



RELEASE NOTES

Software 5.0.1 | October 2013 | 3804-11530-501

Polycom[®] UC Software 5.0.1

Applies to the Polycom[®] VVX[®] 300/310, VVX[®] 400/410, VVX[®] 500, VVX[®] 600, VVX[®] 1500 Business Media Phones and Polycom[®] SoundStructure[®]



Copyright ©2013, Polycom, Inc. All rights reserved. No part of this document may be reproduced, translated into another language or format, or transmitted in any form or by any means, electronic or mechanical, for any purpose, without the express written permission of Polycom, Inc.

6001 America Center Drive
San Jose, CA 95002
USA

Trademarks



Polycom®, the Polycom logo and the names and marks associated with Polycom products are trademarks and/or service marks of Polycom, Inc. and are registered and/or common law marks in the United States and various other countries. All other trademarks are property of their respective owners. No portion hereof may be reproduced or transmitted in any form or by any means, for any purpose other than the recipient's personal use, without the express written permission of Polycom.

End User License Agreement

By installing, copying, or otherwise using this product, you acknowledge that you have read, understand and agree to be bound by the terms and conditions of the [End User License Agreement](#) for this product.

Patent Information

The accompanying product may be protected by one or more U.S. and foreign patents and/or pending patent applications held by Polycom, Inc.

Open Source Software Used in this Product

This product may contain open source software. You may receive the open source software from Polycom up to three (3) years after the distribution date of the applicable product or software at a charge not greater than the cost to Polycom of shipping or distributing the software to you. To receive software information, as well as the open source software code used in this product, contact Polycom by email at OpenSourceVideo@polycom.com.

Disclaimer

While Polycom uses reasonable efforts to include accurate and up-to-date information in this document, Polycom makes no warranties or representations as to its accuracy. Polycom assumes no liability or responsibility for any typographical or other errors or omissions in the content of this document.

Limitation of Liability

Polycom and/or its respective suppliers make no representations about the suitability of the information contained in this document for any purpose. Information is provided "as is" without warranty of any kind and is subject to change without notice. The entire risk arising out of its use remains with the recipient. In no event shall Polycom and/or its respective suppliers be liable for any direct, consequential, incidental, special, punitive or other damages whatsoever (including without limitation, damages for loss of business profits, business interruption, or loss of business information), even if Polycom has been advised of the possibility of such damages.

Customer Feedback

We are striving to improve our documentation quality and we appreciate your feedback. Email your opinions and comments to DocumentationFeedback@polycom.com.



Visit the [Polycom Support Center](#) for End User License Agreements, software downloads, product documents, product licenses, troubleshooting tips, service requests, and more.

Contents

General	7
What's in These Release Notes?	7
Understanding Phone Features and Licenses	7
Downloading the Distribution Files	8
Downloading the Combined ZIP File	8
Understanding the Split ZIP Resource Files	10
What's New for Polycom UC Software 5.0.1?	13
New or Enhanced Features.....	13
Enhanced Capabilities	13
Configuration File Enhancements	15
Updates to Previous Software Releases.....	17
Understanding Updates to UC Software 5.0.0	17
New or Enhanced Features.....	17
Enhanced Capabilities	18
Configuration File Enhancements	22
Understanding Updates to UC Software 4.1.4	25
New or Enhanced Features.....	25
Enhanced Capabilities	25
Understanding Updates to UC Software 4.1.3 Rev G.....	25
New or Enhanced Features.....	26
Enhanced Capabilities	26
Configuration File Enhancements	28
Understanding Updates to UC Software 4.1.3	29
New or Enhanced Features.....	29
Enhanced Capabilities	29
Configuration File Enhancements	30
Understanding Updates to UC Software 4.1.2 Rev B	38
New or Enhanced Features.....	39
Enhanced Capabilities	39
Understanding Updates to UC Software 4.1.2	39
New or Enhanced Features.....	40

Enhanced Capabilities	40
Configuration File Enhancements	41
Understanding Updates to UC Software 4.1.0 Rev B	41
New or Enhanced Features.....	42
Enhanced Capabilities	42
Configuration File Enhancements	42
Understanding UC Software 4.1.0 for the SpectraLink 84xx and VVX 500.....	43
New or Enhanced Features.....	43
Enhanced Capabilities	44
Configuration File Enhancements	45
Understanding Updates to UC Software 4.0.2 Rev B	47
New or Enhanced Features.....	48
Enhanced Capabilities	48
Configuration File Enhancements	49
Understanding Updates to UC Software 4.0.2 (Limited Release)	50
New or Enhanced Features.....	50
Enhanced Capabilities	50
Configuration File Enhancements	52
Understanding Updates to UC Software 4.0.1B	52
Understanding Updates to UC Software 4.0.1	53
New or Enhanced Features.....	53
Enhanced Capabilities	55
Configuration File Enhancements	59
Understanding Updates to UC Software 4.0.0	61
New or Enhanced Features.....	61
Enhanced Capabilities	64
Configuration File Enhancements	69
Understanding Updates to UC Software 3.3.2	84
New or Enhanced Features.....	84
Enhanced Capabilities	84
Configuration File Enhancements	89
Understanding Updates to UC Software 3.3.1F	96
Enhanced Capabilities	96
Understanding Updates to UC Software 3.3.1	96

New or Enhanced Features.....	96
Enhanced Capabilities	97
Configuration File Enhancements	100
Understanding Updates to UC Software 3.3.0	100
New or Enhanced Features.....	101
Discontinued Features	103
Enhanced Capabilities	104
Configuration File Enhancements	112
Understanding Updates to SIP 3.2.5	112
New or Enhanced Features.....	112
Enhanced Capabilities	113
Configuration File Enhancements	115
Understanding Updates to SIP 3.2.4B	115
Enhanced Capabilities	116
Understanding Updates to SIP 3.2.4	116
Enhanced Capabilities	116
Understanding Updates to SIP 3.2.3	116
New or Enhanced Features.....	116
Enhanced Capabilities	117
Configuration File Enhancements	118
Understanding Updates to SIP 3.2.2	120
New or Enhanced Features.....	121
Discontinued Features	121
Enhanced Capabilities	121
Configuration File Enhancements	124
Understanding Updates to SIP 3.2.1B	128
New or Enhanced Features.....	128
Configuration File Enhancements	128
Understanding Updates to SIP 3.2.1	129
Enhanced Capabilities	129
Understanding Updates to SIP 3.2.0	129
New or Enhanced Features.....	130
Discontinued Features	132
Enhanced Capabilities	132
Configuration File Enhancements	140

Understanding Updates to SIP 3.1.7	151
New or Enhanced Features.....	151
Enhanced Capabilities	151
Configuration File Enhancements	154
Known Issues and Suggested Workarounds	155
Reference Documents	168

General

Polycom® Unified Communications (UC) Software 5.0.1 is a general release for all open SIP platforms including Microsoft® Lync® Server 2010 and Microsoft Lync Server 2013.

Polycom UC Software 5.0.1 supports the following Polycom® endpoints:

- VVX® 300/310 business media phone
- VVX® 400/410 business media phone
- VVX® 500 business media phone
- VVX® 600 business media phone
- VVX® 1500 business media phone
- SoundStructure®



Note: Registering with Lync Server

If you are using Polycom UC Software 5.0.1 with Microsoft Lync Server, you can only register one line. You cannot register multiple lines on a phone with Lync Server.

These release notes provide important information on software updates, phone features and feature licenses, and known issues. In addition, these release notes refer to previous UC software versions to assist administrators who are updating to a UC software 5.0.1 release from an earlier software release.

What's in These Release Notes?

The Polycom UC Software 5.0.1 Release Notes contain the following sections:

- [General](#) Read this section to understand how the changes in UC Software 5.0.1 affect Polycom hardware, deployment, and configuration of the software.
- [What's New for Polycom UC Software 5.0.1?](#) This section lists new, enhanced, and discontinued software features.
- [Updates to Previous Software Releases](#) This section lists enhanced and discontinued software features in previous software releases.
- [Known Issues and Suggested Workarounds](#) This section lists existing known issues and suggests workarounds, if available.
- [Reference Documents](#) This section lists all documents relevant to these release notes.

Understanding Phone Features and Licenses

The phone features and licenses required to operate a feature vary by phone model. Use this section to find out which phone features and licenses you require for your phone model.

The following table lists features available for each phone model and indicates whether or not a feature license is required.

Use [Table 1: VVX Series Features and Licenses](#) if you are deploying VVX business media phones. In the following table, (No) indicates that a phone does not support a feature, (Yes) indicates that a phone supports a feature and no license is required, and (Yes*) indicates that the phone requires a feature license to support a feature.

Table 1: VVX Series Features and Licenses

<i>Feature</i>	<i>VVX 300/310</i>	<i>VVX 400/410</i>	<i>VVX 500</i>	<i>VVX 600</i>	<i>VVX 1500</i>
VQMon	Yes*	Yes*	Yes (Audio only)	Yes (Audio only)	Yes (Audio only)
Call Recording	Yes	Yes	Yes	Yes	Yes
Conference Management	Yes	Yes	Yes	Yes	Yes
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes
Customizable UI Background	Yes	Yes	Yes	Yes	Yes
Asian Languages	Yes	Yes	Yes	Yes	Yes
Enhanced BLF	Yes	Yes	Yes	Yes	Yes
H.323 Video	No	No	Yes	Yes	Yes

*You must purchase a feature license from Polycom.

Downloading the Distribution Files

To download UC Software 5.0.1, use either the combined file or the split file in the ZIP file format. For general use, Polycom recommends using the split resource file that corresponds to the phone model(s) for your deployment. To match the correct UC software resource file to your phone model, use [Table 3: Understanding the Split ZIP Resource Files](#). If you are provisioning your phones centrally using configuration files, download the corresponding resource file and extract the configuration files to the provisioning server, maintaining the folder hierarchy in the ZIP file.

The current build ID for the sip.ld and resource files is **UCS 5.0.1.4068**

Downloading the Combined ZIP File

To understand the files distributed in the combined ZIP file, refer to [Table 2: Understanding the Combined ZIP File](#).

Table 2: Understanding the Combined ZIP File

<i>Distributed Files</i>	<i>File Purpose and Application</i>
sip.ld	Concatenated SIP application executable
sip.ver	Text file detailing build-identification(s) for the release
000000000000.cfg	Master configuration template file
000000000000-directory~.xml	Local contact directory template file. To apply for each phone, replace the 0s with the MAC address of the phone and remove '~' from the file name.
applications.cfg	Contains configuration parameters for microbrowser and browser applications
features.cfg	Contains configuration parameters for telephony features
H323.cfg	Contains configuration parameters for the H.323 signaling protocol
reg-advanced.cfg	Contains configuration parameters for line and call registration and advanced phone feature settings
reg-basic.cfg	Contains configuration parameters for line and call registration and basic phone settings
region.cfg	Contains configuration parameters for regional and localization settings such as time and date and language
sip-basic.cfg	Contains configuration parameters for the VoIP server and softswitch registration
sip-interop.cfg	Contains configuration parameters for the VoIP server, softswitch registration, and interoperability configuration
site.cfg	Contains configuration parameters that are set for each site
video.cfg	Contains configuration parameters for video connectivity
video-integration.cfg	Contains configuration parameters for SoundStation IP 7000 and Polycom HDX system integration

<i>Distributed Files</i>	<i>File Purpose and Application</i>
SoundPointIP-dictionary.xml	<p>Includes native support for the following languages:</p> <ul style="list-style-type: none"> Chinese, Traditional (for IP 321, 331, 335, 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) Chinese, Simplified (for IP 321, 331, 335, 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) Danish, Denmark Dutch, Netherlands English, Canada English, United Kingdom English, United States French, France German, Germany Italian, Italy Japanese, Japan (for IP 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) Korean, Korea (for IP 450, 550, 560, 650, 670; IP 5000, 6000, 7000, Duo) Norwegian, Norway Polish, Poland Portuguese, Portugal Russian, Russia Slovenian, Slovenia Spanish, Spain Swedish, Sweden Arabic, UAE
SoundPointIPWelcome.wav	Startup welcome sound effect
LoudRing.wav	Loud ringer sound effect
Warble.wav	Loud ringer sound effect

Understanding the Split ZIP Resource Files

Polycom recommends using the split ZIP file to shorten upgrade times. To find the split resource file for your phone model, use [Table 3: Understanding the Split ZIP Resource Files](#).

Table 3: Understanding the Split ZIP Resource Files

<i>Resource File</i>	<i>File Purpose and Application</i>
3111-46135-002.sip.ld	SIP application executable for VVX 300
3111-46161-001.sip.ld	SIP application executable for VVX 310
3111-46157-002.sip.ld	SIP application executable for VVX 400
3111-46162-001.sip.ld	SIP application executable for VVX 410
3111-44500-001.sip.ld	SIP application executable for VVX 500
3111-44600-001.sip.ld	SIP application executable for VVX 600
2345-17960-001.sip.ld	SIP application executable for VVX 1500
3111-33215-001.sip.ld	SIP application executable for SoundStructure
sip.ver	Text file detailing the build-identification(s) for the release
000000000000.cfg	Master configuration template file
000000000000-directory~.xml	Local contact directory template file. To apply for each phone, replace the 0s with the MAC address of the phone, and remove '~' from the file name.
applications.cfg	Contains configuration parameters for microbrowser and browser applications
device.cfg	Contains Network Configuration device parameters.
features.cfg	Contains configuration parameters for telephony features
firewall-nat.cfg	Contains Firewall parameters. Typical MS LYNC Environment.
lync.cfg	Contains LYNC specific configuration parameters.
pstn.cfg	Contains parameters for PSTN Use.
reg-advanced.cfg	Contains configuration parameters for the line and call registration and advanced phone feature settings
reg-basic.cfg	Contains configuration parameters for the line and call registration and basic phone feature settings
region.cfg	Contains configuration parameters for regional and localization settings such as time and date and language
sip-basic.cfg	Contains configuration parameters for the VoIP server, softswitch registration
sip-interop.cfg	Contains configuration parameters for the VoIP server, softswitch registration, and interoperability configuration
site.cfg	Contains configuration parameters that are set for each site

<i>Resource File</i>	<i>File Purpose and Application</i>
SoundPointIP-dictionary.xml	<p>Includes native support for the following languages:</p> <ul style="list-style-type: none"> • Chinese, Traditional (for IP 321, 331, 335, 450, 550, 560, 650, IP 5000, Duo) • Chinese, Simplified (for IP 321, 331, 335, 450, 550, 560, 650, IP 5000, Duo) • Danish, Denmark • Dutch, Netherlands • English, Canada • English, United Kingdom • English, United States • French, France • German, Germany • Italian, Italy • Japanese, Japan (for IP 450, 550, 560, 650, IP 5000, Duo) • Korean, Korea (for IP 450, 550, 560, 650, IP 5000, Duo) • Norwegian, Norway • Polish, Poland • Portuguese, Portugal • Russian, Russia • Slovenian, Slovenia • Spanish, Spain • Swedish, Sweden • Arabic, UAE
SoundPointIPWelcome.wav	Start-up welcome sound effect
LoudRing.wav	Loud ringer sound effect
Warble.wav	Loud ringer sound effect

What's New for Polycom UC Software 5.0.1?

Polycom Unified Communications (UC) Software 5.0.1 is a general release for all open SIP platforms including Microsoft Lync Server 2010 and Microsoft Lync Server 2013. Note that if you are using Polycom UC Software 5.0.1 with Microsoft Lync Server, you can only register one line. You cannot register multiple lines on one phone with Lync Server.

Polycom UC Software 5.0.1 offers support for the following endpoint platforms:

- VVX® 300/310 business media phone
- VVX® 400/410 business media phone
- VVX® 500 business media phone
- VVX® 600 business media phone
- VVX® 1500 business media phone
- SoundStructure®

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

88687 Added support for selecting the Audio or Video mode when retrieving a parked call.

88387/89097 Added support for a digit map timeout for on-hook and off-hook dialing when the normalization is done from the server side.

87132 Added support for quickly searching the BroadSoft UC-One contacts locally on the fly (applies to VVX 300, 310, 400 and 410).

87086 Added support for BToE auto-pairing.

86378 Added support for Arabic language.

Enhanced Capabilities

89005 Adding contacts to the local directory with an apostrophe (') from the BroadSoft directory is now available.

88983 Navigating through the phone's DHCP menu no longer causes a reboot when the language on the phone is set to Japanese (*applies to VVX 300/310/400 and 410*).

88956/88521 Configuring an enhanced feature key for Directed Call Park is now available.

88932/88773/89201 In a group call pickup scenario, the Group option is now available after selecting the Pickup soft key when the `feature.groupCallPickup.enabled` is set to 1.

88894/88998/89079 The phone now sends the correct option from the DHCP sub option 125.

- 88872/86090** In a shared call scenario, the shared registration configured on the first line of the phone is selected by pressing the New Call soft key when there are multiple lines registered on the phone and the other party of the shared line is busy.
- 88819/88403** In a network outage scenario, the phone no longer sends the invite messages after pressing the Cancel soft key.
- 88803** The phone now displays the default home view by pressing the Home button after the language is set to Japanese.
- 88774** The option to select the idle browser as the phones screen saver from the phone's web interface is no longer available.
- 88641** In a call transfer scenario, the phone no longer displays the Call Park soft key.
- 88531** Muting capabilities of the Voyager Legend Bluetooth headset have been optimized.
- 88528** In a BToE Scenario, the phone now sends the correct DTMF when a bridge number is copied and pasted on the Lync client.
- 88481** In a call center environment, the phone is now optimized to perform multiple operations of hunt group and BLF.
- 88410** The phone now plays the ringback tone when you try to call a mobile number with a SIP-URI header from the call logs.
- 88404** The phone now successfully disables the DND hard key by using the `key.16.VVX1500.function.prim="Null"` parameter and no longer causes a reboot.
- 88334** In a Lync Shared Line Appearance scenario, the MWI of the Delegate's phone no longer gets notified when the Boss phone receives a voice mail.
- 88284** In a Lync Shared Line Appearance scenario, the phone no longer reboots when the Boss signs-in while the admin is placing a call on hold and the Boss has more than 100 contacts (*applies to VVX 300/310*).
- 88259** The soft keys on the phone now follow the order specified with the `softkey.x.insert` parameter.
- 88202** In a Lync Shared Line Appearance scenario, the call status on the Delegate's phone and Boss's phone now updates correctly as "Delegate on behalf of Boss", when the Delegate places an outbound call on behalf of Boss, and the remote party holds and resumes the call.
- 88168/88396** In a Lync environment, the phone now updates the configured location correctly to the emergency 911 operator when the LIS subnet is configured on the Lync server, and the user signs-in with a Pin Authentication (*applies to VVX 300/310*).
- 88161** In a call center environment, the phone now enables or disables the Call Center Status" option by using the parameter `feature.callCenterStatus.enabled`.
- 88147** In a Lync Shared Line Appearance scenario, the Delegate's phone now plays the dial tone when the Delegate selects the Boss's line to place an on-behalf call.
- 88045** Contacts on the Local Contact Directory can be sorted either by first name or last name by enabling or disabling the parameter `dir.search.field`.
- 87894** In a Lync Shared Line Appearance scenario, the Boss's phone now retains the call appearance when the Delegate establishes a call on behalf of Boss, and the far end parks the call.

- 87891/87062/88497** The phone's screen now updates without any issue while using hookswitch in quick succession.
- 87825** The phone now successfully sends the DHCP request message for renewing the lease without turning on the screen and playing a notification tone when the NTP is not configured on the DHCP server.
- 87782** In a BToE scenario, the phone now displays "Presenter muted the audience" and turns the Mute key red when the Lync client mutes its initiated conference call.
- 87207** The SNTP address now displays correctly in the TCP/IP parameters menu when multiple IP addresses are configured in option 42 or option 4.
- 87706/88857** Double tapping the "*" key twice in a quick succession now converts into '+' when the `call.internationalDialing.enabled` is set to 0.
- 87152** In the DHCP parameter request list, the phone now successfully includes VLAN information under option 55.
- 86620** Recording a voice mail greeting no longer causes an issue when the video is enabled for different combinations of parameters when `video.allowWithSource` and `video.enable` are used.
- 85711/86597/88668** In a Lync environment, users now successfully sign in using Pin-Authentication when the pin contains leading zeros.
- 84781/89022** On a shared registration scenario, the phone now displays the calling party number on the phone's screen when you long press the line key after a consultative transfer.
- 81514/89062** In a Lync deployment, phone now fetches the location information from the available resources when it receives incomplete location information from LLDP.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 5.0.1 configuration file parameters.

Table 4: UC Software 5.0.1 Parameter Enhancements

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>Diags.dumpcore.enabled</code>	0 or 1	1	The phone no longer generates the core dump during a reboot/crash when set to 0.
<code>up.IdleViewPreferenceRemoteCalls</code>	0 or 1		<p>If set to 1, the call appearance displays for the remote party (SCA or BLF). The phone switches to idle view after there is no activity on remote party lines.</p> <p>On phone, the call appearance displays for remote party's call status and the phone switches to idle view after remote party answers the call.</p>

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>dialplan.userDial.timeOut</code>	1 to 99	3 for Lync 0 for Open SIP	In a Lync environment after the timeout of 3 seconds, you can enter a number and place the call using the first line regardless of the number of registered lines.
<code>dialplan.conflictMatchHandling</code>	0 or 1	0 for generic profile 1 for Lync profile	If the value is set to '0' when the digits entered match a digitmap, the digits are dialed immediately even though there are conflicting digitmaps. When the digits entered are matching more than one digitmap, the timeout is considered before dialing the digits.
<code>dialplan.x.conflictMatchHandling</code>	0 or 1	0 for generic profile 1 for Lync profile	If the value is set to '0', when the digits entered match a digitmap, the digits are dialed immediately even though there are conflicting digitmaps. When the digits entered are matching more than one digitmap, timeout is considered before dialing the digits.

Updates to Previous Software Releases

This section lists new, enhanced and discontinued software features in previous software releases.

Understanding Updates to UC Software 5.0.0

Polycom Unified Communications (UC) Software 5.0.0 is a general release for all open SIP platforms including Microsoft Lync Server 2010 and Microsoft Lync Server 2013. Note that if you are using Polycom UC Software 5.0.0 with Microsoft Lync Server, you can only register one line. You cannot register multiple lines on one phone with Lync Server.

Polycom UC Software 5.0.0 offers support for the following endpoint platforms:

- VVX[®] 300/310 business media phone
- VVX[®] 400/410 business media phone
- VVX[®] 500 business media phone
- VVX[®] 600 business media phone
- VVX[®] 1500 business media phone
- SoundStructure[®]

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.



Note: The support for Lync video calls with UC Software 5.0.0 is limited to the following scenarios when using Lync 2010:

1. In Lync 2010 environment Point-to-Point video calls between VVX 500/600 IP phone and another VVX 500/600 phone is supported.
2. Point-to-Point video calls between VVX 500/600 and Lync Client 2010 are supported.
3. Video Call transfers and Call forward from Lync Client party to another user with VVX 500/600 or vice-versa are supported.
4. Video sessions between a registered remote user and a video capable in-premise enterprise user (VVX 500/600 or Lync Client) are supported.
5. Video call sessions between federated users with endpoints as VVX 500/600 or Lync Client are supported.
6. Video calls in Lync 2013 environment are not supported due to lack of support to H.264 or RTV in UCS 5.0.0.

New or Enhanced Features

68809 Added support for the Lync Call Park feature.

72577 Added support for Premium extensions to the server-synchronized Automatic Call Distribution (ACD) feature (*applies to VVX 1500*).

75133 Added support for the Lync Better Together over Ethernet (BToE) feature.

75938 Added support for Premium ACD (*applies to VVX 1500*).

- 79861** Added support for the BroadSoft directory search, presence and favorites (*applies to VVX 300/310 and VVX 400/410*).
- 81724** Added support for the Lync Address Book Service (ABS).
- 82025** Added support for software upgrades from a Lync server.
- 82460** Added support for XT9 PinYin input for Chinese characters (*applies to VVX 300/310, VVX 400/410, VVX 500, and VVX 600*).
- 85344/87722/87839** Added a Lync status menu on the phone's web interface.

Enhanced Capabilities

- 88168** In a Lync environment, the phone now updates the configured location correctly to the emergency E911 operator when the LIS subnet is configured on the Lync server, and the user signs in with Pin Authentication.
- 87891/87062** The phone's screen now updates correctly after using hookswitch in quick succession.
- 87890** The Ethernet port information of the phone now displays correctly when the LLDP mode is enabled.
- 87560/87422** The phone's microbrowser invokes a backspace soft key (for non-touch based phones) or an on-screen key board (for touch based phones) when the user tries to enter numeric values.
- 87453/87632/87262** The phone now establishes a two-way audio path after successfully establishing a number of (~50) PTT and paging calls.
- 87440/87098** The digit map value now displays as empty on the phone's web interface when the `reg.1.applyServerDigitMapLocally` is set to 0.
- 87394/87599** In a call center environment, the phone no longer produces a beep sound when one of its monitored phones has its handset lifted, or receives a call, or goes off hook using any termination.
- 87389** The phone now follows the second slow start re-invites from the remote destination and forwards the audio and video stream to the respective ports (*applies to VVX 1500*).
- 87151** The enhanced feature keys Blind Transfer and Voice Mail Transfer now work correctly.
- 87112** Upgrading the touch screen firmware displays only the visual indicators and no longer shows the progress bar.
- 87092** The phone now establishes a two-way audio when the video is disabled, and a BroadWorks Anywhere extension is configured.
- 87076** The enhanced feature key \$Tinvite now works and no longer causes the phone to reboot.
- 87075** Placing an emergency call from an unregistered phone is now available from all the call terminations.
- 87063** The phone now accepts the MIME message without a SDP and no longer sends a 415 error message.
- 87059** Sending a visual voicemail from the phone is now available when the VVX camera is not attached, and the `video.enable` parameter is set to 1.

- 87032** In a Busy Lamp Field (BLF) scenario, the phone shows only the LED activity and does not show the call appearance when `attendant.behaviors.display.spontaneousCallAppearances.normal` is set to 0.
- 87031** The phone now retains the configured language even after a reboot.
- 87029** The MWI LED no longer blinks when the phone is in power-saving mode and has not received any new message.
- 87026** A password field entry page now displays on the phone's screen when the user tries to answer an incoming call in the locked state.
- 87025** A power-saving menu is now available on the phone's web interface (*applies to VVX 300/310 and VVX 400/400*).
- 86908/86909/87873** In a Lync environment, the phone now dials the entered number according to the `dialplan.1.lyncdigitmap.timeout` parameter.
- 86877/84668** The Send soft key no longer displays when there are no entries present on the phone's dialer screen (*applies to VVX 400/410*).
- 86873/86256/86569/85368/85957** In a Lync deployment, the phone's stability is improved for different network environments.
- 86772** Soft key strings on the phone's microbrowser now format correctly for French.
- 86730/86869/87504** The phone now transfers a call successfully when the language is set to German, Japanese, Chinese, Taiwanese, and Korean.
- 86709/85370** The call logs menu on the phone now does not update for the missed calls when the parameter `call.advancedMissedCalls.reasonCodes=487` is set (*applies to VVX 500 and VVX 600*).
- 86636** In a call center environment, the LED status on the phone now updates correctly for an incoming call on a shared line (*applies to VVX 300/310 and VVX 400/410*).
- 86623** The phone now plays the defined custom ringtone for a directory contact.
- 86575/85968/84650/83991** The phone now successfully downloads the updates from the `mac-directory.xml` file and modifies the speed dial list correctly.
- 86568** The user can successfully disable the call forward no answer from the phone's screen when the parameter `reg.1.serverFeatureControl.cf` is set to 1.
- 86425/84572** The phone's status menu now updates the AOR details with the primary server when there are two active servers for a single registered number.
- 86324** The default contrast level on the phone is now set to 1 to improve the video quality (*applies to VVX 500 and VVX 600*).
- 86277** The phone now honors the 302 moved temporary requests and performs the corresponding DNS lookup correctly.
- 86276/86767** The phone no longer updates the `directory.xml` file when the presence of the monitored buddy changes.
- 86261** The phone now establishes a two-way audio path when the BroadWorks Anywhere extensions are configured as mobile or PSTN numbers.

- 86237** The phone gets a grace period of an extra 25% of the validity date of the TLS-DSK certificate and does not deregister during that period.
- 86206** The phone now displays only the Conf soft key in the active call state when the parameter `softkey.feature.basicCallManagement.redundant` is set to 0.
- 86113** The phone now removes the BroadWorks favorite contacts from the speed dial menu on the phone's idle screen when the BroadSoft Directory is disabled by setting the parameter `feature.broadsoftdir.enabled` to 0.
- 86075** Sending a voicemail from the phone is now available when the parameters `msg.mwi.1.callBackMode="contact"` and `msg.mwi.1.callBack="voicemail"` are configured (*applies to VVX 400/410*).
- 86054** The phone now successfully receives the ICMP message "Fragmentation needed" without any failover.
- 85962/85984/86764** The phone now displays both the Start and Ping soft keys when the user tries to trace an IP from the phone's menu for the second time and immediately after cancelling the first attempt.
- 85840/85148/84908** In the locked state, the phone no longer initiates the call to the emergency E911 upon off-hooking the handset twice in quick succession, for example, within 2 seconds.
- 85794/86721** The phone now sends a publish request to the VQMon server when the VQMon server is configured with the fully qualified domain name (FQDN).
- 85728/86157** In a shared call scenario, the LED or line status on the phone now changes and shows a solid red color when the incoming call is answered by one of the remote destinations.
- 85408/87180** The ringback for outbound PSTN calls is now optimized and no longer produces any double ringback.
- 85403** In a Lync call server deployment, the phone now adds the '+' prefix automatically for emergency E911 dialing, irrespective of the dial plan.
- 85373** In a Lync environment, the phone now retains the registration when federated contacts (~60) are added and receives 504 server timeout errors from the server.
- 85109/84725/84428/84892** The subnet mask now correctly updates on the phone's screen and the serial port when the DHCP sends an option 43 vendor class.
- 85105** The Hold/Mute keys are now non-functional when the user tries to place a call in the locked state.
- 85034** Pressing the Select hard key in the phone's corporate directory now displays the details of the contact on the phone's screen.
- 84974** Enabling call forward using the phone's web interface is now available.
- 84931** The power-saving state on the phone is now optimized while using the BroadSoft Event Hoteling package (*applies to VVX 500*).
- 84905** The More soft key no longer displays on the ACD queue status pop-up message when the user acts as an agent.
- 84823** Audio quality on the speaker is improved when the user mutes and unmutes the speaker while receiving a page.

- 84815** When the user tries to edit the Desktop User option on the Polycom Desktop Connector, the << soft key is now available on the phone's screen.
- 84776** In a BLF scenario, the user can now place a call from the filtered view of the monitoring phone when one of the monitored users is handling a large number (~24) of active calls.
- 84717** The web browser now displays the volume index bar on the phone's screen when the user tries to increase the volume for a media file.
- 84664** The phone now returns to the idle screen when the user presses any termination key without initiating a call from the directory menu on the dialer screen.
- 84643** In a Lync environment, the phone now correctly normalizes the emergency E911 dial plan.
- 84534/82075/86543** The phone is now able to park and retrieve the call successfully from the park orbit when the parameters `reg.1.lineKeys` and `reg.1.callsPerLineKey` are set to 1.
- 84518** In a federation call scenario, the phone now transfers the calls successfully between organizations with two-way audio.
- 84471/84341** The graphical user interface response on the phone is improved when the phone is used for a long time.
- 84271/82209** The date and time are now arranged successfully on the phone screen using the parameters `lcl.ml.lang.clock.1.dateTop` and `lcl.datetime.date.dateTop`.
- 84161/80529/87856** The selection of ringtone patterns in multiple scenarios is optimized.
- 83854** In a BLF scenario, the New Call soft key now displays on the incoming call screen of the monitoring user when there is an incoming call to one of the monitored users.
- 83619/86104** The phone now sends 'A' as the standard query to the DNS server when the phone uses IPv4.
- 83806/84304/86642/87028/88324/88325** The "Clear browsing data" and "Return to calls" strings on the phone's menu now display in the configured language.
- 83708** The phone now successfully answers an incoming call when it is transferred through the Exchange Auto Attendant.
- 82781** In a call forward scenario with Lync deployment, the phone now shows the details of both the caller and the call forwarder.
- 82044** The phone's menu is now optimized to hide all the soft key impressions of the previous menu.
- 80852/84844** The phone's touch screen responsiveness after a call or after using an application is improved (*applies to VVX 500 and VVX 500*).
- 79723/80958** The phone now plays 183 early media and 180 local ringback independently and one after the other.
- 76769/82121** Calls without SIP DISPLAY INFO now log under the missed or received call lists.
- 76468** In a Hoteling scenario, the phone now fetches the previous status of the guest after a reboot.
- 73942** In a shared line or a BLF scenario, missed calls listed on the phone no longer increment when the other destination answers the call.
- 37220/54278** In a Message Waiting Indicator (MWI) scenario, the phone now plays a beep sound and lights up the backlight only when the phone receives a new message.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 5.0.0 configuration file parameters.

Table 5: UC Software 5.0.0 Parameter Enhancements

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>mwi.backLight.disable</code>	0 or 1	0	If set to 1, backlight is not illuminated on new voice message arrival.
<code>call.advancedMissedCalls.addToReceivedList</code>	0 or 1	1	Call answered from the remote destination on a shared line is logged in the received call list. If the parameter is set to 0, answering the call from the remote destination on shared line does not log the call on the call lists.
<code>feature.lync.abs.enabled</code>	0 or 1	1	Enables/disables the contact search using Lync 2010 address book service.
<code>feature.lync.abs.maxResult</code>	5 to 50	20	Defines the permitted number of contacts to display for a given contact search.
<code>keyboard.layout.type</code>	0 or 1	0	If set to 1 and the language is changed to French under keyboard layout, the virtual keyboard under all the menus/submenus changes to AZERTY. If set to 0, the virtual keyboard is set to QWERTY by default.
<code>up.numOfDisplayColumns</code>	1 to 4	3 (VVX 500) 4 (VVX 600)	Parameter to limit the number of keys/contacts a VVX 500 or VVX 600 can display. 1 column displays when parameter value is 0 or 1. 2 columns display when parameter value is 2. 3 columns display when parameter value is 3. 3 columns (for VVX500) and 4 columns (for VVX 600) display when parameter value is 4.

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
keyboard.encoding.all	0 or 1	1	If set to 0, the encoding options that are present under menus/submenu (for example, under directories, provisioning server) display only ASCII and Latin. If set to 1, the phone displays the default encoding option.
device.prov.lyncDeviceUpdateEnabled.set	0 or 1	0	If base profile is Generic, the default value is 0. If base profile is Lync, the default value is 1.
device.prov.lyncDeviceUpdateEnabled	0 or 1	0 or 1	To enable/disable the Lync device update feature. If base profile is Generic, the default value is 0. If base profile is Lync, the default value is 1.
lync.deviceUpdate.popUpSK.enabled	0 or 1	1	To enable/disable the pop-up message that shows available software updates to the user at runtime.
lync.deviceUpdate.serverPollInterval	1800 s (30 min) to 28800 s (8 hours)	7200 s (2 hours)	Integer value in seconds indicating the interval at which the phone sends a software update request to the server.
lync.deviceUpdate.userInactivityTimeout	300s to 1800 s	900 s	Integer value in seconds indicating the user inactivity timeout value upon which the software update, if available on the server, triggers from the phone.
feature.btoe.enabled	0 or 1	0	When this parameter is enabled, the phone can initiate pairing with the Lync Client
feature.lyncbtoe.auto signin.signoff.enabled	0 or 1	0	By enabling this parameter, the phone automatically signs out of the Lync registration by unpairing the phone with PC/Laptop. Pre-condition: IP phone should be initially signed in using Logon Information window on Lync PC client.
dialplan.impossiblematch handling	0, 1, 2, 3	3 (Lync profile) 0 (Generic profile)	When this parameter is set to 3, in case of an impossible match, the digits entered dial out after a timeout of <code>dialplan.x.impossibleMatchHandling.timeout</code> or <code>dialplan.impossiblematchhandling.timeout</code> .

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
dialplan.x.impossibleMatch Handling.timeOut	0 to 100	8	x is the line number, for example, 1, 2, 3. When the dialed digits match the impossible match, the digits entered dial out after this time.
dialplan.impossibleMatch Handling.timeOut	0 to 100	8	When the dialed digits match the impossible match, the digits entered dial out after this time.
audioVideoToggle.callMode. persistent	0 or 1	1	If set to 1, the selected call mode is retained for all the subsequent calls even after reboot/restart. If set to 0, the selected mode applies only to that call. All subsequent calls are in default mode.
video.allowWithSource	0 or 1	0	If set to 1 and camera is detached, no video codecs are advertised in SDP. If set to 1 and camera is attached, the video codecs are advertised in SDP. If set to 0 and camera is attached or detached, video codecs are advertised in SDP (default behavior).
device.prov.lyncDeviceUpdateUser.set	0 or 1	0	Indicates if the device.prov.lyncDeviceUpdateUser parameter is to be used.
device.prov.lyncDeviceUpdateUser	String (max length of 256 characters)	""	Stores the Lync account user name to be used while updating from remote network.
device.prov.lyncDeviceUpdatePassword.set	0 or 1	0	Indicates if the device.prov.lyncDeviceUpdatePassword parameter is to be used.
device.prov.lyncDeviceUpdatePassword	String (max length of 256 characters)	""	Stores the Lync account password to be used while updating from remote network
device.prov.lyncDeviceUpdateDomain.set	0 or 1	0	Indicates if the device.prov.lyncDeviceUpdateDomain parameter is to be used.
device.prov.lyncDeviceUpdateDomain	String (max length of 256 characters)	""	Stores the Lync account domain to be used while updating from remote network.

Understanding Updates to UC Software 4.1.4

Polycom Unified Communications (UC) Software 4.1.4 is a general release for all open SIP platforms and is compliant with Microsoft® Lync® Server 2010. Note that if you are using Polycom UC Software 4.1.4 with Microsoft Lync Server, you can only register one line. You cannot register multiple lines on a phone with Lync Server.

Polycom UC Software 4.1.4 offers support for the following phone platforms:

- VVX 300/310 business media phones
- VVX 400/410 business media phones
- VVX 500 business media phones
- VVX 600 business media phones



Note: Visual Voice Mail Not Supported

Visual Voice Mail support for the Metaswitch server is not fully qualified in this release and is currently unavailable for these platforms.

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

71881 Added support for VVX 300/310 and VVX 400/410.

Enhanced Capabilities

No changes were made from the previous UC software release.

Understanding Updates to UC Software 4.1.3 Rev G

Polycom Unified Communications (UC) Software 4.1.3 Rev G is a general release for all open SIP servers including Microsoft Lync Server and is available on the following endpoint platforms:

- Polycom VVX 500 business media phone
- Polycom VVX 600 business media phone
- Polycom SoundStructure



Note: VVX Camera Support

UC Software 4.1.3 Rev G offers video support for the VVX Camera for all the open SIP servers except Microsoft Lync. UC Software 4.1.3 Rev G supports all other features of Microsoft Lync.



Web Info: Polycom® BroadSoft UC-One Features and Configuration Details

For information on Polycom BroadSoft UC-One features and configuration procedures, available for the VVX 500 and VVX 600 business media phones, see [Feature Profile 84393: Using the Polycom BroadSoft UC-One Application on Polycom VVX Business Media Phones](#) on the Polycom Support web site.

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 79861** Added support for BroadSoft directory search, presence and favorites.
- 84273** A video mute icon now displays on the phone's local video pane when the VVX camera shutter is closed.

Enhanced Capabilities

- 85148/84908** In the locked state, the phone no longer initiates the call to the emergency E911 upon off-hooking the handset twice in quick succession, for example, within two seconds.
- 84954** SoundStructure Studio now shows the correct status when the SoundStructure card is restarted from the web configuration page.
- 84943/84991** The phone now supports long URIs up to 256 characters in length.
- 84651** In a BLF scenario, the phone is now able to answer an incoming call using a headset when one of the monitored users is in an active or altering state (*applies to VVX 500*).
- 84608/84751** The new parameter `voIpProt.SIP.supportFor100rel` handles 1XX provisional responses.
- 84439** When both call servers fail, the phone first tries the primary server. It then fails over to the secondary server and back to the primary server. This loop continues until the phone registers with one of the servers.
- 84407/84390** In a conference call scenario, audio path loss no longer occurs when a Lync PC client is re-added to a conference call with CX 600 and a video-enabled VVX phone (*applies to VVX 500 and VVX 600*).
- 84299** Call appearance details on the studio UI console now display when a call is made from a private line (*applies to SoundStructure*).

- 84221** Establishing a federated call using the long URI from the phone contacts or quick dialed list is now available.
- 84138/84139** After a call, the presence status of the user now updates correctly on the phone when the parameter `reg.1.callsPerLineKey` is set to 1.
- 84019** The line key now updates with a 10 digit number immediately after a successful Lync registration (*applies to SoundStructure*).
- 83945** The phone is now able to auto answer an incoming call with multiple call-info headers when the parameters `voIpProt.SIP.alertInfo.1.value="auto-answer"` and `voIpProt.SIP.alertInfo.1.class="autoAnswer"` are set.
- 83781** The phone lock option is removed (*applies to SoundStructure*).
- 83776** The idle browser on the phone appears automatically after a reboot/restart without displaying an "Invalid Hostname" message.
- 83564/84405** On the Russian SKUs, only the SIP signaling is encrypted during in-band provisioning, leaving the media unencrypted.
- 83460** The phone is now able to resume an outbound PSTN held call successfully when the user dials the PSTN number without any area code.
- 83326** Scrolling through the contact directory's menu using * or # now works.
- 83308** During Do Not Disturb (DND), the phone's missed call lists are now correctly updated.
- 83069** Music on Hold (MOH) no longer disconnects when the phone uses SRTP traffic.
- 83022** The phones now send SIP INFO packet with a new line between content attributes.
- 82915** Added the configuration parameter `tcpIpApp.snmp.AQuery`. If set to 1, the phone sends a DNS query directly for NTP requests.
- 82319/83255/83806/83834** The phone now displays all the new pop-up messages and strings in the configured language instead of the default language.
- 81639** The phone is now able to handle the bit rate mismatch during a call for G.722.1 codec by setting the parameter `voIpProt.SIP.fmtPMustForG7221` to 1.
- 81554** In a call center scenario, the phone now displays the correct status and soft keys on the phone's screen after a reboot.
- 81327** Changing the static BLF line key configurations on the web interface no longer affects the configured soft keys.
- 80578** The user can now enable the exchange calendar feature when connecting to the hosted exchange server.
- 80293** The "Service Unavailable" message no longer displays on the phone's screen when a paging soft key is enabled and then disabled.
- 80125** Pressing ACD Sign-In/Sign-Out keys quickly now works correctly and does not take the phone to the unavailable state.
- 79846** The phone now retains the server-based DND even after the phone is locked.

77877 Missed or received calls are now logged in the respective configured list when the parameters `call.advancedMissedCalls.enabled` and `call.advancedMissedCalls.addToReceivedList` are set.

77097 The phone's user interface can now handle a 486 busy response.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 4.1.3 Rev G configuration file parameters.

Table 6: UC Software 4.1.3 Rev G Parameter Enhancements

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>feature.qml.enabled</code>	0 or 1	0	If 1, the QML viewer is enabled on phone. If 0, the viewer is disabled. The viewer is used to load the QML applications.
<code>feature.broadsoftdir.enabled</code>	0 or 1	0	If 1, the BroadSoft Enterprise directory is enabled. If 0, the directory is disabled.
<code>feature.broadsoftUcOne.enabled</code>	0 or 1	0	If 1, the BroadSoft UC-One feature is enabled. If 0, the feature is disabled.
<code>dir.broadsoft.xsp.address</code>	Dotted-decimal IP address or hostname or FQDN	Null	The IP address or hostname of the BroadSoft XSP server. For example, <code>http://xsp-idc1.asia.polycom.com</code> .
<code>dir.broadsoft.xsp.username</code>	String	Null	The username used to authenticate the BroadSoft XSP server.
<code>dir.broadsoft.xsp.password</code>	String	Null	The password used to authenticate the BroadSoft XSP server.
<code>feature.presence.enabled</code>	0 or 1	0	To enable the BroadSoft presence.
<code>dir.broadsoft.qml.home</code>	String	<code>"file:///usr/local/apps/BSD/BSD.qml"</code>	To set the BroadSoft Directory QML home address
<code>xmpp.1.auth.password</code>	True/false	True (1)	Password used for XMPP registration. When provisioned from CMA, the value is set to the CMA account password.
<code>xmpp.1.dialMethod</code>	String min 0, max 256	sip	For SIP dialing, the destination XMPP URI is converted to a SIP URI, and the first available SIP line is used to place the call.

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
xmpp.1.enable	True/false	False(0)	Flag to determine if XMPP presence is enabled.
xmpp.1.jid	String min 0, max 256	""	Jabber identity used to register with presence server.
xmpp.1.jid	Duplicate		
xmpp.1.server	String min 0, max 256	""	Presence server IP or FQDN.
xmpp.1.verifyCert	True/false	True(1)	Flag to determine if the phone should verify the TLS certificate provided by the XMPP presence server.

Understanding Updates to UC Software 4.1.3

Polycom Unified Communications (UC) Software 4.1.3 is a general release for all open SIP servers and offers video support for the VVX Camera on the following phone platforms:

- VVX 500 business media phone
- VVX 600 business media phone

UC Software 4.1.3 does not support Lync video on Microsoft Lync Server. It does support all other Microsoft Lync Server features available in UC Software 4.1.2.

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

62655 Added support for the Polycom VVX Camera.

82204 Added video support (SIP and H323) on the VVX 500 and VVX 600 business media phones.

Enhanced Capabilities

84089 The phone now does not show the incoming call appearance pop-up message for the second time when the call is answered after the initial pop-up message has timed out.

83752 The phone's UI/display is improved when switching between active call and other menus.

83725/81683 The phone is now able to transfer the location information to E911 successfully using NTLM credentials in MS Lync Server.

83716 In a BLF scenario, phone is now able to retain the monitored user when the registration switches between private and shared.

-
- 83513** The phone is now equipped to display the E911 disclaimer pop-up message when set on the Lync server.
- 83273** Muting and unmuting the call by double-tapping the mute area on Jabra 9465 USB headset/touch control is now working correctly.
- 82898** Changing the SIP line to a "SIP + H.323" line using the phone's web interface during an active call does not cause any call drop or reboot.
- 82282** The calendar parser on the phone is now able to correctly identify the 'dial in numbers' for conference calls.
- 82149** In a BLF scenario, the phone no longer displays the call appearance of the monitored parties when the parameter `attendant.behaviors.display.spontaneousCallAppearances.normal="0"` is set.
- 82073** Answering a call using EHS is now available on the phone when the BLF status is active for any monitored user.
- 82051/81016** Blind transfer from the contact directory/recent call list/favorites is now working correctly.
- 82005** When URL dialing is disabled and the user tries to transfer an active call made to a BLF-monitored line, the prompt screen no longer displays.
- 81541** In a call center scenario, the guest user is now able to sign out from the host phone in a single attempt after the guest has already signed in or out once.
- 81504** In a call center scenario, the phone now updates the previous state of the user after a line re-registration is performed.
- 81439** In a call center scenario, the phone no longer displays the Back soft key during an incoming call to an ACD line.
- 81245** An incoming call pop-up message no longer flashes on the phone's screen when the call is answered after 10 seconds.
- 80959/79673/79242** Provisioning the phone using the SIP-TLS option now functions correctly.
- 79915** Local contact search on the phone now displays all the contacts when there is no search criterion.
- 79330** The phone now allows the STP traffic from the phone's PC port.
- 76769** Missed and received calls are now updated on the phone when the SIP DISPLAY INFO portion is missing in the initial SIP INVITE.
- 75937** The phone's browser now updates the content correctly during scrolling when the toolbar is in a hidden state.
- 74871** In a PTT call, a beep sound now plays on the phone from the first call, indicating the start of voice transmission.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 4.1.3 configuration file parameters.

Table 7: UC Software 4.1.3 Parameter Enhancements

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>call.autoAnswer.H323</code>	0 or 1	0	If 0, auto-answer is disabled for H.323 calls. If 1, auto-answer is enabled for all H.323 calls.
<code>call.autoAnswer.videoMute</code>	0 or 1	0	If 0, video begins transmitting (video Tx) immediately after a call is auto-answered. If 1, video transmission (video Tx) is initially disabled after a call is auto-answered.
<code>call.autoRouting.preference</code>	line or protocol	line	<p>If set to line, calls are placed on the first available line, regardless of its protocol capabilities. If the first available line has both SIP and H.323 capabilities, the preferred protocol is used (<code>call.autoRouting.preferredProtocol</code>). If set to protocol, the first available line with the preferred protocol activated is used, if available. If not available, the first available line is used.</p> <p>Note: Auto-routing is used when manual routing selection features (<code>up.manualProtocolRouting</code>) are disabled.</p>
<code>call.autoRouting.preferredProtocol</code>	SIP or H323	SIP	If set to SIP, calls are placed through SIP if available, or through H.323 if SIP is not available. If set to H323, calls are placed through H.323 if available, or through SIP if H.323 is not available.
<code>call.enableOnNotRegistered</code>	0 or 1	1	<p>If 1, users can make calls when the phone is not registered. If 0, calls are not permitted without registration.</p> <p>Note: Setting this parameter to 1 allows VVX phones to make calls using the H.323 protocol even though an H.323 gatekeeper is not configured.</p>
<code>feature.audioVideoToggle.enabled</code>	0 or 1	0	If 0, the audio/video toggle feature is disabled. If 1, the feature is enabled.
<code>reg.x.protocol.H323</code>	0 or 1	0	If 0, H.323 signaling is not enabled for registration x. If 1, H.323 signaling is enabled.
<code>reg.x.server.H323.y.address</code>	dotted-decimal IP address or hostname	Null	Address of the H.323 gatekeeper.

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>reg.x.server.H323.y.port</code>	0 to 65535	0	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used.
<code>reg.x.server.H323.y.expires</code>	positive integer	3600	Desired registration period.
<code>sec.H235.mediaEncryption.enabled</code>	0 or 1	1	If 0, H.235 Voice Profile RTP media encryption is disabled. If 1, H.235 media encryption is enabled and negotiated when such encryption is requested by the far end.
<code>sec.H235.mediaEncryption.offer</code>	0 or 1	0	If 0, media encryption negotiations are not initiated with the far end. If 1 and <code>sec.H235.mediaEncryption.enabled</code> is also 1, media encryption negotiations are initiated with the far end. However, successful negotiations are not a requirement for the call to complete.
<code>sec.H235.mediaEncryption.require</code>	0 or 1	0	If 0, media encryption negotiations are not required. If 1 and <code>sec.H235.mediaEncryption.enabled</code> is also 1, media encryption negotiations are initiated or completed with the far end; if negotiations fail, the call is dropped.
<code>sec.srtp.lifetime</code>	0, positive integer minimum 1024 or power of 2 notation	0	<p>The lifetime of the master key used for the cryptographic parameter in SDP. The value specified is the number of SRTP packets. If 0, the master key lifetime is not set. If set to a valid value (at least 1024, or a power such as 2^{10}), the master key lifetime is set. When the lifetime is set, a re-invite with a new key is sent when the number or SRTP packets sent for an outgoing call exceeds half the value of the master key lifetime.</p> <p>Note: Setting this parameter to a non-zero value might affect the performance of the phone. If you are using a VVX Camera on a VVX 500 and 600 phone, set to 2^{31}.</p>

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>up.manualProtocolRouting</code>	0 or 1	1	If 1, the user is presented with a protocol routing choice in situations where a call can be placed using either protocol (for example, with SIP and H.323 protocols). If 0, the default protocol is used, and the user does not choose.
<code>up.manualProtocolRouting.softKeys</code>	0 or 1	1	Enables soft keys that control Manual Protocol Routing options. When enabled, you can use soft keys to choose between the SIP or H.323 protocol. When disabled, soft keys for protocol routing do not display. Enabled by default.
<code>video.autoFullScreen</code>	0 or 1	0	If 0, video calls only use the full screen layout if it is explicitly selected by the user. If 1, video calls use the full screen layout by default, such as when a video call is first created or when an audio call transitions to a video call.
<code>video.autoStartVideoTx</code>	0 or 1	1	When enabled, video transmission to the far side begins when you start a call. When disabled, video transmission does not begin until you press the Video > Start Video soft keys. This parameter controls video sent to the far side. Video from the far side always displays if it is available, and far side users can control when to send video.
<code>video.callMode.default</code>	audio or video	audio	Allows you to select the mode to use when using SIP protocol only.
<code>video.callRate</code>	128 to 2048	512	The default call rate (in kbps) to use when initially negotiating bandwidth for a video call.
<code>video.camera.brightness</code>	0 to 6	3	Set brightness level. The value range is from 0 (dimpest) to 6 (brightest).
<code>video.camera.contrast</code>	0 to 4	0	Set contrast level. The value range is from 0 (no contrast increase) to 3 (most contrast increase), and 4 (noise reduction contrast).

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
video.camera.flicker Avoidance	1 to 2	1	Set flicker avoidance. If set to 1, 50Hz AC power frequency flicker avoidance (Europe/Asia). If set to 2, 60Hz AC power frequency flicker avoidance (North America).
video.camera.frameRate	5 to 30	25	Set target frame rate (frames per second). Values indicate a fixed frame rate, from 5 (least smooth) to 30 (most smooth). Note: If video.camera.frameRate is set to a decimal number, the value 25 is used.
video.camera.saturation	0 to 6	3	Set saturation level. The value range is from 0 (lowest) to 6 (highest).
video.camera.sharpness	0 to 6	3	Set sharpness level. The value range is from 0 (lowest) to 6 (highest).
video.codecPref.H264	1 to 4	1	Specifies the video codec preferences.
video.codecPref.H263		3	
video.dynamicControlMethod	0 or 1	0	If 1, the first I-Frame request uses the method defined by video.forceRtcpVideoCodecControl, and subsequent requests alternate between RTCP-FB and SIP INFO.
video.enable	0=Disable, 1=Enable	0	If 0, video is not enabled and all calls—both sent and received—are audio only. If 1, video is sent in outgoing calls and received in incoming calls (if the other device supports video).
video.enable for VVX 500		0	
video.enable for VVX 600		0	
video.forceRtcpVideoCodec Control	0 or 1	0	If set to 1, the phone is forced to send RTCP feedback messages to request fast update I-frames for all video calls (the phone includes a=rtcp-fb in the SDP). If 0, RTCP feedback messages are not forced.
video.iFrame.delay	0 to 10, seconds	0	When non-zero, an extra I-frame is transmitted after video starts. The amount of delay from the start of video until the I-frame is sent is configurable up to 10 seconds. Use a value of 2 seconds if you are using this parameter in a Microsoft Lync environment.

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>video.iFrame.minPeriod</code>	1 - 60	2	After sending an I-frame, the phone always waits this minimum amount of time before sending another I-frame in response to requests from the far end.
<code>video.iFrame.onPacketLoss</code>	0 or 1	0	If 1, an I-frame is transmitted to the far end when a received RTCP report indicates that video RTP packet loss has occurred.
<code>video.localCameraView.fullscreen.enabled</code>	0 or 1	1	Determines whether the local camera view is shown in the full screen layout. If set to 0, the local camera view is not shown. If set to 1, the local camera view is shown.
<code>video.localCameraView.fullscreen.mode</code>	pip, side-by-side	pip	Determines how the local camera view is shown. If set to pip, the local camera view displays as a picture-in-picture with the far end window. If set to side-by-side, the local camera view displays side-by-side with the far end window.
<code>video.maxCallRate</code>	128 to 2048 kbps	768	The maximum call rate allowed. This allows the administrator to limit the maximum call rate that the users can select. If <code>video.callRate</code> exceeds this value, this value is used as the maximum.
<code>video.profile.H263.CifMpi</code>	1 to 32	1	Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-4. To disable, enter '0'.
<code>video.profile.H263.jitterBufferMax</code>	(<code>video.profile.H263.jitter BufferMin</code> + 500ms) to 2500ms	2000ms	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size always causes lost packets. This parameter should be set to the smallest possible value that supports the expected network jitter.
<code>video.profile.H264.jitterBufferMax</code>	(<code>video.profile.H264.jitter BufferMin</code> + 500ms) to 2500ms		

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
video.profile.H263. jitterBufferMin video.profile.H264. jitterBufferMin	33ms to 1000ms	150ms	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth is achieved, the depth may fall below this point and play out still continues. This parameter should be set to the smallest possible value, which is at least two packet payloads and larger than the expected short-term average jitter.
video.profile.H263. jitterBufferShrink video.profile.H264. jitterBufferShrink	33ms to 1000ms	70ms	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (33 ms) to minimize the delay on known good networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).
video.profile.H263. payloadType	96 to 128	34	RTP payload format type for H263 MIME type.
video.profile.H264. payloadType	96 to 128	109	RTP payload format type for H264/90000 MIME type.
video.profile.H264. profileLevel	1, 1b, 1.1, 1.2, and 1.3 (1 to 2)	1.3	Specify the highest profile level within the baseline profile supported in video calls. The phone supports the following levels: 1, 1b, 1.1, 1.2, 1.3. The default level is 1.3. For more information, refer to ITU-T H.264.
video.profile.H263.QcifMpi	1 to 32	1	Specify the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter '0'.
video.profile.H263.SqcifMpi	1 to 32	1	Specify the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. You can enter a value between 0-32. To disable, enter '0'.

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>video.quality1</code>	motion, sharpness	Null	The optimal quality for video that you send in a call or a conference. Use motion if your outgoing video has motion or movement. Use sharpness or Null if your outgoing video has little or no movement. Note: If motion is not selected, moderate to heavy motion causes some frames to drop.
<code>video.screenMode</code>	normal, full, crop	normal	The screen mode for the video window shown in non-full screen mode. If set to normal or Null, the entire view displays and horizontal or vertical black bars might appear on the edges to maintain the correct aspect ratio. If set to full, the entire view stretches linearly and independently to fill the video frame. If set to crop, black bars do not show, the image re-sizes and enlarges to cover the entire video frame, and parts of the image that do not fit in the display are cropped (removed).
<code>video.screenModeFS</code>	normal, full, crop	normal	The screen mode for the video window shown in full screen mode. If set to normal or Null, the entire view displays and horizontal or vertical black bars might appear on the edges to maintain the correct aspect ratio. If set to full, the entire view stretches linearly and independently to fill the screen. If set to crop, black bars do not show, the image re-sizes and enlarges to cover the entire screen, and parts of the image that do not fit in the display are cropped (removed).
<code>voIpProt.H323.autoGateKeeperDiscovery</code>	0 or 1	1	If set to 1, the phone attempts to discover an H.323 gatekeeper address through the standard multicast technique, provided that a statically configured gatekeeper address is not available. If set to 0, the phone does not send out any gatekeeper discovery messages.
<code>voIpProt.H323.blockFacilityOnStartH245</code>	0 or 1	0	If set to 1, facility messages when using H.245 are removed.

<i>Parameter</i>	<i>Permitted Value</i>	<i>Default</i>	<i>Description</i>
<code>voIpProt.H323.dtmfViaSignaling.enabled</code>	0 or 1	1	If set to 1, the phone uses the H.323 signaling channel for DTMF key press transmission.
<code>voIpProt.H323.dtmfViaSignaling.H245alphanumericMode</code>	0 or 1	1	If set to 1, the phone supports H.245 signaling channel alphanumeric mode DTMF transmission. Note: If both alphanumeric and signal modes are used, the phone gives priority to DTMF.
<code>voIpProt.H323.dtmfViaSignaling.H245signalMode</code>	0 or 1	1	If set to 1, the phone supports H.245 signaling channel signal mode DTMF transmission.
<code>voIpProt.H323.enable</code>	0 or 1	0	A flag to determine if the H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing. If set to 1, the H.323 protocol is used.
<code>voIpProt.H323.local.port</code>	1 to 65535	1720	Local port to be used for H.323 signaling. Local port for sending and receiving H.323 signaling packets. If set to 0, 1720 is used for the local port but is not advertised in the H.323 signaling. If set to another value, that value is used for the local port and is advertised in the H.323 signaling.
<code>voIpProt.H323.local.RAS.port</code>	1 to 65535	1719	Local port for RAS signaling.
<code>voIpProt.server.H323.x.address</code>	dotted-decimal IP address or hostname	Null	Address of the H.323 gatekeeper. Note: Only one H.323 gatekeeper per phone is supported. If more than one is configured, only the first is used.
<code>voIpProt.server.H323.x.port</code>	0 to 65535	1719	Port to be used for H.323 signaling. Note: The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP.
<code>voIpProt.server.H323.x.expires</code>	positive integer	3600	Desired registration period.

Understanding Updates to UC Software 4.1.2 Rev B

Polycom Unified Communications (UC) software 4.1.2 Rev B is a general release for the VVX 500 and VVX 600 products with all supported SIP servers including Microsoft Lync Server 2010 and qualified

compatible Microsoft Lync Server 2013. For Microsoft Lync Server, this release supports a single registered line.



Note: UC Software 4.1.x - New and Enhanced Features

For information on new and enhanced features and capabilities available with Lync-compatible UC software, Microsoft Lync users can refer to [Understanding Updates to UC Software 4.1.0 Rev B](#).

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

82450 The phone now sends SIP INVITE and REGISTER messages in the MS-Subnet header.

Enhanced Capabilities

81683 The phone now fetches the Location Information Service by using either NTLM credentials or the Lync-signed user certificate.

83754 Restarting the phone after a network outage automatically registers and signs in the user.

83564 On the Russian SKUs, only the SIP signaling is encrypted during in-band provisioning, leaving the media unencrypted.

84105 Frequent multiple Sign in- Sign out no longer causes the phone to reboot.

84218 The phone no longer caches the call history of the previously signed user.

84221 The phone can now dial to a federated contact with a long URI from the quick dial list.

84298 Phone calls now comply with the enterprise Lync server bandwidth policy.

84305 The risk of accidentally dialing an emergency number in the locked state is reduced. The user now has to select a number from the authorized contact list and press the send button to initiate a call.

Understanding Updates to UC Software 4.1.2

Polycom Unified Communications (UC) software 4.1.2 is a general release for the VVX 500 and VVX 600 products with all supported SIP servers, including Microsoft Lync Server 2010. For Microsoft Lync Server, this release supports a single registered line.



Note: UC Software 4.1.X - New and Enhanced Features

Microsoft Lync users can refer to [Understanding Updates to UC Software 4.1.0 Rev B](#) for information on new and enhanced features and capabilities available with Lync-compatible UC software.

New or Enhanced Features

- 62651** Added Support for the Polycom VVX 600 product.
- 82042** Added support for BroadSoft Hoteling Event Package.
- 80220** Added support for Pin Authentication on Microsoft Lync.
- 82162** Added support for Verizon (Baltimore CA 2048) certificate.

Enhanced Capabilities

- 82101** The “mute” hard key and mute display on the phone while initiating a conference call are now synchronized.
- 81878** On the phone’s virtual keyboard, the input mode key no longer overlaps with other keys on the screen.
- 81536** On the contact entry/edit screen, the default encoding “Abc” mode is changed to “abc” by pressing the “Caps” key on the phone virtual keyboard.
- 80790** The phone no longer deregisters when trying to install an unknown certificate.
- 80761** When using Polycom Desktop Connector (PDC) the “Alt + numeric” keys are configured to off-hook the lines correctly.
- 79968** Users can now scroll up/down in the phone voice recording menu while deleting recorded files.
- 79680** The phone displays a pop-up message when low bandwidth is detected while hosting a centralized conference.
- 79625** When the phone tries to park a call using blind transfer and fails, the call is retained between the caller and called phone.
- 78256** In a Lync environment, the contacts screen now displays five soft keys.
- 75808** When the phone is configured with one or more BLF lines, the first incoming call to the BLF lines during the prompt stage displays in the main display and the second stacks on top.
- 74607** The Select soft key label is now correct under the unavailable reason codes in the ACD menu.
- 73618** Changing the phone’s web UI language reflects on all the contents except ringtones.
- 72197** After you select the “check for updates” button, the phone web UI now populates all the available software versions from the Polycom hosted server in the combo box.
- 72055** Enhanced the pop-up message to “Failed to fetch available software from the Polycom Hosted server. Please try again later or contact your Network Administrator if the problem persists” when trying to connect to the Polycom hosted server from the phone’s web UI.
- 64091** Resetting the local configuration on the phone no longer reboots the phone.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 4.1.2 configuration file parameters.



Web Info: Understanding Polycom Configuration Parameters

You can find detailed descriptions of these parameters and their values in the [Polycom UC Software 4.1.0 Administrators' Guide](#) on the support web site.

Table 8: UC Software 4.1.2 Parameter Enhancements

<i>File</i>	<i>Modification</i>	<i>Parameter</i>	<i>Modification Description</i>
Feature	Added	<code>reg.X.auth.usePinCredentials</code> = 0 or 1 X= line, for example, 1, 2, 3, etc.	0 or 1 to enable or disable the Pin authentication. This is disabled by default and enabled when the base profile is set to Lync.
Feature	Added	<code>reg.X.auth.useLoginCredentials= 0 or 1</code> X= line, for example, 1, 2, 3, etc.	To enable or disable the login credentials. The treatment of this parameter has not changed by the Pin Authentication Feature.
Feature	Added	<code>reg.X.auth.loginCredentialType="usernameAndPassword"</code> X= line, for example, 1, 2, 3, etc.	To enable the type of login credentials. Below are the available options: 1. loginCredentialNone 2. usernameAndPassword 3. extensionAndPIN The default value is loginCredentialNone.

Understanding Updates to UC Software 4.1.0 Rev B

Polycom UC Software 4.1.0 Rev B is for use only with Microsoft® Lync® Server using a single registered line.

UC Software 4.1.0 Rev B supports the following Polycom devices:

- SoundPoint IP 321, 331, 335, 450, 550, 560, and 650 (The SoundPoint IP 670 is not supported.)
- SoundStation IP 5000 conference phones
- SoundStation Duo conference phones

- VVX 500 business media phone



Note: UC Software 4.1.0 Rev A with SpectraLink 84xx Series Wireless Handsets

If you are using SpectraLink 84xx series wireless handsets, you must use UC Software 4.1.0 Rev A. For limitations and enhancements for UC Software 4.1.0 Rev B, see the section [Understanding UC Software 4.1.0 for the SpectraLink 84xx and VVX 500](#) in this document.

For further details about support for SpectraLink series, see [SpectraLink 8400 Wireless Telephones](#) on the Polycom Support web site.

New or Enhanced Features

78286 Added support for enabling/disabling auto discovery per line.

74716 Add support for dialing by extension

Enhanced Capabilities

80554 When the phone language is set to any non-English language and locked, the language on the phone is preserved even after a reboot.

80519 When a new user is added to an existing IM conversation on a PC client, the added user's phone does not ring after a join request.

80144 On a private line, the phone remains quiet after the ring type is set to silent.

80081 Enable/disable of Polycom Desktop Connector (PDC) from the phone's web user interface now works correctly.

79955 The phone automatically installs the CA certificate and registers to Lync server after reboot when the certificate is deleted from the phone menu.

79916 Line labels are now updating after performing update configuration on the phone.

79797 The message waiting indicator (MWI) no longer blinks after the user signs out of the phone.

79225 The phone logs out the non-default user automatically as designated by the set time in the automatic logout parameter.

79070 Enabling login credentials of the user on the phone's web interface automatically disables the user domain id/credentials on the phone.

77192 Adding a '+' sign to the Line Identification address through the phone's Web Configuration Utility now displays correctly.

Configuration File Enhancements

For information on enhancements to UC Software 4.1.0 Rev B configuration files, see [Configuration File Enhancements](#) for UC Software 4.1.0.

For details on all UC Software 4.1.0 configuration files and parameters, refer to the *Polycom UC Software 4.1.0 Administrators' Guide* on the [Polycom PartnerConnect web site](#).

Understanding UC Software 4.1.0 for the SpectraLink 84xx and VVX 500

Polycom UC Software 4.1.0 (Rev A) was released to support only the SpectraLink 84xx series wireless handsets and the VVX 500 business media phones. UC Software 4.1.0 Rev B supports the VVX 500. If you are using SpectraLink 84xx series wireless handsets, you must use UC Software 4.1.0 Rev A. This section explains enhanced features and capabilities of UC Software 4.1.0 Rev A.

UC Software 4.1.0 is a software upgrade for phones that are qualified to deliver direct interoperability for Lync.

New or Enhanced Features

- 42163** Added support for simplified best-effort SRTP.
- 66597** Added support for Microsoft STUN/TURN/ICE.
- 68649** Added support for Lync Certificate Provisioning using MS Web Ticket.
- 68652** Added support for Microsoft E911.
- 68653** Added support for Lync Call admission control.
- 68802** Added support for Lync In-band provisioning.
- 68803** Added support for Lync server address discovery.
- 68654** Added support for Lync Media bypass.
- 69089** Added support for Lync Private incoming line.
- 69094** Added support to switch over to local ring when early media fails.
- 69096** Added support for Lync dial plans.
- 69106** Added support for Branch Office Resiliency (BOR) feature.
- 70673** Added support for alternative call forwarding identities.
- 74216** Added support for video synchronization with Lync client.
- 74567** Added support for Microsoft Web Ticket Client Protocol.
- 74510** Added ability to route all outbound requests through Outbound Proxy Server with different callee and caller URI domains.
- 74557** Added support for a manual configuration re-sync with the Lync provisioning server.
- 74616** Added support for extension-based dialing.
- 74894** Added ability to retrieve Lync server root certificate automatically using DHCP Option 43.
- 75141** Added Base Profile menu option for easy out-of-the-box experience.
- 77281** Added support for Lync mode on phones (Lync Base Profile).

Enhanced Capabilities

- 79321** Significant improvements made on the battery threshold levels (*applies to SpectraLink 8400*).
- 79063** The phone now updates the presence status information for all watched buddies (*applies to VVX 500*).
- 78107** The phone now displays a warning message upon reaching the maximum number of buddies.
- 78100** On the phone's Presence - Idle timeout settings, changing the *office hours* timeout value does not change the *Off hours* timeout value.
- 78087** When a call is made from a SpectraLink Wi-Fi phone, the receiving phone now displays the complete number of the caller.
- 77089** In a Lync environment, any changes to the buddy presence state are now immediately reflected on the presence icons and status.
- 76995** In a Lync environment, phone call list details are no longer uploaded to the provisioning server when setting the configuration parameter `feature.callList.enabled="0"` (*applies to VVX 500*).
- 76981** The phone now displays a call forwarding icon and forwards calls correctly when the function is enabled from the phone web user interface.
- 76446** Calls to a PSTN network now work correctly, even when Media Bypass is enabled on the PSTN gateway.
- 76261** The phone password field for Lync configuration now remains empty when no password is set, allowing the user to set a new password.
- 76194** Incoming calls now first display as a pop-up message before being minimized to a call appearance/filter view on the phone.
- 76028** The Polycom Desktop Connector (PDC) has been improved to connect to the user's laptop/PC after installation.
- 75874** Only the internal headset ring tone is played on Jabra PRO 9450 headsets (*applies to VVX 500*).
- 75872** The phone ringer now functions correctly when the user attaches or detaches a USB headset during an incoming call.
- 75743** Corrected the usage of a configured outbound proxy server address for Voice quality monitoring feature on phones.
- 75715** Local directory can now save contacts searched from the phone's corporate directory, even when the local directory is disabled.
- 75694** Adding/removing a USB headset no longer affects phones configured with EHS headsets (Jabra, Plantronics, or Sennheiser).
- 75674** URL dialing is now possible between unregistered phones.
- 75643** "Call list display" now functions correctly when the configuration parameter `feature.callList.enabled="0"`.
- 75605** Contacts containing long information fields, such as first name, last name, now display correctly in the detail view on the phone.

- 75431** PTT and Paging feature has now been enhanced to user iLBC codec.
- 75355** Improved the synchronization of contacts and addition/deletion between the Lync MOC client on PC and Phone IM.
- 75245** Buddy presence status is now updated when the phone is disconnected from a network.
- 74901** When the lock feature is enabled after a phone reboot, the emergency/authorized call list displays when the user tries to place a call using the headset/speaker key.
- 74888** The host status display now functions correctly in a multiparty Lync conference call.
- 74175** When the phone presence state is set to Be Right Back, it no longer changes to Offline when the phone is left idle for a long period of time.
- 73797** Caller details now display correctly for participants in a Lync consultative transfer call.
- 72518** On the Lync soft client, only mobile platforms are now categorized as mobile.
- 70723** The phone now fetches the correct available software from the Polycom provisioning server when the parameter `upgrade.plcm.server.url` is set correctly.
- 57864** Changes to SRTP parameters now take effect immediately, and without rebooting the phone.

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 4.1.0 configuration file parameters.



Web Info: Find Detailed Descriptions of Parameters and Values

You can find detailed descriptions of these parameters and their values in the *Polycom UC Software 4.1.0 Administrators' Guide* on the [Polycom PartnerConnect web site](#).

Table 9: UC Software 4.1.0 Parameter Enhancements

<i>File</i>	<i>Modification</i>	<i>Parameter</i>
feature	Added	<code>apps.telNotification.appInitializationEvent</code>
feature	Added	<code>apps.telNotification.networkUpEvent</code>
feature	Added	<code>apps.telNotification.uiInitializationEvent</code>
feature	Added	<code>apps.telNotification.taInitializationEvent</code>
feature	Added	<code>apps.telNotification.uiInitializationEvent</code>
feature	Added	<code>device.baseProfile device.baseProfile.set</code>
feature	Added	<code>dialplan.applyToForward</code>
feature	Added	<code>dialplan.x.applyToForward</code>

<i>File</i>	<i>Modification</i>	<i>Parameter</i>
feature	Added	locInfo.x.A1
feature	Added	locInfo.x.A3
feature	Added	locInfo.x.country
feature	Added	locInfo.x.HNO
feature	Added	locInfo.x.HNS
feature	Added	locInfo.x.label
feature	Added	locInfo.x.LOC
feature	Added	locInfo.x.NAM
feature	Added	locInfo.x.PC
feature	Added	locInfo.x.POD
feature	Added	locInfo.x.PRD
feature	Added	locInfo.x.RD
feature	Added	locInfo.x.STS
feature	Added	log.level.change.afe
feature	Added	log.level.change.ice
feature	Added	log.level.change.loc
feature	Added	log.level.change.tickt
feature	Added	log.level.change.xml
feature	Added	np.custom1.ringing.privateLine.tonePattern
feature	Added	np.custom1.ringing.privateLine.vibration
feature	Added	np.meeting.ringing.privateLine.tonePattern
feature	Added	np.meeting.ringing.privateLine.vibration
feature	Added	np.normal.ringing.privateLine.tonePattern
feature	Added	np.normal.ringing.privateLine.vibration
feature	Added	np.silent.ringing.privateLine.tonePattern
feature	Added	np.silent.ringing.privateLine.vibration
feature	Added	prov.login.lcCache.domain
feature	Added	prov.login.lcCache.user

<i>File</i>	<i>Modification</i>	<i>Parameter</i>
feature	Added	reg.x.dialPlanName
feature	Added	reg.x.lisdisclaimer
feature	Added	reg.x.lync.autoProvisionCertLocation
feature	Added	reg.x.ringType.privateLine
feature	Added	reg.x.serverAutoDiscovery
feature	Added	reg.x.srtp.simplifiedBestEffort
feature	Added	sec.srtp.simplifiedBestEffort
feature	Added	softkey.feature.simplifiedSignIn
feature	Added	tcpIpApp.ice.mode
feature	Added	tcpIpApp.ice.password
feature	Added	tcpIpApp.ice.realm
feature	Added	tcpIpApp.ice.username
feature	Added	tcpIpApp.ice.stun.passwordServer
feature	Added	tcpIpApp.ice.stun.server
feature	Added	tcpIpApp.ice.stun.udpPort
feature	Added	tcpIpApp.ice.tcp.enabled
feature	Added	tcpIpApp.ice.turn.server
feature	Added	tcpIpApp.ice.turn.tcpPort
feature	Added	tcpIpApp.ice.turn.udpPort
feature	Added	voice.page.handsfree.rxag
feature	Added	voice.ptt.handsfree.rxag

Understanding Updates to UC Software 4.0.2 Rev B

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 72403** Added support for DHCP renews after the loss and recovery of a Wi-Fi LAN connection (*applies to SpectraLink 8400*).
- 76730** Enhanced the digitmap by removing the prepending '+' to the outbound calls and giving the option of configuring the '+' in the dial plan (*applies to Lync mode only*).
- 77038** Added support for early media followed by local ringback.

Enhanced Capabilities

- 73667** The syslog counter on SpectraLink 8400 phones are updated correctly after audio packets are received.
- 75557** On the phone's web interface, the logging module parameter Wi-Fi Manager log value is updated in the field help section when the mouse is dragged over the parameter name and its value (*applies to SpectraLink 8400*).
- 76057** When a phone is registered with a single shared line, the Join soft key no longer displays incorrectly when the monitored phone puts the call on hold.
- 76084** In a shared BLF line scenario, the monitoring phone no longer resumes the call from the monitored phone without playing a busy tone.
- 76095** In a shared BLF line scenario, phone now displays the incoming call status of the monitored phone (*applies to VVX 500*).
- 76131** The phone default call appearance eliminates the display of remote on hold calls (*applies to VVX 500*).
- 76321/76515** The phone does not reboot when multiple URIs are pushed to the phone in frequent intervals. (*Customer issue ID VESC-1635, VESC-1650*).
- 76427** The phone displays only the incoming call received from a line monitored using BLF (used to display as two calls in the list) (*applies to VVX 500*).
- 76467** Improved Call List display for caller ID now includes long names (*applies to VVX 1500*).
- 76591** The phones display the correct Asian language fonts for Lync contacts.
- 76679** Unauthorized requests for configuration files through the phone web interface are now restricted.
- 76754** The filtered call view of a BLF-monitored line is shown correctly when the phone is in an off-hook state.
- 76862** During an active BLF call session, frequent pressing of the BLF key on the origination phone no longer causes the caller ID information to be blanked out.
- 76889** The PPT key can now be configured as a Speed dial key on the SpectraLink 8400 phones.
- 76911** The dialer screen UI on the phone refreshes when an incoming ringing call terminates.
- 76934** On-hook dialing works correctly when the phone has an incoming call and the remote party ends the call.
- 76939** The phone highlights the last incoming call as per the order in call appearance screen (*applies to VVX 500*).

- 76940** Incorrect soft key options no longer display on the BLF-monitoring phone when there is an incoming call in certain scenarios.
- 76944** Auto Answer now works correctly when alert information header carrying the string within the angle brackets '< >' is received.
- 76946** The phone current draw is optimized as per its state (*applies to SpectraLink 8400*).
- 76947** The audio/ sound effect termination is always on the dock station when there is an active call on the phone (*applies to SpectraLink 8400*).
- 76949** The browser application on the phone times out correctly (*applies to SpectraLink 8400*).
- 76976** The phone no longer goes off-hook on line1 instead of line 2 when the user presses the second line key while lifting the handset with call hold on line1.
- 77099** Manual configuration of the IP address no longer causes a “network is down” message to display (*applies to SoundStation IP 6000, 7000*).
- 77135** Corrected some Directed call pickup failures in certain situations.
- 77173** Turning the backlight OFF on the phone no longer sends the browser to the session list (*applies to SpectraLink 8400*).
- 77226** When the phone is registered to a Lync line and another call server, the Lync contact presence subscription is now correctly sent through the Lync registered line.
- 77363** Corrected the issue that could cause a reboot when the user has configured many buddy contacts (*applies to VVX 500*). (*Customer issue ID VESC-1763*)
- 77396** Corrected a phone reboot issue in a certain ACD Call Center configuration (*Found in UC Software 3.3.2*).
- 77549** Corrected an issue relating to the use of the Join key which displayed even when only one call was in progress (*applies to SoundPoint IP 33x, SoundStation Duo and SpectraLink 8400 models*).
- 77626** Two-way audio between the phones now works correctly after resuming the call from MOH when `voIpProt.SIP.musicOnHold.uri` or `reg.X.musicOnHold.uri` is used for the address of the MoH server.
- 77749** The registration failing issue with the Microsoft Lync server front end, due to an error in Subject Alternative Name (SAN) validation implementation, is corrected.
- 77830** Corrected an issue when the phone is configured with certain languages options and it attempts to select LLDP or CDP using Ethernet and causes a reboot (*applies to SoundStation IP 6000*).

Configuration File Enhancements

Refer to the following table for a list of all enhancements made to the UC Software 4.0.2 Rev B configuration file parameters.

Table 10: UC Software 4.0.2 Rev B Parameter Enhancements

File	Modification	Parameter	Modification Description
Wireless	Added	device.dhcp.releaseOnLinkRecovery	1 Phone performs a DHCP release on network link recovery (Default) 0 Phone does NOT perform a DHCP release on network link

Note: Also refer to configuration parameter changes in UC Software 4.0.2.



Note: Ensuring Digitmap Compatibility between UC Software Versions 4.0.1 and 4.0.2 Rev B

The existing digitmap `dialplan.digitmap = [2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxx|[2-9]xxxxxxxx|[2-9]xxxT` in the UC Software 4.0.1 release where the phones were automatically pre-pending a + to outbound calls is now removed in UC Software 4.0.2 Rev B.

For UC Software 4.0.2 Rev B to be backwards compatible to UC Software 4.0.1, the digitmap should be `dialplan.digitmap=RR+R[2-9]11|0T|RR+R011xxx.T|RR+R[0-1][2-9]xxxxxxxx|RR+R[2-9]xxxxxxxx|RR+R[2-9]xxxT`, or if the digitmap is to apply for a certain line, use `dialplan.1.digitmap=RR+R[2-9]11|0T|RR+R011xxx.T|RR+R[0-1][2-9]xxxxxxxx|RR+R[2-9]xxxxxxxx|RR+R[2-9]xxxT`.

Understanding Updates to UC Software 4.0.2 (Limited Release)

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

Note: UC Software 4.0.2 is a limited release that was distributed only to select partners and customers and is shipping with SoundStation Duo. The build-ID for this release was UC Software 4.0.2.8017.

New or Enhanced Features

40451/75448 Added support for XT9 PinYin input for Chinese characters (*applies to VVX 1500*).

52485/66494 Added support for BroadSoft Hoteling Event Package.

57167/66494/76023 Added support for BroadSoft Call Center Status Event Package.

54576 Added support for the new SpectraLink 8452 Wi-Fi handset with 2D barcode reader.

Enhanced Capabilities

69469 The display name with special character < or > causes phones to respond with a bad request.

73614 An unintentional touch on the phone initiates a call automatically (*applies to VVX 500*).

- 73946** On the Trapeze/Juniper infrastructure, when multiple SpectraLink phones are involved in a call, one or more phones might lose wireless connectivity.
- 74292** The Bluetooth radio can now be activated on SpectraLink 8400 phones.
- 74427** On a redirected call, the phone now sends a PRACK (acknowledgement) message.
- 75299** The dialpad key presses are now captured when hot dialing from the idle browser screen (*applies to VVX 500*).
- 75419** The ADHOC conference call now works when there is a + sign (for example, SIP:voip+world@voipworld.com) in the SIP URI contact header.
- 75632** Slowness in dialing many phone contacts is resolved (*applies to VVX 500*).
- 75686** The up and down arrow keys are now available on the phone during call transfer (*applies to SoundPoint IP 450*).
- 75726/75716** The phone numbers dialed using auto complete remembers the line information on which the calls were placed earlier (*applies to SpectraLink 8400, VVX 500 and VVX 1500*).
- 75811** Reassigning the line keys preserves the presence information.
- 75888** A scrolling status message now displays when a line is unregistered on the phone.
- 75945** When the phone is off hook, auto dialing now remembers the line information (*applies to SpectraLink 8400, VVX 500 and VVX 1500*).
- 75954** The overlap of idle browser on the call list screen is now set (*applies to VVX 1500*).
- 75949/76258** The SoundStation Duo phone complies with clause 5.5.1.4.1 of Australian spec S002 of the Australian analog telephony specification as per clause 5.5.1.4.1 of Australian spec S002.
- 76141** The call lists on the VVX 1500 phone get updated as required.
- 76171** The phone no longer continuously reboots on reassignment of the line keys.
- 76229** A breakdown observed on the phone monitoring a BLF contact is now corrected. (*Customer issue ID VESC-1670*)
- 76315** The pop-up message “Error Line: Unregistered” no longer appears if there is no register request.
- 76316** The default hookflash timing is set on the SoundStation Duo when the country is set to Australia.
- 76329** Soft keys on the VVX 500 are now responsive to touch.
- 76376** An unregistered pop-up message no longer appears on the SoundStation Duo in PSTN mode.
- 76379** Double quotes appended to the calling party display name on a shared line are now removed.
- 76408** Shared lines continue to ring when another phone with the same shared line answers an incoming call on another line appearance.
- 76420** After reboot, the phone correctly displays incoming calls of a monitored BLF contact even on the first call.
- 76422** All the incoming call appearances on a BLF monitored phone display when the monitoring phone cancels the call to the BLF contact.
- 76423** In a BLF monitored scenario, multiple calls to a monitored phone display the incoming call appearance and call counter appropriately.

- 76428** The phone correctly displays incoming call appearance after terminating a call on the phone using the End Call soft key.
- 76535** The phone now correctly displays a warning icon and a pop-up message for unregistered H.323 lines. (*applies to VVX 1500*)
- 76600** The blank call appearance on the monitoring BLF enabled phone is set.
- 76601/76685** In a multi-party BLF enabled call, the widget displaying the call appearance, counter, icons, and the indicator is updated with the appropriate incoming and outgoing call status.
- 76690** The phone now functions normally with the call appearance of the monitored BLF contacts.
- 76694** The phone no longer crashes when a monitored line ends the call that is associated with the remote call appearance screen.
- 76838** On a PSTN line the invocation of redial is restricted by pressing the # key after entering the digits (*applies to SoundStation Duo*).
- 76849** On a PSTN line pressing the # key is now restricted to send the dialed number only once (*applies to SoundStation Duo*).
- 76875** The phone now picks up the call forwarding settings from the override file after a reboot.

Configuration File Enhancements

Refer to the following table for a list of new parameters. Note that these configuration parameters are detailed in *Feature Profile 76179: Hoteling and ACD*, which is available on [Polycom Profiled UC Software Features](#).

Table 11: UC Software 4.0.2 Parameter Enhancements

<i>File</i>	<i>Modification</i>	<i>Parameter</i>	<i>Modification Description</i>
Feature	Added	<code>feature.callCenterStatus.enabled</code>	Call feature parameter
Feature	Added	<code>feature.hoteling.enabled</code>	Call feature parameter
Feature	Added	<code>hoteling.reg</code>	Call feature parameters
wireless	Added	<code>barcode.X.Y</code>	Parameters used to configure the 2D barcode scanner*

*For a detailed description of new parameters specific to the SpectraLink 8400 product family, their properties and values, refer to the Polycom SpectraLink 8400 Series Wireless Telephone Deployment Guide.

Understanding Updates to UC Software 4.0.1B

There are no functional differences between Polycom UC Software 4.0.1B and Polycom UC Software 4.0.1. Polycom UC Software 4.0.1B was released to include the VVX 500 and SoundStructure sip.id files in the software release package.

Understanding Updates to UC Software 4.0.1

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 48734** In a server-based, centralized conference, the phone can now send parallel REFERs without waiting for a 202 Accepted.
- 67081/70634** Added support for phones to interoperate with a limited set of Microsoft® Lync™ server features (*applies to SoundPoint IP 321, 331, 335, 450, 550, 560, 650, 670, VVX 500, 1500, SoundStation IP 5000, SoundStation Duo, and SpectraLink 84xx*).
- 67090** Syslog now includes the ability to identify multiple audio streams (*applies to SpectraLink 84xx*).
- 67594** Added interoperability between the Message Waiting Indicator (MWI) and Microsoft Lync.
- 68500** The SpectraLink 84xx handsets now display the X-Loader version information in the Phone menu (Menu > Settings > Status > Platform > Phone).
- 68602** Added support for SS RTP.
- 68798** Added support for Microsoft SRTP extensions.
- 69962/70924** Added Microsoft OCS/Lync Presence functionality to phones (*applies to SoundPoint IP 321, 331, 335, 450, 550, 560, 650, 670, VVX 500, 1500, SoundStation IP 5000, SoundStation Duo, and SpectraLink 84xx*).
- 70122** The phones now display the Away presence status after a period of user inactivity specified by the following parameters: `pres.idleTimeout.offHours.period`, `pres.idleTimeout.officeHours.period`, `pres.idleTimeout.offHours.enabled`, and `pres.idleTimeout.officeHours.enabled`.
- 70232** Added a parameter `call.transfer.blindPreferred` to control whether the Transfer soft key on the SpectraLink 84xx should be a consultative transfer or blind transfer.
- 70614** Added support for Microsoft 2008 Radius (802.1X).
- 71025** Added new per-registration configuration options for several SRTP parameters: `reg.x.srtp.enable`, `reg.x.srtp.offer`, `reg.x.srtp.require`.
- 71183** Added missing barcode symbologies (*applies to SpectraLink 84xx*).
- 71198** Added an option in the Web Configuration Utility for SIP and Provisioning TLS applications to make the Common Name of Subject test configurable.
- 71424** Updated the presence icon on the phones to be consistent with the Microsoft Lync/OCS style (*applies to SpectraLink 84xx*).
- 71439** Not including the parameter `oai.userID` in the configuration file or setting the value to NULL both result in the phone using its MAC address to check in into the OAI server (*applies to SpectraLink 84xx*).
- 71660** Enhanced the Reset to Default option in the Updater to match the option in the application software.

-
- 71774** The call forward status on the status bar now displays when Forward – No Answer or Forward – Busy is enabled (*applies to SpectraLink 84xx*).
- 71997** Added full support for RFC2782 (DNS load balancing).
- 72074** In the Web Configuration Utility, the Country Code field has been renamed to Regulatory Domain (*applies to SpectraLink 84xx*).
- 72193** In the phone menu, the Country Code field has been renamed to Regulatory Domain (*applies to SpectraLink 84xx*).
- 72279** Enabled an EFK to allow a user to invoke the Call Back feature while on hook (*applies to VVX 1500*).
- 72304** The default value for the configuration parameter `up.useDirectoryNames` is now 1 (enabled).
- 72310/74129** The SpectraLink series handsets can now display the BootL1 version information in Phone menu (Menu > Status > Platform > Phone).
- 72319** The phone displays a warning icon when the WLAN Network Manager detects an invalid Regulatory Domain request (*applies to SpectraLink 84xx*).
- 72320** The phone displays a warning triangle when the WLAN Network Manager detects an invalid Regulatory Domain limit setting (*applies to SpectraLink 84xx*).
- 72367** The phone automatically publishes an Inactive (Idle) presence status after 5 minutes of user inactivity.
- 72554** Added the ability to configure the `pres.idleTimeout` parameters through the phone menus.
- 72555** Added the ability to configure the `pres.idleTimeout` parameters through the Web Configuration Utility.
- 72654** The Exchange Calendaring feature on the SpectraLink handsets has been improved with the following enhancements:
- The Calendar icon is shown in the main menu once the calendar is authorized.
 - The phone displays a “Calendar: synchronizing” scrolling message in the status bar.
- 72791** The microbrowser on the SoundPoint phones has been functionally improved with the following enhancements:
- The audio tag element injects a Play soft key when in focus.
 - The user can now specify additional attributes to the audio tag which is interpreted as a soft key, thus allowing the user to use elements such as the Details soft key.
 - The audio tag has a descriptive label which is used as the button label for the audio element in the page. This enables each audio tag to render as a single element in the page with an icon and a descriptive text. The user no longer needs to switch to the text to see the details.
 - The descriptive label can also be used for the title of the playback screen.
- 72823** In the media player, the Exit soft key has been renamed to Back.
- 72824** Playback automatically starts when selecting an audio element from the browser.
- 73420** When the phone language is set to Japanese, the phone now uses the English AM/PM string for the time/date display (*applies to all SoundPoint IP, all SoundStation IP, and SoundStation Duo*).

- 73500** The 'Connect/disconnect from the server' option has been moved to the Calendar menu (Features > Calendar) in the SpectraLink 84xx handsets.
- 73510** The Web Configuration Utility language now supports multiple default language labels and help text in English, with the option to add/access other languages.
- 73669** Updated the 2048-bit Trusted CA Root Certificate List from VeriSign.
- 73670** Added new VeriSign Intermediate CA certificates.
- 73671** Added RSA 2048 V3 Root Certificate to Root Store to all phones.
- 73805** The phone can now display up to 4 Chinese characters in the soft keys (*applies to SoundPoint IP 450*).
- 73907/74289** Added the ability to automatically upgrade the BootL1 and BootBlock (*applies to SpectraLink 84xx*).
- 74247** In the Web Configuration Utility, the default utility languages that are available depend upon the platform.
- 74417** The Updater (BootROM) now supports Basic Authentication with HTTP/HTTPS.
- 75308** Added the ability to upload encrypted call lists to the provisioning server (*applies to SpectraLink 84xx*).
- 75469** The volume of PTT audio has been increased and setting the parameter `voice.handsfree.rxag.SL8440=10`, then updating the phone using Update Configuration does not cause the phone to restart (*applies to SpectraLink 84xx*).

Enhanced Capabilities

- 55237** Dialing a semicolon using on-hook dialing no longer displays the off-hook dialing dialog (*applies to VVX 1500*).
- 61038** The phone no longer becomes unresponsive to hard key and touch screen presses for several seconds (*applies to VVX 1500 phones provisioned with CMA using UC Software 3.3.1*).
- 66864** Using a dial plan containing #, when a user dials #1#2#, the phone now sends out an invite message containing %231%232%23 (*applies to SoundStation Duo*).
- 68356** The phones can now fragment packets when instructed to by an ICMP message (*applies to SoundStation IP 6000 and 7000*).
- 68501** When using Exchange Calendaring, the passcode now enters automatically (*applies to SpectraLink 84xx*).
- 68835** The phone can now correctly sniff EAP type frames (*applies to VVX 1500*).
- 69020** SoundPoint IP phones capable of downloadable fonts now correctly display certain Czech characters.
- 69540** A call dropped by the other party no longer displays as a held call.
- 69558** An Avaya 10x0 and a VVX 1500 can successfully establish a video call.
- 69882** In PSTN only mode, a received call is now recorded in the *Received Calls* call list when the call is finished (*applies to SoundStation Duo*).

-
- 69883** In PSTN only mode, an incoming call is now recorded in the *Missed Calls* call list when the call is not answered (*applies to SoundStation Duo*).
 - 70228** The phone no longer reboots when attempting a conference through an SCA line (*applies to SoundPoint 321, 331*).
 - 70542** The registered line icon and BLF icon are no longer corrupted in the SpectraLink 84xx handsets.
 - 70944** A dialed number no longer overwrites a registered line's label if the label is very long (*all SoundStation phones*).
 - 71041** The phones can now play audio from the Lync voicemail system (*applies to SpectraLink 84xx*).
 - 71348** Calls between a VVX and RMX no longer cause the VVX to crash and reboot.
 - 71368** Remote shared line activity no longer affects local phone presence.
 - 71433** The phones crash after loading `se.pat.callProg.dialTone` parameters and pressing the New Call button (*applies to SoundPoint IP 450, 650, and SpectraLink 84xx*).
 - 71604** The configuration parameter `sec.TLS.SIP.strictCertCommonNameValidation` can be updated without requiring a phone reboot.
 - 71616** The conference feature can now correctly handle a 480 response to a BroadSoft SCA line seize SUBSCRIBE.
 - 71763** The phone no longer sends a re-INVITE to the conference server after sending a REFER for each leg of the conference (*applies to SoundPoint 321, 331*).
 - 71764** Sennheiser and Jabra headsets can now go off-hook after switching to a different headset type (*applies to SoundPoint IP 335, 450, 550, 650*).
 - 71802** Warning icons and records are now removed after the error condition is removed (*applies to all SoundPoint IP*).
 - 71829** An EFK has been created to display the corporate directory Advanced Find menu from the idle screen (*applies to VVX 1500*).
 - 71910** The text 'Enter password' in the Advanced menu is now translated when switching phone languages.
 - 72113** The Polycom Quick Barcode Connector icon now appears and disappears for both multiple and single endpoint modes (*applies to SpectraLink 84xx*).
 - 72250** The phone no longer reboots when queued messages are accessed on the phone (*applies to SpectraLink 84xx*).
 - 72368** URL Dialing from the call list is now fully disabled when `feature.urlDialing.enabled=0` (*applies to SpectraLink 84xx*).
 - 72442** The SoundPoint IP 321 and 331 phones now display the correct call x/y widget when `filterReflectedBlaDialogs=0`.
 - 72298** The Join soft key is no longer missing after establishing the maximum number of calls on all the lines using when using a Sylantro call server (*applies to SoundPoint IP 650 and VVX 1500*).
 - 72469** When `feature.urlDialing.enabled` is set to 0, the phone accepts contact entries with a contact number longer than 10 digits (*applies to SpectraLink 84xx*).

- 72739** The phones can now play the audio files using the microbrowser (*applies to all SoundStation phones*).
- 72800** Lines registered to a Microsoft Lync 2010 server now display in the Ring Type menu (Menu > Settings > Basic > Ring Type).
- 72820** The VVX 1500 phone can be registered with up to 29 lines, but a maximum of 24 can be displayed.
- 72822** The phone no longer requires an auto-reboot after receiving two consecutive 401s to a line-seize SUBSCRIBE during conference initiation.
- 72837** The speed dial icon no longer disappears after a Reset Local Config or Reset Web Config option is selected (*applies to SoundPoint IP 321, 331, 331C, 335, 335C*).
- 72895** The phone override file is no longer created on the provisioning server when values are not changed through the phone (*applies to VVX 1500*).
- 72907** Improved the phone's conference call management when a cell phone is connected to the 2.5mm port (*applies to SoundStation Duo and SoundStation IP 7000*).
- 72961** Bellcore Caller ID detection in PSTN mode works reliably now (*applies to SoundStation Duo*).
- 72996** A conference call between three parties now successfully connect all parties after there is a no response to line seize SUBSCRIBE.
- 73027** A Plantronics Savi740 EHS headset no longer has intermittent pairing issues with a VVX 1500.
- 73054** Call appearances now display correctly when pressing the New Call soft key during an incoming call (*applies to SoundStation Duo*).
- 73084** Key in Keypad Diagnostics menu is now translated on SoundPoint IP 321, 331, and 335 phones for all languages.
- 73145** Regulatory Domain Error when radio set to 802.11a and band1 is set to P6 (*applies to SpectraLink 84xx*).
- 73153** Hot dial window now disappears after auto answering a call (*applies to SoundPoint IP 450, 650*).
- 73183** The phone now displays the error message 'Network link is down' when the DHCP server is down (*applies to SoundPoint IP 335, 450, 560, 670*).
- 73195** The bootloader menu for WEP has the correct spelling of Encryption (*applies to SpectraLink 84xx*).
- 73227** The phone UI now displays the proper network parameters such as IP address, subnet mask, and IP gateway when DHCP is enabled (*applies to SoundStation Duo*).
- 73238** Changing the options in the directory search, then logging into the phone no longer causes the phone to restart (*applies to SoundPoint IP 321, 331, 331C, 335, 335C*).
- 73247/74912** In the Quick Setup menu, user entry fields are now set to numeric as default.
- 73259** Simultaneous incoming and monitored BLF calls now both display on the unified call appearance list (*applies to SoundPoint IP 450, 550, 560, 650, 670*).
- 73263** Ending a PTT or page no longer causes the active call appearance to disappear for 10 seconds (*applies to SoundPoint IP 450, 550, 560, 650, 670*).

- 73264** A phantom call appearance no longer displays when there is an active BLF monitored call and the phone has another call appearance.
- 73290** The User Profiles Log In soft key is no longer missing when the feature presence is enabled (*applies to SoundPoint IP 670*).
- 73401/74688** With intercom configured, the handset now rings once and incoming calls are answered automatically (*applies to SpectraLink 84xx*).
- 73405** The Missed Calls list no longer displays on a peer shared line if the call was barged in from a remote shared line (*applies to SoundPoint IP 321, 331, and 335*).
- 73409** In the Web Configuration Utility, all instances of the text 'extension module' have been replaced with 'expansion module'.
- 73419** When the IP 670 is configured for 34 lines using three expansion modules, a large contact directory, LDAP, idle browser, and a microbrowser, it no longer crashes due to a lack of memory.
- 73459** The User Profiles feature is now fully supported on the SoundStation Duo.
- 73527** If barge-in is enabled on a shared line, remote active calls do not appear.
- 73561** When using the Web Configuration Utility, the phone now updates the line key icons correctly when static BLF is configured (*applies to SoundPoint IP 450, 550, and 650*).
- 73697** The flash timing is now correct for France and Singapore (*applies to SoundStation Duo*).
- 73749** Plugging in (or in and out) a 2.5mm mobile phone or PC cable no longer causes the phone to reboot (*applies to SoundStation Duo*).
- 73910** The phone's LCD contrast no longer turns darker after upgrading to UC Software 4.0.0 (*applies to SoundPoint IP 450*).
- 74018** The phone no longer locks up and reboots when there are a large number of incoming calls (*applies to SoundPoint IP 650*).
- 74045** The phone can now boot up to the idle screen correctly after being issued a check sync to enable the paging feature soft key.
- 74050** The phone no longer segfaults on boot due to language dictionary files (*applies to all SoundPoint IP*).
- 74060** OAI PT Select Connections are now accepted by phone before an OAI call is answered by the Start key (*applies to SpectraLink 84xx*).
- 74167** In the Web Configuration Utility, the authentication password can no longer be seen in clear text when opening the line page source code.
- 74259** In PSTN mode, the phone now displays the date and time information on the idle screen after a reboot (*applies to SoundStation Duo*).
- 74431** The phone no longer generates a beep sound when a monitored user goes off-hook (*applies to SoundPoint 650*).
- 74956** The phone no longer goes into an INVITE loop and reboot if it receives a 503 response to its initial INVITE message (*applies to all SoundPoint IP*).
- 74991** A call between a VVX 1500 and CX series phone now resumes after being held for longer than 30 seconds.

- 75151** Audio files are now directly downloaded to the ramdisk.
- 75194** The phones now use an outbound proxy when an outgoing call's URI domain is different from the caller's domain.
- 75334** Fail over on a 503 response can now be disabled.
- 75458** The maximum values for the DNS TTL parameters in the static cache have changed to 2147483647.
- 75485** The default input type for the Unavailable Code field is now numeric.
- 75491** Ampersand characters are now escaped correctly and are no longer stripped from a URL (*applies to VVX 500, 1500*).
- 75743** The Voice Quality monitoring feature now uses an outbound proxy server address for a SIP Publish.
- 75600** The call appearance for an outgoing call no longer displays the *transport* string.
- 75618** Users can now place a call from the Placed Calls call list when the original call was placed using a Click-To-Dial Refer message with Refer-To: header as sip:*number* ext. *number* @*IPaddress*; transport=TCP (*applies to SpectraLink 84xx*).

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the UC Software 4.0.1 configuration file parameters.



Web Info: Parameters Changed in UC Software 4.0.1

The following table includes parameters modified in UC Software 4.0.1. You can find detailed descriptions of the parameters and their values in the [Polycom UC Software 4.0.1 Administrators' Guide](#).

Table 12: UC Software 4.0.1 Parameter Enhancements

<i>File</i>	<i>Modification</i>	<i>Parameter</i>	<i>Modification Description</i>
sip-interop	Added	call.transfer.blindPreferred	Call feature parameter
Debug	Added	feature.lyncDebug	Call feature parameter
Site	Added	reg.x.srtp.enable reg.x.srtp.offer reg.x.srtp.require	Call feature parameters
wireless	Added	np.custom1.ringing.toneVolume.usbHeadset	Notification profiles parameter
wireless	Added	np.meeting.ringing.toneVolume.usbHeadset	Notification profiles parameter

<i>File</i>	<i>Modification</i>	<i>Parameter</i>	<i>Modification Description</i>
wireless	Added	np.normal.ringing.toneVolume. usbHeadset	Notification profiles parameter
wireless	Added	np.silent.ringing.toneVolume. usbHeadset	Notification profiles parameter
Site	Added	sec.encryption.upload. callLists	Security parameter
sip-interop	Added	sec.srtp.mki.length	Security parameter
sip-interop	Added	sec.srtp.padRtpToFourByte Alignment	Security parameter
reg-advanced, site	Added	up.headset.phoneVolumeControl	User preferences parameter
Debug	Added	up.headset.AlwaysUseIntrinsic Ringer	User preferences parameter
reg-advanced, site	Added	up.idleStateView	User preferences parameter
Video	Added	video.iFrame.delay	Video parameter
Debug	Added	video.iFrame.period	Video parameter
techsupport	Added	voice.usb.headset.rxdg	Audio parameter
techsupport	Added	voice.usb.headset.tx dg	Audio parameter
Site	Added	voice.volume.persist.usb Headset	Audio parameter
sip-interop	Added	voIpProt.SIP.conference. parallelRefer	Call feature parameter
Site	Added	webutility.language.plcm ServerUrl	Web Configuration Utility parameter
techsupport	Removed	voice.gain.rx.digital.headset.IP _330	Audio parameter
techsupport	Removed	voice.gain.rx.digital.headset.IP _335	Audio parameter
Site	Changed	dns.cache.NAPTR.x.ttl dns.cache.SRV.x.ttl dns.cache.A.x.ttl	The maximum value increased from 65535 to 2147483647
reg-advanced, site	Changed	up.useDirectoryNames	The default value changed from 0 (disabled) to 1 (enabled).

<i>File</i>	<i>Modification</i>	<i>Parameter</i>	<i>Modification Description</i>
Pstn	Changed	up.operMode	The default value changed from 0 to auto.
techsupport	Changed	voice.headset.rxag.adjust. IP_335	The default value changed from -11 to 4 and the maximum value changed from -11 to 90.
techsupport	Changed	voice.headset.rxag.adjust. IP_330	The default value changed from -5 to 4 and the maximum value changed from -5 back to 90.
sip-interop	Changed	VoIpProt.SIP.failoverOn503 Response	The default value changed from 1 (enabled) to 0 (disabled).

Understanding Updates to UC Software 4.0.0

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 26549** Enhanced the local missed call feature for shared line appearances. This feature supports RFC 3326 Reason Header.
- 28514** Enhanced the method of selecting a ring type on the menu screen.
- 29056** Enhanced the method of notifying the user of unregistered lines.
- 30251** Added support for non-volatile call lists (*applies to SpectraLink 84xx*).
- 30887** Added support for 802.1X authentication. Authentication methods include MD-5, EAP-PEAP, EAP-FAST, EAP-TLS, and EAP-TTLS.
- 32169** The user is now notified with a confirmation when deleting contact information.
- 33546** Added a host name field to the DHCP registration.
- 35170** Added support for User Profiles. Users may log into and out of the phone using a server-independent, configuration file-based, authentication method. When successfully authenticated, the user's personal configuration files are applied as well as the user's personal local contact directory and call lists.
- 35171** Updated most configuration parameters to be updated without the need of a reboot. Only a select number of configuration parameters require a reboot to be invoked.
- 36166** Added the option for the user to allow ringer volume levels to persist after the phone reboots.
- 38201** The Web-based configuration utility no longer requires the user to submit changes along with a reboot after each page has been modified.
- 41429** Added the ability in the microbrowser to manage allowable characters into the input field.

- 41430** Added the ability in the microbrowser for the user to select an item from a list using the dial-pad.
- 44258** Enhanced the API by enhancing the HTTP Push capability by supporting mutual TLS.
- 44699** Added a Reset to Factory capability.
- 45777** Added user accessible diagnostic functions ping and traceroute.
- 47730** The scrolling status bar has been enhanced. The time between scrolled lines has been increased (*applies to SoundPoint IP 3xx*).
- 47766** The Trusted CA Pool Management capability has been enhanced. The number of supported customer certificates has been increased to six.
- 48714** Added ability for the phone to mute the microphone when auto-answering a call.
- 48750** The Web-based configuration utility now enables the user to configure outbound proxies for each line.
- 48757** Contacts added to the list now highlight and display without the need to scroll up or down to view the addition.
- 50258** Enhanced the method of notifying the user of error and warning indications.
- 51101** Added the ability to use an Emergency Location Identification Number (ELIN) from LLDP-MED to add a P-Asserted-Identity when using emergency routing:
`dialplan.routing.emergency.preferredSource=[ELIN|Config]` (default ELIN)
`dialplan.routing.emergency.outboundIdentity=xxxxxxx` (default null)
`dialplan.routing.emergency.outboundIdentity.lldp=xxxxxxx` (default null).
- 51471** Added a configuration option to disable the test of subject's CommonName against the registration address (associated with CA management).
- 52844** Added certificate validation for 802.1X.
- 53128** Added a configuration option to modify the Backlight timeout duration.
- 53360** Added the ability to display the phones current ARP table in the diagnostic menu (*applies to SpectraLink 84xx*).
- 53908** The Web-based configuration utility now offers the ability to configure soft keys and line-keys.
- 54301** The timestamp now displays alongside the caller in the Call Lists.
- 54648** Added HTTPS support in the Updater that was previously called BootROM.
- 54680** Introduced the ability to import and export local and global configuration files using a PC browser.
- 54683** The browser-based SW Upgrade button that enables user to upgrade phones with one of multiple compatible software versions is available on the Polycom provisioning server.
- 54730** Noticeable enhancement from the time the phone is powered up and when it is ready for use.
- 56150** Added Data Link Layer L2 Discovery between phones and PC.
- 56187** Added ToID and FromID in SIP Publish packets for VQMon reports.
- 56274** Added multicast group paging based on the SpectraLink PTT solution.
- 56942** Configuring Soft key (EFK) settings no longer require a reboot.
- 57392** Added support to the microbrowser for HTTP proxy authentication (*applies to SpectraLink 84xx*).

- 57981** Added support for custom device certificates.
- 58007** Added the ability to revoke certificates used in SSL transactions by using OCSP.
- 58336** Added SHOULD SDP answer behavior as per RFC 3264.
- 58507** Enhanced the Web-based configuration and provisioning utility.
- 60297** Added the ability of random distribution of polling to check for software upgrades.
- 60907** Added the ability to disable Call Waiting while still allowing further outgoing calls.
- 61051** Added the ability to display custom soft keys on input forms in the microbrowser.
- 61138** Added support for incoming TLS connections on the Web server.
- 61343** Added the ability to disable authentication verification for received SRTP packets.
- 62671** Added a time-stamped log event indicating when the phone is ready to be used.
- 63592** Added API calls to the microbrowser for Media Player.
- 63629** Added Sennheiser EHS configuration menus and options.
- 64144** The alerting LED and associated line-key animation for second and subsequent incoming calls are now disabled when the Call Waiting feature waiting is disabled.
- 64243** The API Data push message size limit has been increased to 2048 bytes from 1024 bytes.
- 64359** Converted the BLA dialog rendering from *No* to *Yes* for user agents that are a remote party to the existing call dialog.
- 65287** Added the ability to prevent a phone from being provisioned at start-up.
Configuration parameter `prov.startupCheck.enabled` [default = 1 (enabled)]
- 66212** Added support for setting the syslog server address from DHCP.
- 66323** Added an administrator operations menu in the Updater to the setup menu: Reboot, Reset Settings, Format File System, and Install BootBlock.
- 66604** The phone reports connectivity event notifications to an 802.1x enabled switch port when a non-authenticated PC disconnects or reconnects to the phone.
- 67600** Password and other security entry fields now perform a brief echo of entered characters before being obscured from view.
- 68110** Added control of available telephony features on the Office Communications Server (OCS) using the `reg.x.telephony` configuration parameter.
- 68899** Added the ability on the microbrowser to enter two-digit dial pad values for selecting entries in a list.
- 69225** Added the ability to allow a hard key to be directly assigned an Enhanced Feature Key (EFK) style macro.
- 69442** Productivity features such as LDAP, Local Call Recording, and Visual Conference Management are enabled without the requirement of a license file. Note that VQMon remains a licensable feature.
- 71633** The Reset setting in the Updater menu does not erase the CA and Device Certificates.

Enhanced Capabilities

- 31158** Instant messages can now be sent if `msg.bypassInstantMessage=1`. The phone menu is no longer bypassed after pressing the Messages button.
- 38407** After reboot, the phones now transmit a TCP message to the outbound proxy address.
- 39249** Distortion on sound from a gateway call back has been removed (*applies to SoundPoint IP 330, 550, 560, and 650*).
- 40103** The file name of a file copied to a full USB stick full no longer displays (*applies to SoundPoint IP 650 and 670*).
- 40393** The backlight adjustment has been adjusted to work correctly when the incoming call times out.
- 41662** The buzzing sound heard by the far-end user in hands-free mode when a call is answered and while ringing has been removed (*applies to SoundPoint IP 550, 650, and 670*).
- 41897** The sound heard on the phone when attempting to cancel a conference or transfer has been removed.
- 42427** On the phone microbrowser, the thin line over the data field has been removed.
- 42442** The Select hard key has been made functional in the speed dial menu (*applies to SoundPoint IP 330*).
- 43822** When there is an active call, the backlight now adjusts correctly.
- 43846** The menu widget now scales to the correct size of the menu (*applies to SpectraLink 84xx*).
- 43864** The Soft key, line key, and status widgets can now be scaled (*applies to SpectraLink 84xx*).
- 44981** The phone now seizes the correct line for speed dial when `call.stickyAutoLineSeize.onHookDialing=1`.
- 45411** The SoundPoint IP 550 and 650 speaker phone volume can now be adjusted through gain settings for Rx audio.
- 45806** An unsupported format message no longer appears when trying to play a short WAV file.
- 45889** Lighten and Darken soft keys now display for the selected background image only in the Background menu (*applies to SoundPoint IP 450, 550, 560, and 650*).
- 45900** The time and date now display on the multiple call appearances screen (*applies to SoundPoint IP 450*).
- 46134** The phones now play a default ringtone when the ringtone size is larger than the tone quota or the ringtone is not in the cache.
- 46170** On the phone menu, the Local Directory or Corporate Directory cursor can now reach the end of the highlighting bar (*applies to SoundPoint IP 450*).
- 46773** On the microbrowser, the backspace soft key is now made available when the user enters a character to the input box (*applies to SoundPoint IP 320, 330, 430, and 450*).
- 48153** In the phone menu password settings, deleting a character before the character timeout now clears the last asterisk symbol.
- 48217** When `ramdisk.nBlocks=0` is set, `ramdisk.nBlocks` no longer consumes extraneous memory.

- 48753** The XML dictionary download no longer fails when the dictionary file size exceeds the defined size.
- 50234** The phone no longer crashes when starting a native application (*applies to SpectraLink 84xx*).
- 50633** When the user enters text for a contact in the Contact Directory, the backspace soft key now appears before the character selection widget for the first character has closed (*applies to SoundPoint IP 330 and 331*).
- 50735** After a conference call, redial now dials the last dialed number (*applies to SoundPoint IP 330 and 335*).
- 50745** Pressing the hookswitch toggle quickly no longer creates a phone and headset mismatch.
- 50766** During a save confirmation screen, if there is a missed call, the contact can now be saved. (*SoundPoint IP 330, 331, and 335*).
- 50788** When the user tries to play a recorded audio file from Menu/Status/Diagnostics/Test Hardware/Audio Diagnostics on different termination points (Handsfree/Headset/Handset), the icons are now updated (*applies to SoundPoint IP 450*).
- 50972** SIP call format is no longer missing for URL dialing (*applies to SoundPoint IP 330 and 335*).
- 51238** In the Install Custom CA Certificate menu on the phone, the 1/A/a soft key is no longer missing (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).
- 51301** A loud ring has been removed from the speaker when canceling a conference call or switching between calls.
- 51493** When call forward is enabled on the second registered line, phones now display the text string Call Forward Enabled on the top of the UI (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).
- 51751** When there are multiple calls waiting, dropping one remote party now plays the call waiting ring on the originating phone.
- 51767** The phone no longer crashes after trying to add a large number of contacts.
- 51930** The phone now shows *Connecting:* instead of *Transfer to:* on the top UI when initiating a transfer (*applies to SoundPoint IP 330 and 335*).
- 51994** When the contact is highlighted in the Contact Directory, pressing the character selection widget no longer erases the highlight bar (*applies to SoundPoint IP 330, 331, and 335*).
- 51996** Trying to dial an invalid speed dial number no longer causes slight corruption in the phone UI (*applies to SoundPoint IP 330, 331, and 335*).
- 52006** The call waiting tone no longer changes to a single beep when a double beep is configured on the phone.
- 52007** When a call is automatically disconnected at the far end phone after time out, the current active call no longer goes on hold inadvertently.
- 52103** The phones no longer display the first line label in scrolling status bar when it is visible (*applies to SoundPoint IP 330, 331, and 335*).
- 52112** In the phone Contact Directory menu, when the 1/A/a soft key is pressed to change the capitalization of the letters, the encoding no longer resets to ASCII (*applies to SoundPoint IP 450, 560, and 670*).

- 52219** In the phone Audio Diagnostics Menu, pressing the left arrow to exit now restores the previously selected termination state (*applies to SoundPoint IP 450, 560, and 670*).
- 52270** Backlight values now match the phone menu option and override file parameters.
- 52276** When a call is answered automatically while a recorded WAV file is playing the audio player screen now exits correctly (*applies to SoundPoint IP 650*).
- 52380** When the phone lines are configured to call server and presence server respectively, the presence information now displays on the first line as well as the presence line.
- 52489** Character encoding is no longer available while entering the contact information in the contact directory (*applies to SoundPoint IP 430, 550, 650, and 670*).
- 52552** The momentarily observed UI corruption/flickering during an incoming call with a long caller ID has been resolved (*applies to SoundPoint IP 330, 331, and 335*).
- 52560** Menu titles now fit in the phone UI (*applies to SoundPoint IP 330, 331, and 335*).
- 52688** Enhanced the `mb.main.idleTimeout` parameter behavior.
- 52692** The character entry mode now shows consistently when entering passwords in the phone menu (*applies to SoundPoint IP 450, 560, and 670*).
- 52734** The password field now clears after entering an invalid password (*applies to SoundPoint IP 330, 331, and 335*).
- 52766** The speaker/headset key LED no longer remains ON when the active call is on hold while there is an incoming call on the same line (*applies to SoundPoint IP 550, 560, 650, and 670*).
- 52789** The phone settings menu now has an appropriate label for the menu item in Handsfree Mode.
- 52917** In the phone Install Custom CA Cert menu, the character selection widget no longer remains visible when closed (*applies to SoundPoint IP 330, 331, and 335*).
- 52920** When a custom CA cert URL is unreachable, an appropriate message now displays on the phone.
- 52945** Inappropriate characters no longer display in the quick search menu (*applies to SoundPoint IP 330, 331, and 335*).
- 52948** When long text is entered in the quick search menu, the text now moves forward and displays the last character (*applies to SoundPoint IP 330, 331, and 335*).
- 53055** When composing a new instant message and if there is a new incoming instant message, the UI no longer becomes corrupted (*applies to SoundPoint IP 560 and 670*).
- 53101** VQMon values displayed on the SQmediator are now the same as the single SIP-Publish packet values.
- 53138** Instant messaging strings now have spaces in between words in all instances.
- 53142** Extra spaces at the beginning and end of the phone labels have been removed.
- 53235** Extra spaces in the phone exit menu have been removed.
- 54147** The phone conference screen titles are no longer truncated when using certain languages (*applies to SoundPoint IP 330, 331, and 335*).
- 54449** The phone no longer displays an error message when trying to edit a long contact number.

- 54553** When the user tries to use two lines to dial, the user no longer sees a scrolling message instead of Enter Number text (*applies to SoundPoint IP 450, 650, and 670*).
- 54913** In a BLF scenario, when the monitored phone places a call to another phone, the Dialog Event Package no longer contains repeated remote identity when its INVITE has received an initial 407 or 401 response.
- 55639** When the enhanced BLF feature is enabled on an active call, the call state no longer changes in the call appearance for monitored users (*applies to SoundPoint IP 650*).
- 55655** Pressing the message hard key now leads the user to voicemail (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).
- 55669** When the phone is configured as TCP only, and the phone receives a REFER in UDP, the phone now sends an INVITE in TCP.
- 55681** When transport is set to TCP, Refer-Based Click-To-Dial now works when the phone has an active call.
- 55852** When the phone is in the setup menu, pressing * key now always moves a page up (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).
- 55957** When the BLF feature is enabled, the remote call appearance screen now correctly times out and does not wait until the call is ended by the monitored user.
- 57538** In the phone menus, field names no longer truncate when the user tries to make edits (*applies to SpectraLink 84xx*).
- 57625** Applications are now loading as per the order specified (*applies to SpectraLink 84xx*).
- 58860** When the forward feature is enabled, the number of rings set now matches the actual ring cadences.
- 59086** When the phone is configured to an external server like CMA, the phone clock format (12 hrs – 24 hrs) does not get affected until the phone reboots.
- 59202** When the phone loses an active call on hold, pressing resume no longer drops the call.
- 59285** The phone no longer shows a delete (<<) soft key when there are no numbers to delete (*applies to SoundPoint IP 330, 331, and 335*).
- 59355** The phone no longer logs error messages when it is unable to connect to any of the telephony notification URLs.
- 59463** When using the phone's Web configuration, the phone no longer restarts when updating telephony notification event or URL.
- 59478** In the phone digit map, segments longer than 40 characters no longer truncated to 40 when applied.
- 61100** Added the ability to override complex audio codec instance count definition for each individual codec type.
- 61553** The phone no longer crashes when trying to split a conference service which is unavailable.
- 63190** The macro \$FServerACDSignIn\$ now works when configuring the soft key using EFK to exercise the `ServerACDSignIn` function.

- 63582** The excessively long boot time resulting from FTP errors and failures has been noticeably decreased (*applies to SpectraLink 84xx*).
- 64389** Added the ability to set the correct TLS Profile using the Updater and/or Application UI menus.
- 64455** There is no longer a delay between the time the Push URL is sent to the phone and the time it takes the browser to execute the fetch URL (*applies to SpectraLink 84xx*).
- 64464** The phones no longer wait to auth/re-associate to AP until AP starts the full security exchange (*applies to SpectraLink 84xx*).
- 64693** The payload settings specified by the phone are now used by the receiving phone.
- 64932** When the server side Call Forward No Answer (CFNA) is enabled, the user no longer has the option to select the number of rings on the local phone.
- 65081** The user is now able to navigate on the phone menus having select options available (*applies to SpectraLink 84xx*).
- 65082** The radio performance has been improved to reduce the reported number of missed packets and high retry rates (*applies to SpectraLink 84xx*).
- 65255** Selected options on the menus no longer disappear when selected from the phone navigation right hard key (*applies to SpectraLink 84xx*).
- 65275** When persistence login is enabled for default, the user log out no longer reboots the phone.
- 65309** Incorrect soft key options no longer show up when certain selections are invoked in the menu/UI (*applies to SpectraLink 84xx*).
- 65337** The phones now provision correctly through HTTPS.
- 65622** In case of emergency failover, the correct information with Invite request is routed (*applies to SoundPoint IP 450, 650, and 335*).
- 66582** In the quick setup menu, the unwanted Ok soft key no longer appears while changing the Boot server options (*applies to SpectraLink 84xx*).
- 66975** The voicemail icon now displays when there is a voicemail notification on the phone (*applies to SpectraLink 84xx*).
- 67928** In the phone buddy list, the pop-up message *Contact already exists* has been resolved.
- 68345** The phone dial pad keys now wake/light up when the phone comes back from the power save mode (*applies to SpectraLink 84xx*).
- 68533** A reboot is no longer necessary to change the parameter `voIpProt.SIP.conference.address` in the configuration files.
- 69372** During an IM chat session, the phone now displays all messages using the same quick note string (*applies to SpectraLink 84xx*).
- 69373** When provisioning TLS applications, the CommonName of a server certificate is now configurable with the new configuration option.
- 69374** User interface changes have been made in the Application menu and BootROM menu for SIP and Provisioning applications respectively to make CommonName configurable.
- 69929** When the BLF feature is enabled, the monitored BLF line status now updates as busy after the attendant phone transfers a call to that line (*applies to SoundPoint IP 650*).

- 69965** When the soft key is configured with capital HTTP in the URL, it no longer tries to dial instead of using the phone microbrowser.
- 70258** Resetting the phone to factory settings no longer clears the password (*applies to SpectraLink 84xx*).
- 70458** Text on the input fields and soft keys now correctly display.
- 70547** Download Custom CA Cert screen now disappears after saving a certificate (*applies to all SoundPoint IP*).
- 70633** The phone no longer mishandles 403 responses to first REFER sent to phone to centralized conference server.
- 70732** The links on the microbrowser page are no longer hidden when the links are highlighted and moves to another link.
- 70908** When using user profiles, the phone no longer powers off at the user login prompt (*applies to SpectraLink 84xx*).
- 71046** When the phone is in a conference call and a roaming attempt is made, the access point no longer rejects the reassociation request (*applies to SpectraLink 84xx*).
- 71103** When the phone is configured as per call center mode, call hold is no longer treated as a missed call (*applies to SoundPoint IP 650*).
- 71156** When there is an active call between two parties, accepting another call no longer makes the phone vibrate continuously when the mode is set to Ring and Vibrate (*applies to SpectraLink 84xx*).
- 71913** Registered lines now display on Expansion modules (*applies to SoundPoint IP 650, 670*).

Configuration File Enhancements

Certain groups of configuration parameters have been modified in UC Software 4.0.0. In these cases, instead of listing every parameter, the following table specifies a group of related parameters with an abbreviated XML path name ending with (.*)

For example, suppose the following parameters are modified: `device.wifi.enabled`, `device.wifi.ipAddress`, and `device.wifi.ssid`. Since these parameters all begin with `device.wifi`, the following table abbreviates these parameters as `device.wifi.*`



Note: Parameters with .set

Most `device` parameters have identical parameters ending with `.set`. The `.set` parameters are not included in the following table.



Web Info: Parameters Changed in UC Software 4.0.0

The following table includes parameters changed in UC Software 4.0.0. You can find the new descriptions and values in the [Polycom UC Software Administrators Guide](#).

Table 13: UC Software 4.0.0 Parameter Enhancements

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Discontinued	apps.x.Label	Productivity Applications parameter
Discontinued	apps.x.Url	Productivity Applications parameter
Discontinued	apps.ucdesktop.IP	Productivity Applications parameter
Discontinued	apps.ucdesktop.name	Productivity Applications parameter
Discontinued	apps.ucdesktop.port	Productivity Applications parameter
Discontinued	device.auth.* The parameters device.auth.localAdminPassword and device.auth.localUserPassword have not been removed	Provisioning parameters
Discontinued	device.dhcp.offerTimeout	Provisioning parameter
Discontinued	device.prov.appProvString	Provisioning parameter
Discontinued	device.prof.appProvType	Provisioning parameter
Discontinued	device.sec.SSL.*	Provisioning parameters
Discontinued	device.sec.deviceCertEnabled	Provisioning parameter
Discontinued	exchange.server.address	Productivity Applications parameter
Discontinued	httpd.lp.port	Provisioning parameter
Discontinued	lcl.datetime.date.digitFormatEnable	User preference parameter
Discontinued	log.level.change.lp	Log parameter
Discontinued	log.level.change.nwmgr	Log parameter
Discontinued	log.level.change.sync	Log parameter
Discontinued	reg.x.filterReflectedBlaDialogs	Call feature parameter
Discontinued	reg.x.server.H323.y.register	Call feature parameter
Discontinued	sec.TLS.customDeviceCert.enable	Security parameter
Discontinued	sec.dot1x.eapollogoff.pcforscelanlinkreset	Security parameter
Discontinued	voIpProt.SDP.useLegacyPayloadType Negotiation	Call feature parameter
Discontinued	voIpProt.server.H323.x.register	Call feature parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Discontinued	voice.aec.hd.* The parameter voice.aec.hd.enable has not been removed	Audio parameters
Discontinued	voice.aec.hf.* The parameter voice.aec.hf.enable has not been removed	Audio parameters
Discontinued	voice.aec.hs.* The parameter voice.aec.hs.enable has not been removed	Audio parameters
Discontinued	voice.aes.hd.duplexBalance	Audio parameter
Discontinued	voice.aes.hf.* The parameter voice.aes.hf.enable has not been removed	Audio parameters
Discontinued	voice.aes.hs.duplexBalance	Audio parameter
Discontinued	voice.gain.rx.digital.ringer.* The parameter voice.gain.rx.digital.ringer has also been removed	Audio parameters
Discontinued	voice.handset.wideband	Audio parameter
Discontinued	voice.rxEq.hf.postFilter.* The parameter voice.rxEq.hf.postFilter.enable has not been removed	Audio parameters
Discontinued	voice.rxEq.hf.preFilter.* The parameter voice.rxEq.hf.preFilter.enable has not been removed	Audio parameters
Discontinued	voice.rxEq.hs.postFilter.* The parameter voice.rxEq.hs.postFilter.enable has not been removed	Audio parameters
Discontinued	voice.rxEq.hs.preFilter.* The parameter voice.rxEq.hs.preFilter.enable has not been removed	Audio parameters
Added	apps.x.label	Productivity Applications parameter
Added	apps.x.url	Productivity Applications parameter
Added	apps.push.secureTunnelEnabled	Productivity Applications parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	apps.push.secureTunnelPort	Productivity Applications parameter
Added	apps.push.secureTunnelRequired	Productivity Applications parameter
Added	apps.statePolling.responseMode	Productivity Applications parameter
Added	apps.telNotification.callStateChangeEvent	Productivity Applications parameter
Added	apps.telNotification.userLogInOutEvent	Productivity Applications parameter
Added	apps.ucdesktop.* The parameter apps.ucdesktop.enabled has existed in previous versions	Productivity Applications parameters
Added	bg.color.*	Background parameter
Added	bluetooth.radioOn	Call feature parameter
Added	call.advancedMissedCalls.*	Call feature parameters
Added	call.callWaiting.enable	Call feature parameter
Added	call.localConferenceEnabled	Call feature parameter
Added	callLists.*	Call feature parameters
Added	device.hostname	Provisioning parameter
Added	device.net.dhcpBootServer	Provisioning parameter
Added	device.net.dot1x.*	Provisioning parameters
Added	device.pacfile.*	Provisioning parameters
Added	device.prov.upgradeServer	Provisioning parameter
Added	device.sec.TLS.*	Provisioning parameters
Added	device.usbnet.*	Provisioning parameters
Added	device.wifi.*	Provisioning parameter
Added	dialplan.applyToPstnDialing	Call feature parameter
Added	dialplan.routing.emergency.outboundIdentity	Call feature parameter
Added	dialplan.routing.emergency.outboundIdentity.lldp	Call feature parameter
Added	dialplan.routing.ermgency.preferredSource	Call feature parameter
Added	exchange.meeting.*	Productivity Applications parameter
Added	exchange.server.url	Productivity Applications parameter
Added	feature.audioVideoToggle.enabled	Call feature parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	<code>feature.bluetooth.enabled</code>	Call feature parameter
Added	<code>feature.enhancedCallDisplay.enabled</code>	Call feature parameter
Added	<code>feature.exchangeCalendar.enabled</code>	Call feature parameter
Added	<code>feature.nonVolatileRingerVolume.enabled</code>	Call feature parameter
Added	<code>httpd.cfg.secureTunnelEnabled</code>	Web Configuration Utility parameter
Added	<code>httpd.cfg.secureTunnelPort</code>	Web Configuration Utility parameter
Added	<code>httpd.cfg.secureTunnelRequired</code>	Web Configuration Utility parameter
Added	<code>httpd.ta.secureTunnelEnabled</code>	Web Configuration Utility parameter
Added	<code>httpd.ta.secureTunnelPort</code>	Web Configuration Utility parameter
Added	<code>httpd.ta.secureTunnelRequired</code>	Web Configuration Utility parameter
Added	<code>ind.pattern.blink.*</code>	LED indicator parameter
Added	<code>ind.pattern.flashSlow2.*</code>	LED indicator parameter
Added	<code>lcl.x.pstnCountry</code>	Multilingual parameter
Added	<code>lcl.aidt</code>	Multilingual parameter
Added	<code>lcl.callerId</code>	Multilingual parameter
Added	<code>lcl.callerIdType</code>	Multilingual parameter
Added	<code>lcl.country.*</code> The parameter <code>lcl.country</code> has also been added	Multilingual parameters
Added	<code>lcl.dtmfLevel</code>	Multilingual parameter
Added	<code>lcl.dtmfTwist</code>	Multilingual parameter
Added	<code>lcl.flashTiming</code>	Multilingual parameter
Added	<code>lcl.pstnCountryIndex</code>	Multilingual parameter
Added	<code>lineKey.*</code>	Flexible line key assignment parameters
Added	<code>log.level.change.barcd</code>	Log parameter
Added	<code>log.level.change.bluet</code>	Log parameter
Added	<code>log.level.change.clist</code>	Log parameter
Added	<code>log.level.change.daa</code>	Log parameter
Added	<code>log.level.change.dock</code>	Log parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	log.level.change.drvtb	Log parameter
Added	log.level.change.oaip	Log parameter
Added	log.level.change.ocsp	Log parameter
Added	log.level.change.pdc	Log parameter
Added	log.level.change.pres	Log parameter
Added	log.level.change.pstn	Log parameter
Added	log.level.change.ptt	Log parameter
Added	log.level.change.rtls	Log parameter
Added	log.level.change.tls	Log parameter
Added	log.level.change.wifi	Log parameter
Added	log.render.stdout.* The parameter log.render.stdout has existed in previous versions	Log parameter
Added	mb.main.toolbar.autoHide.* The parameter mb.main.toolbar.autoHide.enabled has existed in previous versions	Microbrowser parameters
Added	messaging.*	SpectraLink instant messaging parameters
Added	np.*	SpectraLink notification profiles parameters
Added	oai.*	SpectraLink Open Application Interface parameters
Added	prov.login.*	Distributed polling parameters
Added	prov.loginCredPwdFlushed.enabled	Distributed polling parameter
Added	prov.polling.timeRandomEnd	Distributed polling parameter
Added	prov.startupCheck.enabled	Provisioning parameter
Added	pstn.*	
Added	ptt.*	Paging and push-to-talk parameters
Added	qbc.*	SpectraLink quick barcode connector parameters

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	qos.ethernet.* The parameters qos.ethernet.callControl.user_priority, qos.ethernet.other.user_priority, qos.ethernet.rtp.user_priority, and qos.ethernet.rtp.video.user_priority existed in previous versions	SpectraLink QoS parameters
Added	reg.x.auth.domain	Call feature parameter
Added	reg.x.auth.useLoginCredentials	Call feature parameter
Added	reg.x.gruu	Call feature parameter
Added	reg.x.server.y.specialInterop	Call feature parameter
Added	reg.x.server.y.useOutboundProxy	Call feature parameter
Added	reg.x.srtp.enable	Call feature parameter
Added	reg.x.srtp.offer	Call feature parameter
Added	reg.x.srtp.require	Call feature parameter
Added	reg.x.telephony	Call feature parameter
Added	se.pat.misc.customX.*	Sound effects parameters
Added	se.pat.misc.miscX.*	Sound effects parameters
Added	se.rt.answerMute.*	Sound effects parameters
Added	se.rt.autoAnswer.micMute	Sound effects parameter
Added	se.rt.autoAnswer.videoMute	Sound effects parameter
Added	se.rt.customX.micMute	Sound effects parameter
Added	se.rt.customX.videoMute	Sound effects parameter
Added	se.rt.default.micMute	Sound effects parameter
Added	se.rt.default.videoMute	Sound effects parameter
Added	se.rt.emergency.micMute	Sound effects parameter
Added	se.rt.emergency.videoMute	Sound effects parameter
Added	se.rt.external.micMute	Sound effects parameter
Added	se.rt.external.videoMute	Sound effects parameter
Added	se.rt.internal.micMute	Sound effects parameter
Added	se.rt.internal.videoMute	Sound effects parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	se.rt.precedence.micMute	Sound effects parameter
Added	se.rt.precedence.videoMute	Sound effects parameter
Added	se.rt.ringAnswerMute.*	Sound effects parameters
Added	se.rt.splash.micMute	Sound effects parameter
Added	se.rt.splash.videoMute	Sound effects parameter
Added	se.rt.visual.micMute	Sound effects parameter
Added	se.rt.visual.videoMute	Sound effects parameter
Added	sec.TLS.SIP.strictCertCommonNameValidation	Security parameter
Added	sec.TLS.customCaCert.*	Security parameters
Added	sec.TLS.customDeviceCert.*	Security parameters
Added	sec.TLS.customDeviceKey.*	Security parameters
Added	sec.TLS.profile.*	Security parameters
Added	sec.TLS.profileSelection.*	Security parameters
Added	sec.hostMoveDetect.*	Security parameters
Added	sec.srtp.holdWithNewKey	Security parameter
Added	sec.srtp.mki.length	Security parameter
Added	sec.srtp.resumeWithNewKey	Security parameter
Added	softkey.x.insert	Security parameter
Added	tcpIpApp.fileTransfer.waitForLinkIfDown	IP parameter
Added	tone.chord.misc.A3Major.*	Tone parameters
Added	tone.chord.misc.C3Major.*	Tone parameters
Added	tone.chord.misc.Db3Major.*	Tone parameters
Added	tone.chord.misc.E3Major.*	Tone parameters
Added	tone.chord.misc.cs12.*	Tone parameters
Added	up.25mmRealTime	Default 1 Used to configure whether a mobile phone or a PC is connected to the 2.5 mm port in the conference phones. min="1" max="2"

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	up.backlight.timeout.* The parameter up.backlight.timeout has also been added	User preferences parameters
Added	up.cfgWarningsEnabled	User preferences parameter
Added	up.displayOperMode	Default 0 Phone can display the status i.e., Up/Down when it is working on PSTN mode(Applicable to SoundStation duo).
Added	up.headsetOnlyAlerting	User preferences parameter
Added	up.hearingAidCompatibility.enabled	User preferences parameter
Added	up.hideDateTimeWhenNotSet	User preferences parameter
Added	up.multiKeyAnswerEnabled	User preferences parameter
Added	up.operMode	User preferences parameter
Added	up.pstnSetup	Default 0 User can setup the PSTN line by selecting the country when the phone initially boots up
Added	up.warningLevel	User preferences parameter
Added	upgrade.*	Provisioning parameter
Added	video.callMode.default	Video parameter
Added	video.debug	Video parameter
Added	voIpProt.SIP.dialog.strictXLineId	Call feature parameter
Added	voIpProt.SIP.IM.autoAnswerDelay	Call feature parameter
Added	voIpProt.SIP.mtls.enable	Call feature parameter
Added	voIpProt.SIP.pingMethod	Call feature parameter
Added	voIpProt.server.x.specialInterop	Call feature parameter
Added	voIpProt.server.x.useOutboundProxy	Call feature parameter
Added	voice.aec.bt.hd.enable	Audio parameter
Added	voice.aec.usb.hf.enable	Audio parameter
Added	voice.aes.bt.hd.enable	Audio parameter
Added	voice.aes.usb.hf.enable	Audio parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Added	voice.agc.bt.hd.enable	Audio parameter
Added	voice.agc.usb.hf.enable	Audio parameter
Added	voice.bt.*	Audio parameters
Added	voice.handset.rxag	Audio parameter
Added	voice.handset.rxdg	Audio parameter
Added	voice.handset.st.	Audio parameter
Added	voice.handset.txag	Audio parameter
Added	voice.handset.txdg	Audio parameter
Added	voice.handsfree.*	Audio parameters
Added	voice.headset.rxag	Audio parameter
Added	voice.headset.rxdg	Audio parameter
Added	voice.headset.st	Audio parameter
Added	voice.headset.txag	Audio parameter
Added	voice.headset.txdg	Audio parameter
Added	voice.ns.bt.*	Audio parameters
Added	voice.ns.usb.*	Audio parameters
Added	voice.ringer.rxag	Audio parameter
Added	voice.rxEq.usb.*	Audio parameters
Added	voice.rxQos.ptt.*	Audio parameters
Added	voice.rxQos.wireless.*	Audio parameters
Added	voice.txEq.usb.*	Audio parameters
Added	voice.usb.*	Audio parameters
Added	voice.volume.persist.bluetooth.headset	Audio parameter
Added	voice.volume.persist.usb.handsfree	Audio parameter
Added	wifi.*	Wifi parameters
Changed	apps.push.messageType	Productivity Applications parameter
Changed	apps.uc.desktop.enabled	Productivity Applications parameter
Changed	call.autoRouting.preferredProtocol	Call feature parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Changed	device.auth.*	Provisioning parameters
Changed	device.cma.mode	Provisioning parameter
Changed	device.dhcp.* The parameter device.dhcp.bootSrvOpt has not been changed	Provisioning parameters
Changed	device.dns.*	Provisioning parameters
Changed	device.em.power	Provisioning parameter
Changed	device.line.*	Provisioning parameters
Changed	device.net.* The parameters device.net.dhcpBootServer, device.net.IPgateway, device.net.subnetMask, and device.net.vlanId have not been changed	Provisioning parameters
Changed	device.ntlm.versionMode	Provisioning parameter
Changed	device.prov.* The parameters device.prov.password, device.prov.serverName, device.prov.upgradeServer, and device.prov.user have not been changed	Provisioning parameters
Changed	device.serial.enable	Provisioning parameter
Changed	device.sntp.gmtOffset	Provisioning parameter
Changed	device.syslog.* The parameter device.syslog.serverName has not been changed	Provisioning parameters
Changed	dialplan.x.digitmap	Call feature parameter
Changed	dialplan.x.routing.server.y.transport	Call feature parameter
Changed	dialplan.digitmap	Call feature parameter
Changed	dialplan.digitmap.timeOut	Call feature parameter
Changed	dialplan.impossibleMatchHandling	Call feature parameter
Changed	dialplan.removeEndOfDial	Call feature parameter
Changed	dialplan.routing.server.x.transport	Call feature parameter
Changed	dir.H350.dev.attribute.x.type	Directory parameter
Changed	dir.H350.dev.transport	Directory parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Changed	dir.H350.group.attribute.x.type	Directory parameter
Changed	dir.H350.group.transport	Directory parameter
Changed	dir.H350.person.attribute.x.type	Directory parameter
Changed	dir.H350.person.transport	Directory parameter
Changed	dir.corp.attribute.x.type	Directory parameter
Changed	dir.corp.transport	Directory parameter
Changed	dir.local.nonVolatile.maxSize	Directory parameter
Changed	dir.local.volatile.maxSize	Directory parameter
Changed	divert.x.autoOnSpecificCaller	Call feature parameter
Changed	divert.x.contact	Call feature parameter
Changed	divert.x.sharedDisabled	Call feature parameter
Changed	divert.busy.*	Call feature parameters
Changed	divert.dnd.*	Call feature parameters
Changed	divert.fwd.x.enabled	Call feature parameter
Changed	divert.noanswer.*	Call feature parameters
Changed	ind.led.x.index	LED indicator parameter
Changed	keypadLock.*	Phone lock parameters
Changed	lcl.ml.lang.clock.x.format	Multilingual parameter
Changed	lcl.ml.lang.list	Multilingual parameter
Changed	lcl.ml.lang.menu.*	Multilingual parameters
Changed	lcl.ml.lang.tags.*	Multilingual parameters
Changed	log.level.change.slog	Log parameter
Changed	log.render.file.size	Log parameter
Changed	log.render.file.upload.append.limitMode	Log parameter
Changed	log.render.file.upload.period	Log parameter
Changed	log.render.stdout	Log parameter
Changed	log.sched.*	Log parameters
Changed	msg.bypassInstantMessage	Voicemail parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Changed	nat.* The parameter nat.keepalive.interval has not been changed	IP parameters
Changed	phoneLock.enabled	Phone lock parameter
Changed	pnet.remoteCall.dtmfDuration	Peer networking parameter
Changed	powerSaving.enable	Power-saving parameter
Changed	prov.fileSystem.ffs0.minFreeSpace	Provisioning parameter
Changed	prov.polling.mode	Distributed polling parameter
Changed	prov.polling.time	Distributed polling parameter
Changed	qos.ethernet.*	Quality of Service parameters
Changed	qos.ip.callControl.* The parameters qos.ip.callControl.dscp.* have not been changed	Quality of Service parameters
Changed	qos.ip.rtp.* The parameters qos.ip.rtp.dscp.* and qos.ip.rtp.video.dscp.* have not been changed	Quality of Service parameters
Changed	reg.x.callsPerLineKey	Line registration parameter
Changed	reg.x.outboundProxy.transport	Line registration parameter
Changed	reg.x.ringType	Line registration parameter
Changed	reg.x.server.y.transport	Line registration parameter
Changed	res.finder.minFree	Phone memory parameter
Changed	res.finder.sizeLimit	Phone memory parameter
Changed	res.quotas.background	Phone memory parameter
Changed	res.quotas.bitmap	Phone memory parameter
Changed	res.quotas.cache	Phone memory parameter
Changed	res.quotas.font	Phone memory parameter
Changed	res.quotas.tone	Phone memory parameter
Changed	res.quotas.xmlui	Phone memory parameter
Changed	roaming_buddies.reg	SpectraLink call feature parameter
Changed	roaming_privacy.reg	SpectraLink call feature parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Changed	se.destination	Sound effects parameter
Changed	se.pat.callProg.msgWaiting.name	Sound effects parameter
Changed	se.pat.misc.instantMessage.name	Sound effects parameter
Changed	se.pat.misc.localHoldNotification.name	Sound effects parameter
Changed	se.pat.misc.messageWaiting.name	Sound effects parameter
Changed	se.pat.misc.negativeConfirm.name	Sound effects parameter
Changed	se.pat.misc.positiveConfirm.name	Sound effects parameter
Changed	se.pat.misc.remoteHoldNotification.name	Sound effects parameter
Changed	se.pat.misc.welcome.name	Sound effects parameter
Changed	sec.H235.*	Sound effects parameters
Changed	tcpIpApp.keepalive.*	IP parameters
Changed	tcpIpApp.port.*	IP parameters
Changed	up.25mm	User preferences parameter
Changed	up.analogHeadsetOption	User preferences parameter
Changed	up.backlight.idleIntensity	User preferences parameter
Changed	up.oneTouchVoiceMail	User preferences parameter
Changed	up.useDirectoryNames	User preferences parameter
Changed	up.welcomeSoundEnabled	User preferences parameter
Changed	up.welcomeSoundOnWarmBootEnabled	User preferences parameter
Changed	video.camera.frameRate	Video parameter
Changed	video.localCameraView.fullScreen.mode	Video parameter
Changed	video.maxCallRate	Video parameter
Changed	video.screenMode	Video parameter
Changed	video.screenModeFS	Video parameter
Changed	voIpProt.H323.dtmfViaSignaling	Call feature parameter
Changed	voIpProt.H323.enable	Call feature parameter
Changed	voIpProt.H323.local.port	Call feature parameter
Changed	voIpProt.SIP.local.port	Call feature parameter

<i>Modification</i>	<i>Configuration Parameter</i>	<i>Description</i>
Changed	voIpProt.SIP.outboundProxy.transport	Call feature parameter
Changed	voIpProt.SIP.specialEvent.lineSeize.nonStandard	Call feature parameter
Changed	voIpProt.server.x.transport	Call feature parameter
Changed	voIpProt.server.dhcp.option	Call feature parameter
Changed	voice.audioProfile.Lin16.16ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.32ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.44_1ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.48ksps.payloadSize	Audio parameter
Changed	voice.audioProfile.Lin16.8ksps.payloadSize	Audio parameter
Changed	voice.codecPref.iLBC.*	Audio parameters
Changed	voice.gain.rx.analog.*	Audio parameters
Changed	voice.gain.rx.digital.*	Audio parameters
Changed	voice.gain.tx.analog.*	Audio parameters
Changed	voice.gain.tx.digital.*	Audio parameters
Changed	voice.handset.rxag.adjust.*	Audio parameters
Changed	voice.handset.sidetone.adjust.*	Audio parameters
Changed	voice.handset.txag.adjust.*	Audio parameters
Changed	voice.headset.sidetone.adjust.*	Audio parameters
Changed	voice.headset.txag.adjust*	Audio parameters
Changed	voice.ns.hd.* The parameter voice.ns.hd.enable has not been changed	Audio parameters
Changed	voice.ns.hf.signalAttn	Audio parameter
Changed	voice.ns.hf.silenceAttn	Audio parameter
Changed	voice.ns.hs.signalAttn	Audio parameter
Changed	voice.ns.hs.silenceAttn	Audio parameter

Understanding Updates to UC Software 3.3.2

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 56249** Added user confirmation on the phone before placing outgoing calls as part of the click to dial behavior.
- 64548** Added missed-call synchronization. When local call lists are disabled on the phones, Missed call notifications are sent from the call server to the respective users.
- 66464** Provides simplified display option by removing protocol tag and host details (*applies to VVX 1500D*).
- 66466** When the parameter `up.simplifiedSipCallInfo` is enabled, the caller ID does not display the host name for incoming and outgoing calls and the protocol information is not shown.
- 66624** Geographical redundancy enhancements.
- 68836** Added support for Sennheiser EHS headset to the phone menus and configuration.
- 70475** Extended the `dialplan.digitmap` String to support up to 100 from 30 segments.

Enhanced Capabilities

- 27469** Local Conferencing is no longer disabled if G.729 is in the Codec preference list (*applies to SoundStation IP 4000*).
- 27777** The phone now plays a local hold reminder tone (*applies to SoundStation IP 4000*).
- 34454** If the microbrowser is enabled and it is refreshed too frequently and the pages contain large images, the phone no longer locks up.
- 34743** A phone no longer freezes when it receives a check-sync to indicate if the resources on the phone are heavily used by downloaded wave files or by large/complex microbrowser pages.
- 39630** When using the SoundPoint IP 330/320 phone with LCS2005, blocking a roaming buddy from the privacy list no longer prevents the user from viewing the blocked buddy's status.
- 45247** The SoundPoint IP 430 no longer reboot when viewing microbrowser pages and the internal memory is being used for other function/operations.
- 47827** The Voice Quality Monitor feature on SoundPoint IP now uses the correct units for jitter in SIP PUBLISH VQSession Reports.
- 49324** When dialing 99* from the phone with an integrated Polycom HDX system, the * is no longer changed to a dot on the HDX system (*applies to SoundStation IP 7000*).
- 51904** When a Polycom HDX system is configured for SIP through UDP, it now makes a video connection with the VVX 1500 phone.
- 52592** The phone now provisions when using the combined **sip.ld** file and a TFTP provisioning server that does not support the bulk size option (*applies to SoundStation IP 6000*).

- 52782** Resolved some video issues that occur when VVX 1500 phones are bridged on Polycom HDX and VSX MCUs.
- 55910** The phone no longer freezes/stops while preparing to boot up (*applies to SoundPoint IP 430*).
- 57838** When there is a call between two SoundStation IP 7000 phones and an HDX system, the HDX 9004 system added a video call to an HDX 9002 system. There is no longer a hold between the SoundPoint IP 7000 phones.
- 58177** When attempting to do a blind transfer from a PSTN line to an internal extension, three beeps are no longer heard after pressing the Send soft key. Cancelling the operations enables the call transfer.
- 60086** The SoundPoint IP 650 phones now send an Off-Hook or On-Hook notification when set to Auto Answer.
- 60131** Reassignment of the speed dial keys now functions correctly (*applies to SoundStation IP 5000, 6000, and 7000*).
- 60186** Call connection bandwidth between the HDX and VVX 1500 now synchronizes when the VVX 1500 is in CMA provisioning mode.
- 60255** A noticeable high-frequency flicker has been removed from the display when an update for BLA remote hold/resume status occurs (*applies to SoundPoint IP 650/670*).
- 60729** The phone now honors a BLA NOTIFY when the version number in the message body has increased by more than 1.
- 60733** Can now establish a local conference bridge by using a speed-dial key, BLF line key or using call lists.
- 60984** The `dir.local.contacts.maxNum` parameter no longer accepts 0 as stated in the Administrators' Guide; `dir.local.contacts.naxNum` accepts 1 to 99 OR 1 to 9999.
- 61013** On the VVX 1500, the Call Rate value can no longer be set higher than Max Call Rate value when configured through the Web Configuration Utility.
- 61067** On the phone interface menu, pressing the Back soft key in the Authentication menu now restores the menu title correctly.
- 61089** A pop-up no longer appears when adjusting the ringer volume while the call is on hold. Hands-free Volume no longer appears instead of Ringer Volume (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 61145** The `dialplan.digitmap` uses up to a maximum of 767 characters. The last character is no longer truncated.
- 61147** The phone no longer reboots when a GET request is sent to the phone to `/TA/getParam?paramName=reg.1.ringType` (*applies to SoundPoint IP 331, 335, 450, 550, 560, 650, 670 and SoundStation IP 5000*).
- 61316** The message waiting indicator on the VVX 1500 shows up on the correct line after the registrations are moved to different line keys.
- 61955** An RTP audio delay is no longer detected when calling or receiving calls from a PSTN line.

- 62389** When the VVX 1500 phone is left idle for an extended period of time and a call is made between the CMA users, the pop-up on the phone user interface is no longer delayed for both incoming and outgoing H.323 calls.
- 62450** When the value in configuration parameter `mb.idleDisplay.home` is set to point to a URL with an image, the phones idle display no longer shows a break in the border located at the bottom-left corner.
- 62664** The VVX 1500 phones no longer drop the domain information when sending call setup to the gatekeeper.
- 62675** Calls placed with the VVX 1500 through the URL dialing now show up in the missed calls list.
- 62687** Disabling local call forward now stops the phone from forwarding calls.
- 62974** Local call forwarding rules are now disabled when the feature is disabled.
- 63070** Call Forward CF messages such as Call Forward *destination:Fwd:<number>* now display when DND is active.
- 63262** When using the SoundPoint 650 to dial a call through the Out of Dialog REFER based method, the user no longer needs to press the speakerphone key twice to terminate the call.
- 63277** During a call, a macro which is set to simulate a soft key press no longer displays an error (*applies to the VVX 1500*).
- 63452** When URL dialing is disabled on the phone, transferring call initiated by BLF speed dial key no longer prompts again for the URL.
- 63746** During fail-back attempt, line icon shows as unregistered.
- 64036** Caller ID on the VVX 1500 now works correctly when both Contact Directory Matching and Chinese characters are enabled.
- 64190** Enhanced video quality during conferences with RMX 1500.
- 63850/64430** Initial dialog event NOTIFY after subscribe has the correct version.
- 64859** When a call is made through SRTP and TLS, the far end can now hear audio even when the SRTP packet sequence counter rolls over to zero.
- 64896** When Call Forwarding is on, the phone no longer updates the display before the server has confirmed operation using NOTIFY.
- 65014** When a VVX 1500 phone is provisioned through CMA 5.3, LDAP directory searches now return meeting rooms names when searching the CMA directory.
- 65207** Resolved a memory leak on the phones in specific call server environments when a call is answered from a hunt group.
- 65288** When server-side DND is enabled, an incoming call from a white list phone number can now be picked.
- 65345/64862** One of the callers is no longer dropped when trying to set up a conference between PSTN users.
- 65617** When Auto answer is enabled, the phone no longer sends a 180 response to the server.
- 65754** The XML string `<key key.25.VVX 1500.function.prim=null/>` no longer disables the Menu soft key or the Menu hard key (*applies to the VVX 1500*).

- 65758** An extra space has been removed from each side of an unlabeled character in the microbrowser idle display: G Ä rtner instead of GÄrtner.
- 65979** The SoundPoint IP phone quick setup menu user name entry has been set to numeric characters as the default.
- 66027** When the phone lock feature is enabled, trying to dial any number no longer dials the emergency number.
- 66106** When a call is made from the phone from the dial pad and pressing the speakerphone key it correctly selects line 2 instead of line 1.
- 66217** Directed Call Pick-up soft key now works correctly (*applies to the SoundPoint IP 650 and VVX 1500*).
- 66554** During a video call on the VVX 1500, the phone no longer drops video momentarily when there are periodic offer less re-invites/ session refresh messages from server.
- 66593** When trying to dial an extension of four digits by pressing only two digits, a prompt displays on the phone to enter more digits. After entering the other two digits, it no longer appends to the earlier two digits thereby resulting in a failed call (*applies to SoundPoint IP 3xx running UC Software 3.3.x*).
- 66621/66619** Created configuration parameters that allow phones to perform fail-over when a 503 response is received.
- 66625** The phone no longer sends three extra registration requests to primary proxy during fail-over.
- 66626** DNSTTL no longer counts down during a fail-back that fails. TTL is reset after re-registering to secondary.
- 66666** The SoundStation IP 7000 phone is now being provisioned through FTP when the Windows 2003 server path MTU Discovery is disabled.
- 66964** Local call forward behavior is now working, as mentioned in the Admin Guide for the parameter `voIpProt.SIP.serverFeatureControl.localProcessing.cf`.
- 67455** The phone no longer crashes during registration with TLS (*applies to SoundPoint IP 650/670*).
- 67622** The phone displays the correct caller name in the user interface where there is a call from a group pickup number.
- 67633** Added support for the Zero-Touch Provisioning (ZTP) feature. Note that feature should not be enabled unless the phone has been registered for use with the associated provisioning system offered by your service provider.
- 67641** Pressing the Directory soft key on the VVX 1500 phone no longer redirects to the Advanced Find screen automatically.
- 67642** When using the VVX 1500 to dial contacts from the corporate directory for which the dialing entries are not filled appropriately, the phones no longer displays the attributes and no longer skips null values.
- 67753** The phone no longer plays ringback after the call is timed out.
- 67867** The phone seizes the correct line after transferring an incoming call to the line when going off-hook.

- 67966** When a call is made between two VVX 1500s the answering VVX 1500 which comes up after first reboot no longer plays a noise pattern on the screen before playing video.
- 68063** The phone no longer reboots when DHCP failover occurs.
- 68195** Directed Pickup using star codes, for example, *200,*300 now works on the SoundPoint IP 650.
- 68267** When the BLF feature is enabled on the SoundPoint IP 650 and there is an active call, a ringtone heard in low sound within the handset has been removed.
- 68344** **The** Request Validation feature no longer rejects requests from another (second) server listed in the configuration files.
- 68376** The VVX 1500 phone no longer reboots when the DND button is pressed repeatedly during the idle state.
- 68382** The phone shows the correct time when the IP address on the NTP server is 12 digits.
- 68446** The parameter `call.hold.localReminder.startDelay` value is now honored when the value is less than 60 seconds.
- 68476** The SoundPoint IP 331 phone is now bootable when set with Option 60, ASCII String and DHCP Server Option 43.
- 69166** Placing a call is on hold in which RFC 3264 directionality attributes are present no longer results in failure of terminating music on hold session.
- 69421** The VVX 1500 phones no longer fail to pass special character \$ in the password using DHCP Option 66.
- 69671** The phones accept VLAN ID from DHCP option 129.
- 70027/62203** Conference with Genband CS2000 now works correctly.
- 70988** The SoundPoint IP 550, when powered by external AC power, no longer reboots when certain audio plays on full volume.
- 70233** When URL dialing is disabled, the user is no longer prompted for a URL when attempting to transfer calls after using directed call pickup of a monitored BLF line.
- 70317** During an active IVR call, if there is a second incoming call to the same line, the phone now sends DTMF.
- 70456** The SoundPoint IP 450 phones now show the complete text on the microbrowser screen.
- 71071** The inbound caller ID on the SoundPoint IP 33x now displays for new calls during an active call on the phone. Observed this in version 3.2.5 and 3.3.1
- 71328** Modifying the phone username through phone Web user interface no longer changes the password from numeric values to ???? (four question marks).
- 71947** When the phone lock feature is enabled on the SoundPoint IP 650, outbound calls can no longer be dialed using line key, handset, headset and speakerphone key.
- 72766** When the configuration parameter `device.set` is set on the SoundPoint IP 650/670 with a new boot server IP address, it now forces the phone to reboot.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the UC software version 3.3.2 configuration file parameters.

Table 14: UC Software 3.3.2 Parameter Enhancements

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
call.clickToDial.referBased. userConfirm	added	accessType		Admin
		callback		DefaultCbNone
		cfgParamType		param
		default		0
		help		
		templates		sip-interop
		type		Bool
call.shared.notifyTransferHold AsActive	added	accessType		Admin
		callback		DefaultCbRestart
		cfgParamType		param
		default		0
		help		
		max		1
		min		0
		templates		hidden
		type		Bool
device.prov.ztpEnabled	added	callback		DefaultCbRestart
		cfgParamType		param
		default		
		help		
		max		256
		min		0

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
		templates		device, site, new
		type		String
device.prov.ztpEnabled.set	added	callback		DefaultCbRest art
		cfgParamType		param
		default		0
		help		
		templates		device, site, new
		type		Bool
dialplan.x.routing.server.y.transport	change d	enum		transport
		max	256	
		min	0	
		type	String	Enum
dialplan.routing.server.x.transport	change d	enum		transport
		max	256	
		min	0	
		type	String	Enum
dir.H350.dev.transport	change d	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	
		type	String	Enum
dir.H350.group.transport	change d	default		TCP
		enum		dirCorpTransport

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
		max	256	
		min	0	
		type	String	Enum
dir.H350.person.transport	changed	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	
		type	String	Enum
dir.corp.transport	changed	default		TCP
		enum		dirCorpTransport
		max	256	
		min	0	
		type	String	Enum
mb.main.toolbar.autoHide.enabled	added	accessType		User, Admin
		callback		CbMicroBrowser
		cfgParamType		param
		default		1
		help		Enable/Disable browser toolbar auto hide feature
		templates		applications
		type		Bool
pnet.joinOnAutoAnswer	added	accessType		Admin
		callback		DefaultCbNone
		cfgParamType		param

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
		default		0
		help		
		templates		video-integration
		type		Bool
reg.x.outboundProxy.failOver. onlySignalWithRegistered	added	callback		CbReg
		cfgParamType		paramArray
		defaultAll		1
		help		
		numReplace1 Max		Const_NumLineReg
		templates		reg-advanced
		type		Bool
reg.x.outboundProxy.transport	changed	enum		transport
		max	256	
		min	0	
		type	String	Enum
reg.x.server.y.failOver.onlySignal WithRegistered	added	callback		CbReg
		cfgParamType		param2DArray
		defaultAll		1
		help		
		numReplace1 Max		Const_NumLineReg
		numReplace2 Max		Const_NumServers
		templates		site
reg.x.server.y.transport		type		Bool
	changed	enum		transport
		max	128	

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
sec.srtp.newContextOnResume	added	min	0	
		type	String	Enum
		callback		DefaultCbRestart
		cfgParamType		param
		default		1
		help		
		templates		hidden
up.simplifiedSipCallInfo	added	type		Bool
		accessType		Admin
		callback		DefaultCbNone
		cfgParamType		param
		default		0
		help		
		templates		new
voIpProt.SIP.dialog.strictVersionValidation	added	type		Bool
		accessType		Admin
		callback		CbSipStack
		cfgParamType		param
		default		1
		help		
		templates		new
voIpProt.SIP.failoverOn503Response	added	type		Bool
		callback		CbSipStack
		cfgParamType		param
		default		1
		help		
		templates		sip-interop

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
		type		Bool
voIpProt.SIP.outboundProxy.failOver. onlySignalWithRegistered	added	callback		CbReg
		cfgParamType		param
		default		1
		help		
		templates		sip-interop
		type		Bool
voIpProt.SIP.outboundProxy.transport	changed	enum		transport
		max	256	
		min	0	
		type	String	Enum
voIpProt.SIP.serverFeatureControl. missedCalls	added	accessType		Admin
		callback		DefaultCbRest art
		cfgParamType		param
		default		0
		help		
		templates		sip-interop
voIpProt.server.x.failOver.onlySignal WithRegistered	added	type		Bool
		callback		CbReg
		cfgParamType		paramArray
		defaultAll		1
		help		
		numReplace1 Max		Const_NumSe rvers
voIpProt.server.x.transport	changed	templates		sip-interop
		type		Bool
		enum		transport

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
		max	256	
		min	0	
		type	String	Enum
voice.gain.rx.digital.headset.IP _330	added	callback		DefaultCbRest art
		cfgParamType		param
		default		-12
		help		
		max		1000
		min		-1000
		templates		techsupport
		type		SInt
voice.gain.rx.digital.headset.IP _335	added	callback		DefaultCbRest art
		cfgParamType		param
		default		-9
		help		
		max		1000
		min		-1000
		templates		techsupport
		type		SInt
voice.gain.vol.ringer.gain.adjust	added	callback		DefaultCbRest art
		cfgParamType		param
		default		-9
		help		
		max		1000
		min		-1000
		templates		new

<i>Configuration Parameter</i>	<i>Action</i>	<i>Property</i>	<i>Old Value</i>	<i>New Value</i>
		type		SIInt
voice.headset.rxag.adjust.IP _335	added	callback		DefaultCbRestart
		cfgParamType		param
		default		-11
		help		
		max		-11
		min		-1000
		templates		new
		type		SIInt

Understanding Updates to UC Software 3.3.1F

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

Enhanced Capabilities

66743 The phones may be vulnerable to Denial of Service attacks when used in certain configurations. Sending HTTP GET requests with a broken authorization header can produce a device restart under certain circumstances in certain models of phones. For full details, refer to [Technical Bulletin 66743](#).

Understanding Updates to UC Software 3.3.1

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

50065 Added support to the VVX 1500 for CMA presence.

52476 Added support for Premium extensions to server synchronized ACD feature.

- 55061** Added support for the Team Function feature. This feature extends the compatibility of statically configured Busy Lamp Field (BLF) to operate in a system requires the use of two URIs, one for call operations and another one to subscribe for notification of dialog events. It also provides Ringing Indication and a Directed call pick-up capability in a system that does not generate RFC 4235 compliant dialog-info+xml documents.
- 58888** Added the ability to trigger a reboot (or configuration update) from the microbrowser. For example, `<softkey index=3 label=Reboot action=Action:UpdateConfig />`
- 59000** The phones now ignore BLA dialog documents (through NOTIFY) that are reflected to User Agents that are party to the dialog.
- 60306** The server certificate Serial Number SN is now verified against the server/proxy's A record domain names if the SRV record domain does not match the SN.
- 61343** The phones now provide a configurable parameter that allows the verification of the authentication tag to be disabled for received SRTP packets. The purpose of this is to allow system administrators to resolve defects in other endpoints where the authentication tag is not computed correctly. Supported parameter: `sec.srtp.noAuthRxRTP`
- 61389** During the 802.1x - EAPOL Logoff, the phone recycles the LAN link (for example, it will bring it down and up in an interval of one second) upon detecting a PC link down event. This forces the 802.1X switch to refresh the authorized port state and start sending requests for identity challenge messages. The associated configuration parameter is `sec.dot1x.eapollogoff.pcforgelanlinkreset` with values 0 - never recycle LAN link and 1 - the phone unconditionally recycles the LAN link upon detecting PC link down event.
- 61861** Corporate Directory LDAP initialization supports the bind authentication.
- 62115** The phones now display the full text strings of the Phone Lock feature (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 62259** The phones now display the Call Forward destination on Idle Display.
- 62775** The toolbar slide-out option is now configurable on the VVX 1500. The associated configuration parameter is `mb.main.toolbar.autohide.feature`
- 1 feature is enabled (default)
 - 0 feature is disabled. The Autohide enable/disable buttons are no longer visible to the user in the toolbar.

Enhanced Capabilities

- 44337** Configured characters ;, /, ?, &, =, ~, %, \ are now escaped in INVITE messages.
- 55794** The SoundStation IP 6000 and 7000 phones no longer reboot upon receiving a call with incorrect SRTCP indices.
- 56491** As of SIP 3.2.x, the screen no longer displays the IP address of the server when disabling the Call Forwarding feature using a # code.
- 59824** The phone now changes all of the menu option labels into the selected language.
- 59843** On the VVX 1500, the caller ID correctly displays during an active call after switching (exchanging) valid logon credentials between two phones.

- 60015** The phone no longer sends RTP media for 2.4 seconds after call is declined with 603 Decline.
- 60175** When using the Contact Directory speed dial, the left and right arrow keys no longer increment and decrement the index unexpectedly (applies to *SoundPoint IP 320, 321, 330, 331, and 335*).
- 60514** The user password can now be changed by an administrator if the old password is unknown.
- 60572** An EFK soft key no longer requires at least one valid entry in `<efk.efklist />` configuration to be enabled.
- 60645** The VVX 1500 phone no longer resets to previous values in the Edit contact menu when the mode is changed from Tel to Url.
- 60761** The Transfer and Conference soft keys on the SoundStation IP 5000 and 6000 are no longer absent upon the 8th active outgoing call.
- 60788** When operating with a sip X server, there is now music on resume from a double Music On Hold (MOH) between two phones.
- 60814** The Login soft key on the SoundStation IP 7000 now displays when `feature.acdLoginLogout.enabled` is set.
- 60831** Ringback tone no longer continues to play for 30 seconds after the SoundStation IP 6000 phone sends a BYE message.
- 60848** After invoking the Update Configuration menu option, the phone now returns to the idle screen.
- 60897** The Custom Ringer Types menu no longer uses the file name rather than configured name.
- 61030** The Buddy Watch presence now works on the VVX 1500 phone after it boots initially with `voIpProt.H323.enable=1`.
- 61031** Active call now has a timer when attempting to transfer or conference the call (*applies to SoundPoint IP 450, 550, 560, 650, 670*).
- 61041** The Call Server Configuration Menu now displays Options (1, 2, ...) within the Menu items (*applies to the VVX 1500*).
- 61042** The Directed call pick-up feature now works when the SUBSCRIBE message expires=0.
- 61046** The Saved Certificate prompt is now shown when a new CA certificate is downloaded (*applies to SoundPoint IP 320, 321, 330, 331, 335, and SoundStation 5000 and 7000*).
- 61088** The VVX 1500 phone no longer freezes and reboots after making a call with `tcpIpApp.port.rtp.forceSend=1024`.
- 61090** The configuration parameter `voIpProt.SIP.musicOnHold.uri` is now updated after a configuration change.
- 61095** While dialing a URL using the on-screen keyboard on the VVX 1500, the first entered character is no longer unexpectedly deleted.
- 61102** The Handset or Speaker icon no longer appears instead of the Ringer icon when you adjust the ringer volume while the SoundStation 500 and 6000 phones are idle.
- 61104** With a shared line configured on the VVX 1500 phone, activity on the remote shared line no longer causes the idle browser content to cycle off then on.
- 61114** The VVX 1500 phone boots-up with the DHCP VLAN 256 DVD option.

- 61115** You can now answer calls after a configuration update is invoked.
- 61242** The configuration parameter `voIpProt.SIP.useCompleteUriForRetrieve` updates after a configuration change.
- 61246** The `voIpProt.SIP.allowTransferOnProceeding` XML schema lists the correct type as stated in the Administrators' Guide.
- 61273** Joining calls into a local conference when 1 leg is a remotely held BLA line now maintains audio between both remote users.
- 61314** The number of characters for custom names has been extended to 127 from 12.
- 61367** When dialing a number with a + sign (for example, +492101099210) `user=phone` is now added to the To header.
- 61677** The VVX 1500 phone no longer escapes the % character as %25 when is present in the destination of a call.
- 61723** The VVX 1500 phone is no longer missing the first string `<?xml version=1.0 encoding=utf-8 ?>` in FAST UPDATE requests.
- 61779** The phone no longer reboots spontaneously from the idle state or in-use state.
- 61904** A call is placed with the correct signaling protocol on the VVX 1500 when the line is configured as dual line protocol.
- 62036** The SoundPoint IP 320 and 330 phones now send DTMF RTP EVENTS when receiving a second incoming call during an active primary call.
- 62114** The user can now unlock the VVX 1500 phone after the phone is locked with a password containing letters.
- 62325** Chinese characters no longer cause the VVX 1500 phone to become unresponsive to user requests.
- 62333** The correct Chinese characters display in the reboot menu on the VVX 1500.
- 62417** When an off-hook event is received from the headset base station, the phone no longer sends three events to the base station in DHSG headsets and platforms.
- 62453** The phone no longer displays a ghost call appearance labeled Unknown Party if a remote party is held while the reorder tone is played locally.
- 62490** Enabling the Screen capture function with `httpd.enabled=0` no longer causes the phone to freeze and reboot.
- 62576** The phone now reboots to pick up new **sip.ld** file after an Update Configuration is invoked from the menu.
- 62642** The phone no longer plays a dial tone and RTP media when resuming on a call held at another phone.
- 62704** BLA presence now recovers correctly on the monitoring phone when the LAN cable is disconnected and then re-connected.
- 62906** The phone correctly provisions using the HTTPS protocol option when using a server certificate with an older MD2 digest message algorithm.
- 63076** The phones with BLA lines are now able to establish more than 10 outgoing calls.

63214 The phone no longer reboots if it receives more REFERS that `reg.x.callsPerLineKey` is configured for.

64742 Using a dial plan containing #, when a user goes off-hook and dials #1#2#, the phone now sends out an invite message containing %231%232%23.

Configuration File Enhancements

Refer to the following table for a list of changes in configuration parameters. The list applies only to the changes made since UC Software 3.3.0.

The configuration files, their respective parameters and defaults, as well as the provisioning methods have been simplified but extensively modified starting from UC Software 3.3.0.

Before installing the software, it is highly recommended that you first familiarize yourself with the changes outlined in the [Administrators Guide for the Polycom® UC Software – 3.3.0](#) and [Technical Bulletin 60519](#).

Table 15: UC Software 3.3.1 Parameter Enhancements

<i>Configuration Parameter</i>	<i>Action</i>
<code>attendant.resourceList.x.callAddress</code>	added
<code>attendant.resourceList.x.proceedingIsRecipient</code>	added
<code>device.cma.disableTlsForDebug</code>	added
<code>device.cma.disableTlsForDebug.set</code>	added
<code>dir.H350.dev.bindOnInit</code>	added
<code>dir.H350.group.bindOnInit</code>	added
<code>dir.H350.person.bindOnInit</code>	added
<code>dir.corp.bindOnInit</code>	added
<code>feature.acdPremiumUnavailability.enabled</code>	added
<code>reg.x.filterReflectedBlaDialogs</code>	added
<code>sec.dot1x.eapollogoff.pcforgelanlinkreset</code>	added
<code>sec.srtp.noAuthRxRTP</code>	added
<code>voIpProt.SIP.allowTransferOnProceeding</code>	changed

Understanding Updates to UC Software 3.3.0

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 23335** Configuration parameter values can now be updated at run-time.
- 24111** Enhanced the user interface for selecting a distinctive ringtone associated with a contact in the local directory. You can now review the ring name and play the ringtone before accepting and associating the ringtone for specific contacts.
- 23394** Configuring parameters are now self-contained (default parameter values) and the configuration process is more fault-tolerant.
- 35245** Line key behavior (configurable) has changed so that keys can now be used to hang-up/terminate calls as well as establishing calls. The associated configuration parameter is `up.lineKeyCallTerminatetype=Booldefault=0 min=0 max=1`.
- 38826** Added configuration parameters to expand the range of ports as well as to randomize port selection for the purpose of downloading configuration files to the phone through TCP connections.
- 48138** Added support for dynamic support of G.729AB and iLBC codecs. G.729AB / iLBC (*applies to SoundPoint IP 320, 321, 330, 331, 335, 450, 550, 560, 650, 670*).
- 48526** Simplified selection of codec configuration preferences. See [Technical Bulletin 60519](#) for more information. This change is not backward compatible to configuration files used with previous software releases.
- 48690** Added the ability for users to lock the phone and restrict its access from unauthorized users. Users must enter a PIN to access and use the phone. Refer to [Quick Tip 57215](#) for more information regarding this feature and configuration.
- 49658** Added configuration parameter to allow the phone to obtain Caller ID from the From header instead of the P-Asserted-Identity segment. The associated configuration parameter is `voIpProt.SIP.CID.sourcePreference = P-Asserted-Identity, Remote-Party-ID, or From`.
- 50067** Local contact directory now matches the Polycom CMA products style and user experience.
- 50151** Removed redundant levels of abstraction associated with arrays in configuration files.
- 50644** Enhanced the visual indicator of incoming calls on the VVX 1500 for the hearing impaired. Upon receiving an incoming call, the phone rings and the display flashes on and off with a bright orange and white screen. This visual indicator is visible even when viewing the display at an indirect angle. The associated configuration parameter is `up.accessibilityFeatures=1`.
- 51121** RAM disk configuration parameters have been optimized.
- 51314** Added a configuration option to allow for minimal latency to meet JITC requirements. The associated configuration parameter is `voice.txPacketDelay`. Normal or NULL (default) = no change to Tx latency; low = low delay
- 51523** Added the ability to scroll horizontally caller ID information (if it is truncated when the number of characters in the caller ID string exceeds the capacity of the display).
- 51446** Added configuration parameters supporting TLS cipher suites.
- 51594** Digit map replacements no longer need to be reflected in the placed calls list.
- 51725** Added support for G.719 audio codec in H.323 calls (*applies to VVX 1500*).

- 51979** Added support for asymmetric audio codecs.
- 52253** Configuration parameter values modified by an administrator logon credential using the phones Web server are not permitted to be altered by user level logon credentials.
- 52493** Added support for MD4 encryption key (OpenSSL).
- 52532** The phones no longer invoke a reboot during the uploading of override files as a result of an unresponsive provisioning server (after a timeout).
- 52864** Enhanced the user experience of confirming a Local Directory Search (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 53021** Added support for NTLM version 2 authentication [using XMPP, LDAP and HTTP(s)] for use with CMA.
- 53023** Edit fields have been expanded to display additional content (*applies to VVX 1500*).
- 53231** Added a configuration parameter to control the behavior of terminating a 3-way conference by the conference initiator. Options now include either terminating all conference legs or allowing the other parties to stay connected.
- 53417** Implemented a slider bar on the VVX 1500 for adjusting levels in various menu screens.
- 53703** Added the ability for phones to send an 802.1x EAPOL Logoff message on behalf of an attached PC when the PC is disconnected from the data port.
- 53932** Presence and BLF is supported on Avaya CS2100 soft switches.
- 54037** Attempting a Transfer/Conf of a held party while in active call is now consistent with all phones.
- 54045** Registration parameters can now be modified and activated without requiring the phone to restart or reboot.
- 54098** Added the ability to automatically upgrade the BootBlock section of the BootROM.
- 54167** The BootROM and application software versions may now be obtained by using the on-board Web interface.
- 54301** A timestamp displays in Call Lists alongside the Caller ID.
- 54308** The navigation keys can now be used as a spin box control (ability to select values using the up and down arrow keys) for numeric fields (*applies to SoundPoint IP 320, 321, 330, 331, 335*).
- 54678** The phones can now be deployed with a pre-set language. This supports out-of-the box localization.
- 54928** Added a new API Telephony Event (XML) which is sent to the attached application upon a successful line registration with a PBX.
- 55028** The maximum size of the contact directory contact field has been increased to 128 to accommodate complex dialing scenarios.
- 55040/57981** Added the ability for administrators to install custom device certificates. The administrator can add private and public keys (certificate) through TLS links.
- 55068** Added support for Null Ciphers to be used with TLS Authentication.
- 55318** The Advanced LDAP Search screen now supports languages other than English.
- 55334** Added the ability for the toolbar to hide automatically on the VVX 1500.

- 55490** The configuration Web interface has been expanded to include parameters associated with security.
- 55508** When a precedence call is offered to the phone, it now rings with a corresponding precedence ringtone.
- 55509** When a precedence outgoing call is initiated, a precedence style ring-back tone is generated.
- 55510** The DSCP Differentiated Services Code Point levels for standard and precedence level calls are aligned.
- 55513** The current precedence level of a call now displays.
- 55546** The following diacritic letters and ligature are now supported (language option selection) and display without having to change the character encoding scheme: ä, ö, ü / Ä, Ö, Ü ß.
- 55745** The phones now generate a MLPP resource-priority Header based on the dialed number.
- 55985** The SoundPoint IP 7000 now displays the LogOut soft key when configured to be enabled.
- 56666/56668** Added dynamic codec switching.
- 56790** Enhanced the computation of jitter buffer parameters based on received Quality of Service QoS and expected payload size values.
- 56944** Enhanced the ability for application developers to implement changes to the phone's configuration. Configuration parameters can be modified through the Web interface. The Enhanced method also eliminates the need to reboot the phone to register the changes.
- 57504** A new Warble.wav file can be configured as an audible ringer for incoming calls. This file generates a loud ringer tone for phones deployed in areas with a high ambient noise background.
- 58103** The default maximum call data rate has been increased to 768 Kbps from 512 on the VVX 1500.
- 58156** The user video call rate setting parameter value options have been shortened on the VVX 1500.
- 58758** Enhanced the rendering performance of the browser on the VVX 1500.
- 58764** Added the ability of uploading configuration files representing the phones current set of configured parameter values to the provisioning server.
- 59307** Added a diagnostic menu option that enables the display of configuration file statistics.
- 60316** Added an option in the user interface that allows the user to invoke the phone to force it to re-configure itself based on newly administered configuration file parameter values.
- 60353** Custom ring classes (`se.rt`) can now be set to a maximum value of 17.
- 60363** Custom ringer chords (`tone.chord.ringer.spareX`) can now be set to a maximum value of 19.

Discontinued Features

- 50200/53590** Removed configuration parameters that are no longer required for custom bit-mapped graphic indicator icons.
- 56209** Removed support for the SoundPoint IP 430.
- 59917** Removed support for the animated idle display images (static idle display images are still supported).

Enhanced Capabilities

- 33425** On the SoundStation IP 7000, users can now reply to instant messages.
- 42509** Can now invoke the speed dial list using the Up Arrow key when first call is kept on hold (*applies to VVX 1500*).
- 43660** URL addresses on the SoundPoint IP 330 are now saved in the call list entry. When the phone receives a URL call from SPIP @xxx.xxx.xxx.xxx, the phone now saves the incoming URL call address into call list entry.
- 44034** On the SoundPoint IP 330, the cursor now blinks in hot dial prompt.
- 44278** The phone number now displays correctly on a line key when the number of digits exceeds 10.
- 44478** Configurable soft key features now work on the VVX 1500.
- 44889** The Polycom bitmapped logo now displays on the SoundPoint IP 330 idle screen.
- 45013** The phones no longer reboot after a check-sync request when a call is held and a new call is initiated and then cancelled.
- 47135** Casing of current encoding indication at title bar now match corresponding soft keys on the VVX 1500.
- 47542** The URL entry field on the VVX 1500 allows for 32 instead of only 28 characters.
- 48228** The Contact Directory now has a functional <New Entry> option and a correct Navigation Cluster Guide NCG while dialing (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 48257** The default background image on the VVX 1500 now displays after the following sequence of events: select an image file, followed by selecting an invalid image (file not found) and select the default background image.
- 48463** Can now view JPEG images with file extensions .jpe or .jif (*applies to VVX 1500*).
- 48701** The touch-screen no longer disables during keypad diagnostics (*applies to VVX 1500*).
- 48776** Scrolling in the Ethernet menu no longer causes the selected highlighted item to be positioned at the bottom of the screen (*applies to VVX 1500*).
- 48840** Pressing the Slower and Faster soft keys no longer causes the update cursor to advance immediately.
- 49331** Audio is no longer lost when disabling the hands-free mode while on a speakerphone call (*applies to VVX 1500*).
- 50812** Changes to configuration options are no longer lost without warning if you exit from the Settings menu without passing through confirmation dialog.
- 50855** An error message is no longer shown when a contact is saved with an empty contact number (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 50920** The phones display a correct contact upon pressing the speed dial line key while editing the contact entry (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 50922** The dial plan now applies after editing a call list item and attempting to dial the number.
- 50969** The language displayed for a Missed Call notification now changes when the option is changed to another language setting (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

-
- 50997** Upon pressing a line key, the phone now dials the stored hot dial number (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 51152** The back arrow now works as backspace when pressed in the Display and Touch Screen Diagnostics or Media Statistics screens (*applies to VVX 1500*).
- 51237** In the Server Menu, the Server Password option no longer accepts digits instead of characters as the default (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 51656** The interactive microbrowser now times out if `mb.main.idleTimeout > 600`.
- 51664** The VVX 1500 phone no longer enters LCD Power-down mode in 3 to 4 minutes instead of the time set by `powerSaving.userDetectionSensitivity.officeHours=0`.
- 51669** After both SIP and H.323 Call Server parameters in Admin Settings are reconfigured on the VVX 1500, only one dialog method is now offered to exit. A reboot is no longer required.
- 51947** You can now delete the URL on the VVX 1500 by selecting it right-to-left and pressing the backspace key.
- 51993** Cancelling the deletion of a contact no longer appends an ellipsis to that contacts entry in the list (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 52212** The phone now restarts while another extension on a shared line is in use. The phone no longer thinks it is active on a call.
- 52374** Options in the Forwarding menu are no longer appended with an ellipsis after returning from the selected option.
- 52438** Typing in a fully filled field no longer prevent the cursor from advancing and overwriting existing content (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 52447** After placing 21 encrypted calls on hold, the VVX 1500 phone no longer locks up and reboots on the 22nd multiple encrypted call.
- 52590** The Add Video soft key is no longer accessible when flashing the POTS line to make a second POTS call. While playing dial tone for second POTS call, pressing the Add Video soft key and dialing a video number no longer causes the HDX system to lock-up and reboot (*applies to SoundStation IP 7000*).
- 52629** The phones accept tel URIs as `tel://number` and `tel:number`.
- 52655** Upon disabling the directory, saving a contact from the corporate directory to the directory file no longer causes the saved contact to reuse the speed dial index starting from 1.
- 52690** The Add Phone soft key no longer appears while the Call Type is set to Conference-SIP and the phone is rebooted without a network connection (*applies to SoundStation IP 7000*).
- 52772** In the Corporate Directory, when `sortControl=1`, a quick search on multiple searchable attributes no longer causes the entry list to display items that are not starting from the beginning.
- 52851** Cancelling the Directory Search configuration change no longer appends an ellipsis to menu item label (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 52895** Enabling the Call Forwarding feature without entering a contact number no longer causes it to fail (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 52896** The Forwarding status field in the Forward menu option screen now correctly corresponds to the actual call forwarding status (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

- 52945** When performing a quick search in the Corporate Directory, the Select/Submit indicator in the Navigation Cluster Guide displays correctly.
- 53066** When hot dialing on the VVX 1500, the white screen no longer flashes after pressing the Dial soft key, termination key, and dual line key.
- 53104** When an attempt to change the language option fails, the list of available language options are now sorted correctly.
- 53447** Initiating a URL based hot-dial by pressing the # or * key on the VVX 1500 no longer causes an invalid character to be inserted in the SIP URL.
- 53679** The Back soft key on the VVX 1500 is no longer always present in the APP menu screen.
- 53953** In the LDAP feature on the VVX 1500, the Scroll icons for navigating up and down pages now display when the last contact in the search list is reached.
- 54131** The SoundStation IP 5000 phone now displays the Volume control while the ringer volume is being adjusted in Quick Search mode.
- 54175** The Swedish Group soft key is no longer truncated; the visible portion translates correctly.
- 54219** The phones can now establish a link when connected to some switches when both phone and switch are configured for 100Mbps/Full Duplex (*applies to SoundPoint IP 560 670*).
- 54292** On the VVX 1500, the information in the Line status menu accurately reflects the Gatekeeper address in use by the phone.
- 54343** Added the ability to save changes to text or IP entry fields while in the Admin Settings menu after viewing the Web browser on the VVX 1500.
- 54356** The Delete key on the VVX 1500 does not dismiss the character selection control or prevents character entry in the browser.
- 54614** The VVX 1500, upon originating a conference call, no longer shows a blank black screen instead of the No Video - crossed out camera image while the call is on hold.
- 54616** When DND is enabled on the VVX 1500 on both SIP and H.323 lines, a SIP call no longer generates a busy tone and an H.323 call no longer generates a re-order tone.
- 54617** While listening to a fast busy tone, if an incoming call is offered, the speaker LED is no longer turned off even though the fast busy tone is still present.
- 54638** Opening and closing the web browser on the VVX 1500 no longer resets ABC/abc/123 and encoding soft keys.
- 54656** The phone now displays an x/y indicator when multiple calls are active if the Time and date display is disabled.
- 54720** Placing the handset on-hook no longer unexpectedly closes the Audio Diagnostics menu on the VVX 1500.
- 54727** Invoking the Abc/ASCII entry mode on the VVX 1500 capitalizes entered letters correctly in the Corporate Directory search field.
- 54735** Upon pressing the VIDEO key on the VVX 1500, the focus now changes to the Active Conference pane.
- 54756** The VVX 1500 phone can display the dialing screen when an alpha character is configured and entered into the contact field followed by a call to the specified contact.

- 54757** The Call Timer on the VVX 1500 displays a correct duration value.
- 54834** The VVX 1500 no longer connects with audio only when an MGC IVR Video Welcome Slide is used.
- 54876** The inter-digit DTMF signaling interval now matches the `tone.dtmf.offTime` setting.
- 54949** An unassigned soft key no longer operates as a Dir soft key on the SoundStation IP 7000.
- 54966** The Lin16.16kspcs codec now engages if it is the only supported codec.
- 54988** The user can now make additional changes to the selected item in the Prioritize Background menu after making an initial selection (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).
- 54993** The SoundStation IP 7000 phone no longer displays Enter name instead of Enter URL in the Install Custom CA Cert menu.
- 54995** Pressing the # key while in an idle call state no longer displays the character in dial screen.
- 55001** The Backspace soft key no longer shows at left edge of a dialed SIP URL (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55002** The user is now able to press and hold the Backspace soft key to clear the contents of the dialing fields (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55005** The user Interface no longer becomes corrupted when change the language while hot-dialing (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55014** Soft keys no longer disappear from a shared line when a hold/resume operation is performed on another remote shared line (*applies to SoundStation IP 5000 and 6000*).
- 55017** Auto Reject now functions correctly when the feature is enabled through the Contact Directory when an Alpha-character is present in the CONTACT field (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55039** The Navigation Control Group (NCG) Indicator no longer shows the right-pointing arrow even when there are no calls in the call lists.
- 55053** The Page up arrow now functions correctly when the Server menu is highlighted and the DHCP client is set to disabled.
- 55063** It is no longer possible to select a disabled menu item using the * key, resulting in a non-functional Edit soft key.
- 55094** When taking the phone off-hook and dialing the # key, the # no longer displays in the Enter more digits field (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55129** The Transfer and Conference soft keys on the SoundStation IP 7000 now display on an 8th active call.
- 55139** The user can now view the full date when the phone is configured for Norwegian language Norsk (no-no) (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55271** The “Please Enter a Contact” pop-up message no longer displays unexpectedly when adding/editing contacts.
- 55287** The phone no longer drops the incorrect call if the user selects a held call on the phone UI and then attempts to terminate the active call (for example, by placing the handset on hook).

- 55297** An audio and video conference call appears as a single video call (*applies to the SoundStation IP 7000*).
- 55313** The Routing soft keys on the VVX 1500 now display correctly when the Call Park feature is enabled.
- 55339** Resuming a conference while running the Slide Show application on the VVX 1500 no longer causes the user interface to become dysfunctional.
- 55340** The user can no longer launch a picture frame on the VVX 1500 while a recording is in progress, causing the USB busy icon to disappear.
- 55375** The Outgoing Call control interface on the VVX 1500 now displays when the Speed Dial Contact Enhanced Feature Key macro fails to execute.
- 55423** The correct soft keys and user interface displays on the VVX 1500 after exiting the screen that was previously opened from the icon in the status bar; while hot dialing digits.
- 55457** When the dual protocol line is registered only to the gatekeeper and not to the SIP server on the VVX 1500, this no longer causes hot-dialed SIP URL to call via H323 and dialog options to appear when a hot dial URL call is attempted.
- 55477** SRTP Key renewal now occurs during local conference calls.
- 55478** The DHCP VLAN Discovery (DVD) no longer reports as not active when it is active.
- 55485** The Camera Settings Save soft key on the VVX 1500 no longer loses its context-sensitivity upon second visit to the menu option.
- 55514** Calling into a Video Server on the VVX 1500 no longer causes the phone to connect the audio portion of call but does not establish a video connection.
- 55560** On occasion, the VVX 1500 phone no longer displays an incorrect call duration timer value while on an H.323 call to an RMX-2000.
- 55641** The Y-axis auto-scaling of the Network Load graph on the VVX 1500 is now accurate.
- 55697** The phone no longer rejects calls with 486 if NOTIFY:Alerting is received before the INVITE and `reg.x.lineKeys` and `reg.x.callsPerLineKey` is set to 1.
- 55644** The VVX 1500 now uses the correct routing protocol when dialing an LDAP contact from the on-hook state through termination.
- 55907** On the VVX 1500, typing a or # no longer causes the on-screen keyboard to unexpectedly close and discard any edits.
- 55911** Changing the text entry mode no longer causes the backspace soft key to disappear (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55929** Pressing the down arrow no longer affects a change on the Navigational Cluster Guide (NCG) (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 55964** The VVX 1500 phone no longer seizes the only unregistered share line using the New Call soft key, speaker and headset function key.
- 56046** The default value of the Sound Effect Destination parameter setting is now removed from the override file when a new value is selected from the menu option.

- 56057** The phones no longer de-registered upon receiving a large number of NOTIFY messages for watch buddy enabled contacts.
- 56147** Adding Contacts to the SoundPoint IP 550 and 670 that are longer than 10 characters or numbers are no longer truncated on the idle screen.
- 56156** The abc/ASCII string no longer remains in the title bar on the VVX 1500 after leaving edit mode for a menu item.
- 56161** Emergency numbers matched against `dialplan.routing.emergency.x.value` are now sent to servers listed in `dialplan.routing.emergency.x.server.y`.
- 56168** When adding more than 7 characters and/or digits to a local contact directory entry on the SoundPoint IP 450, the characters and/or numbers no longer overlap on the idle screen, when they should be truncated.
- 56294** The `Dutch_Netherlands` localization now displays the correct default 24 hour time format in SIP 3.2.x.
- 56289** Label text is no longer drawn past the edge of the speed dial label on the display next to the key (*applies to SoundPoint IP 550, 560, 650, 670*).
- 56333** The SoundStation IP 5000 phone no longer reboots automatically when lease time expires after disabling and enabling the DHCP server.
- 56334** The phone no longer plays a short ringtone burst upon switching to initiating a call sequence of transfer, conference initiation, and then cancels.
- 56338** The SoundStation IP 7000 phone no longer reboots when the user presses the Manage soft key during an 8-way MP call plus 1 audio EP conference.
- 56381** The conference phones now accept a DHCP offer that do include the terminating END (0xFF) option (*applies to SoundPoint IP 5000, 6000, 7000*).
- 56401** The Admin password length in the boot menu and Menu > Settings > Advanced menu now match.
- 56678** Local contact directory entries now store up to 32 characters instead of only 31.
- 56708** The SIP URL dialing field accepts up to 32 characters instead of 256.
- 56787** The phone no longer plays a short tone while retrieving a parked call using an incorrect contact.
- 56788** The contact field of the local contact directory now accepts 128 characters instead of only 32 (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 56797** The SoundPoint IP 450 Admin Settings submenus correctly display the titles in a white background box.
- 56809** The configuration parameter `voice.audioProfile.Lin16.48ksps.payloadType` has a default value of 118 instead of 119.
- 56827** The soft keys associated with Conference Remote Pickup NewCall, Transfer, and Conf soft keys on the SoundStation IP 7000 are no longer missing when the conference call is split.
- 56868** Published CDP power values in TB 48152 now match actual measured consumption.
- 57146** The phone no longer freezes and reboots when it receives an INVITE message with special characters in the FROM header and the call is placed on hold.

- 57368** The second contact in the Local Contact Directory on the SoundStation IP 5000 is no longer highlighted when it is selected.
- 57369** The Contact entry in the Local Contact Directory no longer takes a long time to display (*applies to SoundStation IP 6000 and 7000*).
- 57398** The phone no longer displays *Please enter a contact* pop-up message after adding a contact in the local contact directory.
- 57443** The display on the SoundStation IP 6000 and 7000 no longer flickers while making an outgoing call.
- 57597** Using the VVX 1500 phone with an HDX system, the phone now transmits video upon resuming a SIP call.
- 57615** The Autohide feature on the VVX 1500 now functions when the PIN is pressed and the toolbar is sliding out of view.
- 57849** The phone now acquires the correct VLAN using LLDP after a boot up.
- 57863** The phone accepts a DHCP END (0xFF) option in a DHCP INFORM response.
- 57958** In the fail-over feature, while re-registering, a 32-second delay no longer occurs before sending an INVITE to the third server.
- 58023** A call into a 3COM VCX audio conference server when using the VVX 1500 no longer causes the phone to reboot.
- 58172** Hot-dial numbers no longer disappear from the screen if there is an incoming call during the outgoing hot-dialing state (*applies to SoundStation IP 5000 and 6000*).
- 58197** After upgrading from version 3.0.0 to 3.1.3 Rev C, a delay no longer occurs in the audio signal when using the speakerphone to answer a call.
- 58296** The H.323 digit-map no longer routes files when the `reg.1.lineKeys` configuration parameter has a value of greater than 1 and `reg.1` is assigned a SIP number (*applies to VVX 1500*).
- 58362** Initiating a URL hot-dial call by pressing the # or * key on the VVX 1500 no longer causes the Enter URL dialog to pop-up with the # character already inserted into the field, even though the # character is not a valid SIP URL character.
- 58464** A contact can now be saved from a Corporate Directory search result into a local directory.
- 58498** Within the re-registration on fail-over feature, Subscribe now triggers the fail-over. The phone now sends the register request to the second server after received an ICMP from the primary server.
- 58509** Within the Re-registration on failover feature, the phone no longer sends an extra Register request to primary server after the first fail-over.
- 58520** Resolved a uni-directional Video Streaming interoperation issue with a Siemens Video Desktop Client ODC (*applies to VVX 1500*).
- 58574** The SoundPoint IP 650 phone now validates an existing registration when it is registered with a BroadSoft server.
- 58619** The line no longer becomes unregistered when an invalid name and password is entered from the menu options on the phone. The line becomes unregistered until the phone is rebooted.

-
- 58782** The phone sets the Call Control 802.1Q Priority correctly when using TCP. The value is set correctly when using UDP.
- 58785** The VVX 1500 phone now appends the MAC address to HTTP user agent headers when configured to do so.
- 58787** The VVX 1500 phone no longer reboots immediately after making a call to an RMX when the Camera Target Frame Rate is set to minimum.
- 58874** When using TCP preferred transport, the phone now resends a 200 OK message after answering a call without receiving an ACK.
- 58906** The phone now clears its BLA state table when receiving a NOTIFY message with state = full after a SUBSCRIBE message.
- 58907** The VVX 1500 phone now sends an INVITE SIP packet when the configuration parameter `msg.mwi.1.callBack=voicemail` and the user presses the Messages key.
- 58908** With BootROM 4.2.1.0334, the VVX 1500 phone no longer sends a truncated Option 60 message.
- 58913** The phone no longer reboots when pressing the Messages key while Message Waiting Indicator is disabled. When the phone has more than one registration and `msg.mwi.1.callBackMode=disabled` and `msg.mwi.2.callBackMode=disabled`, the phone no longer freezes when the Messages key is pressed. The phone no longer responds to any further key presses.
- 59129** The Centralized Conference feature no longer fails when a URI is incorrectly assigned to `voIpProt.SIP.conference.address`.
- 59262** A conference notification no longer causes the phone to lock-up and then reboot.
- 59308** A retransmitted INVITE message no longer results in a 400 response.
- 59430** Calls received from a mobile to the VVX 1500 no longer cause the phone to display SIP +86...@. The @ no longer displays.
- 59561** The VVX 1500 phone no longer displays an incorrect time after the configuration parameter `tcpIpApp.sntp.daylightSavings.enable` is set to disabled.
- 59737** The Line Label now displays on the top line of the screen when using the HTML idle display micro-browser page (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 59777** When using NN# speed dial feature, the title no longer displays Directory instead of Speed Dial (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 59949** An idle bitmap graphic no longer displays on the bottom of the screen so that only half of the display is used when `ind.idleDisplay.mode=2` or `3` (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 59954** The phone no longer locks up and reboots when a Re-INVITE message within same dialog is sent to the phone immediately after sending a CANCEL message for the initial INVITE.
- 59967** When an incorrect CA certificate is installed, the phone does not attempt to retry a TLS handshake (*applies to LDAP on the VVX 1500*).
- 60013** The phone no longer locks up and reboots when accessing the contact directory if `dir.local.readonly=1` (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).

- 60126** Gateways no longer reject an INVITE message when `reg.1.csta=1`. The INVITE includes the header *Acceptapplication/sdp/application/csta+xml*.
- 60145** The SoundPoint IP 650 phone now correctly presents 2 BLA call appearances. The second call appearance now correctly indicates a remotely held line, when it is not.
- 60264** When a BLA line is showing the dialing screen, remote call appearances no longer display when the remote BLA line resumes a call (*applies to SoundPoint IP 450, 550, 560, 650, and 670*).
- 60266** When a phone is in dialing screen, if a remote SCA/BLA line holds and resumes, the dialing icon no longer changes between the animation arrow and the termination (speaker) icon. The termination icon displays continuously and no longer changes (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 60267** You can now change a checked item twice in the Prioritize Background menu (*applies to SoundPoint IP 550, 560, 650, and 670*).
- 60340** The Join soft key on the SoundPoint IP 650 no longer displays on a phone with a BLA line when there is only one call on the phone.
- 60650** The idle browser on the VVX 1500 no longer alternates between current content and earlier content when it the display is refreshed.
- 62621** SoundPoint IP 321 and 331 phones running SIP 3.2.3.3122 and configured for HTTPS no longer display the error messages *Alert: Fatal, Description, Decode Error*.

Configuration File Enhancements

Note that the configuration files, their respective parameters and defaults, as well as the provisioning methods have been simplified but extensively modified compared to previous releases.

Before installing the software, it is highly recommended that you first familiarize yourself with the changes outlined in the [Administrators Guide for the Polycom® UC Software – 3.3.0](#) and [Technical Bulletin 60519: Simplified Configuration Enhancements in Polycom® UC Software 3.3.0](#).

Understanding Updates to SIP 3.2.5

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 59000** The phones now ignore BLA dialog documents sent within NOTIFY messages that are reflected to User Agents that are party to the dialog.
- 62939** Various enhancements to the Geo-Redundancy (multiple server fail-over support) feature. For full details, refer to the list of documents in [Reference Documents](#).
- 64359** Bridged Line Appearance BLA line dialog rendering is now converted from No to Yes on User Agents that are a remote party to the dialog.

Enhanced Capabilities

- 54219** The SoundPoint IP 560 and 670 phones now establish a data link when connected to some switches when both phone and switch are configured for 100Mbps/Full Duplex.
- 57570** A fail-over is now performed as a result of a SIP Response code 503.
- 60851** Dialing through the Speaker or Headset key no longer drops the initial call appearance.
- 60973** Entering a username and password using the Quick Setup (QSetup) soft key followed by a request to save, now automatically invokes the phone to reboot the phone to the changes to be applied.
- 61248** After configuring a phone with 3 line registrations, while the second line is on hold, if a user hot-dials using the speaker/headset termination key, the phone no longer seizes line 3 to dial out.
- 61283** The phone no longer sends a NOTIFY with `<param pname=+sip.rendering pvalue=no />` when a user attempts to place a conference call on hold and the phone receives a 400 Bad request.
- 61541** When a user attempts to place a conference on hold and the phone receives a 400 Bad request, the phone correctly sends a NOTIFY with `I=no`. This no longer causes the incorrect presence to display on the other Bridged Line Appearance line.
- 62206** The phone no longer displays Service Unavailable upon lifting the handset and pressing the Line 2 key (*applies to SoundPoint IP 320, 321, 330, 331, and 335*).
- 62226** The phones no longer join a conference after receiving a 403 Forbidden from the switch.
- 62383** A held call on a Bridged Line Appearance with remote phones is now presented (*applies to SoundPoint IP 601*).
- 62567** SoundPoint IP 3xx phones monitoring each other in a 2x2 BLA configuration are now able to pick up held calls.
- 62621** SoundPoint IP 3xx phones configured for HTTPS no longer display the error messages *Alert: Fatal, Description, Decode Error*.
- 62642** The phones no longer play a dial tone and RTP audio when resuming a call held at another phone.
- 62643** When the user presses both line keys (Line 1-hold and Line 2-Active call) simultaneously on the SoundPoint IP 3xx, the active call on Line 2 is no longer dropped.
- 62669** Multiple phones no longer try to resume a held Bridge Line Appearance BLA line at the same time.
- 62672** Directed Call Pickup DCP or Group Call Pickup feature (using soft keys instead of *53 and *54 feature access codes) no longer fail when the user enters an account code. The account code is not appended to the user portion of the URI.
- 62855** Invoking either the Group Call Pickup or Directed Call Pickup feature, using its corresponding soft key, now functions correctly. The display shows Unknown and the call is not picked up (*applies to SoundPoint IP 3xx*).
- 62902** The phone now accepts inbound SIP requests from an RROFO (Geo-redundancy) server that is not registered with that phone.

- 62926** The Resume soft key on the SoundPoint IP 3xx is now presented when the line key is pressed continuously while the line is in a remote held call state. This occurs when the line is configured as `callsPerLineKey=1`.
- 63099** The phones monitoring Bridged Line Appearance BLA line, configured for one call per line, can now pick up the held call after the call on a BLA line has been put on hold using the Transfer/Conference key.
- 63280** Regarding Geo-redundancy RROFO, calls on hold are now released when pressing the Resume soft key after the IP BE fail-over occurs while using the geo-redundancy feature. The user no longer needs to press the End Call soft key to complete the intended result.
- 63388** If a phone's SIP lines are not registered with a call server, and the Emergency Call Routing Feature is enabled (by configuring the `dialplan.routing.emergency.x.value` and `dialplan.routing.emergency.x.server.y` parameters) dialing the configured emergency number now works when you use on-hook dialing and enable URL dialing.
- 63536** The Redial feature functions correctly after invoking an outgoing call accompanied with an account code.
- 63631** The counting down aspect of the Geo-redundancy RROFO-DNSTTL feature no longer fails during fail-back. The Time-To-Live TTL timer should be reset after re-registering to the secondary server.
- 63704** Regarding Geo-redundancy RROFO, the phone no longer sends three extraneous registration requests to the primary proxy server during a fail-over.
- 64093** Regarding Geo-redundancy RROFO, a fail-over using either the Conference or Transfer feature now stops a consultative call when the primary call is terminated.
- 64212** Invoking the Call Park feature on the SoundPoint IP 3xx with the soft key now functions correctly when the soft key is configured as 1 line and 1 call per line.
- 64219** The SoundPoint IP 3xx phone sends a proper hold NOTIFY message after a consultative transfer is canceled when the configuration parameter `notifyTransferHoldAsActive` is disabled.
- 64274** In an attempt to resume a held call, the held call is no longer terminated when the user accidentally seizes two line keys simultaneously.
- 64327** In an attempt to answer an incoming call, the user can no longer inadvertently press two line keys. The user is now not connected to both lines, that include an incoming caller and a dial tone.
- 64340** The indicator, on a Bridged Line Appearance BLA line that is monitoring other lines, no longer remains on continuously after the monitored phone performs the following sequence transfer > split > endcall > resume > hold.
- 64356** The display on the SoundPoint IP 3xx showing a remote call appearance now times out correctly when the user presses continuously a BLA line key followed by pressing a down arrow key while there are multiple calls on hold on the remote BLA.
- 64360** The state of the indicator of a BLA line appearance is now correctly reported after the phone receives an INVITE containing replaces.
- 64762** When special characters in the FROM field are received, they no longer prevent the SoundPoint IP 430 phone from displaying Caller ID information.
- 64862** Joining an internal extension with an external PSTN call no longer causes one call to drop.

- 65119** When a Bridged Line Appearance BLA line is presented in a dialing screen, the remote call appearance now displays when the remote BLA line resumes a call.
- 65207** A slow memory leak no longer occurs in the SIP stack due to the receipt of hunt group INVITE containing replaces with phones using ADTRAN switches.
- 65368** When the configuration parameter `signalWithUnregistered=0`, the phone now always ignores all of the messaging traffic.
- 65842** Call waiting tone no longer continues to play after an inbound call has been forwarded and answered by the PSTN.
- 67178** Centralized conference can now be established when `reg.1.lineKeys` is 5 or greater (*applies to all SoundPoint IP*).

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP 3.2.5 configuration file parameters.

Table 16: Software Version 3.2.5 Parameter Enhancements

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>
phone1	added	<code>reg.n.server.m.failOver.onlySignalWithRegistered</code>	N/A	Null
phone1	added	<code>reg.n.outboundProxy.failOver.onlySignalWithRegistered</code>	N/A	Null
phone1	added	<code>reg.n.filterReflectedBlaDialogs</code>	N/A	Null
sip	added	<code>voIpProt.server.n.failOver.onlySignalWithRegistered</code>	N/A	Null
sip	added	<code>voIpProt.SIP.CID.sourcePreference</code>	N/A	Null
sip	added	<code>voIpProt.SIP.failoverOn503Response</code>	N/A	1
sip	added	<code>voIpProt.SIP.outboundProxy.failOver.onlySignalWithRegistered</code>	N/A	Null
sip	added	<code>call.localConferenceEnabled</code>	N/A	1

Understanding Updates to SIP 3.2.4B

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

Enhanced Capabilities

- 66743** The phones may be vulnerable to Denial of Service attacks when used in certain configurations. Sending HTTP GET requests with a broken authorization header can produce a device restart under certain circumstances in certain models of phones. For full details, refer to [Technical Bulletin 66743](#).

Understanding Updates to SIP 3.2.4

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

Enhanced Capabilities

- 59308** A retransmitted INVITE message causes a 400 Bad Response reply. This is in violation of RFC 3261 section 17.2.1.
- 65207** A consistent but slow memory leak occurs as a result of receiving INVITE messages containing replaces.
- 65435/65725** With reference to IEC 60268-1, the default and maximum values for the headset and headphone audio levels have been adjusted to ensure compliance with the IEC 60268-1 TUV safety requirements (*applies to SoundPoint IP/VVX 1500*).
- 65660** The BootBlock may become corrupted as a result of accessing unprotected section of flash memory.

Understanding Updates to SIP 3.2.3

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 43099** Added support for the SoundStation IP 5000 conference phone.
- 43297** Sound effects can now be played out of a destination based on user configuration. The available destinations are: *chassis*, *handset*, *headset* or *active*. The default is *chassis*.
- 45462** All SoundPoint and SoundStation phones now comply with retry-after instructions embedded in SIP Response codes 500 and 503 as part of REGISTER and other requests.
- 50739** On a multi-leg conference on the SoundStation IP 7000, when the End Call soft key or the On Hook hard key is pressed, the conference phone asks the user if the entire call should terminate. A negative response guides the user to the conference manage menu to allow the user to terminate the individual legs of the call. The dialog only appears for multi-leg conference calls.
- 51753** Enhanced the appearance on the SoundPoint IP 450 of anti-aliased characters.

- 51940** All SIP phones now have a fail-over feature that enables phones to re-register before diverting SIP signaling to an alternate server. This feature will be formally released and documented in a future release.
- 54041** Format of DHCP Option 60 Data is now configurable and added support for Option 125 as per RFC 3925.
- 54983** Internal IP address of the VVX 1500 phone (instead of an alias) is no longer being sent in the Facility Message.
- 55524** Logs no longer display Cannot set 802.1Q VLAN ID for TCP protocol messages to default when running on a VLAN.
- 56272** Network Configuration DHCP submenu now supports Option 60 format. The new options include setting either RFC 3925 Binary [default] or ASCII String.

Enhanced Capabilities

- 45188** The minimum acceptable amount of free RAM has been increased on the SoundPoint IP 320, 330, and 430 so that functions such as ringtones are not affected.
- 47897** The Back soft key works when a user tries to exit from Instant Message menu.
- 52119** VVX 1500 phones no longer reboot during G.729 packet loss concealment such as when the remote phone is placed on hold.
- 52787** The configuration parameter `voIpProt.SIP.requestValidation.x.method=source` does works with DNS SRV Static Cache
- 53473** When the SoundStation IP 7000 is used with an HDX system, the parameter `voice.volume.persist.handsfree=0` also affects the HDX system.
- 54549** Changes in the display color palette on the SoundPoint IP 450 no longer cause contrast problems.
- 54751** SIP INVITE messages can be sent when dialing a number containing the period character.
- 54832** The phone enables user to add more than 32 characters in Hot Dial screen (*applies to VVX 1500, 321, 325, 330, 331, and 335*).
- 54867** In the Contact Directory, the text fields scroll to the left to reveal the first character as you move the text cursor left (*applies to SoundPoint IP 321, 325, 330, 331, and 335*).
- 54908** An unexpected colon has been removed in the scrolling status line during an incoming call (*applies to SoundPoint IP 321, 325, 330, 331, and 335*).
- 55099** In a long SRTP conference, steering video on the VVX 1500 between active and inactive no longer causes the video leg to fail.
- 55120** Dialing numbers in the Contact Directory no longer opens contacts for editing (*applies to SoundPoint IP 550, 560, 650, and 670*).
- 55296** On the VVX 1500, the dial pad widget displays when attempting to conference or transfer a held call while in a ringback state.
- 55378** The VVX 1500 phone can invoke LCD power down mode after a remote end places the call on hold.

- 55415** The phone enables the user to enter more characters than it is capable of saving in the Contact Directory fields.
- 55420** The VVX 1500 phone can play back video after a SIP re-INVITE message is sent to an RMX meeting room.
- 55560** The VVX 1500 phone displays correct call timer values while in an H.323 call to an RMX-2000.
- 55618** Switching to Katakana characters before the character selection widget times out no longer produces random characters that on occasion cause the phone to malfunction (*applies to SoundPoint IP 450, 550, 560, 650, 670; SoundStation 5000 and 7000*).
- 55844** Proceeding outgoing call state on one line is adversely affected by an outgoing call on another line (*applies to SoundPoint IP 321, 325, 330, 331, and 335*).
- 55884** The displays on a SoundPoint IP 650 with expansion modules no longer freeze during a consultative transfer.
- 56032** SoundPoint IP 650 phones with two expansion modules no longer reboot while monitoring continuous BLF traffic.
- 56488** In packets sent from the client, the Parameter Request List option no longer contains two duplicate requests for the options Router (3) and Domain Name (15) (*applies to SoundStation IP 6000 and 7000*).
- 56641** The phone no longer ignores the LLDP broadcast from a switch. It defaults to the data VLAN instead of the voice VLAN. There is a LOSS of LINK during the boot process causing LLDP to fail (*applies to SoundStation IP 6000 and 7000*).
- 56836** After dialing and then adjusting the volume, lifting the handset no longer dials the last hot-dialed number immediately (*applies to SoundPoint IP 550, 560, 650, and 670*).
- 57133** The SoundPoint IP 321, 330, and 331 phones can display a customer supplied logo.
- 57457** The LoudRing.wav audio file has been distributed in release 3.2.2.
- 57796** Invalid Message-Summary Event no longer results in invalid MWI notification.
- 57849** The SoundPoint IP 330 and 550 phones can acquire the correct VLAN via LLDP.
- 58024** The Hold function on the VVX 1500D no longer fails in a specific customer scenario.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP 3.2.3 configuration file parameters.

Table 17: Software Version 3.2.3 Parameter Enhancements

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>New Value</i>
phone1	added	reg.n.server.1.failOver.reRegisterOn	
phone1	added	reg.n.server.1.failOver.failBack.mode	
phone1	added	reg.n.server.1.failOver.failBack.timeout	
phone1	added	reg.n.server.2.failOver.reRegisterOn	

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>New Value</i>
phone1	added	reg.n.server.2.failOver.failRegistrationOn	
phone1	added	reg.n.server.2.failOver.failBack.mode	
phone1	added	reg.n.server.2.failOver.failBack.timeout	
phone1	added	reg.n.outboundProxy.failOver.reRegisterOn	
phone1	added	reg.n.outboundProxy.failOver.failRegistrationOn	
phone1	added	reg.n.outboundProxy.failOver.failBack.mode	
phone1	added	reg.n.outboundProxy.failOver.failBack.timeout	
phone1	added	reg.n.useCompleteUriForRetrieve	1
sip	added	voIpProt.server.1.failOver.reRegisterOn	
sip	added	voIpProt.server.1.failOver.failRegistrationOn	
sip	added	voIpProt.server.1.failOver.failBack.mode	
sip	added	voIpProt.server.1.failOver.failBack.timeout	
sip	added	voIpProt.server.2.failOver.reRegisterOn	
sip	added	voIpProt.server.2.failOver.failRegistrationOn	
sip	added	voIpProt.server.2.failOver.failBack.mode	
sip	added	voIpProt.server.2.failOver.failBack.timeout	
sip	added	voIpPort.SIP.useCompleteUriForRetrieve	1
sip	added	voIpProt.SIP.outboundProxy.failOver.reRegisterOn	
sip	added	voIpProt.SIP.outboundProxy.failOver.failRegistrationOn	
sip	added	voIpProt.SIP.outboundProxy.failOver.failBack.mode	
sip	added	voIpProt.SIP.outboundProxy.failOver.failBack.timeout	
sip	added	voIpProt.H323.blockFacilityOnStartH245	0
sip	added	se.destination	chassis
sip	added	voice.codecPref.IP _5000.G711Mu	2
sip	added	voice.codecPref.IP _5000.G711A	3
sip	added	voice.codecPref.IP _5000.G729AB	4
sip	added	voice.codecPref.IP _5000.G722	1
sip	added	voice.codecPref.IP _5000.iLBC.13_33Kbps	

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>New Value</i>
sip	added	voice.codecPref.IP _5000.iLBC.15_2Kbps	
sip	added	voice.gain.rx.analog.chassis.IP _5000	0
sip	added	voice.gain.rx.analog.ringer.IP _5000	0
sip	added	voice.gain.rx.digital.chassis.IP _5000	11
sip	added	voice.gain.rx.digital.ringer.IP _5000	-12
sip	added	voice.gain.tx.analog.chassis.IP _5000	0
sip	added	voice.gain.tx.digital.chassis.IP _5000	15
sip	added	voice.aes.hf.duplexBalance.IP _5000.0	10
sip	added	voice.aes.hf.duplexBalance.IP _5000.1	9
sip	added	voice.aes.hf.duplexBalance.IP _5000.2	8
sip	added	voice.aes.hf.duplexBalance.IP _5000.3	7
sip	added	voice.aes.hf.duplexBalance.IP _5000.4	6
sip	added	voice.aes.hf.duplexBalance.IP _5000.5	5
sip	added	voice.aes.hf.duplexBalance.IP _5000.6	4
sip	added	voice.aes.hf.duplexBalance.IP _5000.7	3
sip	added	voice.aes.hf.duplexBalance.IP _5000.8	2
sip	added	voice.ns.hf.IP _5000.enable	1
sip	added	voice.ns.hf.IP _5000.signalAttn	-6
sip	added	voice.ns.hf.IP _5000.silenceAttn	-9
sip	added	voice.rxEq.hf.IP _5000.preFilter.enable	1
sip	added	voice.rxEq.hf.IP _5000.postFilter.enable	0
sip	added	voice.txEq.hf.IP _5000.preFilter.enable	0
sip	added	voice.txEq.hf.IP _5000.postFilter.enable	1

Understanding Updates to SIP 3.2.2

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

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

Discontinued Features

- 52522** Removed Launchpad Feature (*applies to VVX 1500*).

Enhanced Capabilities

- 44782** Improved phone UI response when a local conference is active (*applies to VVX 1500*).
- 44980** The phone falls back to configured video codec configuration for Tx video when the incoming signaling lacks codec modifiers (*applies to VVX 1500*).
- 47023** Text font no longer randomly changes (*applies to VVX 1500*).
- 47476** Using the XML API, when the user is inside an XHTML Form Field, the Submit soft key displays correctly.
- 47768** CDP power usage advertisement matches the peak power conditions (*applies to SoundPoint IP 450*).
- 48175** The EFK feature can now establish conference calls (*applies to VVX 1500*).

- 48784** Soft keys are restored after rejecting a call from within the Applications UI context (*applies to VVX 1500*).
- 48857** Recording (R) no longer stops or reboots phone in various high load scenarios such as (a) recording during SRTP conference call, or (b) recording while browsing the application menu during non-SRTP conference call (*applies to VVX 1500*).
- 48921** Digit key presses are no longer missed in certain scenarios (*applies to VVX 1500*).
- 50152** Change non-null sticky primary filter, search (filtered) bar remains on old data (*applies to VVX 1500*).
- 50192** Media Statistics menu displays correctly for several languages (*applies to VVX 1500*).
- 50286** Pressing the page down key # does not move entry list after pressing the page up key * in the quick search menu (*applies to VVX 1500*).
- 50531** The SoundStation IP 7000 phone can startup without network connection when using the PIC cable.
- 50624** The phone sends a 603 Decline message when an inbound call times out.
- 51141** A small number on the left side of the scrolling status bar has been removed.
- 51449** The Out of Dialog Refer-based dialing on the VVX 1500 no longer fails. SDP on INVITE from VVX is missing media attributes, generating a 606 response.
- 51533** Backlight intensity change updates appropriately in Overrides configuration file.
- 51605** VVX 1500 phones correctly handle back-to-back Push requests.
- 51643** Japanese now displays correctly on the SoundStation IP 6000 and VVX 1500.
- 51753** Display text on the SoundPoint IP 450 looks clearer.
- 51959** Handling of Hold re-Invites is correct after one-touch blind transfer to full park orbit.
- 51965** HTTP request messages are directed to proxy.
- 52164** Hot-dial on the VVX 1500 works in headset mode.
- 52360** The Auth Password field can no longer be viewed in the Web configuration page.
- 52365** The phones can easily transition from LLDP to CDP.
- 52370** Removing the Ethernet cable from the SoundStation IP 7000 no longer unmutes the muted phone.
- 52376** The parameter `daylightSavingsTime` can now be disabled. Introduced in SIP 3.2.0 (*applies to SoundStation IP 6000 and 7000*).
- 52381** The Retrieve, Directed, and Group soft keys no longer disappear after entering some digits. This occurs when using the Call Park/Pick-Up feature through SIP signaling. Introduced in SIP 3.2.0 (*applies to SoundPoint IP 430, 450, and SoundStation 6000*).
- 52415** When using enhanced BLF, ringtones are no longer suppressed when a user is parked.
- 52568** The SoundStation IP 7000 phone plays a DTMF tone with the default configuration.
- 52580** Delayed DTMF audio feedback is heard when conferencing third POTS end while using the SoundStation IP 7000 User Interface.

- 52656** The VVX 1500 phone supports transcoding of video codecs that are not included in the far-ends capability set
- 52678** Using the quick/AdvFind search on full last name in the Corporate Directory no longer misses some entries.
- 52709** License menu displays Active to indicate a license with no expiry date.
- 52770** Message-summary SUBSCRIBE is sent when `reg.x.type=shared`.
- 52836** The phone no longer enables the user to enter more than the maximum allowed (32) characters in hot dial and contact directory operations.
- 52860** The split soft key no longer displays while transferring calls if the call per line limit is reached.
- 52883** When a call is placed to a shared line, the ringer for an IP 650 no longer stutters when the call is picked up at another station.
- 52943** LLDP reported power usage in logs indicates appropriate power consumption.
- 52950** Packet Loss and Burst Gap Loss metrics too high when calling IVR, caused by valid gap in audio sent from IVR.
- 52963** The SoundPoint IP 320, 321, 330, and 331 phones no longer reboot when the user presses NN# from idle screen to invoke Contact Directory entry screen for NN speed dial index.
- 52971** The phone no longer reboots when the `efkprompt` label is longer than 32 characters.
- 52977** The Directory soft key on the VVX 1500 does not disappear after selecting Blind transfer mode.
- 53007** VQMon on the VVX 1500 phone computes RFactor and MOS quality scores for the G7221C codec.
- 53034** SUBSCRIBE for BLA with expires: 0 received from server is recognized as terminating the subscription
- 53254** VVX 1500 enables users to change Auth Password for SIP Lines through the phone's user interface.
- 53598** Side-tone disappears after a call hangs up on headset using GN9350e with EHS.
- 53656** The part number in the Phone Status menu now displays proper part number.
- 53855** When a phones extension has an underscore in the name, followed only by numbers, the underscore is no longer removed in SIP signaling and the device can be found.
- 53917** The phone no longer reboots in a certain scenario when using the Join key.
- 53944** SoundPoint IP 320, 330, 321, and 331; SoundStation IP 7000: Phone displays Dir soft key in Korean and Slovenian languages.
- 53946** SoundPoint IP 550, 560, 650, and 670 phones no longer randomly display the time and date behind a custom idle display.
- 53975** The phones send a SUBSCRIBE message in a certain scenario when using SCA with barge-in enabled.
- 54034** The VVX 1500 phone no longer generates loud static when CNG packets are received.
- 54139** Consultative transfer uses the correct URI on REFER.
- 54262** The Ethernet status menu on the SoundPoint IP 320 and 321 displays the correct information.

- 54631** The Voice/Video call type prompt on the SoundStation IP 7000 defaults to Voice.
- 54765** The VVX 1500 phone fails to resend INVITE after 401 from server when second INVITE is roughly 1500 bytes.
- 54768** VVX 1500 phones can establish calls correctly when booted without a network connection.
- 54886** The phones send re-Invite with SDP containing session attribute a=sendrecv upon resuming a call when the call is initiated with a=sendrecv offered.
- 54940** New REQUESTS sent directly to far end; route set ignored after a call is placed on MOH, resulting in a loss of audio.
- 55052** An additional parameter in the From header of INVITE no longer causes 1-way audio when it is not found in the ACK to a 200 OK.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP 3.2.2 configuration file parameters.

Table 18: Software Version 3.2.2 Parameter Enhancements

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>
phone1	added	call.autoOffHook.1.protocol		
phone1	added	call.autoOffHook.2.protocol		
phone1	added	call.autoOffHook.3.protocol		
phone1	added	call.autoOffHook.4.protocol		
phone1	added	call.autoOffHook.5.protocol		
phone1	added	call.autoOffHook.6.protocol		
phone1	added	reg.1.protocol.H323		
phone1	added	reg.1.protocol.SIP		
phone1	added	reg.1.server.H323.1.address		
phone1	added	reg.1.server.H323.1.expires		
phone1	added	reg.1.server.H323.1.port		
phone1	added	reg.2.protocol.H323		
phone1	added	reg.2.protocol.SIP		
phone1	added	reg.2.server.H323.1.address		
phone1	added	reg.2.server.H323.1.expires		
phone1	added	reg.2.server.H323.1.port		
phone1	added	reg.3.protocol.H323		

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>
phone1	added	reg.3.protocol.SIP		
phone1	added	reg.3.server.H323.1.address		
phone1	added	reg.3.server.H323.1.expires		
phone1	added	reg.3.server.H323.1.port		
phone1	added	reg.4.protocol.H323		
phone1	added	reg.4.protocol.SIP		
phone1	added	reg.4.server.H323.1.address		
phone1	added	reg.4.server.H323.1.expires		
phone1	added	reg.4.server.H323.1.port		
phone1	added	reg.5.protocol.H323		
phone1	added	reg.5.protocol.SIP		
phone1	added	reg.5.server.H323.1.address		
phone1	added	reg.5.server.H323.1.expires		
phone1	added	reg.5.server.H323.1.port		
phone1	added	reg.6.protocol.H323		
phone1	added	reg.6.protocol.SIP		
phone1	added	reg.6.server.H323.1.address		
phone1	added	reg.6.server.H323.1.expires		
phone1	added	reg.6.server.H323.1.port		
sip	added	call.autoAnswer.H323		0
sip	added	call.autoAnswer.micMute		1
sip	added	call.autoAnswer.ringClass		4
sip	added	call.autoAnswer.SIP		0
sip	added	call.autoAnswer.videoMute		0
sip	added	call.autoRouting.preference		line
sip	added	call.autoRouting.preferredProtocol		SIP
sip	removed	httpd.lp.port		
sip	removed	httpd.ta.enabled		

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>
sip	added	log.level.change.h323		4
sip	added	log.level.change.poll		4
sip	added	log.level.change.push		4
sip	added	log.level.change.wmgr		4
sip	removed	mb.launchpad.enabled		
sip	removed	mb.main.1.icon		
sip	removed	mb.main.1.text		
sip	removed	mb.main.1.url		
sip	removed	mb.main.2.icon		
sip	removed	mb.main.2.text		
sip	removed	mb.main.2.url		
sip	removed	mb.main.3.icon		
sip	removed	mb.main.3.text		
sip	removed	mb.main.3.url		
sip	removed	mb.main.4.icon		
sip	removed	mb.main.4.text		
sip	removed	mb.main.4.url		
sip	removed	mb.main.5.icon		
sip	removed	mb.main.5.text		
sip	removed	mb.main.5.url		
sip	removed	mb.main.6.icon		
sip	removed	mb.main.6.text		
sip	removed	mb.main.6.url		
sip	added	sec.H235.mediaEncryption.enabled		1
sip	added	sec.H235.mediaEncryption.offer		0
sip	added	sec.H235.mediaEncryption.require		0
sip	added	up.callTypePromptPref		1
sip	added	up.enableCallTypePrompt		1

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>
sip	changed	up.idleBrowser.enabled	0	
sip	added	up.manualProtocolRouting		1
sip	added	up.manualProtocolRouting.softKeys		1
sip	changed	video.autoStartVideoTx		1
sip	added	video.callRate		448
sip	added	video.codecPref.H261		4
sip	changed	video.enable		1
sip	added	video.forceRtcpVideoCodecControl		0
sip	changed	video.maxCallRate		512
sip	added	video.profile.H261.annexD		
sip	added	video.profile.H261.CifMpi		1
sip	added	video.profile.H261.jitterBufferMax		2000
sip	added	video.profile.H261.jitterBufferMin		150
sip	added	video.profile.H261.jitterBufferShrink		70
sip	added	video.profile.H261.QcifMpi		1
sip	changed	video.screenMode		normal
sip	changed	video.screenModeFS		normal
sip	added	voice.audioProfile.G719.32Kbps. payloadType		107
sip	added	voice.audioProfile.G719.48Kbps. payloadType		108
sip	added	voice.audioProfile.G719.64Kbps. payloadType		109
sip	added	voice.audioProfile.G719.jitterBufferMax		200
sip	added	voice.audioProfile.G719.jitterBufferMin		40
sip	added	voice.audioProfile.G719.jitterBuffer Shrink		1500
sip	added	voice.audioProfile.G719.payloadSize		20
sip	added	voice.codecPref.VVX_1500.G719.32Kbps		
sip	added	voice.codecPref.VVX_1500.G719.48Kbps		

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>
sip	added	voice.codecPref.VVX_1500.G719.64Kbps		
sip	changed	voice.gain.tx.digital.chassis.VVX_1500	6	3
sip	added	voIpProt.H323.autoGateKeeperDiscovery		0
sip	added	voIpProt.H323.dtmfViaSignaling.enabled		1
sip	added	voIpProt.H323.dtmfViaSignaling. H245alphanumericMode		1
sip	added	voIpProt.H323.dtmfViaSignaling. H245signalMode		1
sip	added	voIpProt.H323.enable		0
sip	added	voIpProt.H323.local.port		1720
sip	removed	voIpProt.local.port		
sip	added	voIpProt.server.H323.1.address		
sip	added	voIpProt.server.H323.1.expires		
sip	added	voIpProt.server.H323.1.port		
sip	added	voIpProt.SIP.dtmfViaSignaling.rfc2976		
sip	added	voIpProt.SIP.enable		1
sip	added	voIpProt.SIP.local.port		5060

Understanding Updates to SIP 3.2.1B

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

48947 Support for the SoundPoint IP 335 product.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to the SIP configuration file parameters.

Table 19: Software Version 3.2.1B Parameter Enhancements

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>New Value</i>	<i>Description</i>
sip	added	ind.anim.IP_335.42.frame.1.bitmap	Handset	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.42.frame.1.duration	1300	
sip	added	ind.anim.IP_335.42.frame.2.bitmap	PlumHd	
sip	added	ind.anim.IP_335.42.frame.2.duration	1300	See Administrators Guide for SIP 3.2.2 for details
sip	added	ind.anim.IP_335.43.frame.1.bitmap	Headset	
sip	added	ind.anim.IP_335.43.frame.1.duration	1300	
sip	added	ind.anim.IP_335.43.frame.2.bitmap	PlumHd	
sip	added	ind.anim.IP_335.43.frame.2.duration	1300	
sip	added	ind.anim.IP_335.44.frame.1.bitmap	Speaker	
sip	added	ind.anim.IP_335.44.frame.1.duration	1300	
sip	added	ind.anim.IP_335.44.frame.2.bitmap	PlumHd	
sip	added	ind.anim.IP_335.44.frame.2.duration	1300	

Understanding Updates to SIP 3.2.1

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

Enhanced Capabilities

53322 Setting `voIpProt.local.port` to a non-standard port does not send from or advertise that port.

53611 User Language Selection is retained during an upgrade to SIP 3.2.0.

53685 The phones no longer ignore `nat.ip` parameters.

53852 DTMF duration on the SoundStation IP 7000 defaults to 300ms for HDX system integration.

Understanding Updates to SIP 3.2.0

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 22527** Implemented the Scrolling Status Bar on the SoundPoint IP 320, 321, 330, 331, 550, 560, 650, 670, and SoundStation IP 6000 and 7000.
- 26754** Support for the iLBC codec on the SoundPoint IP 320, 321, 330, 331, 450, 550, 560, 650, and 670.
- 30079** Added support for mutual TLS authentication. See [Technical Bulletin 52609](#) for more details on this feature.
- 32259** Microbrowser recognizes multiple mime types.
- 32753** Support for LLDP protocol. To take full advantage of this feature, you need to use BootROM 4.2.0.
- 34782** Replaced libSRTP algorithms with OpenSSL versions.
- 35525** The DND icon contains text identifying that DND is active.
- 37118** Added the ability to take a screen capture.
- 39358** Added a Loud Ringer Ringtone selection. See [Technical Bulletin 39358](#) for instructions on how this can be configured.
- 30855** Create a SoundStation IP 7000 Setup Guide.
- 41579** Met requirements of ETSI TS 102 027-2 v4.1.1 RFC 3261 compliance test for Anatel/Brazil.
- 43141** Support for Statically Configured BLF and Call Park and Retrieve enhancements.
- 43142** Support for single button Blind Transfer and Retrieve of a call designated as an automata in the dialog used for Statically Configured BLF.
- 43646** Improved boot speed in some situations where the boot server is incorrectly configured.
- 45057** Languages selection presented in appropriate language.
- 45174** Upgraded zlib to version 1.2.3.
- 45743** Upgraded curl library to version 7.19.2.
- 45787** Added instructions to the SoundPoint IP 450, 550, 560, 650, and 670 for changing label colors in the User Guides.
- 45791** Added a configuration option on the SoundStation IP 7000 to disable digit-map rules for Remote Dialing when connected to an HDX system.
- 46093** Added the ability for User to enable/disable display of idle browser from menu.
- 46113** Added navigation button shortcuts in Idle Mode consistent with other phone models (*applies to SoundPoint IP 320, 321, 330, and 331*).
- 46248** Added an Admin menu option on the SoundStation IP 7000 to manually specify the value to be used as the extension displayed on the phone screen.
- 46424** Improved readability of Menu items when using Background images on the display.
- 46446** New menu option to view the status of feature licenses.
- 46683** Removed Background from scrolling Status Bar for improved readability.

- 47355** Scrolling the Status Bar gives equal time to each status message.
- 47390** Added configuration parameters for select ETSI SIP compliance requirements.
- 47463** The phones allow for secure entry of passwords in the micro-browser API.
- 47487** Added the ability to enable/disable a Back soft key in the microbrowser
- 47689** Added support for SoundStation IP 7000/HDX 6000 system integration. This feature requires a future update release to the HDX 6000 system software.
- 47749** Support Transmission of Join Header as per RFC 3911.
- 48004** Support for BLF call pick-up using Dialog-info within an INVITE with Replaces header.
- 48055** Improved user experience of the Enhanced BLF feature when an incoming call occurs whilst the user is viewing BLF monitored line call details.
- 48109** Included fmtp attribute specifying Mode=30 in the SDP when 13.33 Kbps iLBC is used.
- 48136** Removed platform specific TFTP code and instead used TFTP support in curl library 7.19.2.
- 48137** Support for BLF call pick-up using Dialog-info within an INVITE with Replaces header.
- 48205** Support for the iLBC Codec (*applies to SoundStation IP 6000 and 7000*).
- 48559** Consistent scrolling status line on various phones (*applies to SoundPoint IP 450, 550, 560, 650, and 670; SoundStation IP 6000 and 7000*).
- 48578** Reduced the local Contact Directory maximum to 99 on the SoundPoint IP 430.
- 48579** Reduced the maximum number of calls supported to 4 (from 8) on the SoundPoint IP 430.
- 48664** Added the user accessible menu option to display whether a device certificate is installed.
- 48678** Media Statistics menu is more easily accessible. Accessed from Menu > Status > Diagnostics > Media Statistics.
- 48738** Added configurable behavior for Directed Call Pick-Up as used for Enhanced BLF.
- 48780** Added an option to apply digit-map rules to tel:URI initiated calls.
- 48846** Added a configuration option for whether the call appearance on a remotely monitored BLF line should be presented on the monitoring/attendant phone.
- 48861** Add a configuration option `voIpProt.SIP.strictReplacesHeader` to control whether the phone requires call-id, to-tag, and from-tag to perform and INVITE with Replaces.
- 48984** The phone populates the display-name field in the To header of responses that it generates.
- 48998** Added a configuration option for the phone to send 486 Busy when a call is rejected.
- 49309** Combined the SoundPoint IP 550 and 560 user guides.
- 49465** Updated Destination of outbound call based on the display name in the SIP To header responses.
- 49660** During call forwarding, *user=phone* should be included in `refer-to` parameter of Refer header.
- 49695** Allow for SDP offer or answer in provisional reliable response and PRACK request and response.
- 49839** RTP Rx detects and corrects for G.722, G.722.1, G.722.1C, and G.719 RTP timestamp increments based on different sample rates.

50769 Added support for Hook-Flash during POTS calls on the SoundStation IP 7000.

50927 Added Equifax Secure eBusiness CA-1 to the trusted CA list.

51419 RFC2543 Hold not working when video SDP present in certain scenarios.

Discontinued Features

48283 Removed support for SoundPoint IP 301, 501, 600, and 601 phones.

48698 Removed support for SoundStation IP 4000.

Enhanced Capabilities

Application load progress bar matches actual progress.

29148 The phone formats the file system when it notes an error on the screen while loading large configuration files.

29344 HTTP Digest Authentication now works on IIS.

30219 Logs are uploaded when phone resets to factory default.

31858 When two phones with a shared line simultaneously resume a held call, the phone which did not retrieve the call shows call in progress on its shared line indicator.

34681 The parameters `stickyAutoLineSeize` and `call.enableOnNotRegistered=0` do not seize correctly if the first line is unregistered.

35288 The Web Configuration Utility uses less memory during initialization.

35991 The Roaming Buddy list with Office Communicator reports the proper status of all buddies.

36969 The SoundStation IP 6000 displays Japanese language correctly.

38348 The SRTP call displays proper line icons in a certain scenario on the SoundPoint IP 320, 321, 330, and 331.

38392 Performing a Blind Transfer from an encrypted phone to an unencrypted private line establishes the new call as encrypted.

38418 The phones no longer show SRTCP authentication failure at log level 0.

38824 After audio diagnostics such as Record and Play in handset, the first call is no longer established in handset mode even if the handset is ON-HOOK.

39013 Attaching a cell phone cable to the SoundStation IP 7000 no longer invokes the Cell phone UI until a physical cell phone is attached.

39143 The P-Asserted-Identity header in initial INVITE message is no longer used for caller ID.

39949 The navigation icon in the Corporate Directory correctly displays the available navigation options when using the keypad to navigate (*applies to SoundPoint IP 320, 321, 330, and 331*).

40679 Changing the status on the MyStatus menu of the SoundStation IP 6000 changes the OC client status when `roaming_buddies.reg= 1`.

40892 The Time/Date displays on the SoundStation IP 7000 when the first phone call is established.

- 41939** The user is not able to play the WAV file when it has a call on hold and is also in a remote busy state. Junk characters appear in audio player.
- 42092** Special Slovenian characters are included in the phone's fonts.
- 42213** The SIP: string displays on the SoundStation IP 7000 when using URL dialing.
- 42611** Recording no longer begins when a full USB drive is attached.
- 42761** Pressing the Content soft key on the SoundStation IP 7000 no longer prompts the user to choose VGA input.
- 43910** The microbrowser can process an http response which contains an image/bmp.
- 43916** Configured sampled wave files can be downloaded onto the phone depending on sufficient RAM Disk size.
- 43990** Missing glyphs in the Katakana bit stream fonts on the SoundStation IP 7000.
- 44100** Call display names containing an @ symbol no longer truncate characters after the @ symbol.
- 44248** The microbrowser displays an error message when unsupported media is configured in the microbrowser URL.
- 44273** The phones can process all contacts in a SIP Contact header containing a comma separated list.
- 44278** The phone numbers display correctly on line keys when the length of a phone number is more than 10 characters.
- 44301** The Date on the SoundStation IP 6000 and 7000 when the idle browser is enabled.
- 44377** The Redial key can be reassigned.
- 44443** The Menu exit through the Menu key is ignored while in Edit mode (*applies to SoundPoint IP 320, 321, 330, and 331*).
- 44635** The SoundStation IP 6000 phone uses the correct configuration parameters to download customizable fonts.
- 44783** The Cipher list is the same for different TLS transactions.
- 44844** USB Call Recording can be stopped using the Stop soft key.
- 44855** When using Call Lists, the missed calls are incremented on Call Forward on Busy.
- 44892** When using SCA Barge-In on the SoundStation IP 6000 and 7000 phones, the user no longer interferes by answering the wrong call in certain scenarios.
- 44962** The phone no longer displays a three-way animation icon on the held screen when conference legs on hold.
- 45143** When the maximum conference size is reached when using Centralized Conference, the phone no longer displays a local conference UI.
- 45327** When the user establishes a call between two phones configured as shared lines, and presses the down arrow key, all soft keys no longer disappear.
- 45428** An unexpected re-INVITE no longer occurs before BYE when removing a leg from a conference call.
- 45650** In a double hold with music on hold and a non-Polycom SIP phone, – MOH no longer fails.

- 45658** The platform string in transmitted CDP packets is consistent across SoundPoint IP products.
- 45716** Text on the SoundPoint IP 450 is consistent as on other phones.
- 45835** Status Bar text on the SoundPoint IP 450 is easier to read on some backgrounds.
- 45943** Correct logic is used when picking line for outgoing call in a multiple registration scenario.
- 46068** Transfer On Proceeding is supported when using a proxy server.
- 46334** DTMF local rendering does not stop. If the far end holds while local digit key is pressed then the far end resumes.
- 46478** On the EFK feature, the phone sends invite when executing `$Cwaitdialtone$`.
- 46513** Dialog Event Package Content Guideline 6B (Local Identity).
- 46514** Dialog Event Package Content Guideline 6C (Local Target).
- 46547** Warning Header Text notification on the SoundStation IP 7000 displays on phone (when configured).
- 46550** Directed-Call-Pickup no longer fails when SIP server is a proxy.
- 46588** Info soft key on the SoundStation IP 7000 is no longer missing in the Contact Directory.
- 46738** The `attendant.ringType` parameter is removed from the override file when default (silent) attendant ring type is selected.
- 46741** Using enhanced BLF, when the watched line hangs up an outgoing call, the remote call appearance screen times out on the console phone.
- 46770** On the microbrowser, the * and # buttons work correctly when the text input mode is set to numeric on input fields.
- 46899** When using the electronic hookswitch, audio is heard during an active call if the user answers by pressing the hookswitch button immediately on a Jabra headset under a specific scenario.
- 47039** The line LED flashes instead of remaining a stable green when an active call is kept on hold during an incoming call.
- 47123** When using the USB Call Recording, the missed call notification no longer displays on the audio player screen if an incoming call is not answered during playback.
- 47207** When the MUTE is active on the SoundStation IP 7000, it no longer covers up the dialing fields.
- 47248** Hot dial works when lifting the handset for the second call when `call.stickyAutoLineSeize=1`.
- 47300** A URL dial-disabled message displays and successfully routes to voicemail from Message Center tab.
- 47336** The Received\Missed call list on the SoundStation IP 7000 no longer shows the IP address of the SIP server instead of the Extension number of a call received/Missed from a SIP extension.
- 47464** When two incoming calls are active on a phone, lifting the handset or pressing the handsfree key to answer the call no longer results in the most recent call being answered, even though the ring tone is played according to the first incoming call (*applies to SoundPoint IP 320 and 330, and SoundStation IP 7000*).

- 47535** The soft keys no longer reset to the default layout on an inbound call in some multiple call handling scenarios.
- 47566** When an internal URI is executed with multiple VolUp and VolDown action URIs, the Ringer horizontal bar is seen and the Volume sound going UP and DOWN is heard.
- 47578** When using the Corporate Directory on the SoundPoint IP 320, 321, 330, 331, the sticky attributes are saved.
- 47612** When using BLF, cancelling a Transfer for a call that was initiated using Directed Call Pick-Up sequence results in the correct caller-id display to the user.
- 47641** The *Network Link Down* message on the SoundStation IP 7000 displays on the screen unless the phone reboots and comes up with Ethernet cable.
- 47695** When the phones have two registrations, the NewCall soft key no longer displays for alerting call appearance when there are max call appearances (*applies to SoundPoint IP 320, 321, 330, 331, 430, and 450*).
- 47699** When using XML API Internal URIs on the SoundStation IP 6000, the Tel URI is works correctly if embedded within a couple of internal URI actions.
- 47712** A local contact directory search on the SoundPoint IP 320, 321, 330, and 331 now works correctly.
- 47724** Mute icon and Call appearance counter on the SoundPoint IP 450 no longer conflict when DND is turned on and multiple call appearances are present on the phone.
- 47729** The on-hook dialing widget no longer uses multi-tap behavior but is not in multi-tap mode.
- 47746** The New Call soft key does not display when phone holds max conference calls.
- 47798** The location of the Transfer and Conference soft keys on the SoundStation IP 7000 are more easily accessible during conference setup.
- 47847** When using BLF, the monitoring phone continues ringing if a shared line is seized while the monitored line has an incoming call.
- 47853** When the headset memory mode is active, the Headset key continues blinking during incoming calls after ending the first active call.
- 47862** The Time and Date on the SoundStation IP 6000 now displays during a call.
- 47863** The phone's HTTP server is no longer sending some HTTP traffic in very small TCP segments.
- 47916** The Resume soft key on the SoundPoint IP 320, 321, 330, and 331 is available for second call appearance after splitting conf established through Join from different shared line registrations.
- 47921** The order of call appearances on the SoundPoint IP 320, 321, 330, and 331 is consistent with other phones after splitting a conference.
- 47929** Rendering special characters like no longer break the hyperlink style display.
- 47932** The Call widget counter (1/n) appears while in the dial tone state.
- 47951** Transfer has precedence over pickup of a ringing BLF line when pressing the line key during a call transfer.
- 47953** Call info display on the SoundStation IP 6000 displays correctly when volume up/down key is pressed.

- 47958** More than one contact can be added when the SoundStation IP 7000 is configured with no Ethernet cable connected + HDX system.
- 47962** An incorrect icon no longer displays when Redialing POTS call on the SoundStation IP 7000.
- 48003** The SoundStation IP 7000 phone no longer dials a POTS call as a video call when dialing from the idle state for a certain configuration.
- 48011** Use of the Idle Browser on the SoundStation IP 7000 no longer interferes with some display elements such as the Mute Icon and the Video/Phone Call pop-up message when connected to an HDX system.
- 48019** The pop-up message "Video or Phone Call?" is no longer overwritten by the idle browser on the SoundStation IP 7000.
- 48045** When using enhanced BLF, the phone holds the first call when pressing the Dial soft key to make the second call to the same party.
- 48049** When using BLF, the attendant phone displays all remote calls on a BLF monitored line if the Monitored Phone has a call in the Ringing state.
- 48061** When using enhanced BLF, the attendant phone updates the 1/x widget when the BLF monitored line has one or multiple incoming calls being ended.
- 48069** When using the SCA Barge-In feature, extra soft keys no longer display on the remote shared phone while viewing the call appearance list by long pressing a line key.
- 48071** Key:Handsfree internal URI action is executed by the phone in a certain scenario.
- 48115** The HDX system no longer plays a ring sound after answering a POTS call on the SoundStation IP 7000.
- 48131** Call Forwarding Status now shows multiple Call Forward Types are selected.
- 48149** The SDP attribute is no longer truncated when first character of the value is a digit.
- 48162** The Boot Server status field no longer shows an incomplete or blank path if a / is included in the setting.
- 48174** A failed call no longer causes subsequent calls to skip URL/Number mode selection.
- 48179** A called Party number is no longer shown overlapped in incoming event notification when IP dialed calls are made between unregistered phones.
- 48209** The left-most character can be deleted before character selection timeout.
- 48213** Key:LineX is executed only if X is a supported line key for that platform.
- 48333** When using the USB Call Recording, the USB busy indicator appears on main screen when recording in progress.
- 48414** The phone no longer occasionally fails to act on the electronic hookswitch up/down signal from Plantronics and Hydra headsets.
- 48700** When using the USB Call Recording, playback can be stopped through a Stop soft key.
- 48745** LDAP Critical Extension Error 0x0c no longer causes the CD Server to not respond to messages from phone.
- 48981** SRTP no longer fails in 3.1.2 when the user presses Hold then Resume during a call. This happens on several different models of the IP phone.

- 48996** The phone tags the correct DSCP value to some packets (Trying, Ringing and OK).
- 49106** The entire dialed URL is saved in the phone's call history.
- 49251** The Polish XML Dictionary includes Polish characters.
- 49300** Ensure that the DTMF tones are being sent through the dtmf start/stop Clink2 API (*applies to SoundStation IP 7000*).
- 49417** The phone no longer reports MOH dialog if SUBSCRIBE received while on hold.
- 49459** Cancel works after entering hot dial digits.
- 49461** DND symbol(X) appears after the DND feature is disabled in a certain configuration.
- 49473** When using the Corporate Directory on the SoundPoint IP 320, 321, 330, 331, using the # key to change text entry mode it resets the Quick Search timeout timer.
- 49476** The scrolling indicators on the Corporate Directory work better.
- 49512** HTTP Refresh header response loads the specified URL on the phones after the specified amount of time has passed, in a certain situation.
- 49516** Hanging up the handset terminates calls in Audio or Display Diagnostics.
- 49523** Asian fonts are clearer on the SoundPoint IP 450 and SoundStation IP 7000.
- 49548** The Edit and Delete soft keys on the SoundPoint IP 320, 321, 330 and 331 disappear after deleting the last contact.
- 49572** When using the Corporate Directory on the SoundStation IP 7000, numeric characters can be entered in the Quick Search entry field.
- 49617** The phone plays a dial tone after a hold reminder is played in certain scenarios.
- 49619** The call waiting beep plays on phone when call hold reminder is set.
- 49620** Volume settings for Recording work in handsfree mode.
- 49639** The handsfree dial tone is no longer interrupted by the hold reminder and call waiting ringtones.
- 49641** Call info display on the SoundStation IP 6000 and 7000 displays correctly while changing volume.
- 49677** The phone complies with RFC4475 3.1.2.3 Negative Content-Length.
- 49685** On the SoundPoint IP 320, 321, 330, and 331, you can enter URLs with uppercase letters.
- 49692** The seconds colon in the time display blinks for every second on the SoundPoint IP 450.
- 49693** The ACD icon displays when the parameter `voIpProt.SIP.serverFeatureControl.cf=1` is enabled.
- 49696** After a long LAN outage while downloading a new application, when the phone re-connects to the network, it displays an error message.
- 49701** The SoundStation IP 7000 phone response with `reg.1.server.1.expires=5` setting is consistent.
- 49706** The SIP Extension display on the SoundStation IP 7000 is no longer disabled after disconnecting from an HDX system with an HDX-Preference option set.
- 49757** The SoundStation IP 7000 phone displays Network Link is Down after the cable is disconnected from a hub.

- 49758** The SoundStation IP 7000 phone no longer gets into a bad state and can recover from temporarily unplugging network connection during an active call.
- 49776** If `dir.corp.user` is misconfigured, the phone displays Login Error.
- 49813** When using the Corporate Directory, the phones no longer display *Enter More Chars...* when submitting a string that returns no results in the Quick search mode.
- 49825** When using the Corporate Directory, the black background for the Search bar displays consistently on different platforms.
- 49829** NTP Time synchronization is reliable in a particular scenario.
- 49834** When using the Corporate Directory, if VLV indexing is configured and an Advanced Find yields more results than the configured page Size (Default is 64), scrolling through the entries works correctly.
- 49836** If the Corporate Directory is down and the phone reboots, the phones display a static "Please try again" message.
- 49911** Incoming ring tones are played on the phone in a certain enhanced BLF use case.
- 49926** The SoundPoint IP 320, 321, 330, and 331 phones no longer auto-increment the new contacts speed dial index to 100 even though the maximum amount of entries is 99.
- 49927** After an AdvFind search, exit and re-enter Corp Dir menu, phone displays search bar as Search: not Search (Filtered) (*applies to SoundPoint IP 320,321,330,331 and VVX 1500*).
- 49929** The SoundStation IP 7000 displays an HDX system Extension when voice call type is set to Auto and phone is not registered to SIP server.
- 49981** After rebooting the SoundStation IP 7000, the proper HDX system extension displays.
- 49982** The SoundPoint IP 320, 321, 330, and 331 phones reconfigure when DHCP lease expires.
- 49989** The SoundStation IP 7000 phone is no longer adding contact directories from the call list with the existing speed dial number.
- 49977** The SoundPoint IP 320, 321, 330, and 331 phones display the selected status under MyStat menu.
- 50090** The SoundStation IP 7000 phone displays an Active Conference screen on joining a remotely held SLA call without first holding the local call.
- 50099** Consultative Transfer no longer fails if the second leg is forwarding and its 302 response is handled by proxy.
- 50109** Volume levels on the SoundStation IP 7000 are in Sync when Dialing a Video call.
- 50110** An *Enter number* message displays for Video and audio calls once the Ethernet is removed on the SoundStation IP 7000.
- 50115** The DTMF tone of the first digit on the SoundStation IP 7000 plays at the SoundStation IP 7000 volume instead of the HDX system volume.
- 50118** Dial tone volume and Hands Free volume are in sync on the SoundStation IP 7000.
- 50137** The volume no longer resets to default on the SoundStation IP 7000 after a POTS call is connected if `voice.volume.persists.handsfree=0`.

- 50153** When using the Corporate Directory, setting the Primary Attribute as sticky `dir.corp.attribute.1.sticky=1` gives a clearer user interface behavior.
- 50159** When using the Corporate Directory, a Quick search on a non-null sticky primary filter is no longer missing records.
- 50189** SIP responses are no longer missing the to-tag after the phone challenges INVITE.
- 50212** Scrolling upward on the Corporate Directory sorts the order of the phone entry list.
- 50253** When using the Corporate Directory on the SoundStation IP 7000 and the edit phone number attribute in AdvFind menu, pressing on the 1/A/a soft key creates an Encoding soft key.
- 50254** The phone does honors SDP sent in PRACK.
- 50255** SIP Reliable Provisional responses are retransmitted.
- 50256** When not yet registered, phones experience a random delay of 30-60 seconds between registration attempts.
- 50264** Global prefix +present on calls made from Placed Calls list.
- 50299** When using the Corporate Directory on the SoundStation IP 7000, Quick search text input starts at the first multi tap character.
- 50381** Pressing the left navigation key on the SoundPoint IP 320, 321, 330, and 331 before the character selection timeout no longer moves cursor 2 spots.
- 50397** The SoundStation IP 7000 phone displays licenses correctly in the status screen.
- 50407** When the Corporate Directory server is down with phone connecting to LDAP server, a quick search results in the phone displaying a proper error message.
- 50523** When using the Corporate Directory on the SoundPoint IP 320, 321, 330, 331, the phone displays the Contact title in the View menu.
- 50546** With URL dialing disabled, a BLIND soft key appears in the third soft key slot after pressing TRANSFER.
- 50811** P-Asserted ID display name is a sticky on UI call appearance and in the placed call list.
- 50869** The phone only offers SRTP when SRTP crypto suite is selected.
- 50891** The Resume soft key on the SoundStation IP 6000 and 7000 displays when the phone is put on hold on another shared line phone.
- 50989** Receiving a 603 Decline by a BLF monitored user plays a reorder tone.
- 51041** Regarding X-IdleBrowserSelectUrl, `http://url` is no longer remembered by the phone.
- 51245** The BLF state is updated on receipt of the first full state NOTIFY after a reboot.
- 51320** The message "Conference in Another Video or phone call?" no longer displays in a loop for each press on the Conf hard key (*applies to SoundStation IP 7000*).
- 51432** The Conference hard key pop-up message on the SoundStation IP 7000 does not display any messages. It directly allows the user to make a video call.
- 51554** The phones no longer add an additional CRC to some 802.1X packets received on the PC port.
- 51567** Server based CFWD/DND sync no longer fails on 3.1.2.0392.

- 51605** API push request is no longer lost if it immediately follows another push request.
- 51631** The phone releases the first assigned IP address when VLAN is set through DHCP.
- 51633** The phone plays a busy/reorder tone upon a refer-based transfer and when it gets a 603 or 486 response.
- 51644** Certain Japanese strings now display correctly.
- 51690** The EFK feature is used for one touch voicemail dialing. When using EFK with SIP 3.1.3, the phone honors the `stickyautolineseize`.
- 51718** The phone no longer continues to ring after a call has been answered with a certain call signaling sequence.
- 51763** When adding video to an existing call on a SoundStation IP 7000, pressing the Mute key successfully mutes the far end.
- 51838** Japanese characters now correctly display.
- 52014/53597** In SIP 3.x.x, when an IP phone picks up a transferred call in a certain scenario, the call is connected instead of being placed on hold.
- 52017** The web interface issue Password entry is masked when entered.
- 52108** The phone successfully restores destination to Asserted Identity or Remote ID after a transfer fails.

Configuration File Enhancements

Refer to the following table for a list of the parameters that have been added, changed, or deleted from the template **phone1.cfg** and **sip.cfg** files. You can find further descriptions of parameters in [Administrators' Guide for the SIP 3.2.0 Release](#).

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

Table 20: Software Version 3.2.0 Parameter Enhancements

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	added	<code>call.directedCallPickup</code> Method		native or legacy	See Administrators Guide for SIP 3.2.0 for details.
sip	added	<code>call.parkedCallRetrieve</code> Method		native or legacy	
sip	added	<code>call.parkedCallRetrieve</code> String		Star code	See Administrators Guide for SIP 3.2.0 for details.
sip	added	<code>dialplan.applyToRemote</code> Dialing		0 or 1; Default is 0	A flag to determine if the dial plan applies to calls made through the Polycom HDX system.

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	added	dialplan.applyToTelUriDial		0 or 1 Default is 1	A flag to determine if the dial plan applies to uses of the tel:// URI.
sip	added	ind.class.2.state.35.index		44	Changes Relating to screen layout modifications.
sip	added	ind.class.2.state.36.index		42	
sip	added	ind.class.2.state.37.index		43	
sip	changed	ind.gi.IP_400.4.physX	122	0	
sip	changed	ind.gi.IP_400.5.physX	112	10	
sip	changed	ind.gi.IP_4000.6.physH	12	0	
sip	changed	ind.gi.IP_4000.6.physW	14	0	
sip	changed	ind.gi.IP_4000.6.physX	16	0	
sip	changed	ind.gi.IP_4000.6.physY	2	0	
sip	changed	ind.gi.IP_450.16.physX	176	196	
sip	changed	ind.gi.IP_450.17.physX	176	196	
sip	changed	ind.gi.IP_450.18.physX	176	196	
sip	changed	ind.gi.IP_450.19.physX	176	196	
sip	changed	ind.gi.IP_450.2.physX	40	20	
sip	changed	ind.gi.IP_450.3.physH	20	0	
sip	changed	ind.gi.IP_450.3.physW	20	0	
sip	changed	ind.gi.IP_450.3.physX	20	0	
sip	changed	ind.gi.IP_450.3.physY	2	0	
sip	changed	ind.gi.IP_600.13.physH	103	111	
sip	changed	ind.gi.IP_600.13.physY	0	25	
sip	changed	ind.gi.IP_600.4.physY	105	3	
sip	changed	ind.gi.IP_600.6.physH	20	0	
sip	changed	ind.gi.IP_600.6.physW	20	0	
sip	changed	ind.gi.IP_600.6.physX	113	0	
sip	changed	ind.gi.IP_600.6.physY	110	0	
sip	changed	ind.gi.IP_7000.3.physH	20	0	

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	changed	ind.gi.IP_7000.3.physW	20	0	Language selection displays in the correct language.
sip	changed	ind.gi.IP_7000.3.physX	20	0	
sip	added	lcl.ml.lang.menu.1.label		简体中文 (zh-cn)	
sip	added	lcl.ml.lang.menu.10.label		日本語 (ja-jp)	
sip	added	lcl.ml.lang.menu.11.label		한국어 (ko-kr)	
sip	added	lcl.ml.lang.menu.12.label		Norsk (no-no)	
sip	added	lcl.ml.lang.menu.13.label		Polski (pl-pl)	
sip	added	lcl.ml.lang.menu.14.label		Português (pt-br)	
sip	added	lcl.ml.lang.menu.15.label		Сский (ru-ru)	
sip	added	lcl.ml.lang.menu.16.label		Slovenski (sl-si)	
sip	added	lcl.ml.lang.menu.17.label		Español (es-es)	
sip	added	lcl.ml.lang.menu.18.label		Svenska (sv-se)	
sip	added	lcl.ml.lang.menu.2.label		Dansk (da-dk)	
sip	added	lcl.ml.lang.menu.3.label		Nederlands (nl-nl)	
sip	added	lcl.ml.lang.menu.4.label		English (en-ca)	
sip	added	lcl.ml.lang.menu.5.label		English (en-gb)	
sip	added	lcl.ml.lang.menu.6.label		English (en-us)	
sip	added	lcl.ml.lang.menu.7.label		Français (fr-fr)	
sip	added	lcl.ml.lang.menu.8.label		Deutsch (de-de)	

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	added	lcl.ml.lang.menu.9.label		Italiano (it-it)	
sip	added	log.level.change.lldp		4	Control the logging detail level for the LLDP feature.
sip	added	mb.main.autoBackKey		1	See Administrators Guide for SIP 3.2.0 for details.
sip	changed	ramdisk.minfree	3072	3150	Minimum amount of free space that must be left after the RAM disk has been created.
sip	changed	se.pat.ringer.13.name	Sampled 1		Customer ringer file names.
sip	changed	se.pat.ringer.14.name	Sampled 2		
sip	changed	se.pat.ringer.15.name	Sampled 3		
sip	changed	se.pat.ringer.16.name	Sampled 4		
sip	changed	se.pat.ringer.17.name	Sampled 5		
sip	changed	se.pat.ringer.18.name	Sampled 6		
sip	changed	se.pat.ringer.19.name	Sampled 7		
sip	changed	se.pat.ringer.20.name	Sampled 8		
sip	changed	se.pat.ringer.21.name	Sampled 9		
sip	changed	se.pat.ringer.22.name	Sampled 10		
sip	added	sec.srtp.requireMatchingTag		0 or 1	A flag to determine whether or not to check the tag value in the crypto attribute in an SDP answer.

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	changed	tone.dtmf.rfc2833Payload	101	127	The phone-event payload encoding in the dynamic range to be used in SDP offers.
sip	added	up.idleBrowser.enabled		0 or 1; default is 0	A flag to determine whether or not the background takes priority over the idle browser. Used in conjunction with up.prioritizeBackgroundund.enable.
sip	added	up.prioritizeBackgroundMenuItem.enabled		0 or 1; default is 1.	If set to 1, the Prioritize Background menu is available to the user. The user can then decide whether or not the background takes priority over the idle browser. Used in conjunction with up.idleBrowser.enabled.
sip	added	up.screenCapture.enabled		0 or 1; Default is 0	A flag to determine whether or not the user can get a screen capture of the current screen shown on a phone. The flag is cleared when the phone reboots.
sip	added	voice.audioProfile.iLBC.13_33Kbps.payloadSize		30	See Administrators Guide for SIP 3.2.0 for details.
sip	added	voice.audioProfile.iLBC.15_2Kbps.payloadSize		20	
sip	added	voice.audioProfile.iLBC.jitterBufferMax		160	
sip	added	voice.audioProfile.iLBC.jitterBufferMin		40	
sip	added	voice.audioProfile.iLBC.itterBufferShrink		500	

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	added	voice.audioProfile.iLBC. payloadType		110	
sip	removed	voice.audioProfile.Lin16. 44_1ksps.payloadType	120		Parameter renamed.
sip	added	voice.audioProfile.Lin16. 44_1ksps.payloadType		120	See Administrators Guide for SIP 3.2.0 for details
sip	added	voice.audioProfile.Lin16. 8ksps.payloadType		116	See Administrators Guide for SIP 3.2.0 for details
sip	added	voice.codecPref.iLBC. 13_33Kbps			
sip	added	voice.codecPref.iLBC. 15_2Kbps			
sip	added	voice.codecPref.IP_6000. iLBC.13_33Kbps			
sip	added	voice.codecPref.IP_6000. iLBC.15_2Kbps			
sip	added	voice.codecPref.IP_650. iLBC.13_33Kbps			
sip	added	voice.codecPref.IP_650. iLBC.15_2Kbps			
sip	added	voice.codecPref.IP_7000. iLBC.13_33Kbps			
sip	added	voice.codecPref.IP_7000. iLBC.15_2Kbps			
sip	added	voIpProt.SDP.early.answer OrOffer			If set to 1, an SDP offer or answer is generated in a provisional reliable response and PRACK request and response. If set to 0, an SDP offer or answer is not generated.
sip	added	voIpProt.SDP.offer.iLBC. 13_33Kbps.includeMode			See Administrators Guide for SIP 3.2.0 for details.

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
sip	changed	voIpProt.server.1.port	5060		The port of a SIP server that accepts registration.
sip	added	voIpProt.server.2.address			
sip	added	voIpProt.server.2.expires			Minimum is now 10
sip	added	voIpProt.server.2.expires. lineSeize		30	
sip	added	voIpProt.server.2.expires. overlap			
sip	added	voIpProt.server.2.lcs			
sip	added	voIpProt.server.2.port			
sip	added	voIpProt.server.2.register		1	
sip	added	voIpProt.server.2.retryMax Count		0	
sip	added	voIpProt.server.2.retry TimeOut		0	
sip	added	voIpProt.SIP.compliance.RFC 3261.validate.contentLength			If set to 1, validation of the SIP header content language is enabled.
sip	added	voIpProt.SIP.compliance.RFC 3261.validate.uriScheme			If set to 1 or Null, validation of the SIP header URI scheme is enabled.
sip	added	voIpProt.SIP.strictReplaces Header			This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.
sip	added	voIpProt.SIP.use486for Reject			If set to 1 and the phone shows a ringing inbound call appearance, phone transmits a 486 response to the received INVITE when the Reject soft key is pressed.

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
phone1	added	attendant.behaviors.display.remoteCallerID.automata		1	Flags to determine whether or not remote party caller ID information is presented to the attendant.
phone1	added	attendant.behaviors.display.remoteCallerID.normal		1	
phone1	added	attendant.behaviors.display.spontaneousCallAppearances.automata		0	Flags to determine whether or not a call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource
phone1	added	attendant.behaviors.display.spontaneousCallAppearances.normal		1	
phone1	added	attendant.resourceList.x.address		The value of x depends on the phone. For IP 450 x=1-2; IP 550, 560 x=1-3; IP 650, 670 x=1-47	The user referenced by attendant.reg= subscribes to this URI for dialog.
phone1	added	attendant.resourceList.x.label			Text label to appear on the display adjacent to the associated line key
phone1	added	attendant.resourceList.x.type		normal	Type of resource being monitored.
phone1	changed	attendant.ringType		1	
phone1	added	dialplan.1.applyToTelUriDial		1	When present, and if dialplan.x.digitmap is not Null, this attribute overrides the global dial plan defined in the sip.cfg configuration file.
phone1	added	dialplan.2.applyToTelUriDial		1	
phone1	added	dialplan.3.applyToTelUriDial		1	
phone1	added	dialplan.4.applyToTelUriDial		1	
phone1	added	dialplan.5.applyToTelUriDial		1	

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
phone1	added	dialplan.6.applyToTelUriDial		1	
phone1	changed	divert.noanswer.1.timeout	60	55	Modified No Answer Timeout
phone1	changed	divert.noanswer.2.timeout	60	55	
phone1	changed	divert.noanswer.3.timeout	60	55	
phone1	changed	divert.noanswer.4.timeout	60	55	
phone1	changed	divert.noanswer.5.timeout	60	55	
phone1	changed	divert.noanswer.6.timeout	60	55	
phone1	added	reg.1.server.2.address			See Administrators Guide for SIP 3.2.0 for details
phone1	added	reg.1.server.2.expires			
phone1	added	reg.1.server.2.expires.lineSeize			
phone1	added	reg.1.server.2.expires.overlap			
phone1	added	reg.1.server.2.lcs			
phone1	added	reg.1.server.2.port			
phone1	added	reg.1.server.2.register			
phone1	added	reg.1.server.2.retryMaxCount			
phone1	added	reg.1.server.2.retryTimeOut			
phone1	added	reg.2.musicOnHold.uri			
phone1	added	reg.2.server.1.lcs			
phone1	added	reg.2.server.2.address			
phone1	added	reg.2.server.2.expires			
phone1	added	reg.2.server.2.expires.lineSeize			
phone1	added	reg.2.server.2.expires.overlap			
phone1	added	reg.2.server.2.lcs			
phone1	added	reg.2.server.2.port			
phone1	added	reg.2.server.2.register			

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
phone1	added	reg.2.server.2.retryMax Count			
phone1	added	reg.2.server.2.retryTimeOut			
phone1	added	reg.2.tcpFastFailover			
phone1	added	reg.3.musicOnHold.uri			
phone1	added	reg.3.server.1.lcs			
phone1	added	reg.3.server.2.address			
phone1	added	reg.3.server.2.expires			
phone1	added	reg.3.server.2.expires.line eize			
phone1	added	reg.3.server.2.expires. overlap			
phone1	added	reg.3.server.2.lcs			
phone1	added	reg.3.server.2.port			
phone1	added	reg.3.server.2.register			
phone1	added	reg.3.server.2.retryMax Count			
phone1	added	reg.3.server.2.retryTimeOut			
phone1	added	reg.3.tcpFastFailover			
phone1	added	reg.4.musicOnHold.uri			
phone1	added	reg.4.server.1.lcs			
phone1	added	reg.4.server.2.address			
phone1	added	reg.4.server.2.expires			
phone1	added	reg.4.server.2.expires.line Seize			
phone1	added	reg.4.server.2.expires. overlap			
phone1	added	reg.4.server.2.lcs			
phone1	added	reg.4.server.2.port			
phone1	added	reg.4.server.2.register			
phone1	added	reg.4.server.2.retryMax Count			

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
phone1	added	reg.4.server.2.retryTimeOut			
phone1	added	reg.4.tcpFastFailover			
phone1	added	reg.5.musicOnHold.uri			
phone1	added	reg.5.server.1.lcs			
phone1	added	reg.5.server.2.address			
phone1	added	reg.5.server.2.expires			
phone1	added	reg.5.server.2.expires.line Seize			
phone1	added	reg.5.server.2.expires. overlap			
phone1	added	reg.5.server.2.lcs			
phone1	added	reg.5.server.2.port			
phone1	added	reg.5.server.2.register			
phone1	added	reg.5.server.2.retryMax Count			
phone1	added	reg.5.server.2.retryTimeOut			
phone1	added	reg.5.tcpFastFailover			
phone1	added	reg.6.musicOnHold.uri			
phone1	added	reg.6.server.1.lcs			
phone1	added	reg.6.server.2.address			
phone1	added	reg.6.server.2.expires			
phone1	added	reg.6.server.2.expires.line Seize			
phone1	added	reg.6.server.2.expires. overlap			
phone1	added	reg.6.server.2.lcs			
phone1	added	reg.6.server.2.port			
phone1	added	reg.6.server.2.register			
phone1	added	reg.6.server.2.retryMax Count			
phone1	added	reg.6.server.2.retryTimeOut			

<i>File</i>	<i>Change</i>	<i>Configuration Parameter</i>	<i>Old Value</i>	<i>New Value</i>	<i>Description</i>
phone1	added	reg.6.tcpFastFailover			

Understanding Updates to SIP 3.1.7

The following sections list Polycom tracking numbers and the corresponding changes and enhancements in this release.

New or Enhanced Features

- 61028** Added support for SoundPoint IP 430.
- 61547** The phones now send a 486 (Busy) response to a received INVITE message when a call is rejected.

Enhanced Capabilities

- 51718** Under certain configurations, phone no longer continues to ring after the call has been answered.
- 52968** Deleted instant messages can be removed from the main screen.
- 53975** The phones send a SUBSCRIBE message in a certain scenario when using an SCA with barge-in enabled.
- 55884** The displays on a SoundPoint IP 650 with expansion modules no longer freeze during a consultative transfer.
- 58689** The phones no longer send a 486 if an INVITE is received after a NOTIFY for the alerting state and the configuration parameter `callsPerLineKey` is set to 1.
- 58728** The phone presents the New Call soft key and the End Call soft key to allow the user to release the call and place the phone into idle state after hanging up the call during a consultative transfer.
- 59789** On the SoundPoint IP 650, the user is able to correctly resume a held call after answering a different call.
- 60051** On the SoundPoint IP 650 using a BLA, the display shows the status of the remotely held call while there is an active call currently displayed. Press the Down Arrow key followed by the Up Arrow key to refresh the display and to correctly show the status of the held call.
- 60141** On the SoundPoint IP 650, on a Bridged Line Appearance BLA line, the display incorrectly indicates 2 call appearances when there should only be one for the active call. The second call appearance is for the previously held remote call that is no longer on hold.
- 60145** On the SoundPoint IP 650 using a BLA, the display on the phone correctly presents 2 call appearances instead of only one.
- 60177** The display on the SoundPoint IP 5xx and 6xx presents hot-dialed digits when the idle display feature is enabled.

- 60264** During a call using a BLA line, when the display shows the dialing screen, remote call appearances no longer display when the remote phone's BLA line resumes a call.
- 60340** The Join soft key no longer displays for phones with BLA lines when there is only one call active on the phone.
- 60480** A phone monitoring other BLA lines show the presence (LED goes out) of a BLA line when that monitored line joins two other calls.
- 60756** A phone monitoring a Shared Call Appearance line presents a correct presence indication of a BLA line when that monitored line joins two other calls in a centralized conference.
- 61264** Calls placed on hold using a shared BLA line timeout when a remote phone picks up the held call (on the BLA line).
- 61283** When a user attempts to place a conference call on hold and the phone receives a 400 Bad request. The phone no longer sends a NOTIFY with `<paramname=+sip.rendering pvalue=no />`.
- 61298** When 1.2Mbps of multicast traffic is passed through the PC port on the SoundPoint IP 601 phone, the data port no longer experiences a packet loss of 17%.
- 61299** When a phone has established a centralized conference call, the user is able to transfer a third incoming call.
- 61321** When a phone joins a centralized conference bridge, other monitoring phones correctly show the BLA line as being on hold instead of being in use.
- 61547** The phone sends a 486 Busy message when a call (INVITE) is rejected. A binary configuration parameter is added to **sip.cfg** called `voIpProt.SIP.use486forReject`. By default, (parameter is 0) the feature is disabled. If the parameter is 1, the feature is enabled. If enabled and the phone shows a ringing inbound call appearance, then upon pressing the Reject soft key, the phone transmits a 486 response to the originator of the received INVITE message.
- 61725** Users can pick up a held call after multiple hold/resume interactions on the phone.
- 61950/62024** The phone honors a retry-after header in a 500 Glare message responding to a BLA re-SUBSCRIBE message.
- 62036** The SoundPoint IP 3xx phone continues sending DTMF RTP EVENTS when receiving a second incoming call while it is already active on a previously established call.
- 62050** The SoundPoint IP 650 phone correctly updates the number of held calls after sending 200 OK messages as part of the notifications process.
- 62127** The Blind transfer soft key on the SoundPoint IP 650 is presented on the display when the Transfer soft key is pressed on the second call.
- 62223** The phone no longer crashes after resuming a held call using a BLA.
- 62226** The phone no longer proceeds to join a conference after receiving a 403 Forbidden from the switch.
- 62262** The phone no longer establishes a 1-way audio path after it has re-established a centralized conference call with the dropped third party. This behavior is observed with Sylantro switches.
- 62279** The presence indicator on a Bridged Line Appearance displays correctly after the phone receives a 486 message.

- 62313** Using a BLA configuration, a dial tone is present when pressing the second line key followed by lifting handset after holding a call on first line appearance.
- 62361** The call status on a BLA Bridged Line Appearance (configured for 1 call per line appearance) of a monitoring phone is updated correctly when transfer/conference soft key is pressed.
- 62435** The SoundPoint IP 650 phone correctly displays a call appearance labeled Unknown Party if the remote party is held while reorder tone is played locally.
- 62511** In certain situations, the monitored Busy Lamp Field line invokes an incoming call notification (icon and tone).
- 62514** In certain situations, the status of the monitored Busy Lamp Field lines on the SoundPoint IP 670 is removed from the display even though the status has been updated by the switch.
- 62569** The phone no longer generates a redundant NOTIFY message when triggered by a 100 response during a re-INVITE.
- 62669** When multiple phones try to resume a held Bridge Line Appearance BLA line at the same time, the presence indicator on the BLA line is preserved on the trailing phone when the reorder tone is played.
- 62672** Either Directed Call Pickup DCP or Group Call Pickup feature (through soft keys instead of *53 and *54 feature access codes) no longer fail when the user enters an account code. The account code is appended to the user portion of the URI.
- 62704** The presence indicator of a Bridged Line Appearance BLA is updated correctly on monitoring phones when the phones LAN data cable is disconnected and then re-connected.
- 62926** The Resume soft key on the SoundPoint IP 3xx displays when the line key is pressed continuously while the line is in a remote held call state. This occurs when the line is configured as `callsPerLineKey=1`.
- 63099** The phones monitoring Bridged Line Appearance BLA line, configured for one call per line, can pick up the held call after the call on a BLA line has been put on hold using the Transfer/Conference key.
- 63286** The phone's Part Number is listed correctly instead of YYYY-YYYYY-YYY.
- 64212** Invoking the Call Park feature with the soft key on the SoundPoint IP 3xx functions correctly when the soft key is configured as 1 line and 1 call per line.
- 64219** The SoundPoint IP 3xx phone sends a proper hold NOTIFY message after a consultative transfer is canceled when the configuration parameter `notifyTransferHoldAsActive` is disabled.
- 64271** In an attempt to answer an incoming call, the call is no longer unintentionally terminated. This occurs when the incoming calls line key is pressed simultaneously as the handset is lifted.
- 64274** In an attempt to resume a held call, the held call is no longer unintentionally terminated when the user accidentally seizes two line keys simultaneously.
- 64327** In an attempt to answer an incoming call after the user inadvertently pressed 2 line keys, the user is no longer connected to both lines one with an incoming caller on one and a dial tone on the other.
- 64340** The indicator, on a Bridged Line Appearance BLA line that is monitoring other lines, blink after the monitored phone performs the following sequence: Transfer > Split > EndCall > Resume > Hold.

- 64356** The display on the SoundPoint IP 3xx showing a remote call appearance times out when the user presses continuously a BLA line key followed by pressing a down arrow key while there are multiple calls on hold on the remote BLA.
- 64822** When configuring the SoundPoint IP 3xx phones using **sip_att.cfg**, the phone no longer shows Service Unavailable when the speed dial key is pressed while the phone is off-hook.
- 64862** Joining an internal extension with an external PSTN call no longer causes one call to drop.
- 65119** When a Bridged Line Appearance (BLA) line presents in a dialing screen, the remote call appearance correctly displays when the remote BLA line resumes a call.
- 65207** A slow memory leak due to the receipt of a hunt group INVITE that contains replaces no longer occurs in the SIP stack.
- 67186** All soft keys on the SoundPoint IP 301, 501, and IP 601 no longer disappear on the assistant phone when pressing down the arrow key after placing multiple calls on hold with the boss line appearance.

Configuration File Enhancements

Refer to the following table for a list of enhancements made to software version 3.1.7 configuration file parameters.

Table 21: Software Version 3.1.7 Parameter Enhancements

<i>File</i>	<i>Action</i>	<i>Parameter</i>	<i>Modification Description</i>
sip	added	voIpProt.SIP.use486forReject	Defaults to null
sip	added	call.localConferenceEnabled=1	Defaults to 1

Known Issues and Suggested Workarounds

The following issues are known to be present in the current release. They will be reviewed for possible fixes in a future release if no reasonable workaround is available.

- 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. This issue exists in SIP 1.4.x.

Workaround: Use the correct subnet mask.

- 26920** Centralized conference fails due to a RTP port opening too slowly in some cases.

Workaround: No workaround is currently available.

- 30086** Boot servers running explicit FTPS are not supported.

Workaround: Use implicit FTPS or HTTPS.

- 30371** The pattern generator for tones does not work well in the case of a single repeating chord.

Workaround: Start the pattern with a short period of silence followed by the desired initial chord. Loop back to the desired initial chord instead of the initial silence.

- 33445** LCS Presence and dialing from Buddy Lists does not work across federations.

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

- 37175** If configuration files are used to set the SNTP server address, date validity checking on CA certificates are ignored for https provisioning.

Workaround: Set the SNTP server address through the phone UI or use DHCP to inform the phone of the SNTP server address.

- 37273** If the custom idle display and idle browser features are both enabled, the phone UI displays incorrectly.

Workaround: Do not set `ind.idleDisplay.enabled=1` and enable the Idle Browser at the same time.

- 37984** Enabling the Idle bit-map on SoundPoint IP 330 and 320 phones causes the Line Key labels and dialed digits to be invisible.

Workaround: Do not use the idle bit-map on 330/320 phones; instead, set `ind.idleDisplay.enabled=0`.

- 41993** Scrolling through the Corporate Directory may not return complete results if results contain Unicode character values > 127 (server does not support sorting).

Workaround: Start the search in a different location or avoid use of Unicode characters >127 in directories.

- 42027** In certain scenarios, the time-stamping in log files of a SoundStation IP 7000 that is used as a secondary/slave device is incorrect.

Workaround: As of SIP 3.1.0 the occurrence of this issue only relates to the treatment of Daylight savings Time settings.

- 44764** SRTP processing may cause performance degradation with certain video/audio codec combinations on the VVX 1500.
Workaround: If SRTP is being used, limit the video bit rate to 384 Kbps.
- 46997** Camera brightness adjustment does not work between levels 3 to 6 on the VVX 1500.
Workaround: No workaround is currently available.
- 48905** The jitter parameter is not correctly computed on the SoundStation IP 6000/7000 as per RFC3550.
Workaround: No workaround is currently available.
- 52141** During software upgrades to daisy-chained SoundStation IP 7000 phones, the upgrades sometimes stop.
Workaround: Press any key on the phone to continue the upgrade.
- 52142** Video connections with CounterPath Eyebeam client on the VVX 1500 do not work if H.263-1998 codec is selected on an Eyebeam version 1.5.19.5 build 52345.
Workaround: Try using a different codec. Try other versions of Eyebeam client as some do work.
- 53514** H.264 calls to an HDX 9002 system using an MGC 50 Gateway that uses a H.320 connection results in lip sync issues (*applies to VVX 1500*).
Workaround: Set the call for transcoding on the MGC.
- 54027** The receiving phone does not re-invite with a new key at the half-life of the key life-time.
Workaround: Ensure that both ends use the same key life time so that the sending phone initiates a key re-negotiation.
- 54028** Key changes do not function correctly when multiple crypto suites are enabled.
Workaround: Configure a single crypto suite on the phone.
- 54321** The VVX 1500 does not receive video (does receive audio) when calls are initiated from a Tandberg C20 (running 2.0.0.191232) device using SIP.
Workaround: No workaround is currently available.
- 54799** The VVX 1500 transmits H.264 QCIF video to Tandberg MXPs in H.323 calls.
Workaround: Set the video bit rate on the VVX 1500 to 512 Kbps to avoid the issue.
- 54976** H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway using encrypted media (offered but not required) results in distorted audio and no video on the VVX 1500.
Workaround: Configure system for encryption required.
- 54977** H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway result in lip sync issues on the VVX 1500.
Workaround: No workaround is currently available.
- 59812** Blind transfer to a URL is not successful on the SoundStation IP 7000. Eventually, the URL soft key becomes unavailable.
Workaround: No workaround is currently available.
- 61091** The configuration parameter `tcpIpApp.port.rtp.forceSend=1024` works only for the SoundStation IP 6000, 7000 and VVX 1500. It does not work correctly for SoundPoint IP phones.
Workaround: No workaround is currently available.

- 62387** Adding a new line registration to a phone with BLF causes the notifications (ringing) for the BLF line to display on the previous line. Introduced in UC Software 3.3.1
Workaround: A phone reset resolves the issue.
- 62482** Server certificate Serial Number is checked against the host name if the outbound proxy is configured.
Workaround: No workaround is currently available.
- 63123** Instead of initiating a new call, an attendant phone plays a reorder tone when a BLF line key is pressed for the second time.
Workaround: No workaround is currently available.
- 63527** The phone sends out INVITE and CANCEL messages if no provisional response is received.
Workaround: No workaround is currently available.
- 63609** You cannot answer a call using the speaker soft key when DND is enabled and `call.rejectBusyOnDnd` is set to zero (*applies to SpectraLink 84xx*).
Workaround: No workaround is currently available.
- 66251** British Telecom Caller ID type is not correctly supported (*applies to SoundStation Duo*).
Workaround: No workaround is currently available.
- 68815** The phone does not send a `CallState=CallConference` notification when a conference is established (*applies to all SoundPoint IP and SpectraLink 84xx*).
Workaround: No workaround is currently available.
- 69502** The confirm Click-to-dial text does not appear on the SoundPoint IP 331 phone when SNTP fails.
Workaround: Configure SNTP.
- 69552** The music on hold (MOH) call dialog does not get terminated when there is an update from the MOH server.
Workaround: End the call to restore normal state.
- 69735** When the phone is registered with a H.323 line, DTMF digits are not sent in the Tel URI call with Ext and Postd options (*applies to VVX 500, 1500*).
Workaround: No workaround is currently available.
- 69898** The quick search bar on the SoundPoint IP 321, 330, 331, and 355 only accept 15 characters when the corporate directory is configured.
Workaround: No workaround is currently available.
- 70480** When the phone uses the Polycom Desktop Connector, the keyboard arrow keys do not support active and inactive call navigation (*applies to VVX 500*).
Workaround: No workaround is currently available.
- 70728** Software Upgrade does not work if `<partnumber>.xml` file is not specified as a part of `upgrade.custom.server.url` configuration value.
Workaround: Ensure the part-number.xml file is part of the `upgrade.custom.serverurl` configuration value.
- 71386** Soft key URIs do not function when the phone is in the Enter Number screen (*applies to VVX 1500*).

Workaround: No workaround is currently available.

- 71800** Users cannot change the user password in the Web Configuration Utility.

Workaround: Use the phone's user interface to change the user password.

- 72082** The phones do not detect a server certificate status change from REVOKED to GOOD until the phone is rebooted (*applies to SoundPoint IP 321, 331, 450, 550, 560, 650, and 670, and SoundStation IP 5000.*)

Workaround: No workaround is currently available.

- 72211** An explicitly trusted Intermediate CA fails TLS verification when it is the issuer of a server certificate.

Workaround: No workaround is currently available.

- 72242** The phone cannot connect to a radius server when configured with EAP method as PEAP and inner authentication as GTC (*applies to VVX 500*).

Workaround: Recommend to use Cisco ACS server 5.1 or higher.

- 72299** When the SoundPoint IP 450, 560, and 650 phones are registered with BLA lines, they continue to display remote hold appearances even after the remote BLA resumes the call.

Workaround: No workaround is currently available.

- 72387** After pressing the Transfer soft key, the remote BLA line does not show remote hold status when `call.shared.exposeAutoHolds` is set to 1.

Workaround: No workaround is currently available.

- 72601** The SoundPoint IP 33x phones fail to dial authorized call when in the Phone Locked state.

Workaround: No workaround is currently available.

- 72677** When a NOTIFY message with a higher version is sent, the phone re-subscribes to the server and gets a NOTIFY with the correct version, but fails to update the dialog with the state (*applies to SoundPoint IP 450/560/650*).

Workaround: No workaround is currently available.

- 72898** Hard key external URL mapping requires EFK enabled on the SoundPoint IP 650.

Workaround: Enable EFK using configuration files.

- 73015** The LifeSize Team 220 incorrectly remains in a connecting state when there is a call from VVX 1500 over H323.

Workaround: No workaround is currently available.

- 74120** Plantronics Audio 646 DSP USB headset volume control does not work (*applies to VVX 500*).

Workaround: Adjust the volume using the volume keys on the phone.

- 74533** A phone configured with a Synergy call server displays the incorrect caller ID on the UI for an incoming call (*applies to VVX 1500*).

Workaround: No workaround is currently available.

- 74650** In an active audio-only call between a PC client and a VVX 1500, the far-end video never starts on the PC client when a user presses *Add Video*.

Workaround: No workaround is currently available.

- 74763** The MKC5 key to upload logs does not work (*applies to SoundStation Duo*).

- Workaround:* No workaround is currently available.
- 74901** When the lock feature is enabled after a phone reboot, the emergency/authorized call list does not display when the user tries to place a call using headset/speaker key.
Workaround: No workaround is currently available.
- 74958** When DND is enabled, the phone is missing the call forward message Fwd:<number> (*applies to VVX 500, VVX 1500, and SpectraLink 84xx*).
Workaround: No workaround is currently available.
- 75157** A phone configured with a Synergy call server displays the incorrect soft keys after a “Conference service unavailable” error is shown in UC Software 3.3.3.
Workaround: No workaround is currently available.
- 75195** Hold/Transfer/Conference does not display when the parameter `softkey.feature.basicCallManagement.redundant = 0` (*applies to SoundStation Duo*).
Workaround: No workaround is currently available.
- 75229** A phone configured with a Synergy call server displays the local conference UI when establishing a centralized conference using the Join soft key.
Workaround: No workaround is currently available.
- 75427** The Unified Call Appearance List (UCAL) filtered view times out to the default UCAL view when a user scrolls the filtered list and does not change the focus (*applies to VVX 500*).
Workaround: No workaround is currently available.
- 75591** In the Lync environment, when the user logs out, the phone does not logout all the user login credential-dependent applications.
Workaround: No workaround is currently available.
- 75661** The multi-key combination shortcuts for uploading logs and rebooting the phone sometimes do not work (*applies to VVX 500*).
Workaround: No workaround is currently available.
- 75671** When parking a call from the Favorites menu, the call park input dialog (where users enter a park extension) disappears (*applies to VVX 500*).
Workaround: No workaround is currently available.
- 75759** Numeric data entered using the dialpad on the phone browser cannot be deleted on the dialpad.
Workaround: Use the virtual keyboard.
- 75778** Using Microsoft Lync, if a user dials an invalid extension, the entry is sometimes not logged in the Placed Calls call list.
Workaround: No workaround is currently available.
- 75869** Changing the local contact directory search option from first name to last name and vice versa causes the Restart and Save soft keys to disappear on the phone.
Workaround: Exiting and re-entering the directory.
- 75898** Pressing the App hard key on the phone and trying to dial the highlighted/focused SIP/Tel URI does not work with the microbrowser (*applies to VVX 1500 and VVX 500*).
Workaround: No workaround is currently available.

- 76522** In the hoteling call center feature, the phone does not display the status of the call center when a special character is in the call center name.
Workaround: The call center administrator can set the call center name.
- 76655** Using a star (*) in the dial string on the SoundStation IP 7000 causes the phone to send the star as a dot (.) to HDX systems.
Workaround: Two stars (**) should be used.
- 76753** Removing a BLF line from the server causes the speed dial icon to disappear.
Workaround: Restart or reboot the phone, and the icon re-appears.
- 76881** On a shared call, the reorder tone is not played to the user when a Resume attempt fails.
Workaround: No workaround is currently available.
- 76977** Adding a new registration line changes the BLF-monitored lines label from first/last name to its extension number.
Workaround: Reboot the phone.
- 77039** When PTT is enabled, sender name/ID, updated through the parameter `reg.x.displayname`, does not update during the PPT call.
Workaround: No workaround is currently available.
- 77076** When the XT9 input mode is enabled, the phone displays unmatched UIMA-focused items in the first position during XT9 (PinYin) input.
Workaround: No workaround is currently available.
- 77195** Reboots occur occasionally if the roaming contacts exceed 100 on the SoundPoint IP phones and 200 on the VVX phones.
Workaround: No workaround is currently available
- 78232** During a remote conference pickup on a shared line, the phone does not display the call appearance and call indicator.
Workaround: No workaround is currently available.
- 78340** Sending several MWI NOTIFY messages within a few seconds of each other might cause the phone to reset.
Workaround: Avoid sending multiple MWI messages close together.
- 79634** During paging, the receiving phone displays the MAC address of the sender instead of the caller ID.
Workaround: Try restarting the phone.
- 79735** Changing the language of the phone from German to any other language (other than English) might result in a display of diacritic letters (*applies to VVX 500 and SoundPoint 331*).
Workaround: Try changing the language to English first.
- 80212** In a Lync environment, when the corporate directory and parameter `dir.corp.sortcontrol` are enabled, the contact search does not fetch any contacts.
Workaround: Set the parameter `dir.corp.sortcontrol=0`.
- 80227** The phone does not display the saved name of the contact in the local contact directory.
Workaround: Use the full URI while adding the contacts in the local contact directory.

- 81272** When the held call is transferred to a CX 600 phone, the call is established as a one-way call on the far end.
Workaround: Try hold/resume on the CX 600 to establish a two-way call.
- 81315** The call logs of the first user are available on the phone when a new user logs in without signing out the first user.
Workaround: No workaround is currently available.
- 81968** In a hoteling scenario, the phone displays a non-functional Sign-in soft key when a guest is associated using “search and add a Hoteling host number” for the first time.
Workaround: Try pressing any key to update the soft key correctly.
- 82030** When the Calendar is configured on the phone and the active directory credentials are changed by the user/admin, the phone fails to register to the Lync server.
Workaround: The user needs to register the phone manually with the correct credentials.
- 82043** When a Lync profile is used along with the bootserver, any changes performed to the MAC.cfg file using XML notepad and uploaded to the phone cause the phone to deregister. The xml notepad adds an extra space in the certificate which makes the certificate invalid and causes the phone to deregister.
Workaround: Use VI editor or Edit Plus editor.
- 82212** Immediately answering a call on a phone which is outside the enterprise (remote worker/federation scenario) when the UDP is blocked by a firewall, may result in a reboot (*applies to SoundPoint IP 321/331*).
Workaround: No workaround is currently available.
- 82302** In a CAC (Call Admission Control) scenario, when a call transfer fails from the phone to remote Lync client, the phone is unable to resume the call.
Workaround: Try doing consultative transfer.
- 82401** The call order widget disappears on the phone screen after scrolling through five of the maximum number of calls (24).
Workaround: No workaround is currently available.
- 82873/82877** The phone fails to update its presence state when trying to dial the emergency call number 911.
Workaround: No workaround is currently available.
- 83101** In a federated environment, when the UDP traffic is blocked on the firewall, the phone might fail to connect the calls.
Workaround: No workaround is currently available.
- 83157** The phone does not display the protocol field for the local contacts.
Workaround: No workaround is currently available.
- 83330** In a call center scenario, an incoming call during a guest sign-in displays some non-functional soft keys.
Workaround: No workaround is currently available.
- 83442** The call forward icon continues to display on the phone’s scroll bar when the call forward configuration parameters are added and removed using an XML file.

Workaround: Use the phone's user interface to enable the call forward feature.

- 83782** The phone stays in the active call state and does not move to the idle screen when the far end crashes or powers off during an active call.

Workaround: Reboot or restart the phone.

- 83875** In a conference call scenario, the first phone connected to the conference does not transmit video when joined in a H.323 video conference call to a Cisco SX20 IMCU (*applies to VVX 500 and VVX 600*).

Workaround: No workaround is currently available.

- 83884** The VVX phone displays a gray image when a video call is established with Grandstream Video phone using DMA server.

Workaround: Use a bit rate of 512 Kbps or 384 Kbps, or reduce the packet size to 1200.

- 83888** In a conference call scenario, the first phone connected to the conference does not transmit video when joined in a H.323 conference call to an HDX 8006 system at a bit rate of 768 Kbps.

Workaround: Use any other bit rate except 768 Kbps, for example, 384, 512, and 1024

- 83887/83889** A VSX displays a blank or reduced image in a video call with a VVX when the phone transmits at a bit rate of 384 Kbps or 786 Kbps.

Workaround: Use H.263 video codec with a bit rate greater than 1500 Kbps.

- 83910** An elongated video displays on the phone when it receives video at bit rates greater than 512 Kbps from a Real Presence Mobile client.

Workaround: No workaround is currently available.

- 84061** In a call center scenario, the phone does not display the call center information on the default screen when the VVX Camera is attached.

Workaround: Press the call center info soft key to retrieve the call center information.

- 84103** When the user tries to navigate back from the diagnostics menu, a colored screen appears on the phone (*applies to VVX 300/310*).

Workaround: No workaround is currently available.

- 84125** The phone cannot switch the call mode from audio-video to audio only in SIP protocol when auto-routing is enabled and `feature.audioVideoToggle.enabled=1` is set (*applies to VVX 500 and VVX 600*).

Workaround: Select the SIP protocol manually from the protocol menu to switch the phone from video mode to audio only mode.

- 84179** The phone automatically changes the cursor focus when you try to edit, for the second time, the last characters in any of the network IPs on the phone's UI.

Workaround: No workaround is currently available.

- 84189** An incorrect pop-up error message displays on the phone's screen if the user tries to log in using Pin Authentication when the server is down.

Workaround: No workaround is currently available.

- 84289** When the EDGE server is down, the phone takes slightly longer to establish a call with CX 3000 within the same organization.

Workaround: No workaround is currently available.

-
- 84450** An incorrect pop-up error message "DHCP failed" displays on the phone's screen, instead of "Duplicate IP", when the phone detects a duplicate IP.
Workaround: No workaround is currently available.
- 84598** When a Lync user saves contacts locally on the phone, the contacts display on the screen even after the user signs out and a second user signs in.
Workaround: Reboot the phone after the second user signs in.
- 85011** In a Push-to-Talk (PTT)/Paging scenario and when navigating with the arrow keys, the active page does not go to the waiting state when the Talk soft key is released.
Workaround: Browse through the menu again after releasing the Talk soft key.
- 85154** An error pop-up message does not display on the phone's screen when the user tries to play an unsupported media file on the phone's microbrowser.
Workaround: No workaround is currently available.
- 84795** A pop-up message covers the details view of the contacts on the phone when the user tries to add a contact to favorites (*applies to VVX 300/310*).
Workaround: No workaround is currently available.
- 84774** Calls display in the Call Logs menu according to the logging time.
Workaround: No workaround is currently available.
- 84692** The sign-in pop-up message takes slightly longer (~30s) to display when a Lync user reboots the phone after a few contacts (~15) are pinned to 'frequent contacts' (*applies to VVX 300/310*).
Workaround: No workaround is currently available.
- 86172** Adding, deleting, or editing the BroadSoft directory contact from the phone is not available.
Workaround: No workaround is currently available.
- 85606** Setting the DND presence state from the "UC-One Application" or "My status" menu doesn't set the local DND to ON.
Workaround: No workaround is currently available.
- 86478/88053** In a BToE scenario, placing or receiving video calls from or to Lync 2013 client from the phone is not supported as Polycom phones currently does not support H.264 (Lync) and RTV codecs.
Workaround: No workaround is currently available.
- 86901** In a BToE scenario, the call control window is sometimes not available when there is an active call on the Lync client and the user tries to pair the phone with the computer.
Workaround: No workaround is currently available.
- 87129** The network administrator or user has to manually set the base profile of the phone to Lync before establishing a BToE connection.
Workaround: No workaround is currently available.
- 87214** In a Lync call park scenario, the phone displays "call is parked up by sip URI" for a shorter period, i.e., 2 to 3 seconds, and the call is parked successfully when the log levels are set to 0.
Workaround: No workaround is currently available.
- 87285** In the Bluetooth device-pairing scenario, the phone sometimes does not enable the Bluetooth after it has been reset to default using the file format system option (*applies to VVX 600*).

Workaround: No workaround is currently available.

- 87292** In a BToE Scenario, phone is not updating the manually configured location information of set on the Lync client when the location information is removed from the server.

Workaround: Try to configure the location information manually on the phone.

- 87294** The phone fails to pair with the Plantronics Voyager Pro UC v2 Bluetooth headset for the first time after a reset to factory default or a format of the file system (*applies to VVX 600*).

Workaround: Try to search and pair the headset one more time.

- 87328** In a Lync deployment, placing an outgoing PSTN call from the auto-complete list is not available immediately after a local Lync call transfer.

Workaround: Try selecting the first entry in the auto-complete list.

- 87335** In a BToE scenario, a few audio quality issues occur on the laptop when the default audio is set to USB-IP (Polycom BToE virtual USB device).

Workaround: Remove the Polycom device as default audio device on your PC.

- 87337** In a Lync Shared Line Appearance scenario, the phone does not display “Answered by delegate (name) on behalf of Boss (name)” on the caller’s phone when a delegate answers the incoming call of the boss.

Workaround: No workaround is currently available.

- 87338** In a BToE scenario and while installing the Polycom BToE Connector application, some PCs/laptops request a reboot twice.

Workaround: No workaround is currently available.

- 87342** In a Lync environment, observed that admin phone is displaying the mediation call server URL under call logs when the boss retrieves a parked call and holds it, and the admin picks that held call from his phone.

- 87552** In a BToE scenario, a Lync client reboot occurs when the paired phone does not have the correct timestamp in the absence of NTP server.

Workaround: Ensure that the phone displays the correct date and time before connecting to the PC.

- 87655** In a Lync environment, the phone displays the complete SIP URI for outgoing PSTN calls.

Workaround: No workaround is currently available.

- 87785** In a BToE scenario, issues arise sometimes when the call is answered using the phone and content sharing is enabled using the Lync client.

Workaround: No workaround is currently available.

- 87814** In a Lync call park scenario, the phone’s screen displays two parked call images when the parked call is not retrieved before reaching the maximum timeout.

Workaround: No workaround is currently available.

- 87815** While installing the Polycom BToE Connector application, you may observe two pop-up messages display at the same time.

Workaround: No workaround is currently available.

- 87847** The phone currently plays the same sound for reboot, restart, and calendar notification.

Workaround: No workaround is currently available.

- 87848** In a Lync environment, the phone is unable to place calls when an administrator moves the user from one pool to another in P-preferred-register.
Workaround: Reboot the phone.
- 87892** When auto-answer is enabled, the phone automatically answers the call after two seconds.
Workaround: No workaround is currently available.
- 87908** The Polycom BToE Connector application does not work if the computer is running in IPv6 mode.
Workaround: No workaround is currently available.
- 87909** In a BLF scenario, the monitoring phone's screen sometimes does not time out and return to the idle screen from the filtered view of the monitored party.
Workaround: No workaround is currently available.
- 87976** The trace route menu on the phone displays truncated information on the phone's screen when there are very long strings.
Workaround: No workaround is currently available.
- 87987** When performing a trace route on the phone, the order of the network components, like MAC ID and IP address of the router in the Network Neighbors menu, differs among the phone platforms.
Workaround: No workaround is currently available.
- 88029** When there are more than 250 contacts on the phone and you try to delete contacts from the contact directory in a very quick succession results in a blurred screen (*applies to VVX 500 and VVX 600*).
Workaround: Delete the contacts with a time delay of 3 to 4 seconds.
- 88034** The Polycom BToE Connector application is not supported on a Windows XP platform.
Workaround: No workaround is currently available.
- 88062** In a BToE scenario, the phone does not always fetch the call when BToE pairing is initiated during an active call on Lync client.
Workaround: No workaround is currently available.
- 88139** In a BToE scenario, the phone rings for a couple of seconds before going to the active state when the user answers an incoming call on Lync client and there is already an active call.
Workaround: No workaround is currently available.
- 88174** Creating a mixed environment using UC Software 5.0.0 and previous Lync-supported software versions for a Shared Line Appearance feature is not supported.
Workaround: No workaround is currently available.
- 88182** Placing an outgoing call to a phone which has the simultaneous ring option with a PSTN number displays only the End Call soft key when the media by-pass is enabled on the server and video is enabled on the phone.
Workaround: No workaround is currently available.
- 88209** In the CMA server, the CMAD icon on the phone displays a DND icon instead of the On-Call icon when a call is placed to the buddy.
Workaround: No workaround is currently available.
- 88233** Running the Polycom BToE Connector application on your computer decreases the media volume on YouTube videos in the web browser.

Workaround: On your computer, in the Start menu, select **Control Panel > Hardware and Sound > Sound -> Communications**, and select **Do Nothing**

- 88252** Launching the Polycom BToE Connector application on your computer while a media file is playing on the Windows Media Player pauses the media player.

Workaround: On your computer, in the Start menu, select **Control Panel > Hardware and Sound > Sound -> Communications**, and select **Do Nothing**

- 88254** In a Lync BToE scenario, auto sign-in of the Lync client on the phone is not currently available when the phone is already registered with a different Lync user.

Workaround: No workaround is currently available.

- 88273** Leaving a voicemail directly without first calling the user is not available.

Workaround: No workaround is currently available.

- 88276** In a Shared Line Appearance scenario, the Delegate's phone does not display "On behalf of Boss" when the Delegate answers the Boss's call and the caller transfers the call.

Workaround: No workaround is currently available.

- 88278** In a shared line scenario, the phone does not display the initial incoming call screen pop-up message for the fourth incoming call when there are calls on the remote destination and the parameters `reg.1.linekeys = 2` and `reg.1.callsPerLineKey = 6` are configured.

Workaround: No workaround is currently available.

- 88290** In a server-based DND scenario, the phone displays the DND active state after locking and unlocking when the "DND when locked" option is selected.

Workaround: Press the DND soft key to disable DND.

- 88308** The phone plays the ringtone on the speakerphone for a fraction of a second before playing it on the headset when the user plays a video file from the microbrowser using a headset (*applies to VVX 500*).

Workaround: No workaround is currently available.

- 88309** The phone's micro browser closes when you press the Next and Back soft keys in quick succession while downloading HTML pages that contain audio files (*applies to VVX 300/310 and VVX 400/410*).

Workaround: No workaround is currently available.

- 88313** In a Lync BToE and Call Park scenario, the phone displays "Call Park Service" instead of the parked caller's name when a call is made from a CX300 or Lync client to a BToE-connected phone. The Lync client answers the call and parks it.

Workaround: No workaround is currently available.

- 88618** The line label is not displayed properly when you set a long user name mixed with numbers when the language is set to Arabic.

Workaround: No workaround is currently available.

- 88643** In a Lync Shared Line Appearance scenario, the phone loses the "on behalf of boss" information when a delegate places an on behalf of call and another delegate answers the Boss's call and places it on hold

Workaround: No workaround is currently available.

88678 In a Lync environment, the phone is not updating the presence status as DND when the Lync client is presenting and the Lync client and phone are logged in as the same user.

Workaround: No workaround is currently available.

88749/89308 You need administrator privileges to install the Polycom BToE Connector application.

Workaround: No workaround is currently available.

89004 In a BToE scenario, establishing a local conference is not available when the phone is paired with your computer.

Workaround: You can make conference calls from the Lync client, or you can disable BToE on your phone by pressing the **BToE** soft key and selecting **Disable** before making a conference call on your phone.

89018 Observed voice echo issues at times when the Plantronics EHS headset is used.

Workaround: No workaround is currently available.

89082 The call list icon on the phone is not displayed when “DND when locked” is set in a locked state .

Workaround: No workaround is currently available.

89132 The display name on the phone is truncated during a video call when the language is set to Arabic on the phone.

Workaround: No workaround is currently available.

89393 After successfully upgrading the latest BToE connector application, user has to manually run the application for the first time

Workaround: No workaround is currently available.

89402 Observed few issues when you try repair the Polycom BToE Connector during the new client installation

Workaround: Uninstall the Polycom BToE Connector application on your computer and reinstall the latest BToE application installer 1.1.0.

89543 In a BToE scenario, phone shows “Successfully Paired” pop-up message and remains unusable when it is already signed-in and connected to the Lync client of a different user.

Workaround: No workaround is currently available.

Reference Documents

This section lists all documents referenced in these release notes and other relevant documents.

For information and support for all Polycom voice products and software, and for access to supporting documentation, see [Polycom UC Software Support Center](#).

Polycom UC Software Administrators' Guide

[Polycom UC Software 5.0.1 Administrator's Guide](#)

[Polycom UC Software 4.1.0 Administrator's Guide](#)

Technical Bulletins, Quick Tips, White Papers, and Engineering Advisories

[Feature Profile 84393: Using the Polycom BroadSoft UC-One Application on Polycom VVX Business Media Phones](#)

[White Paper: Configuration File Management on Polycom SoundPoint IP, SoundStation IP, and VVX Phones](#)

[Technical Bulletin 35311: Maintaining Older Polycom Phones Beyond Their Last Supported Software Release](#)

[Technical Bulletin 39358: Using Custom Ringtones on Polycom SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones](#)

[Technical Bulletin 52609: Mutual Transport Layer Security Provisioning Using Microsoft Internet Information Services 6.0](#)

[Technical Bulletin 56449: Polycom SoundPoint IP /SoundStation IP /VVX Software Changes in the Next Release](#)

[Quick Tip 57215: Phone Lock Feature on Polycom Phones Running Polycom UC Software](#)

[Technical Bulletin 60519: Simplified Configuration Enhancements in Polycom UC Software 3.3.0](#)

[Engineering Advisory 64731: Polycom UC Software 4.0.x: Upgrade and Downgrade Methods](#)

[Technical Bulletin 66743: Security Advisory Relating to Denial of Service Vulnerability on Polycom SoundPoint IP and SoundStation IP Phones](#)

[Feature Profile 72430: Using Polycom Phones with Microsoft Lync Server 2010](#)

User Guides

[SoundPoint IP Phones](#)

[SoundStation IP Phones](#)

[VVX Business Media Phones](#)

Miscellaneous

[SIP/UC Software Downloads Matrix](#)