Release Notes

SIP Application

SoundPoint® and SoundStation® IP

Version 2.1.2
14 June 2007

Part Number 3804-11530-212

Table of Contents

1. General

These release notes apply to version 2.1.2 of the SoundPoint IP SIP application.

This release is a maintenance patch release of the SIP software for Polycom SoundPoint IP phones and replaces the SIP 2.1.1RevC release as the most recent GA release available.

WARNING: The Server Redundancy Behavior in SIP2.1 has changed from that implemented in prior releases. Prior to SIP 2.1 the reg.x.server.y parameters (see section 4.6.2.1 of the SIP 2.0 Administrator's Guide) could be used for fail-over configuration The older behavior is no longer supported. Customers that are using the reg.x.server.y. configuration parameters where y>=2 should take care to ensure that their current implementations are not adversely affected. For example the phone will only attempt advanced SIP features such as Shared Lines, Missed Calls, Presence with the Primary Server (y=1). Refer to Tech Bulletin TB5844 SIP Server fallback for more details.

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

1.1 System Requirements

Platform	BootROM version
SoundPoint IP 300	2.6.1 or greater
SoundPoint IP 301	2.6.1 or greater
SoundPoint IP 320	3.2.3RevB or greater
SoundPoint IP 330	3.2.3RevB or greater
SoundPoint IP 430	3.1.3 or greater
SoundPoint IP 500	2.6.1 or greater
SoundPoint IP 501	2.6.1 or greater
SoundPoint IP 550	3.2.3 or greater
SoundPoint IP 600	2.6.1 or greater
SoundPoint IP 601	3.1.0 or greater
SoundPoint IP 650	3.2.2RevB or greater
SoundStation IP 4000	3.1.2 or greater

1.2 Distribution Files

The following files constitute the 2.1.2 distribution of the SoundPoint / SoundStation IP SIP application. For centrally provisioned systems, copy these files to the boot server, maintaining the folder hierarchy present in the zip file.

Some of the configuration files must be modified. Refer to the *Administrator Guide* for details.

Files	Description
sip.ld	SIP application executable, App Version 2.1.2.0049 for SoundPoint IP 320, 330 2.1.2.0078 for all other platforms
sip.cfg	main core and SIP configuration file
phone1.cfg	example per-phone SIP configuration
000000000000.cfg	example master configuration file
000000000000-directory~.xml	example per-phone local contact directory XML file (edit and then remove '~' from name to seed phones which have no directory)
SoundPointIP-dictionary.xml	dictionary files for multilingual support include (no IP 30X support): Chinese, China (for IP 6XX, IP 550 and 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 6XX, IP 550 and IP 4000 only) Korean, Korea (for IP 6XX, IP 550 and IP 4000 only) Norwegian, Norway Portuguese, Portugal Russian, Russia Spanish, Spain Swedish, Sweden
SoundPointIPWelcome.wav	start up welcome sound effect

2. Changes

2.1 Version 2.1.2

2.1.1 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**

2.1.2 Removed Features

None.

2.1.3 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**

2.1.4 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
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.

2.2 Version 2.1.1 C

2.2.1 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**

2.2.2 Removed Features

None.

2.2.3 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**

2.2.4 Configuration File Parameter Changes

None.

2.3 Version 2.1.1

2.3.1 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**

2.3.2 Removed Features

None.

2.3.3 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**

2.3.4 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.
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.

2.4 *Version 2.1.0*

2.4.1 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**

2.4.2 Removed Features

None.

2.4.3 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**

2.4.4 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”	
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.

2.5 Version 2.0.3 B

2.5.1 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**

2.5.2 Removed Features

None.

2.5.3 Corrections

The following issues have been resolved with this release:

None.

2.5.4 Configuration File Parameter Changes

None.

2.6 Version 2.0.3

2.6.1 Added or Changed Features

None

2.6.2 Removed Features

None.

2.6.3 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**

2.6.4 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).
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.
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.

.cfg File	Action	Parameter	Description
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.
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.

2.7 Version 2.0.2

2.7.1 Added or Changed Features

- 8428: **Split call signaling processing from "lamp management" processing**

- 18356: **Emergency routing is not supported on shared lines**

2.7.2 Removed Features

None.

2.7.3 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**

2.7.4 Configuration File Parameter Changes

None.

2.8 Version 2.0.1 B

2.8.1 Added or Changed Features

None.

2.8.2 Removed Features

None.

2.8.3 Corrections

The following issues have been resolved with this release:

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

2.8.4 Configuration File Parameter Changes

None.

2.9 Version 2.0.1

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

2.9.1 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**

2.9.2 Removed Features

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

2.9.3 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**

2.9.4 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.

2.10 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

2.10.1 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**

2.10.2 Removed Features

- 12109: **Removed configuration parameters for localized call progress tones menu**
  In order to still use this feature, see details in 3.1 Upgrading.
- 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**

2.10.3 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**

2.10.4 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.
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.

.cfg File	Action	Parameter	Description
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.
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.
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).

.cfg File	Action	Parameter	Description
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.
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’.
phone1	added	reg.x.outboundProxy.address	Same interpretation as voipProt.SIP.outboundProxy.address for registration ‘x’.

.cfg File	Action	Parameter	Description
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.

2.11 Version 1.6.7

2.11.1 Added or Changed Features

- 15930: **Added ability to set Ethernet link mode on SoundPoint IP 601**
- 15981: **Added menu options for setting Ethernet link mode on SoundPoint IP 601**
- 16376: **Improved response time of phone to SIP messages**
- 16482: **Added option for phone to be more assertive in negotiating the preferred codec**
- 16500: **Added configurable line-seize behavior**

2.11.2 Removed Features

None.

2.11.3 Corrections

- 16027: **When connecting to voicemail in specific scenario, phone may have no audio**

- 16075: **Phone plays re-order tone when taking call off hold in specific scenario**
- 16100: **BLA line key status is not maintained in specific scenario**
- 16116: **Cannot register lines 7 to 12 from SIP configuration menu**
- 16149: **Line key LEDs for BLA lines can switch from one line key to another in specific scenario**
- 16250: **Comfort noise received by phone is handled incorrectly**
- 16374: **Phone keeps sending NOTIFY if 481 received in early NOTIFY**
- 16388: **Removed DC bias from Tx signal**
- 16429: **Web interface does not have configuration options for lines 7 to 12**
- 16459: **Phone is unable to park a call that is received via ACD final destination**
- 16480: **BLA Led gets stuck and there is a phantom NOTIFY from the phone in a particular scenario.**
- 16485: **Notify Talk is ignored if interval between it and 180 is too brief**
- 16565: **Dialed digits can be lost if they are dialed too quickly after selecting an SCA line**
- 16599: **SoundPoint IP 300 and 301 phones reboot when using G.729 codec in a conference call with SIP 1.6.6 C software**
- 16660: **Failover to backup SIP server does not occur when hostname of primary cannot be resolved via DNS**
- 16691: **Dialog does not get removed after its expiration time in some scenarios. This addresses #16374 and #16480.**
- 16813: **Going on and off hook repeatedly on a shared line may result in the line showing an active call state when the handset is physically on-hook**
- 16915: **Phone sends SIP requests to port 5060 regardless of voIpProt.SIP.outboundProxy.port configuration setting**
- 17014: **When a shared line call is on hold, using on-hook dialing seizes the last used line instead of the first available line**
- 17284: **An unnecessary ACK is sent by the phone if no reply is received within 32 seconds**

2.11.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SDP.answer.useLocalPreferences	Can be 0 or 1. Use this new parameter to have the phone use its own preference list when deciding which codec to use rather than the preference list in the offer. Null default = 0 = disabled.

.cfg File	Action	Parameter	Description
sip	added	call.stickyAutoLineSeize	Can be 0 or 1. Set to 1 to make the phone use "sticky" line seize behavior. This will help with features that need a second call object to work with. The phone will attempt to initiate a new outgoing call on the same SIP line that is currently in focus on the LCD (this was the behavior in SIP 1.6.5). This may fail due to glare issues in which case the phone may select a different available line for the call. Null default = 0 = disabled (this was the behavior in SIP 1.6.6).

2.12 Version 1.6.6 C (Limited Distribution)

2.12.1 Added or Changed Features

None.

2.12.2 Removed Features

None.

2.12.3 Corrections

- 16250: **Comfort noise received by phone is handled incorrectly. Fixed for SoundPoint IP 300, 301, 500, 501, 600 and 601 phones.**
- 16388: **DC bias should be removed from Tx signal on SoundPoint IP 300, 301, 500, 501, 600 and 601 phones**

2.12.4 Configuration File Parameter Changes

None.

2.13 Version 1.6.6 B

2.13.1 Added or Changed Features

- **Add Support for SoundPoint IP 430 hardware platform**

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_430, voice.gain.rx.analog.ringer.IP_430, voice.gain.rx.digital.chassis.IP_430, voice.gain.rx.digital.ringer.IP_430, voice.gain.tx.analog.chassis.IP_430, voice.gain.tx.digital.chassis.IP_430, voice.gain.tx.analog.preamp.chassis.IP_430	New gain parameters for SoundPoint IP 430 platform.
sip	added	voice.rxEq.hs.IP_430.preFilter.enable, voice.rxEq.hs.IP_430.postFilter.enable, voice.rxEq.hd.IP_430.preFilter.enable, voice.rxEq.hd.IP_430.postFilter.enable, voice.rxEq.hf.IP_430.preFilter.enable, voice.rxEq.hf.IP_430.postFilter.enable	New Rx EQ parameters for SoundPoint IP 430 platform.
sip	added	voice.txEq.hs.IP_430.preFilter.enable, voice.txEq.hs.IP_430.postFilter.enable, voice.txEq.hd.IP_430.preFilter.enable, voice.txEq.hd.IP_430.postFilter.enable, voice.txEq.hf.IP_430.preFilter.enable, voice.txEq.hf.IP_430.postFilter.enable	New Tx EQ parameters for SoundPoint IP 430 platform.
sip	added	voice.handset.rxag.adjust.IP_430, voice.handset.txag.adjust.IP_430, voice.handset.sidetone.adjust.IP_430, voice.headset.rxag.adjust.IP_430, voice.headset.txag.adjust.IP_430, voice.headset.sidetone.adjust.IP_430	New handset and headset gain adjustments for SoundPoint IP 430 platform.
sip	added	font.IP_400.1.name	New dynamic font download parameter for SoundPoint IP 430 platform.
sip	added	bitmap.IP_400.61.name	New bitmap parameter for SoundPoint IP 430 platform.
sip	added	ind.anim.IP_400.38.frame.1.bitmap, ind.anim.IP_400.38.frame.1.duration	New animation parameters for SoundPoint IP 430 platform.
sip	changed	ind.gi.IP_400…	Changed the values of some of these indicator parameters for the SoundPoint IP 430 platform.

2.14 Version 1.6.6

2.14.1 Added or Changed Features

- 15491: **Added configurable option to enable phone with BLA to send re-INVITE during conference setup**
- 13315: **Increased the maximum number of buddies to 8 for all platforms except SoundPoint IP 600 and 601 which can watch 48 buddies**

2.14.2 Removed Features

None.

2.14.3 Corrections

The following issues have been resolved with this release:

- 11658: **Phone continues to append to log file on FTP boot server after that file has reached its configured size limit**
- 12613: **SoundPoint IP600 and 601 phones may establish a call with no audio after holding, resuming and ending multiple calls**
- 12949: **If the phone's first line is a shared line and cannot obtain dial tone, pressing the "NewCall" soft key does not activate the first available line**
- 14673: **Special characters such as '@', ':' and '?' are not accepted as part of the FTP or HTTP password**
- 14968: **If the phone reboots, the app.log size can increase past the size limit**
- 15002: **If the phone's first line is unregistered, pressing the "NewCall" soft key does not activate another line**
- 15127: **Phone may have one-way audio in a call after multiple transfers have been done**
- 15218: **If multiple contact header fields contain multiple expire values, the phone does not always pick the lowest non-zero value**
- 15235: **Phone will freeze if the SAS-VP server becomes unavailable when the phone application is starting**
- 15339: **ACK lacks the same authorization credentials as the INVITE which is a failure to comply with RFC 3261**
- 15419: **Blind transfer doesn't work for URL calling**
- 15568: **A comma in quotes in SIP address headers should be interpreted correctly**
- 15596: **Remote phone can force local conference host to resume call unexpectedly in specific scenario**
- 15615: **When a shared line call is on hold, lifting the handset seizes the last used line instead of the first available line**
- 14939: **Shared line user must press "Answer" soft key twice to answer an incoming call in some scenarios**
- 15907: **After a reboot, a phone may show "1 new missed call" which can't be cleared until another call is missed**
- 15982: **The SDP session identifier should not be changed on each re-INVITE**
- 16021: **FTP downloads may fail because incorrect timeouts are used**
- 16141: **Phone with a shared line loses hot dialed digits when remote shared line changes state, such as placing an active call on hold**

- 16161: **Phone with a shared line displays the wrong soft key labels after attempting to hot dial when the remote shared line is in use**

2.14.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	call.shared.exposeAutoHolds	call.shared.exposeAutoHolds="1" means that on a shared line, when setting up a conference, a re-INVITE will be sent to the server. call.shared.exposeAutoHolds="0" means no re-INVITE will be sent to the server. Default is “0”.

2.15 Version 1.6.5

2.15.1 Added or Changed Features

- 8072: **Added support for Nortel MCP NAT traversal**
- 11805: **Changed behavior when a local conference is terminated. The remote conference legs are transferred so that the remote parties can continue the conversation.**
- 13193: **Added configuration options to allow configuration file parameters to override DHCP values for SNTP server address and GMT offset**
- 13527: **Added support for setting SIP server address from DHCP option 151**
- 13509: **Added allowing reg.x.address to contain host part instead of being a user part only**
- 13492: **CA certificate expiry is no longer checked if SNTP has not been configured**
- 14052: **Added flash parameter for SoundPoint IP 601phones to toggle power requirements in CDP between 5W (no Expansion Modules can be connected) and 12W (three Expansion Modules can be connected) with a default setting of 5W**
  This “EM Power” flash parameter is accessible when the SIP application is running under the Network Configuration menu. Note that no Expansion Modules can be connected to the phone when the “EM Power” parameter is disabled. The default setting for this parameter is Enabled (i.e. 12W power requirement). In order for the correct CDP power requirements to be reported at boot time as well, bootROM version 3.1.3 is required. See Tech Bulletin TB14052 for details on how to use this feature.
- 14886: **Changed power reported via CDP to platform-specific values**
  In order for these CDP power requirements to be reported at boot time as well, bootROM version 3.1.3 is required.

- 15012: **Added a workaround to restart the application on the phone if many tasks get unrealistic task delays during startup (Outstanding issue 11653)**

2.15.2 Removed Features

None.

2.15.3 Corrections

The following issues have been resolved with this release:

- 11264: **SoundStation IP 4000 hangs when booting if custom DHCP option 150 of type String is used**
- 11302: **SoundPoint IP 300 and 301 incorrectly truncate displayed line label if the reg.x.label field is empty and reg.x.address is longer than 4 characters**
- 13904: **SoundStation IP 4000 always shows LAN Mode as half-duplex**
- 14077: **Under certain DNS failover conditions, the phone stops sending DNS and SIP requests**
- 14110: **Phone does not reset to using "All Certificates" for CA Certificates after the user chooses the Reset Device Settings menu option**
- 14163: **Phone incorrectly updates Placed Calls list with an empty entry after New Call then End Call are pressed**
- 14166: **Calls answered on a phone with a shared line are incorrectly logged in the Received Calls list of another phone sharing that line**
- 14474: **Phone won't upload all log files to TFTP boot server if LOG_FILE_DIRECTORY specified in <*Ethernet Address*>.cfg doesn't exist**
- 14509: **If the SAS-VP xml response has a blank or missing "contactaddr" element, the phone does not use the "username" field for the contact address and may lock up during reboot**
- 14510: **The "username" field in a SAS-VP xml response is not used as the SIP login name for authentication of SIP messages**
- 14557: **The SAS-VP key is cleared if the user chooses the Reset Device Settings menu option**
- 14634: **Blind transfer fails with certain devices due to NOTIFY behavior**
- 14684: **Problems with text entry interface in custom certificate installation display**
- 14805: **Shared lines behave incorrectly if the line registration contains a '.'**
- 14935: **Phone begins to ring when there is no incoming call in specific shared line scenario**
- 15104: **SoundStation IP 4000 CDP does not advertise new link duplex levels correctly**
- 15122: **Time displayed on phone changes from correct to incorrect shortly after a reboot in some scenarios**

- 15162: **Phone clears application log file during a warm boot even if the upload to the boot server failed**

2.15.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.server.dhcp.available	1 = check with the DHCP server for SIP server IP address. 0 = do not check with DHCP server. Default = 0.
sip	added	voIpProt.server.dhcp.option	Option to request from the DHCP server if voIpProt.server.dhcp.available = 1. Allowable range is 128 – 255. There is no default value for this parameter, it must be filled in with a valid value.
sip	added	voIpProt.server.dhcp.type	0 = IP address 1 = string Type to request from the DHCP server if voIpProt.server.dhcp.available = 1. There is no default value for this parameter, it must be filled in with a valid value.
sip	added	tcpIpApp.sntp.address.overrideDHCP and tcpIpApp.sntp.gmtOffset.overrideDHCP	These parameters determine whether configuration file parameters override DHCP parameters for the SNTP server address and GMT offset. The default is 0 which means that DHCP values will override configuration file parameters. A value of 1 means that configuration file parameters will override DHCP values.

2.16 Version 1.6.4

2.16.1 Added or Changed Features

- 12278: **Added support for SAS-VP v3 XML configuration transactions**
- 12883: **Added sending and processing the “early-only” flag in the “replaces” header to support RFC 3891 in call pickup**
- 12890: **Added accepting SDP with telephone-event on the first line**
- 13492: **Disabled CA certificate expiry checking when SNTP has not been configured**

2.16.2 Removed Features

None.

2.16.3 Corrections

The following issues have been resolved with this release:

- 7707: **LED which shows mute and incoming-call and message-waiting status can show incorrect state**
- 8598: **There is no "1/A/a" soft key when editing Forward contact**

- 12626: **Phone reboots on installation of a custom certificate**
- 12882: **Display of time and date on SoundStation IP 4000 gets truncated during a call if the line label is 10 digits long**
- 13034: **Phone should stop sending further NOTIFY messages if 481 response received**
- 13318: **SoundStation IP 4000 file system is smaller than it should be**
- 13440: **Changes in APP_FILE_PATH cause unnecessary application changes**
  Note: This fix requires bootROM version 3.1.2.
- 13507: **The phone at times incorrectly maintains two SUBSCRIBEs for call-info**
- 13533: **The phone doesn't upload directory or configuration override files to a TFTP server unless they already exist on the server**
- 13553: **The "entity" field in a dialog for private lines can be improperly formatted**
- 13554: **A phone in the offering state should send a NOTIFY response to a dialog SUBSCRIBE request for all lines except Bridged Lines**
- 13582: **"Supported" header in INVITE should contain "replaces" instead of "replace"**
- 13699: **VLAN from CDP may work intermittently on SoundStation IP 4000**
- 14116: **After a blind transfer fails, the call cannot be retrieved**
- 14219: **RTP sequence numbering starts at wrong value after a call is resumed from hold**
- 14220: **Lost packets statistics are incorrect after far end resumes a call**
- 14387: **A display name containing a '.' is not displayed in some scenarios**

2.16.4 Configuration File Parameter Changes

None.

2.17 Version 1.6.3

2.17.1 Added or Changed Features

- 11358: **Added configurable subdirectories for configuration and contact directory override files**
- 12761: **Added support for setting flash parameters from configuration file**
- 13029: **Added support for new dialog event package draft**
  draft-ietf-sipping-dialog-package-06.txt
- 13030: **Added support for new BLA draft**
  draft-anil-sipping-bla-02.txt
- 13222: **Changed maximum number of XML retries for SAS-VP to be equal to 7 days**

- 13931: **Added notice of file system fix for bug 13361 to header of SoundStation IP 4000 binary image**

2.17.2 Removed Features

- 13025: **Disabled url-dialing in main partner configuration files**

2.17.3 Corrections

The following issues have been resolved with this release:

- 11271: **Phone repeatedly tries to upload log file when log.render.file parameter disabled**
- 12449: **Shared line continues to ring after receiving a CANCEL event in some scenarios**
- 12470: **Misplaced comma in date display for two possible date formats**
- 12748: **Caller ID shows IP address when PSTN caller is unknown**
  Note: The “url-dialing” feature must be disabled in order for the IP address to be hidden
- 12842: **Some characters sent in the dial string should be escaped but are not**
- 13089: **Outbound proxy port greater than 6535 does not work**
- 13198: **Long date format gets changed to short date format after first call**
- 13223: **All user agent headers for SAS-VP v3 must include <*Ethernet address*>**
- 13228: **Audio lost for the first call after rejecting the second incoming call if headset or hands free is used**
- 13235: **Repeatedly holding and resuming a call can result in no audio when the call is resumed**
- 13258: **Frequent registration retry to an inactive server after server failover can result in the phone being unable to put a call on hold**
- 13285: **Unverified SSL connections were allowed to SAS-VP server**
- 13289: **Long date format does not work if a shared line calls itself**
- 13361: **IP 4000 security certificate (HTTPS and SAS-VP provisioning) can become corrupt after file system activity.**

  Note: BootROM must be upgraded to version 3.1.2 as instructed in Technical Bulletin TB13361
- 13517: **Hands free dial-tone volume can become very quiet after significant volume adjustment**

2.17.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
000000000000	added	CONTACTS_DIRECTORY, OVERRIDES_DIRECTORY	New fields which can specify a directory on the boot server in which contact overrides (<*Ethernet address*>-directory.xml) and configuration overrides (<*Ethernet address*>-phone.cfg) should be stored.
sip	added	voIpProt.SIP.dialog.useSDP	0 or Null: New dialog event package draft is used (no SDP in dialog body). 1: For backwards compatibility, use this setting to send SDP in dialog body.
sip	changed	feature.9.enabled	The “url-dialing” feature must be disabled by setting feature.9.enabled=”0” in order to prevent unknown callers from being identified on the display by an IP address.

2.18 Version 1.6.2

2.18.1 Added or Changed Features

None.

2.18.2 Removed Features

None.

2.18.3 Corrections

The following issues have been resolved with this release:

- 9580: **Changes in <*Ethernet address*>.cfg will not be detected during configuration polling**
- 11190: **Incorrect time zone is used for one to two minutes after a reboot**
- 12552: **Phone reboots if line keys on Expansion Module are pressed rapidly and continuously**
- 12841: **Far end phone continues to ring if near end phone ends call prior to far end answering in specific shared-line scenario**
- 12951: **Malformed RTP packets received by phone can cause it to crash**

2.18.4 Configuration File Parameter Changes

None.

2.19 Version 1.6.1

2.19.1 Added or Changed Features

- 12296: **Pressing and holding unassigned line key adds a directory contact**
- 12366: **Application log is uploaded shortly after reboot**

2.19.2 Removed Features

None.

2.19.3 Corrections

The following issues have been resolved with this release:

- 11388: **Phone does not get a CDP response reliably in some scenarios**
- 12208: **Indicator for watched contact remains red if speed dial line removed**
- 12247: **Two-stage dialing user interface not correct**
- 12348: **Handsfree and handset buttons do not work correctly to answer call when silent ringer is selected**
- 12364: **Cannot establish a centralized conference from one of the conference legs**
- 12475: **One-Touch Voicemail dialing does not support multiple lines correctly**
- 12506: **INVITE message never tried on backup proxy when primary server fails over**
- 12640: **CDP word on SoundPoint IP 601 needs to advertise maximum power to Cisco switch**
- 12775: **Phone cannot join more than two legs to centralized conference**

2.19.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	changed	voice.audioProfile.xxx parameter values and voice.gain.xxx parameter values	Use the new values for these parameters.

2.20 Version 1.6.0 (Beta only)

2.20.1 Added or Changed Features

- 4614: **Added display of date and time during a call**
- 9046: **Added support for SoundPoint IP Expansion Module**
- 9108, 10480: **Added support for SoundPoint IP 601 hardware platform**
- 9660: **Pressing and holding an assigned speed dial "line key" opens the contact directory to that entry**
- 11540: **Improved speed dial key assignment**
  When perusing the contact directory, pressing and holding an unassigned line key assigns the in-focus directory entry to that key as a speed dial. A confirmation beep is heard.
  When a new directory entry is added, the speed dial index is automatically assigned the next available value.
- 11731: **Calls from more than one SIP registration (line) can be joined**

- 11849: **Added support for transfer dispatch during consultation call proceeding state**
  New parameter for this is voIpProt.SIP.allowTransferOnProceeding which will normally not need to be changed.
- 12093: **Added a Forward menu so that forwarding can be modified at any time**

2.20.2 Removed Features

None.

2.20.3 Corrections

The following issues have been resolved with this release:

- 7521: **Transfer from a shared line can be interrupted**
- 8507: **Directory search does not produce all matches for some last names**
- 9790: **Outbound proxy transport selection should be clear**
  New parameter for this is voIpProt.SIP.outboundProxy.transport.
- 9827: **A keypad-initiated reboot waits for dial tone to time out before starting**
- 11583: **Phone does not upload log file when it exceeds render file size**
- 11738: **Audio Diagnostics don't work for headset mode**
- 11762: **Headset indicator/icon can blink during a call between two phones using the same bridged line which have headset memory enabled**
- 11790: **Multi-tap entry doesn't work for the very first character entered for URL dialing**
- 11846: **484 response should be treated as an error in ringback state**
- 11848: **No stuttered dial tone when a line has a message waiting**
- 11940: **Phone holds the call when a fourth party is added to a centralized conference**
- 11946: **Some clock date format selections do not work**
- 12032: **Pressing headset button in ringing state does not answer call when headset memory is enabled**
- 12066: **After editing contact directory items, the "Save" soft key can get relabeled as "Search"**
- 12191: **The menu produced when the Directories key is pressed should not include the "Messages" option**
- 12221: **'-1' displayed as number of different priority messages for voice message feature when data is missing**
- 12227: **Phone attempts to forward a call to a shared line if Auto Divert is enabled for the contact making the call**
- 12247: **Two-stage dialing does not work**

- 12284: **Time handling for DHCP needs to be improved**
- 12289: **Common audio equalization tables should be grouped together**
- 12323: **Exiting Display Diagnostics with termination key does not stop display diagnostics**
- 12333: **"Direct" and "Group" soft keys can appear when directed and group call pickup features are disabled**
- 12370: **Ringing can be heard during a connected call mixed with audio when there is a high number of unanswered incoming calls**
- 12541: **Error messages can appear in log file after putting two calls on hold**

2.20.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	added	voIpProt.SIP.allowTransferOnProceeding	0 = don’t allow transfer during consultation call proceeding state 1 = do allow it (1 is the default)
sip	added	voIpProt.SIP.outboundProxy.transport	Same function and possible values as existing voIpProt.server.x.transport parameter. Default is DNSnaptr.
sip	added	voice.gain.rx.analog.chassis.IP_601, voice.gain.rx.analog.ringer.IP_601, voice.gain.rx.digital.chassis.IP_601, voice.gain.rx.digital.ringer.IP_601, voice.gain.tx.analog.chassis.IP_601, voice.gain.tx.digital.chassis.IP_601, voice.gain.tx.analog.preamp.chassis.IP_601	Gains specifically for the IP 601 platform.
sip	changed	voice.aec.xxx	Changed parameter values. Do not modify these.
sip	changed	voice.ns.xxx	Changed parameter values. Do not modify these.
sip	added/ removed	voice.rxEq.xxx	This whole section has changed and must be used. Do not modify these.
sip	added/ removed	voice.txEq.xxx	This whole section has changed and must be used. Do not modify these.
sip	added	log.level.change.sotet, log.level.change.ttrs	Added log level control for logging related to Expansion Module.

3. Notes

3.1 Upgrading

This section lists the changes that should be made to configuration files when using the centralized (boot server) provisioning model. For general guidelines, see the Updating and Rebooting information in Section 4.3 of the Administrator Guide.

3.1.1 From Version 2.1.1 C to 2.1.2

3.1.1.1 Mandatory Changes

- **Adding logging of version information for configuration files**
  In order for this new feature to work, the latest version of all configuration files must be used.

3.1.1.2 Optional Changes

- **Using different versions of configurable items in <*Ethernet address*>.cfg for different phone models or platforms**
  Different phone models or platforms can be configured to use different application files, configuration files, log file directory etc. See technical bulletin TB35361 for details.
- **Optimizing failover behavior for authentication signaling**
  Use the new parameters voIpProt.SIP.authOptimizedInFailover in sip.cfg and reg.x.auth.optimizedInFailover in phone1.cfg to change the phone's failover behavior during authentication signaling if desired.
- **Viewing message waiting indicators while still retaining one-touch voicemail access when multiple lines are configured**
  If a phone has multiple lines with just one registration set to have msg.mwi.x.callBackMode = "registration" and all others set to have msg.mwi.x.callBackMode = "disabled" but it is desirable to be able to see message waiting indicators for all lines and still retain one-touch voicemail access, set the new parameter up.mwiVisible to 1 in sip.cfg.

3.1.2 From Version 2.1.1 to 2.1.1 C

3.1.2.1 Mandatory Changes

None.

3.1.2.2 Optional Changes

None.

3.1.3 From Version 2.1.0 to 2.1.1

3.1.3.1 Mandatory Changes

None.

3.1.3.2 Optional Changes

- **Using URI from call's contact header in refer-to header**
  Set the parameter voIpProt.SIP.useContactInReferTo to 1 in sip.cfg if the URI from the initial call's Contact header should be used in REFER's refer-to header when setting up a transfer. The previous and default behavior is to use the URI from the initial call's To header.
- **Supporting G.729 Annex B SDP signalling per RFC 3555**
  If the new parameter voice.vad.signalAnnexB in sip.cfg is set to 1, a new attribute line will be added to SDP. See details in 2.3.4 Configuration File Parameter Changes.

3.1.4 From Version 2.0.3 to 2.1.0

3.1.4.1 Mandatory Changes

- **Using a Microsoft LCS Server**
  It may be required to set the new parameters voIpProt.server.x.lcs (in sip.cfg) and reg.x.server.y.lcs (in phone1.cfg) if the phone registers to a Microsoft LCS server.

3.1.4.2 Optional Changes

- **Using "inactive" stream mode attribute when a call is put on hold**
  The default behavior is for the "sendonly" stream mode attribute to be used when a call is put on hold. This behavior can be changed to use the "inactive" attribute. In order to configure this behavior, the parameter voIpProt.SIP.useSendonlyHold must be set to 0.
- **Digit map extension support**
  The digit map can be configured to remove, add or replace digits. For details see Technical Bulletin 11572.
- **Restricting transport to TCP**
  The transport used by the phone can be restricted to TCP. This means the phone will not attempt to fail over to UDP if TCP fails. A new "TCPOnly" option has been added to all parameters which control the transport used by the phone.
- **Adding "sticky line seize" behavior for hot-dial (on-hook) dialing**
  If sticky behavior is desired for hot dialing this can be configured using the new call.sticky.AutoLineSeize.onHookDialing parameter. Hot dialing sticky behavior can be configured to be different than normal new call sticky behavior. "Stickiness" refers to using the same line for a new call as the last-used line when a call has been put on hold.

3.1.5 From Version 2.0.3 to 2.0.3 B

3.1.5.1 Mandatory Changes

None.

3.1.5.2 Optional Changes

None.

3.1.6 From Version 2.0.2 to 2.0.3

3.1.6.1 Mandatory Changes

None.

3.1.6.2 Optional Changes

None.

3.1.7 From Version 2.0.1 to 2.0.2

3.1.7.1 Mandatory Changes

None.

3.1.7.2 Optional Changes

None.

3.1.8 From Version 2.0.0 to 2.0.1

3.1.8.1 Mandatory Changes

None.

3.1.8.2 Optional Changes

- **Using template support in master configuration file**
  The master configuration file may contain the string "[MACADDRESS]". This will be replaced with the MAC address of the phone. For example, the file 000000000000.cfg may refer to *[MACADDRESS]phone.cfg* which will be replaced with something like *0004f2100137phone.cfg*. This can make provisioning more efficient.
- **Adding Nortel MCP NAT traversal**
  The new parameters voIpProt.SIP.pingInterval and reg.x.proxyRequire should be configured if this feature is needed.
- **Adding NAT keepalive**
  If NAT keepalive is required, the new parameter nat.keepalive.interval should be set to a non-zero value.

3.1.9 From Version 1.6.7 to 2.0.0

3.1.9.1 Mandatory Changes

- **Using the phone's menu to select call progress tones**
  This feature has been removed from the default configuration of the phone. 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. Old configuration parameters are feature.10.name="cpt-settings", feature.10.enabled="1", and the entire localization – multilingual – language – callProgTones section and the entire localization – callProgTones section.

3.1.9.2 Optional Changes

- **Adding IP QoS support for DSCP (DiffServ)**
  Add the parameters qos.ip.rtp.dscp and qos.ip.callContol.dscp for DSCP. A valid value is either a number or string as follows
  1) Any number from 0 to 63
  2) EF
  3) Any of AF11, AF12, AF13, AF21, AF22, AF23, AF31, AF32, AF33, AF41, AF42, AF43
  The rules are:
  1) When qos.ip.rtp.dscp has a valid value, then it overrides the following:
  i) qos.ip.rtp.min_delay
  ii) qos.ip.rtp.max_throughput
  iii) qos.ip.rtp.max_reliability
  iv) qos.ip.rtp.min_cost
  v) qos.ip.rtp.precedence
  2) Similarly when qos.ip.callControl.dscp has a valid value, then it overrides qos.ip.callControl.min_delay etc.

3.1.10 From Version 1.6.6 to 1.6.7

3.1.10.1 Mandatory Changes

- **Selecting "sticky" line seize behavior**
  To have the same line seize behavior as SIP 1.6.5, set call.stickyAutoLineSeize to 1 in sip.cfg.

3.1.10.2 Optional Changes

- **Overriding codec preferences received from far end**
  To allow the phone to override the list of codec preferences received by the phone, set voIpProt.SDP.answer.useLocalPreferences to 1 in sip.cfg.

3.1.11 From Version 1.6.5 to 1.6.6

3.1.11.1 Mandatory Changes

None.

3.1.11.2 Optional Changes

- **Sending re-INVITE to server during conference setup on BLA**
  Set call.shared.exposeAutoHolds to 1 in sip.cfg

3.1.12 From Version 1.6.4 to 1.6.5

3.1.12.1 Mandatory Changes

- None.

3.1.12.2 Optional Changes

- **Getting SIP server address from DHCP**
  The SIP server address can be obtained from a DHCP server if the new parameters

voIpProt.server.dhcp.available, voIpProt.server.dhcp.option and voIpProt.server.dhcp.type are configured correctly.

- **Using configuration file values for SNTP parameters instead of DHCP values**
If the configuration file settings for the SNTP server address or GMT offset should be used instead of the values obtained from a DHCP server, set one or both of the new parameters tcpIpApp.sntp.address.overrideDHCP and tcpIpApp.sntp.gmtOffset.overrideDHCP to 1.
- **Reducing the power requirements reported via CDP for a SoundPoint IP 601**
A new flash parameter “EM Power” is available under the Network Configuration menu of SoundPoint IP 601 phones. If this is set to “Enabled” the phone will report power requirements of 12W which is sufficient to power three Expansion Modules. If the parameter is set to “Disabled” the phone will report power requirements of 5W and no Expansion Modules can be connected to the phone. By default this parameter will be set to “Enabled” when the phone is upgraded to 1.6.5. BootROM version 3.1.3 is required in order for the same power requirements to be reported at boot time. Please refer to Tech Bulletin TB14052 for details on upgrade/downgrade process with respect to this parameter.

3.1.13 From Version 1.6.3 to 1.6.4

3.1.13.1 Mandatory Changes

None.

3.1.13.2 Optional Changes

None.

3.1.14 From Version 1.6.2 to 1.6.3

3.1.14.1 Mandatory Changes

- **Dialog event package draft backwards compatibility**
If the old dialog event package draft behavior is desired (SDP is sent in dialog body), set the new voIpProt.SIP.dialog.useSDP parameter in sip.cfg to 1.

3.1.14.2 Optional Changes

- **Changing the destination of phone-specific override file uploads**
Use the new CONTACTS_DIRECTORY and OVERRIDES_DIRECTORY fields in 000000000000.cfg.
- **Preventing IP address caller ID display when PSTN caller is unknown**
The “url-dialing” feature must be disabled in order for the IP address to be hidden.

3.1.15 From Version 1.6.1 to 1.6.2

3.1.15.1 Mandatory Changes

None

3.1.16 From Version 1.6.0 to 1.6.1

3.1.16.1 Mandatory Changes

- **Voice Configuration Parameters Updated**
  Some parameters in the "voice" section of sip.cfg have been modified and this entire section is required when using SIP 1.6.1.

3.2 Outstanding Issues

The following issues will be fixed in a subsequent release.

Note: Polycom has switched to a different issue tracking system which has caused the reference numbers in these release notes to be different to earlier versions. When the issues are addressed the numbers in this release note can be used to track in which version the issue is addressed.

- 24398: **No Layer 2 QoS support for signaling protocol (TCP)**
  Workaround: The default QOS parameters will still be used for TCP signaling packets, and these may be specified in the sip.cfg configuration file. Layer3 QoS settings are supported.
- 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=””
- 28419: **On SoundPoint IP 601 phone, per-contact directory settings such as auto-divert do not work for calls arriving on lines 7 to 12**
  Workaround: None.
- 28508: **Phone crashes after receiving high call rate (4 unanswered calls every 18 seconds)**
  Workaround: Reduce the incoming call rate.
- 28570: **Stuttered dial tone does not work if first line is shared**
  Workaround: Configure the first line on the phone as a private line
- 29014: **Cannot edit the contact directory on the phone if the phone’s directory file saved on the boot server has been corrupted**
  Workaround: Correct the directory file on the boot server and reboot the phone.
- 29344: **HTTP Digest Authentication does not work on IIS**
  Workaround: Use a different form of authentication, a different protocol or a different server
- 29946: **Log files are not uploaded if an Apache 2.0.X boot server requires authentication**
  Workaround: Turn off authentication or use version 1.3.3X of the Apache 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.
- 30903: **Packet Loss statistics 'jump' if calls are transferred.**
  Workaround: If using the packet loss statistics for troubleshooting purposes make a note of the Packet Loss value after the transfer and apply a correction based on this to subsequent calculations.
- 32476: **IP601 does not work correctly when Presence feature is enabled with LCS server without using Roaming Buddies**
  Workaround: Enable roaming buddies by setting roaming_buddies.reg to the LCS registration number.
- 32611: **BLA line can not place and hold more than 10 calls**
  Workaround: For BLA lines ensure that call.callsPerLineKey is set to 10 or lower.
- 32816: **Phone crashes on subsequent call if using NTLM and received transfer from non-NTLM phone**
  Workaround: Ensure that all phones involved in a transfer use NTLM, or do not use NTLM authentication
- 32994: **SoundPoint IP 650 phone may have an incomplete display with only shades of grey after booting up**
  Workaround: Cycle power to the phone to make it boot again
- 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.
- 33748: **Web server has vulnerability to DOS attacks**
  Workaround: Disable the web server on the phone.
- 33931: **Not all keys on phone can be remapped to Null**
  Workaround: None. Note that Do Not Disturb key can be remapped to Null
- 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**
  Workaround: There are three possible workarounds.

1. Use a fully qualified domain name for the SIP server.
2. Do not turn off the primary DNS service. Note that if the DNS server shuts down the phone behaves well.
3. Configure the secondary DNS server to provide answers to all phone queries. For example if the phone is configured to use DNS NAPTR make sure the secondary DNS server will provide NAPTR and SRV records.

- 34454: **If microbrowser is enabled and refreshes are too frequent and pages contain large images, the phone may crash**
  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.
- 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**
  Workaround: If multiple emergency numbers are required they should be configured as separate entries within the sip.cfg file:
  e.g
  dialplan.1.routing.server.1.address=216.81.162.151
  dialplan.1.routing.server.1.port=5060
  dialplan.1.routing.emergency.1.value=911
  dialplan.1.routing.emergency.1.server.1=1
  dialplan.1.routing.emergency.2.value=9911
  dialplan.1.routing.emergency.2.server.1=1
- 34743: **A phone may freeze when it receives a check-sync if the resources on the phone are heavily used by downloaded wave files or large or complex microbrowser pages**
  Workaround: Reduce the RAM disk size configured in sip.cfg (this will reduce the amount of space available for downloaded wave files and other resources) by setting ramdisk.nBlocks to 3072. Design web pages used by the microbrowser carefully.

4. Reference Documents

- *Administrator Guide – SoundPoint IP SIP –* Version 2.0
- *SIP2.1.0 Addendum to the SoundPoint IP SIP Administrator's Guide*
- *Technical Bulletins 5844, 11572, 17124, 35361 – may be obtained from the Polycom web-site Support Knowledge-Base www.polycom.com/support/voip*

