Release Notes

SIP Application

SoundPoint® IP, SoundStation® IP

Version 3.1.8

March 2012

Part Number 3804-11530-318

Table of Contents

1. General

These release notes apply to version 3.1.8 of the SoundPoint IP/SoundStation IP SIP software. This release is a patch release applicable to ‘Legacy’ products that are not supported in the SIP 3.2.x and above. The phone models to which this release applies are listed in Section 1.3.

1.1 Important Notes

When deploying this release in environments that include a combination of Legacy and other phones, it is essential that the configuration files used by the phones are properly matched based on the software version. Details on how this is achieved may be found in Technical Bulletin 35311.

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 Family of Products (Desktop Phones)

Feature	IP 301	IP 501	IP 600/601	IP 4000
VQMon	No	Productivity License	Productivity License	Productivity License
LDAP Directory	Yes	Yes	Yes	Yes
Call Recording	No	No	No	No
Conference Management	No	No	No	No
4-way local conference	No	No	No	No
Electronic Hookswitch	No	No	No	No
Enhanced Feature Keys	Yes	Yes	Yes	No
Customizable UI Background	No	No	No	No
Local SRTP Conference	No	No	No	Yes
Asian Language	No	No	No	Yes
Configurable Soft-Keys	Yes	Yes	Yes	No
XML API	No	Yes	No	Yes

Enhanced BLF	No	Yes	Yes	No
Warning Field Display	No	Yes	Yes	Yes

Productivity License – licensed as part of the Productivity Suite

1.3 System Requirements

Platform	**BootROM version**
SoundPoint IP 301	2.6.1 or greater
SoundPoint IP 501	2.6.1 or greater
SoundPoint IP 600	2.6.1 or greater
SoundPoint IP 601	3.1.0 or greater
SoundStation IP 4000	3.1.2 or greater

For details on historical software version support by platform refer to the Polycom® UC Software/Polycom SIP Software Release Matrix.

1.4 Distribution Files

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

The current build ID for the sip.ld **SIP 3.1.8.0070** and the BootROM version used is **4.1.4**

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 Reference Documents at the end of these Release Notes.

1.4.1 Release using individual (split) files

Use of 'split 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
2345-11300-010.sip_318.ld	SIP application executable for SoundPoint IP 301 – Version 3.1.8.0070
2345-11500-030.sip_318.ld 2345-11500-040.sip_318.ld	SIP application executables for SoundPoint IP 501 – Version 3.1.8.0070
2345-11600-001.sip_318.ld	SIP application executable for SoundPoint IP 600 – Version 3.1.8.0070
2345-11605-001.sip_318.ld	SIP application executable for SoundPoint IP 601 – Version 3.1.8.0070
2201-06642-001.sip_318.ld	SIP application executable for SoundStation IP 4000 – Version 3.1.8.0070
sip-318.cfg	main core and SIP configuration file
phone1-318.cfg	example per-phone SIP configuration
sip-318.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 (no IP 30X support): Chinese, China (for IP 4000 only) Danish, Denmark Dutch, Netherlands English, Canada English, United Kingdom English, United States French, France German, Germany Italian, Italy Japanese, Japan (for IP 4000 only) Korean, Korea (for IP 4000 only) Norwegian, Norway Polish, Poland (all phones except IP 301) Portuguese, Portugal Russian, Russia Slovenian, Slovenia (all phones except IP 301 and IP 4000) 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 products. This file is required for any phones that may be running a BootROM release older than 4.0.0 (e.g. BootROM 3.2.3RevB).

Files	Description
sip-318.ld	Concatenated SIP application executable, Version 3.1.8.0070.
sip-318.cfg	main core and SIP configuration file
phone1-318.cfg	example per-phone SIP configuration
sip-318.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 (no IP 30X support): Chinese, China (for IP 4000 only) Danish, Denmark Dutch, Netherlands English, Canada English, United Kingdom English, United States French, France German, Germany Italian, Italy Japanese, Japan (for IP 4000 only) Korean, Korea (for IP 4000 only) Norwegian, Norway Polish, Poland (all phones except IP 301) Portuguese, Portugal Russian, Russia Slovenian, Slovenia (all phones except IP 301 and IP 4000) Spanish, Spain Swedish, Sweden
SoundPointIPWelcome.wav	start up welcome sound effect

2. Changes

2.1 Version 3.1.8

2.1.1 Added or Changed Features

- 71997: **Added full support for RFC2782.**
- 73440: **Added local ringback support on the phone when there is a SIP 183 message followed by SIP 180 message.**
- 76563: **Added support for VeriSign 2048 bit certificates.**
- 76869: **Added support for VeriSign Intermediate CA certificates.**
- 76870: **Updated 2048 bit trusted CA root certificate list.**
- 76871: **Added support for RSA 2048 V3 root certificate.**

2.1.2 Removed features

None

2.1.3 Corrections

- 52477: **Display of the authorized password field on the phone web html page now works.**
- 68063: **DHCP failover no longer impacts the phone reboot.**
- 69022: **Call reminder on a transferred call from the phone is properly set.**
- 73503: **When the phone is off hook, a single event is sent to the phone base station.**
- 73659: **The phone UI refreshes when there is an account code challenge to "Enter more digits"** ***(applies to SoundStation IP 4000).***
- 73678: **Random reboot of the phone SoundPoint IP 601 is fixed.**
- 74477: **The phone UI and the dial tone issue is resolved when the user tries to resume a remotely held BLA line by picking up the handset and pressing the line key at the same time** ***(applies to SoundPoint IP 601).***
- 75636: **The phone NTP request to the NTP server when not available has been restored** ***(applies to SoundPoint IP 601 and SoundStation IP 4000).***
- 75829: **The fragmented UDP packet drop observed on the phone is fixed** ***(applies to SoundPoint IP 601).***
- 75835: **The phone shows the correct time when configured to the NTP server.**
- 76741: **Receiving an inbound call when there is a 3-way conference call originating from the 1st line key is now possible.**
- 77572: **The notification status bar is now visible without any overlap with DND, missed call, and new call UI.**
- 77378: **Handset termination works after pressing the line key and ending the call by pressing the soft key.**
- 77346: **When there is a missed call notification, the call widget is updated with the correct call count (x/x).**

- 77247: **Trying to place multiple calls on a phone's private and shared lines keeps the phone responsive for other tasks *(applies to SoundPoint IP 501 & IP 601).***

2.1.4 Configuration File Parameter Changes

None

2.2 Version 3.1.7

2.2.1 Added or Changed Features

- 61547: **Phones now send a 486 (Busy) response to a received INVITE message when a call is rejected.**

2.2.2 Removed Features

None.

2.2.3 Corrections

- 51718 **: Under certain configurations, phone continues to ring after the call has been answered.**
- 52968: **Cannot remove an instant message from the main screen even though it has been deleted.**
- 53975: **Phones will not send a SUBSCRIBE message in a certain scenario when using an SCA with barge-in enabled.**
- 55884: **The phone's display freezes and both extension modules' displays are cleared during a consultative transfer. The phone does not recover and has to be rebooted.**
- 58177: **On rare occasions, two receptionists at one site will receive an incoming PSTN call and attempt to "blind transfer" to an internal extension. They will first hear 3 notification tones after pressing the "Send" soft key. The transfer will proceed is attempted for the second time.**
- 58689: **Phones will send a 486 if an INVITE is received after a NOTIFY for the alerting state and the configuration parameter "callsPerLineKey" is set to 1.**

- 58728: **Phone presents only the "NewCall" soft key and does not present the "EndCall" soft key to allow the user to release the call and place the phone into idle state after hanging up the call during a consultative transfer.**
- 59789: **SoundPoint IP650: The user is unable to properly resume a held call after answering a different call.**
- 60051: **SoundPoint IP650: Using a BLA, the display does not show the status of the remotely held call while there is an active call currently displayed. Pressing the 'down arrow' key followed by the 'up arrow' key refresh the display to properly show the status of the held call.**
- 60141**: SoundPoint IP650: On a Bridged Line Appearance BLA line, the display incorrectly indicates 2 call appearances when there should only be one for the active call. The 2nd call appearance is for the previously held remote call that is no longer on hold.**
- 60145**: SoundPoint IP650: Using a BLA, the display on the phone incorrectly presents 2 call appearances instead of only one.**
- 60177: **SoundPoint IP5xx, IP6xx: The display does not present "hot-dialed" digits when the "idle display" feature is enabled.**
- 60264: **During a call using a BLA line, when the display is showing the "dialing screen", remote call appearances are displayed when the remote phone's BLA line resumes a call.**
- 60340: **The "Join" soft key is presented for phones with BLA lines when there is only one call active on the phone.**
- 60480: **A phone monitoring other BLA lines fails to show the presence (LED goes out) of a BLA line when that monitored line joins two other calls.**
- 60729: **Phones do not honor a BLA NOTIFY with a version number that has been increased by more than 1.**
- 60756**: A phone monitoring a Shared Call Appearance line presents an incorrect presence indication (LED turns off) of a BLA line when that monitored line joins two other calls in a centralized conference.**
- 60973: **Entering a username and password using the "Quick Setup" (Qsetup) soft key followed by a request to save, does not automatically invoke the phone to reboot the phone in order to the changes to be applied.**

- 61264: **Calls placed on hold using a shared BLA line doe not timeout (it does not receive a 200 message) when a remote phone picks up the held call (on the BLA line).**
- 61283: **When a user attempts to place a conference call on hold and the phone receives a 400 Bad request. The phone then incorrectly sends a NOTIFY with <param pname="+sip.rendering" pvalue="no" />.**
- 61298: **SoundPoint IP601: When 1.2Mbps of multicast traffic is passed through the PC port on the phone, the data port experiences a packet loss of 17%.**
- 61299: **When a phone has established a “centralized” conference call, the user cannot transfer a third incoming call.**
- 61321**: When a phone joins a centralized conference bridge, other monitoring phones incorrectly show the BLA line as being on hold instead of being in use.**
- 61547: **Phone does not send a 486 Busy message when a call (INVITE) is rejected. A binary configuration parameter is added to “sip.cfg” called “volpProt.SIP.use486forReject”. By default, (parameter is 0) the feature is disabled. If the parameter equals 1, the feature is enabled. If enabled and the phone is indicating a ringing inbound call appearance, then upon pressing the “Reject” soft-key, the phone will transmit a “486 Response” to the originator of the received INVITE message.**
- 61725: **Users cannot pick up a held call after multiple hold/resume interactions on the phone. The phone uses the “to-tag” from the 401 responses rather than 2xx responses.**
- 61950, 62024: **Phone does not honor a “retry-after” header in a “500 Glare” message responding to a BLA re-SUBSCRIBE message.**
- 61955: **An RTP audio delay is detected when calling or receiving calls from a PSTN.**
- 62036: **SoundPoint IP 3xx: Phone stops sending DTMF RTP EVENTS when receiving a second incoming call while it is already active on a previously established call.**
- 62050: **SoundPoint IP650: Phone does not properly update the number of held calls after sending “200 OK” messages as part of the notifications process.**
- 62127**: SoundPoint IP650: The “Blind” transfer soft-key is not presented on the display when the “Transfer” soft key is pressed on the second call.**

- 62223: **Phone crashes after resuming a held call using a BLA. A race condition exists with other phones when they answer the same call.**
- 62226: **Phones proceed to join a conference after receiving a “403 Forbidden” from the switch.**
- 62262: **The phone establishes a 1-way audio path after it has re-established a centralized conference call with the dropped 3rd party. This behavior is observed with Sylantro switches.**
- 62279: **The presence indicator on a Bridged Line Appearance remains on incorrectly after the phone receives a “486” message.**
- 62313: **Using a BLA configuration, dial tone is not present when pressing the second line key followed by lifting handset after holding a call on first line appearance.**
- 62361: **The call status on a BLA Bridged Line Appearance (configured for 1 call per line appearance) of a monitoring phone is not updated correctly when transfer/conference soft key is pressed.**
- 62435: **SoundPoint IP650: The phone displays incorrectly a call appearance labeled 'Unknown Party' if the remote party is held while reorder tone is played locally.**
- 62511: **In certain situations, the monitored Busy Lamp Field BLF line does not invoke an incoming call notification (icon and tone).**
- 62514: **SoundPoint IP670: In certain situations, the status of the monitored Busy Lamp Field BLF lines is not removed from the display even though the status has been updated by the switch.**
- 62569: **Phone generates a redundant NOTIFY message when triggered by a “100 response” during a “re-INVITE”.**
- 62669: **Multiple phones try to resume a held Bridge Line Appearance BLA line at the same time. As a result, presence indicator on the BLA line is cleared on the trailing phone when the reorder tone is played.**
- 62672: **Either Directed Call Pickup DCP or Group Call Pickup feature (using soft keys instead of *53 and *54 feature access codes) fails when the user must enter an account code. The account code is not appended to the user portion of the URI.**

- 62704: **The presence indicator of a Bridged Line Appearance BLA is not updated correctly on monitoring phones when the phone's LAN data cable is disconnected and then re-connected.**
- 62855: **SoundPoint IP3xx: Invoking either the Group Call Pickup or Directed Call Pickup feature, using its corresponding soft key, does not function properly. The display shows "Unknown" and the call is not picked up.**
- 62926: **SoundPoint IP3xx: The "Resume" soft key is not presented when the line key is pressed continuously while the line is in a remote held call state. This occurs when the line is configured as callsPerLineKey="1".**
- 63099: **The phone's monitoring Bridged Line Appearance BLA line, configured for one call per line, cannot pickup the held call after the call on a BLA line has been put on hold using the Transfer/Conference key.**
- 63286: **The phone's Part Number is listed incorrectly as "YYYY-YYYYY-YYY" (instead of showing actual digits) when viewing from the display by invoking Menu->Status->Platform->Application->Main.**
- 64212: **SoundPoint IP3xx: Invoking the Call Park feature with the soft key does not function correctly when the soft key is configured as 1 line and 1 call per line.**
- 64219: **SoundPoint IP3xx: Phone does not send a proper hold NOTIFY message after a consultative transfer is canceled when the configuration parameter "notifyTransferHoldAsActive" is disabled.**
- 64271: **In an attempt to answer an incoming call, the call is unintentionally terminated. This occurs when the incoming call's line key is pressed simultaneously as the handset is lifted.**
- 64274**: In an attempt to resume a held call, the held call is unintentionally terminated when the user inadvertently seizes two line keys simultaneously.**
- 64327: **In an attempt to answer an incoming call, the user inadvertently presses 2 line keys. The user is then connected to both lines: one with an incoming caller and the other with dial tone.**
- 64340: **The indicator, on a Bridged Line Appearance BLA line that is monitoring other lines, remains on continuously after the monitored phone performs the following sequence: transfer->split->endcall->resume->hold.**

- 64356: **SoundPoint IP3xx: The display showing a remote call appearance never times out when the user presses continuously a BLA line key followed by pressing a down arrow key while there are multiple calls on hold on the remote BLA.**
- 64822: **SoundPoint IP3xx: When configuring the phones using “sip_att.cfg”, the phone shows "Service Unavailable" when the speed dial key is pressed while the phone is off-hook.**
- 64862: **Joining an internal extension with an external PSTN call causes one call to drop. This occurs occasionally.**
- 65119: **When a Bridged Line Appearance BLA line is presented in a dialing screen, the remote call appearance should is incorrectly displayed when the remote BLA line resumes a call.**
- 65207**: A slow memory leak occurs in the SIP stack. This is due to the receipt of hunt group INVITE containing “replaces”. This occurs with ADTRAN switches.**
- 67178: **Centralized conference is established when “reg.1.lineKeys” is set to 5 or greater.**
- 67186**: SoundPoint: IP301, IP501, IP601: All soft keys disappear on the “assistant” phone when pressing down the arrow key after placing multiple calls on hold with the “boss’” line appearance.**

2.2.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.use486forReject	defaults to null
sip	added	call.localConferenceEnabled="1"	defaults to 1

2.3 Version 3.1.6

2.3.1 Added or Changed Features

None.

2.3.2 Removed Features

None.

2.3.3 Corrections

- 54423: **SoundPoint IP 601: Phone reboots under heavy SIP traffic while using Buddy Watch as a BLF.**
- 54479: **SoundPoint IP 601 + 32 member BLF: After upgrading from 2.1.2 to 3.1.3RevB, users experience a delay in transferring calls using the Transfer key.**

2.3.4 Configuration File Parameter Changes

None.

2.4 Version 3.1.5 (Limited Distribution)

2.4.1 Added or Changed Features

None.

2.4.2 Removed Features

None.

2.4.3 Corrections

- 54165: **Phone cannot pick up call off hold after it receives NOTIFY with dialog state="full" in response to its BLA re-subscribe**

2.4.4 Configuration File Parameter Changes

None.

2.5 Version 3.1.4

2.5.1 Added or Changed Features

None

2.5.2 Removed Features

- **Remove support for the SoundPoint IP 320, 321, 330, 331, 430, 450, 550, 560, 650, 670 products.**
- **Remove support for the SoundStation IP 6000, 7000 products**
- **Remove support for the VVX 1500 product.**

2.5.3 Corrections

- 50189: **SIP responses missing to-tag after Phone challenges INVITE**
- 51031: **Cannot change the language to Russian**
- 52237/52017: **Web interface Password entry is not masked when entered (since SIP 3.0.0).**
- 53826/50546: **When URL dialing disabled, BLIND soft key appears in the 4th soft key slot, as opposed to the 3rd slot, after pressing TRANSFER.**
- 53827/51690**: EFK feature is used for one-touch Voicemail dialing. When using SIP 3.1.3 the phone appears not to honour the stickyAutoLineSeize.**
- 53828/52014**: 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.**
- 53829/50254: **Phone does not honor SDP sent in PRACK.**
- 54214/50869: **Phone will only offer SRTP when SRTP crypto suite is selected**

2.5.4 Configuration File Parameter Changes

None.

2.6 Version 3.1.3 C

2.6.1 Added or Changed Features

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

2.6.2 Removed Features

None.

2.6.3 Corrections

None.

2.6.4 Configuration File Parameter Changes

None.

2.7 Version 3.1.3 B

2.7.1 Added or Changed Features

None.

2.7.2 Removed Features

None.

2.7.3 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.**

2.7.4 Configuration File Parameter Changes

None.

2.8 Version 3.1.3 (Limited Release – Version 3.1.3.0336)

2.8.1 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**

2.8.2 Removed Features

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

2.8.3 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 Signaling 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: **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**
- 49115: **SoundStation IP 6000, 7000: Support new revision of Ethernet PHYs.**
- 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.**

2.8.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.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	voIpProt.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	voIpProt.SIP.tcpFastFailover	If set to 1, failover occurs based on the values of `reg.x.server.y.retryMaxCount` `voIpProt.server.x.retryTimeOut`. If set to 0, use old behavior. If `reg.x.tcpFastFailover` is Null, this attribute is checked. If `voIpProt.SIP.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

2.9 Version 3.1.2 B

2.9.1 Added or Changed Features

- Add Support for the VVX 1500 product.

2.9.2 Removed Features

None.

2.9.3 Corrections

None.

2.9.4 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.

2.10 Version 3.1.2

2.10.1 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 □) 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**.

2.10.2 Removed Features

N/A

2.10.3 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: **Ringing stops when users goes on-hook after lifting handset during incoming call when up.offHookAction.none = 1**
- 45799: **XML API: Internal URIs: softkey:Exit, softkey:Submit and softkey:Reset do not work when called from hyperlink anchor tags**
- 46051: **Manage N-way conference menu has overlapping items if long caller-ids are present.**
- 46144: **JPEG decoder fails on some files**
- 46242: **XML API: If an account supports 2 line keys, API notifications of call events are sent for only 1 of them**
- 46293: **Phones may lock up if a CHECK-SYNC is received while a CHECK-SYNC is in progress**
- 46422: **Five to six second delay in UI when using the SPLIT 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.**

2.10.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
phone1	added	acd.reg	See Technical Bulletin34787 for details
phone1	added	acd.stateAtSignIn	
sip	added	voIpProt.SIP.acd.signalingMethod	
sip	added	voIpProt.SIP.compliance.RFC3261.validate.contentLanguage	If set to 1, validation of the SIP header content language is enabled. If set to 0 or Null, validation is disabled.
sip	removed	bg.gray.selection	Modified the method in which the background settings are managed across multiple phone models
sip	added	bg.hiRes.gray.selection	
sip	removed	bg.color.selection	
sip	added	bg.hiRes.color.selection	
sip	added	bg.medRes.gray.selection	
sip	changed	ind.gi.IP_600.13.physH	Changed from 109 to 103
sip	changed	ind.gi.IP_7000.7.physH	Changed from 60 to 76
sip	added	log.level.change.cmr	Control the logging detail level for individual components: call media 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

2.11 Version 3.1.1 B

2.11.1 Added or Changed Features

None.

2.11.2 Removed Features

None.

2.11.3 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.**

2.11.4 Configuration File Parameter Changes

2.12 Version 3.1.1

2.12.1 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**

2.12.2 Removed Features

N/A

2.12.3 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.**

2.12.4 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

2.13 Version 3.1.0 C

2.13.1 Added or Changed Features

- Add Support for the SoundPoint IP 450 product.

2.13.2 Removed Features

None.

2.13.3 Corrections

None.

2.13.4 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.

2.14 Version 3.1.0 B

2.14.1 Added or Changed Features

None.

2.14.2 Removed Features

None.

2.14.3 Corrections

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

2.14.4 Configuration File Parameter Changes

None.

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

This version should be replaced by 3.1.0RevB

2.15.1 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.**

2.15.2 Removed Features

N/A

2.15.3 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.**

2.15.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.strictLineSeize	If set to 1, forces the phone to wait for 200 OK response when receiving a TRYING notify. If set to 0 or Null, this is old behavior.
sip	added	voIpProt.SIP.strictUserValidation	If set to 1, forces the phone to match user portion of signaling exactly. If set to 0 or Null, phone will use first registration if the user part does not match any registration.
sip	added	voIpProt.SIP.lineSeize.retries	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA).
sip	added	voIpProt.SIP.header.diversion.enable	If set to 1, the diversion header is displayed if received. If set to 0 or Null, the diversion header is not displayed.
sip	added	voIpProt.SIP.header.list.useFirst	If set to 1 or Null, the first diversion header is displayed. If set to 0, the last diversion header is displayed.
sip	added	voIpProt.SIP.header.warning.codes.accept	A list of accepted warning codes. If set to Null, all codes are accepted. Only codes between 300 and 399 are supported.
sip	added	voIpProt.SIP.header.warning.enable	If set to 1, the warning header is displayed if received. If set to 0 or Null, the warning header is not displayed.
sip	added	voIpProt.SIP.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. If reg.x.musicOnHold is set to Null, this attribute is checked.

sip	added	lcl.ml.lang.tags.x	The format is: • The first two letters are the ISO-639 language abbreviation. • The next two letters are the ISO-3166 country code. • The next two letters are the ISO-639 language abbreviation. • The remainder of the string is the preference level for the display of the language, or English if the language is not available
sip	added	up.numberFirst CID	If set to 0 or Null, caller ID display will show caller's name first. If set to 1, caller ID display will show caller's number first.
sip	changed	saf.1	The default value is Null. To allow the SoundPoint IP welcome sound to be played on reboots and restarts, set to **SoundPointIPWelcome.wav**
sip	changed	voice.aec.hs.enable	The default value is enabled (1).
sip	changed	voice.aes.hs.enable	The default value is enabled (1).
sip	added	call.directedCallPickupString	The star code to initiate a directed call pickup.
sip	added	dir.corp.pageSize	The maximum number of entries requested from the corporate directory server with each query.
sip	added	dir.corp.cacheSize	The maximum number of entries that can be cached locally on the phone.
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	This is the text displayed with the soft key. If set to Null, the label to display is determined as follows: • If the soft key is mapped to a enhanced feature key macro, the label of the enhanced feature key macro will be used. • If the soft key is mapped to a speed dial, the label of the corresponding directory entry will be used. If this label does not exist as well and the directory entry is an enhanced feature key macro, then the label of the enhanced feature key macro will be used. • If the soft key is mapped to chained actions, only the first one is considered for label, using the rules above. • If no labels are found after the above steps, the soft key label will be blank.
sip	added	softkey.x.action	The same syntax as the enhanced feature key action.
sip	added	softkey.x.enable	If set to 0 or Null, the soft key is disabled. If set to 1, the soft key is enabled.

sip	added	softkey.x.precede	If set to 0 or Null, the soft key replaces any empty space from the leftmost position. If set to 1, the soft key is displayed before the first standard soft key.
sip	added	softkey.x.use.idle	If set to 0 or Null, the soft key is not displayed in the idle state. If set to 1, the soft key is displayed in the idle state.
sip	added	softkey.x.use.active	If set to 0 or Null, the soft key is not displayed in the active call state. If set to 1, the soft key is displayed in the active call state.
sip	added	softkey.x.use.alerting	If set to 0 or Null, the soft key is not displayed in the alerting state. If set to 1, the soft key is displayed in the alerting state.
sip	added	softkey.x.use.dialtone	If set to 0 or Null, the soft key is not displayed in the dialtone state. If set to 1, the soft key is displayed in the dialtone state.
sip	added	softkey.x.use.proceeding	If set to 0 or Null, the soft key is not displayed in the proceeding state. If set to 1, the soft key is displayed in the proceeding state.
sip	added	softkey.x.use.setup	If set to 0 or Null, the soft key is not displayed in the setup state. If set to 1, the soft key is displayed in the setup state.
sip	added	softkey.x.use.hold	If set to 0 or Null, the soft key is not displayed in the hold state. If set to 1, the soft key is displayed in the hold state.

sip	added	softkey.feature.newcall	If set to 0, the New Call soft key is not displayed when there is another way to place a call. If set to 1 or Null, the New Call soft key is displayed.
sip	added	softkey.feature.endcall	If set to 0, the End Call soft key is not displayed. If set to 1 or Null, the EndCall soft key is displayed.
sip	added	softkey.feature.split	If set to 0, the Split soft key is not displayed. If set to 1 or Null, the Split soft key is displayed.
sip	added	softkey.feature.join	If set to 0, the Join soft key is not displayed. If set to 1 or Null, the Join soft key is displayed.
sip	added	softkey.feature.forward	If set to 0, the Forward soft key is not displayed. If set to 1 or Null, the Forward soft key is displayed.
sip	added	softkey.feature.directories	If set to Null, the Dir soft key is displayed on the SoundPoint IP 320/330 phone, but not on any other phone. If set to 0, the Dir soft key is not displayed on any phone. If set to 1, the Dir soft key is displayed on all phones as follows: • In the idle state, it is displayed after the New Call and Callers soft keys. • In the dialtone state, it is displayed after the End Call and Callers soft keys. • During a conference or transfer, it is displayed after the Callers and Cancel soft keys.

sip	added	softkey.feature.callers	If set to Null, the Callers soft key is displayed on the SoundPoint IP 320/330 phone, but not on any other phone. If set to 0, the Callers soft key is not displayed on any phone. If set to 1, the Callers soft key is displayed on all phones as follows: • In the idle state, it is displayed after the New Call soft key and before the Dir soft key. • In the dialtone state, it is displayed after the End Call soft key and before the Dir soft key. • During a conference or transfer, it is displayed before the Cancel soft key.
sip	added	softkey.feature.mystatus	If set to 0, the MyStatus soft key is not displayed. If set to 1 or Null, the MyStatus soft key is displayed.
sip	added	softkey.feature.buddies	If set to 0, the Buddies soft key is not displayed. If set to 1 or Null, the Buddies soft key is displayed.
sip	added	softkey.feature.basicCallManagement.redundant	If set to 0 and the phone has hard keys mapped for Hold, Transfer, and Conference functions (all must be mapped), all of these soft keys are not displayed. If set to 1 or Null, all of these soft keys are displayed.

phone1	added	reg.x.strictLineSeize	If set to 1, forces phone to wait for 200 OK on registration x when receiving a TRYING notify. If set to 0 or Null, this is old behavior. If this parameter is Null, voIpProt.SIP.strictLineSeize is checked. If both parameters are set, this parameter takes precedence.
phone1	added	reg.x.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. When present, and if reg.x.musicOnHold is not Null, this attribute overrides the global Music on Hold defined in the sip.cfg configuration file.
phone1	added	attendant.ringType	The ring tone to play when a BLF dialog is in the offering state. Permitted values are 1 to 22. The default is Null.

2.16 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.

2.16.1 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.**

2.16.2 Removed Features

N/A

2.16.3 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**

2.16.4 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'

2.17 Version 3.0.3 B

Change made applies to the SoundStation IP 7000 product only.

2.17.1 Added or Changed Features

None.

2.17.2 Removed Features

None.

2.17.3 Corrections

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

2.17.4 Configuration File Parameter Changes

None.

2.18 Version 3.0.3

2.18.1 Added or Changed Features

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

2.18.2 Removed Features

None

2.18.3 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**

2.18.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.strictUserValidation	If set to “1”, forces phone to match user portion of signaling exactly. If set to “0”, phone will use first registration if the user part does not match any registration
sip	added	voIpProt.SIP.strictLineSeize	If set to “1”, forces phone to wait for 200 OK when receiving a TRYING notify.
sip	added	voIPProt.SIP.lineSeize.retries	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA). Valid values are 3 to 10. Note that in this release, a value of 3 results in 10. A value of 2 can be used to get 3 retries.
phone1	added	reg.n.strictLineSeize	If set to “1”, forces phone to wait for 200 OK on registration n when receiving a TRYING notify. If this parameter is Null, voIpProt.SIP.strictLineSeize is checked. This parameter takes precedence.

2.19 Version 3.0.2 C

2.19.1 Added or Changed Features

None.

2.19.2 Removed Features

None.

2.19.3 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.

2.19.4 Configuration File Parameter Changes

None.

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

2.20.1 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**

2.20.2 Removed Features

None.

2.20.3 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**

2.20.4 Configuration File Parameter Changes

<table>
<tr><th>.cfg File</th><th>Action</th><th>Parameter</th><th>Description</th></tr>
<tr><td>sip</td><td>added</td><td>voice.codecPref.IP_(6|7)000.*</td><td>Codec support for IP 6000 and IP 7000.</td></tr>
<tr><td>sip</td><td>added</td><td>voice.gain.(r|t)x.analog.*.IP_(6|7)000</td><td>Gain levels for IP 6000 and IP 7000.</td></tr>
<tr><td>sip</td><td>added</td><td>voice.gain.(r|t)x.analog.*.IP_6000</td><td>Gain levels for IP 6000.</td></tr>
<tr><td>sip</td><td>added</td><td>voice.(r|t)xEq.hf.IP_(6|7)000.(pre|post)Filter.enable</td><td>Prefilter and postfilter enable for IP 6000 and IP 7000.</td></tr>
<tr><td>sip</td><td>changed</td><td>dir.corp.backGroundSync</td><td>Changed from 1 to 0, disabling background sync.</td></tr>
<tr><td>sip</td><td>changed</td><td>dir.corp.backGroundSync.period</td><td>Changed value from 43200 (12 hours) to 86400 (24 hours).</td></tr>
<tr><td>sip</td><td>removed</td><td>bg.ranges</td><td></td></tr>
<tr><td>sip</td><td>changed</td><td>bg.color.selection</td><td>Defines which background is used. Default is “1,1”. First (left) index is the type of background. Second is the index into the table of that type.
<table><tr><th>Index</th><th>Type</th></tr><tr><td>1</td><td>Predefined backgrounds</td></tr><tr><td>2</td><td>Solid patterns</td></tr><tr><td>3</td><td>User-defined bitmaps</td></tr></table></td></tr>
<tr><td>sip</td><td>added</td><td>bg.hiRes.color.pat.solid.*.(name|red|green|blue)</td><td>Defines the name and colour of solid backgrounds.</td></tr>
<tr><td>sip</td><td>added</td><td>bg.hiRes.color.bm.*.(em.)?name</td><td>Defines colour backgrounds for the phone’s display and the expansion modules’ displays (em).</td></tr>
</table>

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

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.

2.21 Version 3.0.1RevB

2.21.1 Added or Changed Features

None

2.21.2 Removed Features

None

2.21.3 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”.**

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

2.22.1 Added or Changed Features

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

2.22.2 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.**

2.22.3 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**

2.22.4 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

2.23 Version 3.0.0

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

2.23.1 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-rtcpxr in SIP stack**
- 37256**: Jabra Jx10 electronic hook switch support on SoundPoint IP 320, 330, 430, 550, 560, 650 phones. Requires an "Interface Cable" from the headset base to the phone for use. Refer to technical bulletin 35150 for more details.**
- **37551**: Add enhanced speed dial capability.**
- 38443**: Support full complement of BLF parties on SoundPoint IP 650 plus 3 EMs using UDP**
- 38847**: Line-Key and Soft-Key Labels changed to white text with 3-D appearance on SoundPoint IP 550, 560, 650 phones.**

- 38979: **Make UI background bitmap configurable – SoundPoint IP 550, 560 and 650 phones**
- 39071**: DHCPINFORM should apply if boot server address is null**
- 39072**: Reduce DHCPINFORM retry timeouts**
- 39305**: Increase Handset transmit loudness by 3dB to better meet standards AS/NZS 60950 and AS/ACIF S004**, **as directed by Category C33 of the Telecommunications Labeling Notice (TLN) (for Australia).**
- 39330**: DHCPINFORM should apply if boot server address is 0.0.0.0**
- 39344: **Update XML Dictionaries for SIP 3.0.0**
- 39695**: Lower minimum syslog.renderLevel to 0 (from 1)**

2.23.2 Removed Features

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

2.23.3 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 no allow 24 calls per line**
- 37111: **"Using default certs" log message appears when configuring for "Custom cert" only**
- 37116: **Date and time disappear from the phone's idle screen when browsing menu during call**
- 37184: **Digest Authentication Password used for downloading configuration files is displayed in log files**
- 37227: **The registration icon disappears when IP301 establishes a conference call**
- 37391: **Phone does not start correctly if the contact directory XML syntax is not correct**
- 37420: **SIP Server Fall-back --- IP 320 and IP 330 -- Line Information screen does not show the server info when 3rd SIP server becomes the working server.**
- 37426: **Cannot change selection in Clock Time menu more than once without exiting**
- 37428: **Selecting another language forces exit from language menu**
- 37603: **Key remapping does not show correct values in diagnostics menu on IP 320, IP 330 and IP 4000**
- 37679: **File TX Tries setting in flash could be set to invalid value 0**

- 37690**: Phone does not retry ACK when receiving duplicate 200 OK**
- 37709**: SoundPoint IP 320 and IP 330 phones may re-boot after several days when the idle micro-browser is configured and active.**
- 37711: **Brief audio ‘noise’ due to SRTP encryption key change.**
- 37719**: Pressing Resume soft key on phone after sending an unresolvable hostname during a blind transfer reboots or freezes the far end phone**
- 37726**: DNS SRV queries can incorrectly append search domain when it is already present**
- 37851**: SRTP phone doesn't include crypto suite in group pickup signaling**
- 37855**: Join soft-key is not available when maximum call appearances are used**
- 37906**: IP301 does not show watch buddy icon when peer-to-peer watch buddy is enabled**
- 37915**: Peer-to-Peer Presence: Blocking contact in Watcher List creates extra contact "SPIP" in directory menu**
- 38021**: Ringer type 12 is not playing correctly**
- 38219**: While receiving multiple NOTIFY messages ,the phone may not send an invite to initiate a call.**
- 38279**: If a 403 response is received by the phone when attempting to complete a call transfer in the proceeding state the phone may re-boot.**
- 38308**: Packet Loss count does not increment correctly when a Held call is resumed and the SSRC value changes.**
- 38334**: MKI format in RTP and RTCP packets is incorrect**
- 38540**: Packet channel statistics computation not resetting properly when SSRC changes**
- 38732**: Line status icon does not change back on line 2 after being on speaker or handset – SoundPoint IP 330/320**
- 38902**: UI malfunctions when remote shared line is in hold status and local phone attempts a new call**
- 39041**: Icon may indicate phone is unregistered after successful re-registration if voIpProt.SIP.serverFeatureControl.cf=1 or voIpProt.SIP.serverFeatureControl.dnd=1**

- 39074**: Microbrowser: clicking a link to non-responsive server takes a long time to timeout**
- 39184**: Read-only directory can be edited on IP 320 and IP 330 if phone is in digit collection state when contact directory is opened**
- 39338**: Some of the SRTP session parameters are incorrectly spelled in the SDP (e.g. UNENCRYPTED_SRTCP is represented as UNENCRYPTED_RTCP)**
- 39362**: Phone does not play incoming RTP when offered send-only stream.**
- 39419: **Maximum Backlight Intensity setting has very little effect on SoundPoint IP 560 phones.**
- 39431**: Display Diagnostics shows very minimal changes on the display on IP 550 and IP 650**
- 39438**: Backlight does not update immediately after pressing cancel on the maximum intensity screen**
- 39490**: In some call scenarios the phone may not display the SRTP secure line icon even though the call is encrypted.**
- 39502**: DigitMap: The + character does not get matched in a dial plan.**
- 39601**: In IP 320 and IP 330 phone's local contact edit menu, cursor flashes on the character just entered instead of after the character**
- 39618**: font500Prop_16_U0000_U00FF.fnt has anomalously wide "K"**
- 39629**: When reg.1.callsPerLineKey=1 is set, and a conference is established while transferring the call, the phone hangs and reboots**
- 39631**: Idle browser cuts volume icon**
- 39652**: Some layered windows are incorrectly clipped**

2.23.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SDP. useLegacyPayloadTypeNegotiation	
sip	added	voIpProt.SIP.csta	Enables uaCSTA.
sip	added	up.handsfreeMode	Enables or disables hands-free speakerphone.
phone1	added	up.analogHeadsetOption	Selects optional external hardware for use with a headset attached to the phone's analog headset jack.
sip	changed	tone.chord.callProg.6.offDur	Changed from 0 to 10000.
sip	changed	tone.chord.callProg.6.repeat	Changed from 1 to 2.
sip	changed	se.pat.ringer.12.name="Ringback-style"	Added 100ms of silence to start of pattern.
sip	removed	voice.gain.rx.analog.handset.wideband voice.gain.rx.analog.handset.sidetone. wideband voice.gain.tx.analog.handset.wideband voice.handset.wideband voice.handset.wideband.rxdg.adjust	Controlled gain for wideband handset. This control is now performed through the parameters that do not include ".wideband".
sip	added	voice.qualityMonitoring	The voice.qualityMonitoring section controls the Voice Quality Monitoring feature.
sip	added	tcpIpApp.keepalive.tcp.idleTransmitInterval tcpIpApp.keepalive.tcp. noResponseTrasmitInterval tcpIpApp.keepalive.tcp.sip.tls.enable	Controls TCP keep-alive on SIP TLS connections.
sip	added	call.singleKeyPressConference call.localConferenceCallHold	Enables new conference behaviors.
sip	added	call.disableAutoResumeCentralConference	For use with uaCSTA feature for centralized confrerencing.
sip	added	bg.hiRes.gray.pat.solid.x.name bg.hiRes.gray.pat.solid.x.red bg.hiRes.gray.pat.solid.x.green bg.hiRes.gray.pat.solid.x.blue bg.hiRes.gray.bm.x.name	Sets up color (gray-scale) and graphical backgrounds for IP 550, IP560 and IP 650 phones.

sip	added	feature.x.name	Added new features “nway-conference”, “call-recording” and “corporate-directory”
phone1	added	reg.x.bargeInEnabled	Enables barge in feature for SCAs.
sip	added	dir.corp	The dir.corp section controls the Corporate Directory feature.
sip	added	usb.set1.device.1.vendor usb.set1.device.1.product	Identifies supported USB devices. This list should be populated only with devices that are known to work with the phones. See Technical Bulletin 38084 for details.

Table 2-2

2.24 Version 2.2.2

2.24.1 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.**

2.24.2 Removed Features

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

2.24.3 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.**

2.24.4 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.

2.25 Version 2.2.1 (Limited Release)

2.25.1 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**

2.25.2 Removed Features

None.

2.25.3 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**

2.25.4 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.

2.26 Version 2.2.0

2.26.1 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"**

2.26.2 Removed Features

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

2.26.3 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 voIpProt.SDP.answer.useLocalPreferences parameter is set to default**
- 34891: **SoundStation IP 4000 loudness does not decrease for bottom six volume settings**
- 35320: **If two function keys are remapped to dial specific speed dial numbers, only the first one will work**
- 35480: **SoundPoint IP 320 and 330 phones allow watching only 7 buddies instead of 8 and may 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**

2.26.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.csta	Not currently used, will be used in a future release.
sip	added	voIpProt.SIP.serverFeatureControl.dnd	See Administrator's Guide for SIP 2.2.0 for details
sip	added	voIpProt.SIP.serverFeatureControl.cf	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

.cfg File	Action	Parameter	Description
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 must be updated. Failure to do this will affect the audio performance of the phone.
sip	added	voice.rxEq.hf.IP_7000.xxx voice.txEq.hf.IP_7000	Not currently used, will be used in a future release.
sip	added	call.dialtoneTimeOut	See Administrator's Guide for SIP 2.2.0 for details
sip	added	call.disableAutoResumeCentralConf erence	Not currently used, will be used in a future release.
sip	added	call.singleKeyPressConference	Not currently used, will be used in a future release.
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.

.cfg File	Action	Parameter	Description
Sip	added	sec.srtp.enable sec.srtp.leg.enable sec.srtp.offer sec.srtp.require sec.srtp.key.lifetime sec.srtp.mki.enabled sec.srtp.sessionParams.noAuth.offer sec.srtp.sessionParams.noAuth.require sec.srtp.sessionParams.noEncrypRTP.offer sec.srtp.sessionParams.noEncrypRTP.require sec.srtp.sessionParams.noEncrypRTCP.offer sec.srtp.sessionParams.noEncrypRTCP.require sec.srtp.sessionParams.leg.noAuth.offer sec.srtp.sessionParams.leg.noAuth.require sec.srtp.sessionParams.leg.noEncrypRTP.offer sec.srtp.sessionParams.leg.noEncrypRTP.require sec.srtp.sessionParams.leg.noEncrypRTCP.offer sec.srtp.sessionParams.leg.noEncrypRTCP.require sec.srtp.sessionParams.IP_4000.noAuth.offer sec.srtp.sessionParams.IP_4000.noAuth.require sec.srtp.sessionParams.IP_4000.noEncrypRTP.offer sec.srtp.sessionParams.IP_4000.noEncrypRTP.require sec.srtp.sessionParams.IP_4000.no	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.

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

.cfg File	Action	Parameter	Description
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

2.27 Version 2.1.2

2.27.1 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 voIpProt.SIP.authOptimizedInFailover and/or reg.x.auth.optimizedInFailover
- 36647: **Added configurable option allowing message waiting indicator to be displayed although voicemail cannot be accessed**
  Uses new parameter up.mwiVisible

- 36681: **Added logging of version information for configuration files**

2.27.2 Removed Features

None.

2.27.3 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**

2.27.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.authOptimizedInFailover	This parameter controls failover behavior during authentication signaling. 0 = default behavior which obeys the RFC 1 = optimization enabled, phone first retries a SIP transaction with the server that has just sent a 401 or 407 response
sip	added	up.mwiVisible	0 = same behavior as SIP 2.1.1, this is the default behavior 1 = if msg.mwi.x.callBackMode parameter is set to “disabled”, message waiting indicator is displayed but voicemail cannot be accessed
sip	changed	Changed file header from $Revision: $ $Date: $ to $RCSfile: sip.cfg,v $ $Revision: $	This is required to support the new feature 36681 described above.
phone1	added	reg.x.auth.optimizedInFailover	If this parameter is set, it overrides the global voIpProt.SIP.authOptimizedInFailover parameter. x is the registration index. See the description for voIpProt.SIP.authOptimizedInFailover

.cfg File	Action	Parameter	Description
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.

2.28 Version 2.1.1 C

2.28.1 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**

2.28.2 Removed Features

None.

2.28.3 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**

2.28.4 Configuration File Parameter Changes

None.

2.29 Version 2.1.1

2.29.1 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**

2.29.2 Removed Features

None.

2.29.3 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 voIpProt.SIP.useContactInReferTo has been added to sip.cfg to control the source of the URI used in the refer-to header
- 33647: **The phone may reboot because it detects a suspended task even though that task may have been suspended intentionally**
- 33967: **An error message is logged if a daylight savings time (DST) start or stop time of 0 (12am) is selected (although the selection is correctly used)**
- 34325: **Microbrowser display is closed when shared line is opened on other**

phone

- 34431: **When changing the configuration of a phone via the web interface, the phone may lock up**
- 34443: **A remote-on-hold call on a line is not picked up by the first press of the line key with some SIP servers**
- 34508: **In a G.729 call, SoundPoint IP 50X and 60X phones may reboot with a DSP assertion failure. This problem is more likely in conference calls and can be reliably reproduced within 20 minutes of the call start.**
- 34723: **RTCP transmission interval is not consistent with industry norms**
- 34772: **The value of the DLSR field in RTCP sent by the phone can be wrong by up to about one second**
- 34827: **There are two places to configure the microbrowser from the phone web server**
- 34882: **The configuration page on the phone web server has two "Event 2" entries in the Global Log Level Limit drop-down list**
- 34906: **NOTIFY request without dialog content (an 'empty' NOTIFY request, such as you would get with a subscription renewal when the line is idle) does not extinguish LED's lit as a result of previous active dialogs**
- 35049: **DSP load graph on SoundPoint IP 550 shows slightly incorrect value**
- 35228: **Phone may have one-way audio when SDP is received with c line below m line**
- 35293: **Soft keys have some missing pixels on the SoundPoint IP 430 when the microbrowser is accessed**
- 35308: **A known problem in the SoundPoint IP 430 processor may cause the phone to reboot with a DSP assertion failure instead of restarting the affected driver**
- 35477: **When handset AEC is enabled on SoundPoint IP 50X and 60X phones, echo may occur on speaker phone when switching between handset and speaker phone**
- 35533: **The phone's web server shows the DST start and stop days as Monday by default instead of Sunday**
- 35537: **A saturated transmit signal may cause SoundPoint IP 430 phone to reboot**

- 35573: **After selecting the Russian language and accessing the microbrowser, the phone may freeze**
- 36012: **Conference host may indicate phone is muted but audio is heard by far end after one leg ends call**

2.29.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.useContactInReferTo	0 = default behavior which is the same as previous behavior, use URI from initial call's To header in REFER's refer-to header 1 = use URI from initial call's Contact header in REFER's refer-to header when setting up a transfer
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.

.cfg File	Action	Parameter	Description
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.

2.30 Version 2.1.0

2.30.1 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**

2.30.2 Removed Features

None.

2.30.3 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**

2.30.4 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	voIpProt.server.1.lcs	Refer to Technical Bulletin 5844.

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.useSendonlyHold	Can be set to 0 or 1. Null default is 0. Default in sip.cfg is 1. If set to 1, the phone will send a reinvite with a stream mode attribute of “sendonly” when a call is put on hold. This is the same as the previous behavior. If set to 0, the phone will send a reinvite with a stream mode attribute of “inactive” when a call is put on hold. Note: The phone will ignore the value of this parameter if set to 1 when the parameter voIpProt.SIP.useRFC2543hold is also set to 1 (default is 0).
sip	added	dialplan.applyToUserSend="1" dialplan.applyToUserDial="1" dialplan.applyToCallListDial="0" dialplan.applyToDirectoryDial="0"	Refer to Technical Bulletin 11572.
sip	changed	dialplan.digitmap.timeOut="3" to "3|3|3|3|3|3"	Refer to Technical Bulletin 11572.
sip	changed	tcpIpApp.sntp.daylightSavings.start.month="4" to “3”	Changes to support new daylight savings time rules.
sip	changed	tcpIpApp.sntp.daylightSavings.start.date="1" to “8”	
sip	changed	tcpIpApp.sntp.daylightSavings.stop.month="10" to “11”	
sip	changed	tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth="1" to “0”	
sip	added	call.stickyAutoLineSeize.onHookDialing	Refer to Administrator’s Guide Addendum for SIP 2.1.
sip	changed	voice.gain.rx.digital.chassis.IP_650="-9" to “6”	Gain changes required to match new software load.
sip	changed	voice.gain.rx.digital.ringer.IP_650="-21" to “-12”	

.cfg File	Action	Parameter	Description
sip	changed	voice.handset.sidetone.adjust.IP_430=" -12" to “-13”	
sip	added	voIpProt.server.x.transport and voIpProt.SIP.outboundProxy.transport	Added “TCPOnly” as a possible value for these existing parameters.

2.31 Version 2.0.3 B

2.31.1 Added or Changed Features

- 14874: **Added support for SoundPoint IP 650 platform**
- 15775: **Added support for LCD backlight on SoundPoint IP 650**
- 15852: **Added support for 32 MB of memory on SoundPoint IP 650**
- 15853: **Added support for G.722 audio code on SoundPoint IP 650**
- 16335: **Added support for 8 MB of flash on SoundPoint IP 650**
- 16686: **Added support for USB diagnostics**
- 17132: **Added visual indication of wideband audio**

2.31.2 Removed Features

None.

2.31.3 Corrections

The following issues have been resolved with this release:

None.

2.31.4 Configuration File Parameter Changes

None.

2.32 Version 2.0.3

2.32.1 Added or Changed Features

None

2.32.2 Removed Features

None.

2.32.3 Corrections

The following issues have been resolved with this release:

- 17981: **DHCP initialization incorrect for SoundStation IP 4000 which may cause boot time problems on some servers**
- 18491: **Network load reported by SoundPoint IP 430 phones is affected by traffic which is not destined for the phone**
- 18692: **Presence subscribe has "application/pidf+xml" in Accept header although it is not fully supported**
- 18766: **Ethernet transmit level is low on SoundPoint IP 430 phone**
- 18790: **Some shared line scenarios do not work with Broadsoft R14 and R13 MP13 releases**
- 18919, 11981, 18997: **Time stamp in RTCP packets is incorrect**
- 19016: **SDP containing two "a=" lines causes transfer from a private line to a shared line to fail**
- 19082: **Phone seizes wrong line making outbound call to FAC *55**
- 19210: **Too many messages are logged when "so" is set to level 2**

2.32.4 Configuration File Parameter Changes

The following configuration file changes have been included in this build in preparation for future inclusion of the IP 650 platform in a software release. Support for the IP 650 is not currently included in this release.

.cfg File	Action	Parameter	Description
sip	added	up.backlight.onIntensity	This parameter controls the intensity of the LCD backlight when it turns on during normal use of the phone. Possible values are 0, 1, 2 or 3. 0 = off 1 to 3 = low, medium, high Null default is 3 (high).

.cfg File	Action	Parameter	Description
sip	added	up.backlight.idleIntensity	This parameter controls the intensity of the LCD backlight when the phone is idle Possible values are 0, 1, 2 or 3. 0 = off 1 to 3 = low, medium, high Null default is 1 (low). Note: If idleIntensity is set higher than onIntensity, it will be replaced with the onIntensity value.
sip	added	voice.codecPref.IP_650.G711Mu voice.codecPref.IP_650.G711A voice.codecPref.IP_650.G729AB voice.codecPref.IP_650.G722	These parameters allow the voice codec preference list to be set for the SoundPoint IP 650 phone. By default the G.722 codec is the first choice. The use of these parameters is the same as other voice.codecPref parameters.
sip	added	voice.audioProfile.G722.payloadSize voice.audioProfile.G722.jitterBufferMin voice.audioProfile.G722.jitterBufferMin voice.audioProfile.G722.jitterBufferMin	These parameters configure the G.722 voice codec. The use of them is the same as the other voice.audioProfile parameters.
sip	added	voice.gain.rx.analog.chassis.IP_650 voice.gain.rx.analog.ringer.IP_650 voice.gain.rx.digital.chassis.IP_650 voice.gain.rx.digital.ringer.IP_650 voice.gain.tx.analog.chassis.IP_650 voice.gain.tx.digital.chassis.IP_650	These parameters control gain settings which are specific to the SoundPoint IP 650 phone. The values should not be modified.

.cfg File	Action	Parameter	Description
sip	added	voice.rxEq.hs.IP_650.preFilter.enable voice.rxEq.hs.IP_650.postFilter.enable voice.rxEq.hd.IP_650.preFilter.enable voice.rxEq.hd.IP_650.postFilter.enable voice.rxEq.hf.IP_650.preFilter.enable voice.rxEq.hf.IP_650.postFilter.enable voice.txEq.hs.IP_650.preFilter.enable voice.txEq.hs.IP_650.postFilter.enable voice.txEq.hd.IP_650.preFilter.enable voice.txEq.hd.IP_650.postFilter.enable voice.txEq.hf.IP_650.preFilter.enable voice.txEq.hf.IP_650.postFilter.enable	These parameters control equalization settings which are specific to the SoundPoint IP 650 phone. The values should not be modified.
sip	added	voice.handset.rxag.adjust.IP_650 voice.handset.txag.adjust.IP_650 voice.handset.sidetone.adjust.IP_650 voice.headset.rxag.adjust.IP_650 voice.headset.txag.adjust.IP_650 voice.headset.sidetone.adjust.IP_650	These parameters control gain settings which are specific to the SoundPoint IP 650 phone. The values should not be modified.
sip	added	dir.local.volatile.8meg	This parameter applies only to platforms with 8 Mbytes of flash memory. It can be set to 0 or 1 and is 0 by default. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
sip	added	dir.local.nonVolatile.maxSize.8meg	This parameter applies only to platforms with 8 Mbytes of flash memory. It can be set from 1 to 100. The units are Kbytes and the default is 100. This is the maximum size of non-volatile storage that the directory will be permitted to consume.
sip	added	log.level.change.usb	This parameter is used to set the logging detail level for the “usb” module.

.cfg File	Action	Parameter	Description
sip	added	prov.fileSystem.ffs0.8meg.minFreeSpace	The minimum free space in Kbytes to reserve in the file system when downloading files from the boot server. It is recommended that this value should not be modified. The allowed range for this parameter is 5 to 512 and the default is 512.
sip	added	usb.enable	This parameter enables or disables the USB port on the phone. It can be set to 0 or 1. The Null default is 0.
sip	added	usb.bulkDrive.enable	This parameter enables or disables support for a USB bulk drive (“memory stick”) connected to the USB port on the phone. It can be set to 0 or 1. The Null default is 0.
sip	added	usb.bulkDrive.name	This parameter is a string which specifies the name of the mounted USB drive. The Null default is “usbDrive”.
sip	changed	dir.local.volatile.maxSize prov.fileSystem.rfs0.minFreeSpace ramdisk.bytesPerBlock res.finder.sizeLimit res.finder.minFree res.quotas.x.value mb.limits.nodes mb.limits.cache	For the SoundPoint IP 650 platform only, the values specified by these parameters are replaced internally with double the value. This is because the SoundPoint IP 650 platform has 32 Mbytes of memory instead of 16 Mbytes.

2.33 Version 2.0.2

2.33.1 Added or Changed Features

- 8428: **Split call signaling processing from "lamp management" processing**
- 18356: **Emergency routing is not supported on shared lines**

2.33.2 Removed Features

None.

2.33.3 Corrections

The following issues have been resolved with this release:

- 6527: **Shared line does not ring if incoming call arrives when phone is playing dial tone then subsequently hangs up**
- 8542: **Phone does not display second call appearance in specific bridged line scenario**
- 8547: **Local ringback is not played if far end does blind transfer without going on hold**
- 15671: **Pressing a line key of a shared line when a call is remote-busy ends the call**
- 16662: **Shared line can not establish a call if there are two simultaneous incoming calls**
- 18435: **If two INVITE's come close together with SDP containing "a=ptime", the phone will crash**
- 18471: **Setting NAT IP address causes truncation or corruption of IP address in VIA**
- 18747: **INVITE failover does not work**

2.33.4 Configuration File Parameter Changes

None.

2.34 Version 2.0.1 B

2.34.1 Added or Changed Features

None.

2.34.2 Removed Features

None.

2.34.3 Corrections

The following issues have been resolved with this release:

- 18358: **Malformed RTCP packets can crash Cisco gateways.**

2.34.4 Configuration File Parameter Changes

None.

2.35 Version 2.0.1

The 2.0.1 Release includes all the changes and corrections from Releases 1.6.6 and 1.6.7

2.35.1 Added or Changed Features

- 8072: **Added Nortel MCP NAT traversal parameters to config files**
- 11678: **Added template support in master configuration file**
- 16399: **Changed behavior when there is an incoming call on a phone – idle dial digits are no longer cleared when an incoming call is received**
- 16645: **Added support for NAT keep-alive**
- 17412: **Added ability to set Ethernet link mode to SoundPoint IP 430**
- 17413: **Added ability to set Ethernet link mode to SoundStation IP 4000**

2.35.2 Removed Features

- 14275: **call.callWaiting.prompt has no effect**
  This parameter has been removed from the configuration files because it is no longer used.

2.35.3 Corrections

The following issues have been resolved with this release:

- 7723: **Name of net logging module is sometimes corrupted in log file**
- 12337: **Display of SoundPoint IP 430 flickers under fluorescent lights and may be shifted vertically by a few pixels**
- 12382: **The phone will freeze if the DNS server address is all zeroes and the phone uses a FQDN server name**
- 12647: **Feature keys cannot be reconfigured to perform other functions**
- 12749: **Phone locks up during CERT PROTOS testing**
- 15138: **Text in line labels on SoundPoint IP 430 should be moved one pixel left**
- 15227: **Phone model of SoundPoint IP 430 is incorrect in CDP packets**

- 15311: **Contrast adjustment range on the SoundPoint IP 430 is unsuitable**
- 15729: **Phone does not retry connecting to boot server in specific scenario**
- 15731: **Phone should use Office Communicator model to update LCS presence status when multiple endpoints share same registration**
- 15812: **Phone doesn't handle simultaneous 200/OK and CANCEL race condition**
- 16069: **When using Russian dictionary, phone reboots after exiting the DHCP Menu**
- 16073: **Phone does not clear indicators if BLF removed on server**
- 16311: **Phone with maximum number of line keys configured may have its line key labels overwritten by roaming buddy records**
- 16373: **Local conference host cannot end conference if one leg is put on hold by far end**
- 16562: **Expansion Module may reboot if the Do Not Disturb key on the phone is pressed multiple times while the Expansion Module is booting up**
- 16577: **Local conference host cannot end conference if first leg was put on hold by far end when conference was created**
- 16659: **To: and Refer-to: domains incorrect during failover**
- 16681: **In some scenarios a phone may initiate a call using TCP but send an ACK using UDP**
- 16768: **Inconsistent backlight behavior on SoundStation IP 4000 when resuming a call or conference**
- 16904: **Excessive logging from "soem" module at boot time in some scenarios involving Expansion Module**
- 17009: **Non-numeric characters or an invalid IP address when dialing by IP may cause the phone to reboot**
- 17068: **If the silent ringer is selected, an incoming call can only be answered in hands free mode**
- 17102: **SoundPoint IP 430 phone locks up instead of rebooting after detecting an operating system suspended task [bug 17037]**
- 17188: **"Time" information in placed call list contains incorrect data after a transfer has been done**

- 17257: **Phone loses audio when there is an active call on headset and another incoming call is rejected**
- 17206: **Local conference host cannot end conference if both legs are put on hold by far ends**
- 17242: **Local conference host's state changes to “held” when second leg holds and invalid soft keys are displayed**
- 17271: **Phone will not accept a call with a codec with a dynamic payload identifier**
- 17308: **Phone displays "In a meeting" status as "Away" when using LCS server**
- 17362: **Add or edit directory (speed dial) contact crashes phone when configured for roaming buddies**
- 17370: **Phone may reboot if LCS server is used and presence is enabled without having roaming buddies enabled**
  Note: If the LCS server is used, the roaming buddies parameter should be enabled
- 17457: **Phone may display incorrect soft keys if a digit is pressed then Menu, Directories or Messages is selected then de-selected**
- 17573: **In some scenarios, phone sends 603-Decline after 2 rings on SCA line**
- 17639: **Expansion Module updates should be continuously done in the background**
- 17656: **Phone does not handle outbound fragmented packets that are tagged for VLAN**
- 17706: **Phone may freeze after regaining connection with LCS server**
- 17783: **PRACK message goes directly between phones instead of via LCS server because of no record-route**
- 17797: **In some scenarios, phone sets its own presence status to 'Away' when using the LCS server**
- 17831: **In some scenarios, phone adds itself to its own buddy list when using the LCS server**
- 17976: **NTLM signature should include full "From:" URI**

2.35.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	removed	call.callWaiting.prompt	
sip	removed	sec.srtp.offer, sec.srtp.require, sec.srtp.key.lifetime	
sip	added	voIpProt.SIP.pingInterval	This parameter is used together with reg.x.proxyRequire. It specifies the number of seconds between PING messages sent by the phone. Default = 0 = disabled. Possible range is 0 to 3600. Note: Server support is required before this feature can be used.
sip	added	res.finder.minFree	This parameter is used to ensure that the phone will not download resources which could leave it with insufficient memory to function correctly. A resource will not be downloaded if the phone has less memory free than res.finder.minFree [kBytes]. This parameter can have the values 1 to 2048. The recommended configuration file value is 1200. If the parameter is left empty the default is 800. Notes: Setting this value too small may affect functionality of the phone. Setting this value too large may mean that some resources are not downloaded at boot time.

.cfg File	Action	Parameter	Description
phone1	added	reg.x.proxyRequire	This parameter is used together with voIpProt.SIP.pingInterval. It specifies the string which is put in the "Proxy-Require" header. Default is an empty string which means no "Proxy-Require" will be sent. Note: Server support is required before this feature can be used.
phone1	added	nat.keepalive.interval	This parameter is used to set the interval in seconds at which phones will send a keep-alive packet to the gateway/NAT device to keep the communication port open so that NAT can continue to function as set up initially. Default value is 0 which means the feature is disabled. The allowable range is 0 to 3600.

2.36 Version 2.0.0 (Beta Release Only)

Note: The 2.0.0 Release does not include the changes and corrections from SIP releases 1.6.6 and 1.6.7

2.36.1 Added or Changed Features

- 2236: **Added support for TLS protocol**
- 2307: **When the phone reboots due to a fatal error, it should first log any useful information**
- 5403: **Added support for the NTLM authentication protocol**
- 5404: **Added support for Microsoft Live Communications Server authentication schemes**
- 8817: **Added support for BLF SCA mode**
- 9110: **Added support for platform-specific override strings in dictionaries to allow abbreviated strings for certain platforms**

- 9734: **Added option to select which registration to use for "presence" signaling**
- 11646: **Added IP QoS support for DSCP (DiffServ)**
- 11785: **Added support for multiple redundant provisioning servers**
- 12270: **SIP re-registration interval is now configurable**
- 12419: **Added support for Broadsoft attendant console/BLF feature**
- 12426: **Added support for peer-to-peer calls using Microsoft Live Communications Server 2005**
- 12427: **Added support for calling to and from Windows Messenger 5.1 and Office Communicator using Microsoft Live Communications Server 2005**
- 12938: **Added caching of the state of the message-waiting indicator LED across controlled reboots**
- 13038: **Changed “DNS Lookup” name to “Transport” in SIP Configuration menu and on web interface to match parameter name in sip.cfg**
- 13080: **Added new consultative transfer behavior so that transfer automatically completes when originator hangs up**
- 13100: **Added support for individual configuration of secondary dial tone**
- 13315: **Increased the maximum number of buddies to 8 for all platforms except SoundPoint IP 600 and 601 which can watch 48 buddies**
- 13317: **Increased speed dial menu size limit to 99 for all platforms**
- 13463: **Added IM support with Office Communicator and Windows Messenger 5.1 in Microsoft Live Communications Server 2005 context**
- 13509: **Added support for reg.x.address configuration parameter to contain host part**
- 13552: **Improved boot-up logging**
- 13613: **Improved support for multiple m lines in SDP**
- 13813: **Added the ability for file transfers to attempt to contact multiple IP addresses per DNS name**
- 13893: **Re-enabled idle micro browser configuration**
- 14029: **Lowered CPU load associated with RTP processing**
- 14209: **Added support for getting buddy lists from Microsoft Live Communications Server 2005**

- 14322: **Added per-registration "lcs" parameters**
- 14323: **Added per-registration outbound proxy parameters**
- 14348: **Added support for connection reuse draft**
- 14496: **Added presence support with Windows Messenger 5.1 / Office Communicator in Microsoft Live Communications Server 2005 context**
- 14498: **Added Windows Messenger 5.1 / Office Communicator-compatible presence and IM support in peer-to-peer mode**
- 14556: **Added support for roaming access control lists**
- 14610: **Added ability to store resource files listed in MISC_FILES field in <*Ethernet Address*>.cfg in flash file system. For example a dictionary file can be listed which should be used if the phone reboots when the boot server is unavailable.**
- 14628: **Added support for populating the speed dial list from a roaming buddies list sent by a Microsoft Live Communications Server 2005**
- 14638: **Changed source port for TCP/TLS connection to be a random value above 32766 after each reboot**
- 15180: **Added configurable maximum number of servers for redundant boot server feature (11785)**
- 15363: **Changed call timer format**
- 15644: **Added a configuration parameter to choose the name of "pval" field in Dialog**
- 15987: **Reduced default resource quota limits for tones**
- 16047: **Added configurable ms-forking support and reject IM when it is enabled**

2.36.2 Removed Features

- 12109: **Removed configuration parameters for localized call progress tones menu** In order to still use this feature, see details in **Error! Reference source not found.** REF _Ref146520523 \h * MERGEFORMAT **Error! Reference source not found.**.
- 13447: **Removed presence and IM support for Windows Messenger 4.6, 4.7 and 5.0**
- 12350: **Removed compiled-in Polycom idle display indicator bitmap**

2.36.3 Corrections

The following issues have been resolved with this release:

- 6078: **Cannot adjust the volume of the reorder tone when attempting to seize a shared line which is remotely active**
- 7084: **Transducer indicator is not cleared after blind transfer on some platforms**
- 9292: **IP 4000 reboots upon downloading a wave file with a path containing '\' instead of '/'**
- 9709: **RTCP not sent or received when calls are on hold**
- 9815: **SoundStation IP 4000 cannot change language after already changing language 10 to 12 times**
- 11177: **Fast-Busy sound effect sequencing wrong in specific scenario when call on hold**
- 11588: **The local contact directory feature cannot be disabled**
- 11952: **If destination phone rejects a blind transferred call, the far end does not hear a busy tone**
- 12020: **Bridged line with multiple line keys may have one line indicator left in the remote active state if a peer bridged line hosts a centralized conference**
- 12043: **Label of CPU Load graph does not change when DSP load is displayed**
- 12106: **Address of boot server is truncated in Configuration menu on SoundPoint IP 500 and 501 phones when it exceeds a certain length**
- 12155: **SoundPoint IP 300 and 301 phones have no "Exit" soft key during the ACD login process**
- 12308: **Cannot place a call from the second line on the phone if the first line is a shared unregistered line**
- 12492: **SoundPoint IP 601 phone with Expansion Module(s) attached may fail to load the selected language after rebooting**
- 12630: **When a shared line is being used on another phone, pressing the line key for that line can cause the display to show "Enter number" briefly**
- 12711: **Phone should play default ring tone if Alert-Info URL is invalid**
- 12952: **There is no way to reset the user password back to the factory default password**

- 13230: **No audio on calls resumed from hold in some multiple call scenarios**
- 13253: **An unregistered SoundStation IP 4000 may reboot if an invalid number is dialed**
- 13320: **When the micro browser fetches SSL data this can interrupt audio transmitted by the phone**
- 13358: **My Status menu has two “offline” entries**
- 13477: **Pressing Hold/Resume soft key twice quickly results in three effective state changes**
- 13500: **Phone does not use FTP password stored in flash when OVERRIDES_DIRECTORY and CONTACTS_DIRECTORY are configured in this format: "FTP://usr@IP/directory"**
- 13512: **Parsing of URLs in configuration files does not work for some categories of URLs**
- 13579: **SDP parser applies wrong logic**
- 13793: **cnonce generated by the phone is not random**
- 13933: **Directory menu display is not perfectly cleaned up after deleting all contacts**
- 14069: **Phone may behave incorrectly if an incoming call is answered on a shared line when another phone sharing the line has Do Not Disturb enabled**
- 14083: **Wrong expire time might be used when there are multiple contact header lines**
- 14126: **If a call is placed to a phone with an unread IM, the message-waiting indicator LED stops flashing**
- 14172: **Phone will reboot when a contact is added to the contact directory which already contains over 40 contacts which are being watched**
- 14390: **Changing the DNS server configuration via the phone’s menu does not have any effect**
- 14400: **Phone can take up to 30 minutes to boot when there are TCP timeouts**
- 14408: **Soft key labels do not get updated correctly after hot dial attempt when remote shared line is busy**
- 14467: **If a URL in <*Ethernet Address*>.cfg specifies a protocol and user name but no password, the password in flash is not used**

- 14635: **No welcome sound effect is played on SoundStation IP 4000 phone**
- 14664: **SoundPoint IP 301 and 501 and SoundStation IP 4000 phones fail during a reboot if 12 SAS-VP appearances are configured**
- 14781: **Cannot use special characters for filenames with TFTP boot server**
- 14844: **A failed download of a pre-existing file causes that file to be deleted**
- 14858: **Phone reboots if idle micro browser is running and the Status – Platform - Application menu is displayed**
- 15007: **If the server address flash parameter is a URL which specifies a protocol and user name but not password, the password in flash is not used**
- 15101: **Provisioning of phone stalled forever in specific scenario**
- 15145: **SAS-VP feature does not work correctly when the filename parameter is empty**
- 15154: **Phone does not behave correctly when it is disconnected from the network and is using SAS-VP**
- 15185: **Editing problems exist with long strings**
- 15214: **Headset memory indicator is not restored after adjusting volume on some platforms**
- 15269: **When tcpIpApp.sntp.gmtOffset.overrideDHCP is set but no override value is given, the DHCP based offset is not applied**
- 15351: **Blind transfer does not drop unless server sends signaling to drop the call on the originator's phone. Problem will occur in pure proxy scenarios only.**
- 15368: **Character appears to be deleted during editing**
- 15412: **TFTP URL of configuration file name in log file may be truncated**
- 15455: **Phone should not reboot if parameters are missing from flash file system**
- 15463: **Phone's presence status is not displayed on UI on SoundPoint IP 300 and 301 phones**
- 15554: **Problems with password entry for very long passwords**
- 15561: **Phone may reboot after entering a long incorrect password**
- 15571: **Phone cannot recover in several scenarios involving transferring mixed URL and E.164 calls**
- 15603: **The 'sip:' field name which appears when using IP dialing should not be**

deletable

- 15679: **Ring Type 12 (Ringback-style) sounds incomplete after the first ring**
- 15694: **Phone crashes and reboots when 'Exit' is pressed from Network Configuration menu in Korean Language**
- 15730: **If a menu is displayed when a call is missed on the SoundPoint IP 300 and 301 phones, the missed call count is not updated on the idle display**
- 15766: **Display is incorrect after selecting name dialing then entering and exiting a call list while dial tone is playing**
- 15781: **After putting a local conference on hold then splitting the calls then joining them, the first call may remain on hold**
- 15855: **In the Instant Msg menu of the SoundPoint IP 300 and 301 phones, "x/Ascii" is not displayed after pressing the "1/A/a" softkey**

2.36.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.server.x.expires.overlap	The number of seconds before the expiration time returned by server 'x' at which the phone should try to re-register. The phone will try to re-register at half the expiration time returned by the server if that value is less than the configured overlap value. Default = 60. Minimum = 5, maximum = 65535.

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.ms-forking	Default = 0. Can be 0 or 1. 0 = Support for MS-forking is disabled. 1 = Support for MS-forking is enabled and the phone will reject all Instant Message INVITEs. This parameter is relevant for LCS server installations. Note that if any endpoint registered to the same account has MS-forking disabled, all other endpoints default back to non-forking mode. Windows Messenger does not use MS-forking so be aware of this behavior if one of the endpoints is Windows Messenger.
sip	added	voIpProt.SIP.dialog.usePvalue	Default = 0. Can be 0 or 1. 0 = Phone uses “pval” field name in Dialog. This obeys the draft-ietf-sipping-dialog-package-06.txt draft. 1 = Phone uses a field name of “pvalue”.
sip	added	voIpProt.SIP.connectionReuse.useAlias	Default = 0. Can be 0 or 1. 0 = old behaviour 1 = Phone uses the connection reuse draft which introduces "alias".
sip	added	se.pat.callProg.15.name="secondary dial" se.pat.callProg.15.inst.1.type="chord" se.pat.callProg.15.inst.1.value="1"	Same configuration method as primary dial tone. Allows a different tone to be configured for secondary dial tone.
sip	added	qos.ip.rtp.dscp	This parameter allows the DSCP of packets to be specified. If set to a value this will override the other qos.ip.rtp… parameters. Default is Null which means the other qos.ip.rtp… parameters will be used. Possible values are 0 to 63, EF, AF11, AF12, AF13, AF21, AF22, AF23, AF31, AF32, AF33, AF41, AF42 or AF43.

.cfg File	Action	Parameter	Description
sip	added	qos.ip.callControl.dscp	This parameter allows the DSCP of packets to be specified. If set to a value this will override the other qos.ip.callControl… parameters. Default is Null which means the other qos.ip.callControl… parameters will be used. Possible values are 0 to 63, EF, AF11, AF12, AF13, AF21, AF22, AF23, AF31, AF32, AF33, AF41, AF42 or AF43.
sip	added	pres.reg	Default = 1. Can be 1, 2, 3, …. Must be a valid line/registration number. If the number is not a valid line/registration number, it is ignored. Specifies the line/registration number used to send SUBSCRIBE for presence.
sip	added	mb.idleDisplay.home	mb.idleDisplay.home can be empty or any fully formed valid HTTP URL. Length up to 255 characters. Default is empty. This specifies the URL used for the microBrowser idle display home page. Example: http://www.example.com/ xhtml/frontpage.cgi?page=home. If empty, there will be no micro Browser idle display feature.

.cfg File	Action	Parameter	Description
sip	added	mb.idleDisplay.refresh	Can be 0 or an integer greater than 5. Values from 1 to 4 will be ignored, and 5 will be used instead. Default = 0 This specifies the period in seconds between refreshes of the microBrowser idle display content. 0 = the idle display microBrowser is not refreshed. Note: If an HTTP Refresh header is detected, it will be respected, even if this parameter is set to 0. The use of this parameter in combination with the Refresh HTTP header may cause the idle display to refresh at unexpected times.
sip	removed	voIpProt.SIP.WM50	For selecting between Windows Messenger 4.7 and 5.0 (no longer supported).
sip	removed	lcl.ml.lang.cpt.x, lcl.cpt, lcl.cpt.menu.x, lcl.cpt.chord.cp.x.y.freq.z, feature.10.name = cpt-settings feature.10.enabled = 1	Removed the parameters used to configure the call progress tone localization menu. In order to still use this feature, the old configuration parameters should be added to the sip.cfg file and a new parameter, feature.cpt.enabled, must be added and set to 1.
sip	changed	tone.chord.ringer.46.offDur from 200 to 0, tone.chord.ringer.46.repeat from 1 to 2 Settings for se.pat.ringer.12	Changes to make ring type 12 work as expected.
sip	changed	voice.gain.tx.digital.chassis.IP_430 from -3 to 0 voice.handset.txag.adjust.IP_430 from 24 to 21	Gain corrections for SoundPoint IP 430 platform.

.cfg File	Action	Parameter	Description
sip	changed	bitmap.IP_400.61.name from IdleDefault to “” bitmap.IP_500.61.name from IdleDefault to “” bitmap.IP_600.65.name from IdleDefault to “” bitmap.IP_4000.66.name from IdleDefault to “”	Removed compiled-in Polycom idle display indicator bitmap.
sip	changed	HEADSET_MEM IP_300 indicator to use indicator #50 HEADSET_MEM IP_500 indicator to use indicator #50 ind.class.4.state.6.index from 48 to 50	Changed due to rearrangement of other indicators.
sip	changed	ind.anim.IP_400.38.frame.1.bitmap from IdleDefault to “” ind.anim.IP_500.38.frame.1.bitmap from IdleDefault to “” ind.anim.IP_500.39.frame.1.bitmap from IdleDefault to “” ind.anim.IP_600.38.frame.1.bitmap from IdleDefault to “” ind.anim.IP_600.39.frame.1.bitmap from IdleDefault to “” ind.anim.IP_4000.38.frame.1.bitmap from IdleDefault to “” ind.anim.IP_4000.39.frame.1.bitmap from IdleDefault to “”	Removed compiled-in Polycom idle display indicator bitmap.
sip	changed	res.quotas.1.value from 2000 to 600	Reduced default resource quota limits for tones.
phone1	added	reg.x.lcs	Default = 0. Can be 0 or 1. If set to 1 the LCS server is supported for registration ‘x’.
phone1	added	reg.x.server.y.expires.overlap	Same interpretation as voIpProt.server.y.expires.overlap for registration ‘x’.

.cfg File	Action	Parameter	Description
phone1	added	reg.x.outboundProxy.address	Same interpretation as voipProt.SIP.outboundProxy.address for registration 'x'.
phone1	added	reg.x.outboundProxy.port	Same interpretation as voipProt.SIP.outboundProxy.port for registration 'x'.
phone1	added	reg.x.outboundProxy.transport	Same interpretation as voipProt.SIP.outboundProxy.transport for registration 'x'.
phone1	added	attendant.uri	For attendant console / BLF feature. This specifies the list SIP URI on the server. If this is just a user part, the URI is constructed with the server host name/IP
phone1	added	attendant.reg	For attendant console / BLF feature. This is the index of the registration which will be used to send a SUBSCRIBE to the list SIP URI specified in attendant.uri. For example, attendant.reg = 2 means the second registration will be used.
phone1	added	roaming_buddies.reg	Specifies the line/registration number which has roaming buddies support enabled. Default is empty which means roaming buddies is disabled. If value < 1 then value is replaced with 1. This parameter is relevant for LCS server installations.
phone1	added	roaming_privacy.reg	Specifies the line/registration number which has roaming privacy support enabled. Default is empty which means roaming privacy is disabled. If value < 1 then value is replaced with 1. This parameter is relevant for LCS server installations.

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.
- 27469: **Local Conferencing on IP4000 phones is disabled if G.729 is in the Codec preference list**
  Workaround: Disable G.729 as a Codec option on the phone by setting voice.codecPref.IP_4000.G729AB=""
- 27777: **SoundStation IP 4000 does not play a local hold reminder tone**
  Workaround: None
- 28508: **Phone crashes after receiving high call rate (4 unanswered calls every 18 seconds)**
  Workaround: Reduce the incoming call rate.
- 29344: **HTTP Digest Authentication does not work on IIS**
  Workaround: Use a different form of authentication, a different protocol or a different server
- 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.
- 33063: **Active FTP mode is not supported for phone provisioning**
  Workaround: Configure the ftp server for Passive FTP operation.
- 33445: **LCS Presence and dialing from Buddy Lists does not work across 'Federations'**

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

- 33593: **Shared line does not show remote active for the second incoming call if callsPerLineKey parameter is set to 1**
  Workaround: Set callsPerLineKey parameter to a value greater than 1.
- 34454: **If microbrowser is enabled and refreshes are too frequent and pages contain large images, the phone may crash. Issue is most apparent on SoundPoint IP 601 phones**
  Workaround: Do not refresh Microbrowser too frequently in configuration settings or by rapidly pressing the Refresh softkey. Design the pages so that the content is within reasonable limits.
- 34743: **A phone may freeze when it receives a check-sync if the resources on the phone are heavily used by downloaded 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.
- 36969: **SoundStation IP 4000 and IP 6000 phones do not display Japanese language properly.**
  Workaround: None.
- 37175: **If configuration files are used to set the SNTP server address, date validity checking on CA certificates will be ignored for https provisioning.**
  Workaround: Set the SNTP server address through the Phone UI or use DHCP to inform the phone of the SNTP server address.
- 37273: **If the Custom Idle Display and Idle Browser features are both enabled the phone UI displays incorrectly.**
  Workaround: Do not set ind.idleDisplay.enabled="1" and enable the Idle Browser at the same time.
- 37437: **When SRTP is used with both Authentication and Encryption enabled on SoundPoint IP 301, 501, 600 and 601 platforms, and three-way conferencing is enabled the phone will re-boot when a local conference is attempted.**
  Workaround: Disable local conferencing by setting sec.srtp.leg.allowLocalConf="0" (this is the default setting) or disable SRTP Authentication. See Technical Bulletin 25751 for details.

- 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.
- 45143: **Centralized Conference: When maximum conference size is reached phone displays the local conference UI**
  Workaround: None
- 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.**
  Workaround: None
- 47827: **VQMon: Inter-arrival jitter is being reported in RTP timestamp units instead of in milliseconds.**
  Workaround: None
- 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.**
  Workaround: None
- 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.**
  Workaround: Refine the Advanced Find search criteria until the total entries that match is less than the configured 'pageSize'.
- 50153: **Corporate Directory: Setting the Primary Attribute as 'sticky' (dir.corp.attribute.1.sticky="1") can give confusing user interface behavior.**
  Workaround: Configure primary attribute as 'non-sticky'; dir.corp.attribute.1.sticky="0"
- 54027: **SRTP key lifetime: Phones receiving calls do not re-invite with key at key's half-life.**
  Workaround: None
- 54028: **SRTP key lifetime: Key changes do not appear to be correct when multiple encryption suites are enabled.**
  Workaround: None
- 63123: **Instead of initiating a new call, attendant phone plays reorder tone when the BLF line key is pressed for the second time.**
  Workaround: None
- 65133: **SoundPoint IP3xx: Cannot invoke the Redial feature after making a call**

and entering an account code.
Workaround: None

- 75229: **Phones does not handle the 503 response properly when trying to join more calls on centralized conference beyond the server limit. This could result in displaying the local conference UI on the phone.**
  Workaround: None

4. Reference Documents

1. Administrator*'s Guide for* the Polycom® SIP Software – Version 3.1.0
http://support.polycom.com/PolycomService/support/us/support/voice/index.html

2. 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

3. Technical Bulletins and Quick Tips (including the following that are new or updated relating to this release) – may be obtained from the Polycom Support web-site at:
http://support.polycom.com/PolycomService/support/us/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html

4. User Guides can be downloaded from the following support web pages:
SSIP -
http://support.polycom.com/PolycomService/support/us/support/voice/soundstation_ip_series/index.html

SPIP -

http://support.polycom.com/PolycomService/support/us/support/voice/soundpoint_ip/index.html