POLYCOM®

Release Notes

SIP Application

SoundPoint® and SoundStation® IP

Version 1.6.2
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Part Number 3804-11530-162

Table of Contents

1. General

These release notes apply to version 1.6.2 of the SoundPoint IP SIP application. For more information, refer to the documents listed in § 4 Reference Documents.

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 500	2.6.1 or greater
SoundPoint IP 501	2.6.1 or greater
SoundPoint IP 600	2.6.1 or greater
SoundPoint IP 601	3.1.0 or greater
SoundStation IP 4000	2.6.1 or greater

2. Changes

2.1 Version 1.6.2

2.1.1 Added or Changed Features

None.

2.1.2 Removed Features

None.

2.1.3 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**

2.1.4 Configuration File Parameter Changes

None.

2.2 Version 1.6.1

2.2.1 Added or Changed Features

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

2.2.2 Removed Features

None.

2.2.3 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**

2.2.4 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.

2.3 Version 1.6.0

2.3.1 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**

2.3.2 Removed Features

None.

2.3.3 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**

2.3.4 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.

2.4 Version 1.5.2

2.4.1 Added or Changed Features

- 11356: **Changed configuration of presence and instant messaging features to be disabled by default**
- 11552: **Added phone UI and web interface configuration support for lineKeys and callsPerLineKey**

2.4.2 Removed Features

- 11816: **Pressing a line key will no longer terminate a call**

2.4.3 Corrections

The following issues have been resolved with this release:

- 9491: **Empty "to" header may be sent in some cases**
- 9776: **Parsing errors when dealing with the override file**
- 9817: **Configuration override file gets unnecessary extra parameters**
- 11189: **User can corrupt the directory by editing it when “presence” feature is disabled**
- 11343: **Pressing handsfree or headset button activates handset if handset is off hook**
- 11409: **Provisioning may not work reliably with the proftpd FTP server on Linux**
- 11417: **Phone may not be able to boot from a remote subnet**
- 11426: **Secondary dial tone plays incorrectly on certain digit maps**
- 11466, 11558: **Provisioning may fail using HTTPS if a custom certificate is used**
- 11556: **Stored authentication key from a SAS-VP server is deleted when the phone is reset to factory defaults**
- 11558: **Provisioning may fail using HTTPS if a custom certificate is used**
- 11575: **SoundPoint IP300/301 doesn't give warning message if duplicate IP is detected by DHCP client**
- 11584: **Automatic key repeats do not work**
- 11595: **Phone displays URL encoded digits when dialing**
- 11599: **Check-sync and polled configuration change features do not work**
- 11600: **Phone ignores maximum password length parameters**
- 11608: **Disabling "presence" feature does not remove it from phone's menu**
- 11609: **Disabling “messaging” feature on SoundStation IP 4000 and SoundPoint IP30x disables voice message feature as well**

- 11612: **When Do Not Disturb per-registration is enabled, the Do Not Disturb “clear all” soft key is missing**
- 11616: **CANCEL requests include tag when they shouldn't**
- 11633: **Phone should use flash credentials when boot server URL lacks them**
- 11641: **Phone shows an error message on the display when Hold is invoked on the last available call appearance**
- 11644: **Join does not work from the last available call appearance**
- 11665: **Pressing the headset button in ringing state does not answer call when headset memory is enabled**
- 11685: **Line configuration cannot be changed using web server**
- 11739: **A call can be lost when Split is used under certain circumstances**
- 11760: **Custom certificate gets corrupted if SAS-VP is used**
- 11788: **Pressing "New Call" soft key auto dials the previous number entered using on-hook dialing if the previous call failed**
- 11789: **The "more" soft key for establishing a conference can disappear, hiding the “Join” soft key**
- 11798: **There is an incompatability when using EPSV with proftpd**

2.4.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
sip	changed	feature.1.enabled, feature.2.enabled changed from 1 to 0	Presence and Instant Messaging are disabled by default.
sip	changed	voIpProt.server.x.transport	Explicitly set default to DNSnaptr
phone1	changed	reg.x.server.y.transport	Explicitly set default to DNSnaptr

2.5 Version 1.5.1

2.5.1 Added or Changed Features

- 966: **A single call will always show up in the first call appearance position**
- 1509: **Improved menu hierarchy**
- 1842: **Added visual "status" to contacts assigned to speed dial bins**
- 3924: **Added conference feature enhancement to "join" calls in progress**
- 7204: **Added flashing time/date until successful SNTP response**
- 7663: **Added ability to specify boot server address as URL per RFC 1738**
  This requires bootROM 3.0 or greater.
- 7894: **Added support for having more than one line key associated with the same SIP identity**
  This includes a new feature – pressing and holding down the line key provides call information about a call which is on hold on that line key.

- 7899: **Added support for the application to provision its own configuration files**
- 7900: **Added application support for HTTP and HTTPS boot server transport**
  This requires bootROM 3.0 or greater.
  For HTTPS, if the time on the phone is wrong the SSL certificate may be rejected. Configure SNTP to obtain an accurate time.
- 8055: **Added support for SAS-VP v2 management**
  This requires bootROM 3.0 or greater.
- 8521: **Added a menu entry to format the file system**
- 8786: **Added display of name and number on incoming caller ID**
- 9053: **Added support for displaying a useful CID when display name is uninformative**
- 9096: **Added customization options for SSL certificates**
- 9108, 10480: **Added support for SoundPoint IP 601 hardware platform**
- 9299: **Added allowing all files in <MAC>.cfg to be full URL's**
- 9323: **Removed requirement for at least two audio codecs to be configured**
- 9496: **Merged sip.cfg and ipmid.cfg configuration files into new sip.cfg file**
- 9548: **Added allowing user to disable time and date display**
- 9579: **Added allowing specific master configuration file to be specified in boot server URL**
- 9588: **Changed offering LED animation to continuous 2 Hz flash, rather than intermittent**
- 9659: **Added feature to split conferences and consultation calls into separate calls**
- 9675: **Added feature to allow conference initiation from call hold context**
- 9694: **Changed example directory file to no longer use silent ring type for contacts**
- 9710: **Changed default hold signaling to be the RFC 3261 style**
- 10806: **Added build ID to software revision stamps in User-Agent header**
- 11235: **Added support for arrow-key call-list shortcuts when phone is playing dial tone**

2.5.2 Removed Features

- 11973: **Removed support for port mode FTP server configurations**
  Use an FTP server/firewall that supports passive mode connections.

2.5.3 Corrections

The following issues have been resolved with this release:

- 737: **Phone will not accept IP packets bigger than 38,000 bytes**
- 2311: **Line labels do not line up with line keys on SoundPoint IP 600**
- 3707: **Can't use speed dial when one call already on Hold**
- 7952: **FTP transfers should remove partially written files in a failure scenario**
- 8050: **Parameters which were not changed are saved in configuration override file**
- 8333: **Improve source data for random device**
- 8416: **Bridged Line second call appearance is incorrect in specific scenario**
- 8616: **Incorrect message on display for incoming call on shared line on SoundPoint IP 4000**
- 8674: **Missing remote hold call appearance in specific Bridged Line scenario**
- 8755: **For TCP, the response to a request should try the remote port that sends the request first**
- 8771: **IP 4000 cannot download large directory file**
- 8801: **Phone ignores X-Syl-Line-ID and mixes call appearances**
- 8873: **When a new DHCP lease is obtained, the updated DNS information is not used**
- 8962: **Active Bridged Line cannot switch to incoming call**
- 9090: **Clock date menu choice ending in 'YYYY' not displayed properly**
- 9135: **Random string for CNONCE value for digest authentication should be limited to the base64 character set**
- 9187: **GMT offset and SNTP address set in flash are ignored if parameters exist in configuration file but have no associated value (i.e. are empty)**
- 9243: **Web server buttons not labeled, and some labels are incorrect**
- 9326: **DST not working for Southern Hemisphere**
- 9452: **DTMF tones not recognized by specific IVR after shared line remote resume**
- 9481: **Phone will attempt to download files indefinitely if connection to FTP server lost**
- 9482: **Phone waits for an error response from the FTP server when none is forthcoming**
- 9584: **Call duration missing from placed call list items on IP 4000**
- 9601: **DNS resolution fails when downloading [mac]-phone.cfg**
- 9735: **Web interface of SoundStation IP 4000 phone edits some non-IP 4000 parameters**
- 10825: **Phone should not collect digits after dial tone has timed out**

- 11285: **SIP authentication password stored in [mac]-phone.cfg file**
- 11303: **SoundPoint IP 300 phone loses contrast settings during reboot**
- 11348: **Large DHCP messages get truncated**
- 11350: **SoundStation IP 4000 phone can lock up when a key is pressed**
- 11458: **Audio loss on one leg of conference after second conference automatically put on Hold (first conference is Resumed)**
- 11516: **Off-by-one error when ringTypes are saved**
- 11548: **Cannot change administrator password or user password on SoundPoint IP 300**
- 11559: **ACD login does not work on SoundStation IP 4000**
- 11563: **ACD available/unavailable functions work differently on Bridged and Private lines**
- 11573: **Pressing Handsfree button does not put you back to handset when handset is off hook**

2.5.4 Configuration File Parameter Changes

.cfg File	Action	Parameter	Description
ipmid	removed	All parameters	The contents of this file have been added to sip.cfg and this file is no longer used.
sip	added	All “ipmid.cfg” parameters	The contents of the old ipmid.cfg file have been added to sip.cfg.
sip	added	call.callsPerLineKey	The number of calls or conferences which may be active or on hold per line key on the phone. For the IP 600, range is 1 to 24 and default is 24. For all other phones, range is 1 to 8 and default is 8.
sip	changed	voIpProt.SIP.useRFC2543hold	Changed the default value to "0" (it used to be “1”) which means that RFC 3261-style hold signalling is the default.
sip	changed	voice.gain.rx.analog.chassis.IP300 voice.gain.rx.analog.ringer.IP300	Changed name to voice.gain.rx.analog.chassis.IP_300 voice.gain.rx.analog.ringer.IP_300
sip	changed	call.shared.oneTouchResume	This applies to SoundStation IP 4000 phones only in this build. For all other phones, one-touch resume is the default. In order to view call information about a call on hold on another phone with a shared line – press and hold down the line key for a few seconds.
sip	changed	ind.gi.IP_600.x…	Changed values used for locating line key labels. Update this whole section.
sip	removed	ind.pattern.8.step.3 to 6	An incoming call causes the LED to flash continuously at 2Hz rather than flash intermittently.

.cfg File	Action	Parameter	Description
sip	removed	all .obs parameters from logging section	These parameters are no longer used.
phone1	added	reg.x.ringType	The ring type for each registration can be configured. Range is 1 to 22. **Note**: ring type number 1 is "silent ring".
phone1	added	reg.x.lineKeys	The number of line keys on the phone to be associated with registration 'x'. Range is 1 to the maximum number of line keys on the phone (IP 300 = 2, IP 500 = 3, IP 600 = 6, IP 4000 = 1). Default is 1.
phone1	added	reg.x.callsPerLineKey	The number of calls or conferences which may be active or on hold per line key for a specific registration on the phone. This will override the global call.callsPerLineKey parameter in sip.cfg. Same range and defaults as call.callsPerLineKey above.
000000000000	removed	ipmid.cfg from list of CONFIG_FILES	The ipmid.cfg file is no longer used.

3. Notes

3.1 Distribution Files

The following files constitute the 1.6.2 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 1.6.2.0041
	IP 300 2345-11300-001: 1.6.2 9-Aug-05 17:28 IP 301 2345-11300-010: 1.6.2 9-Aug-05 17:06
	IP 500 2345-11500-001: 1.6.2 9-Aug-05 17:20 2345-11500-010: 1.6.2 9-Aug-05 17:20 2345-11500-030: 1.6.2 9-Aug-05 17:17 2345-11500-020: 1.6.2 9-Aug-05 17:24 2345-11500-040: 1.6.2 9-Aug-05 17:17
	IP 600 2345-11600-001: 1.6.2 9-Aug-05 17:03 IP 601 2345-11605-001: 1.6.2 9-Aug-05 17:03
	IP 4000 2201-06642-001: 1.6.2 9-Aug-05 17:13
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 included (no IP 30X support): Chinese, China (for IP 60X 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 60X and IP 4000 only) Korean, Korea (for IP 60X and IP 4000 only) Norwegian, Norway Portuguese, Portugal Russian, Russia Spanish, Spain Swedish, Sweden
SoundPointIPWelcome.wav	start up welcome sound effect

3.2 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.

3.2.1 From Version 1.6.1 to 1.6.2

3.2.1.1 Mandatory Changes

None

3.2.2 From Version 1.6.0 to 1.6.1

3.2.2.1 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.

3.2.3 From Version 1.5.2 to 1.6.0

3.2.3.1 Mandatory Changes

- **Voice Configuration Parameters Updated**
  Many parameters in the “voice” section of sip.cfg have been modified and this entire section is required when using SIP 1.6.0.
- **Transfer On Proceeding Enabled by Default**
  In SIP 1.5.2 there was no option to complete a transfer during the proceeding state of a consultation call. In SIP 1.6.0 this has been added and it is enabled by default. Set the parameter voIpProt.SIP.allowTransferOnProceeding to 0 if this feature is not wanted.
- **Selecting the Transport for an Outbound Proxy**
  The transport used by an outbound proxy is determined by the new parameter voIpProt.SIP.outboundProxy.transport. If this parameter is missing, the default of NAPTR will be used. In SIP 1.5.X the outbound proxy transport was determined by the voIpProt.server.1.transport or reg.x.server.1.transport parameters but these are no longer taken into account.

3.2.4 From Version 1.5.1 to 1.5.2

3.2.4.1 Mandatory Changes

- **Presence and Instant Messaging Disabled by Default**
  These features have been disabled in sip.cfg by setting feature.1.enabled and feature.2.enabled to 0. If these features are required they must be enabled in sip.cfg.

3.3 Outstanding Issues

The following issues will be fixed in a subsequent release.

- 8921: **Centralized conference fails due to RTP port slow to open in some cases**
  Workaround: None.
- 9292: **IP 4000 reboots upon downloading a wave file with a path containing '\' instead of '/'**
  Workaround: Wave file paths must be specified using '/' e.g. "wavs/ring1.wav"
- 12842: **Some characters sent in the dial string should be escaped but are not**
  Workaround: None.
- 13089: **Outbound proxy port greater than 6535 does not work**
  Workaround: Use an outbound proxy port number smaller than 6535
- 13198: **Long date format does not work if a shared line calls itself**
  Workaround: The short date format will be used in this scenario.
- 13228: **Audio lost for the first call after rejecting the second incoming call if headset or handsfree is used**
  Workaround: Use handset when rejecting a second incoming call while the first call is still active.

4. Reference Documents

- *Administrator Guide – SoundPoint IP SIP* – Version 1.6

