 3.2.0.**
Workaround: None.
- **55477: SRTP Key renewal does not occur during local conference calls.**
Workaround: None.

4. Reference Documents

- *Administrator’s Guide – SoundPoint IP SIP – Version 3.2.0*
- *White paper – Configuration File Management on SoundPoint IP Phones – available from*
http://www.polycom.com/common/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf
- Technical Bulletins and Quick Tips (including the following that are new or updated relating to this release: 39358, 52609) – may be obtained from the Polycom Support web-site at:
http://www.polycom.com/usa/en/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html
- Several Phone user Guides have been updated for the 3.1.X releases, and users should download updated copies from the Polycom Product Support Pages as needed:
http://www.polycom.com/usa/en/support/voice/soundpoint_ip/soundpoint_ip.html
- SoundStation IP 7000 HDX Integration Overview, available from
http://www.polycom.com/global/documents/support/sales_marketing/products/voice/soundstation_ip_7000_hdx_integration_overview.pdf
- SoundStation IP 7000/HDX Integration Guide, available from:
http://www.polycom.com/global/documents/support/setup_maintenance/products/voice/ssip_7000_HDX_Integrat_Guide_English_RevA.pdf