Release Notes

SIP Application

SoundPoint® and SoundStation® IP

Version 1.6.6
28 April 2006

Part Number 3804-11530-166

Table of Contents

1. General

- These release notes apply to version 1.6.5 of the SoundPoint IP SIP application. For more information, refer to the documents listed in § 16041: **After a reboot, a phone with a shared line is occasionally unable to seize the line**
  Workaround: Reboot the phone again.

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	3.1.2 or greater

2. Changes

2.1 Version 1.6.6

2.1.1 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**

2.1.2 Removed Features

None.

2.1.3 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" softkey 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" softkey 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" softkey 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 softkey labels after attempting to hot dial when the remote shared line is in use**

2.1.4 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”.

2.2 Version 1.6.5

2.2.1 Added or Changed Features

- 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)**

2.2.2 Removed Features

None.

2.2.3 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**

2.2.4 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.

2.3 Version 1.6.4

2.3.1 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**

2.3.2 Removed Features

None.

2.3.3 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" softkey 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**

2.3.4 Configuration File Parameter Changes

None.

2.4 Version 1.6.3

2.4.1 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**

2.4.2 Removed Features

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

2.4.3 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 handsfree is used**
- 13235: **Repeatly 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 filesystem activity.**

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

2.4.4 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.

2.5 Version 1.6.2

2.5.1 Added or Changed Features

None.

2.5.2 Removed Features

None.

2.5.3 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**

2.5.4 Configuration File Parameter Changes

None.

2.6 Version 1.6.1

2.6.1 Added or Changed Features

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

2.6.2 Removed Features

None.

2.6.3 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**

2.6.4 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.

2.7 Version 1.6.0

2.7.1 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**

2.7.2 Removed Features

None.

2.7.3 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**

2.7.4 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.

2.8 Version 1.5.2

2.8.1 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**

2.8.2 Removed Features

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

2.8.3 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**

2.8.4 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

2.9 Version 1.5.1

2.9.1 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**
- 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**

2.9.2 Removed Features

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

2.9.3 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**

2.9.4 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.
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.

.cfg File	Action	Parameter	Description
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 1.6.5 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 1.6.6.0036.
	IP 300 2345-11300-001: 1.6.6 IP 301 2345-11300-010: 1.6.6
	IP 500 2345-11500-001: 1.6.6 2345-11500-010: 1.6.6 2345-11500-030: 1.6.6 2345-11500-020: 1.6.6 2345-11500-040: 1.6.6
	IP 600 2345-11600-001: 1.6.6 IP 601 2345-11605-001: 1.6.6
	IP 4000 2201-06642-001: 1.6.6
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 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 1.6.5 to 1.6.6

3.2.1.1 Mandatory Changes

None.

3.2.1.2 Optional Changes

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

3.2.2 From Version 1.6.4 to 1.6.5

3.2.2.1 Mandatory Changes

- None.

3.2.2.2 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.

3.2.3 From Version 1.6.3 to 1.6.4

3.2.3.1 Mandatory Changes

None.

3.2.3.2 Optional Changes

None.

3.2.4 From Version 1.6.2 to 1.6.3

3.2.4.1 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.

3.2.4.2 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.

3.2.5 From Version 1.6.1 to 1.6.2

3.2.5.1 Mandatory Changes

None

3.2.6 From Version 1.6.0 to 1.6.1

3.2.6.1 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.

3.2.7 From Version 1.5.2 to 1.6.0

3.2.7.1 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.

3.2.8 From Version 1.5.1 to 1.5.2

3.2.8.1 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.

- 4310: **No 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.
- 5085: **Cannot answer an incoming call while directory is being saved**
  Workaround: None.
- 6527: **Shared line does not ring if incoming call arrives when phone is playing dialtone then subsequently hangs up**
  Workaround: None.
- 8532: **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.
- 8547: **Local ringback is not played if far end does blind transfer without going on hold**
  Workaround: None.
- 8921: **Centralized conference fails due to RTP port being slow to open in some cases**
  Workaround: None.
- 9176: **Memory leak in phone if it tries to upload log files into a non-existent folder which is specified by LOG_FILE_DIRECTORY**
  Workaround: Specify a valid folder destination in LOG_FILE_DIRECTORY.
- 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".
- 9709: **RTCP not sent or received when calls are on hold**
  Workaround: None.
- 11588: **The local contact directory feature cannot be disabled**
  Workaround: None.
- 11653: **Startup of phone make get delayed indefinitely because of some tasks getting unrealistic delays**
  Workaround: This issue is mitigated by the workaround implemented in 15012. This issue is worse with slow boot servers.
- 12155: **SoundPoint IP 300 and 301 phones have no "Exit" softkey during the ACD login process**
  Workaround: Exit the display by pressing the Menu key or lifting and replacing the handset.
- 12455: **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.

- 12492: **SoundPoint IP 601 phone with Expansion Module(s) attached may fail to load the selected language after rebooting**
  Workaround: Switch to English (Internal) and then back to the desired language after the reboot.
- 12616: **Phone crashes after receiving high call rate (4 unanswered calls every 18 seconds)**
  Workaround: Reduce the incoming call rate.
- 12647: **Feature keys cannot be reconfigured to perform other functions**
  Workaround: None.
- 12722: **Stuttered dial tone does not work if first line is shared**
  Workaround: Configure the first line on the phone as a private line
- 12952: **There is no way to reset the user password back to the factory default password**
  Workaround: None.
- 13076: **Phone can pause at the “Welcome” screen for more than 5 minutes after being rebooted**
  Workaround: Ensure that the boot server can handle the load of multiple phones rebooting.
- 13230: **No audio on calls resumed from hold in some multiple call scenarios**
  Workaround: None.
- 13412: **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.
- 13579: **SDP parser applies wrong logic**
  Workaround: Change the order of lines in the SDP.
- 13786: **HTTP Digest Authentication does not work on IIS**
  Workaround: Use a different form of authentication, a different protocol or a different server
- 14275: **The call.callWaiting.prompt parameter does not have any effect**
  Workaround: None. This functionality changed in SIP 1.5.
- 14400: **Phone can take up to 30 minutes to boot when there are TCP timeouts**
  Workaround: Ensure that the configured boot server is running correctly or do not use a boot server.
- 14466: **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.
- 14467: **If a URL in <*Ethernet Address*>.cfg specifies a protocol and user name but no password, the password in flash is not used**
  Workaround: Specify the password in the configuration file
- 14624: **Boot servers running explicit FTPS are not supported**
  Workaround: Use implicit FTPS or HTTPS.

- 14844: **A failed download of a pre-existing file causes that file to be deleted**
  Workaround: None.
- 14937: **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.
- 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**
  Workaround: Include the password in the server address URL.
- 16041: **After a reboot, a phone with a shared line is occasionally unable to seize the line**
  Workaround: Reboot the phone again.

4. Reference Documents

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