RELEASE NOTES

UC Software 5.4.0 | May 2015 | 3725-49125-001A

Polycom® UC Software 5.4.0

Applies to Polycom VVX® Business Media Phones and Polycom SoundStructure® VoIP Interface

Contents

What's New in Polycom UC Software 5.4.0

Polycom Unified Communications (UC) Software 5.4.0 is a release for all open SIP platforms in addition to Alcatel-Lucent CTS Server and Microsoft® Lync® Server. UC Software 5.4.0 is the first release to support the new VVX 101 and VVX 201 business media phones.

Polycom UC Software 5.4.0 supports the following Polycom endpoints:

- VVX 101/201 business media phones (new phones supported in this release)
- VVX 300/310 business media phones
- VVX 400/410 business media phones
- VVX 500 business media phones
- VVX 600 business media phones
- VVX 1500 business media phones
- SoundStructure VoIP Interface

Polycom UC Software 5.4.0 supports the following Polycom accessories:

- VVX Camera
- VVX Expansion Module

These release notes provide important information on software updates, phone features, and known issues.

New Features

Polycom UC Software 5.4.0 includes the features and functionality of previous releases and includes the following new features:

- Advanced Conference
- Barge In for Busy Lamp Field Lines
- Bridge In for Shared Call Appearance
- Dual-Tone Multi-Frequency (DTMF) Relay
- Shared Call Appearance
- SIP Instance Support
- Visitor Desk Phone
- Comfort Noise
- Opus Codec Support
- DNS Server Address Override
- Global Directory Synchronization
- Basic Menu Lock
- Additional Features in This Release

See the section Configuration File Enhancements for the permitted values and descriptions for each feature's parameters.

Note: Configuring Alcatel-Lucent CTS features

To enable and use any of the listed Alcatel-Lucent CTS-specific features, enter ALU-CTS as the value for the parameter voIpProt.server.1.specialInterop. See the *Administrator Guide for Polycom UC Software 5.4* available on the Polycom Voice Support site for more information.

Advanced Conference

Users registered with the Alcatel-Lucent CTS server can initiate ad-hoc audio conferences with two or more contacts, manage conference participants, view a roster of participants, and join two calls into a conference call. When the Push-to-Conference feature is enabled, users can choose a list of participants to invite to an audio conference.

This feature is not supported on VVX 101, 201, and 1500 phones and SoundStructure VoIP Interface.

This feature is disabled by default. Administrators can configure the following parameters for Advanced Conference:

- `feature.advancedConference.enabled`
- `reg.x.feature.advancedConference.pushToConference`
- `reg.x.advancedConference.maxParticipants`
- `reg.x.advancedConference.subscribeForConfEvents`
- `reg.x.advancedConference.subscribeForConfEventsOnCCPE`

Barge In for Busy Lamp Field Lines

This feature enables users registered with the Alcatel-Lucent CTS server to barge in on active and held calls on Busy Lamp Field (BLF) lines. The following barge-in modes are supported:

- **Normal mode:** Forms a full conference connection with the barged-in party so that all participants have full 2-way communication.
- **Whisper mode:** Enables the barging-in party to speak to the barged-into party, with reduced volume, without the far end hearing the audio of the barged-in party.
- **Listen mode:** Enables the barging-in party to listen to all communication exchanged between the other parties, but other participants cannot hear audio from the barging-in party.

This feature is not supported on VVX 101, 201, and 1500 business media phones and SoundStructure VoIP Interface.

The Barge In feature for BLF lines is disabled by default. Administrators can enable the Barge In feature, the default barge in mode, and if a tone plays when a contact barges in on a call.

Administrators can configure the following parameters for Barge In:

- `attendant.resourceList.x.bargeInMode`
- `attendant.resourceList.x.requestSilentBargeIn`

Bridge In for Shared Call Appearance

This feature enables multiple users in a Shared Call Appearance group registered with the Alcatel-Lucent CTS server to view and bridge into active calls on a shared line. Bridge In is not supported for VVX 101 and 1500 business media phones and SoundStructure VoIP Interface.

This feature is disabled by default. Administrators can enable this feature using the parameter `reg.x.bargeInEnabled`.

Dual-Tone Multi-Frequency (DTMF) Relay

This feature enables users registered with the Alcatel-Lucent CTS server to press DTMF digits during an active audio or conference call to select options or enter codes. The digits are transferred in a SIP INFO messages over the SIP signaling interface. This feature is not supported for H.323 calls.

This feature is disabled by default. Administrators can enable DTMF Relay using the parameter `voIpProt.SIP.dtmfViaSignaling.rfc2976`.

Shared Call Appearance

This feature enables shared line users registered with the Alcatel-Lucent CTS server to monitor and bridge into calls on the shared line. Each line supports 21 call appearances.

This feature is disabled by default. This feature is not supported on VVX 101 and 1500 phones and SoundStructure VoIP Interface.

Administrators can enable the feature and configure the hold request for the line using the following existing parameters:

- `reg.1.address`
- `reg.1.type`
- `reg.1.server.1.specialInterop`

SIP Instance Support

In environments where multiple phones are registered using the same address of record (AOR), the phones are distinguished by their IP address. However, firewalls set up in these environments can change the IP addresses regularly for security purposes. The SIP Instance Support feature provides support for using +`sip.instance`, a parameter used in the contact header, to identify individual phones instead of using IP addresses. This feature complies with RFC 3840.

Administrators can enable this feature using configuration files. The parameter `reg.x.gruu` is a new parameter for this feature.

Visitor Desk Phone

This feature enables users registered with the Alcatel-Lucent CTS server to log into a public phone and access their personal settings. When a user logs in to the public phone, their personal settings are available and any changes the user makes to phone settings are stored. After the user logs out, another user can log in and access their personal settings.

Visitor Desk Phone is not supported on VVX 1500 business media phones and SoundStructure VoIP Interface.

Administrators can configure a common setting for all phones and any user can make calls, including emergency calls, from a phone without having to log in.

Administrators can use the following parameters to configure Visitor Desk Phone:

- `feature.VDP.enabled`
- `prov.vdp.accessCode.login`
- `prov.vdp.accessCode.logout`

Comfort Noise Control

Previously, when Voice Activity Detection was enabled, Comfort Noise was sent using the default payload type of 13, which was not advertised in SDP. Now, when enabled, the Comfort Noise payload type is negotiated in SDP with the default of 13 for 8 KHz codecs and a configurable value between 96 and 127 for 16 KHz codecs.

Administrators can use the following parameters to configure this feature:

- `voice.CNControl`
- `voice.CN16KPayload`

Opus Codec Support

Opus is an adaptive, lossy audio coding format that is suitable for interactive real-time applications over the Internet. This feature also simplifies internetworking between mobile and fixed line networks.

This feature is currently supported on VVX 500 and 600 phones only.

The following are a list of limitations when using the Opus codec on VVX 500 and 600 phones:

- N-way calling is not available.
- Administrators must configure at least one other codec in addition to the Opus codec.
- The Opus codec is not available when you enable video on the VVX 500 or VVX 600 business media phones. If you want to use the Opus codec, disable video using the configuration file parameter `video.enable.VVX500=0` or `video.enable.VVX600=0`. By default, these two parameters are enabled ‘1’, and you must manually set the value to ‘0’ to disable.

Administrators can add the Opus codec to the list of supported codecs using the Web Configuration Utility under the **Settings > Codec Priorities** menu.

DNS Server Address Override

Previously, any DNS server address provided by DHCP is given the highest precedence even if an administrator manually enters a DNS server address. This feature enables an administrator to override the DHCP option and enter any DNS server address or domain.

Administrators can use the following parameters to configure this feature:

- `tcpIpApp.dns.address.overrideDHCP`
- `tcpIpApp.dns.domain.overrideDHCP`

Global Directory Synchronization

The Polycom Contact Directory available on Polycom phones uses two files to generate and maintain a directory. Previously, the Global Directory was loaded onto the phone only once, from the 00000000-directory.xml file, and was copied into a phone specific Personal Directory file (<MAC>-directory.xml). Thereafter, only the Personal Directory file was loaded or modified by the phone. This made it difficult to make changes to the Global Directory and have those changes updated on all phones without manually editing each Personal Directory file.

This feature changes the way directory files are managed and used by VVX phones. It enables administrators to update the Global Directory file and have that update apply to all phones on the network.

With this release, the phone will no longer copy the entire Global Directory file to the Personal Directory file. The Personal Directory file will only contain contacts that have are edited by the user. Any new or modified contacts in the Global Directory file are saved in the Personal Directory file and uploaded to the server. The Personal Directory xml file will not contain any unmodified contacts that come from the Global Directory file.

Both the Global and Personal Directory files are downloaded to the phone after each restart or upon receipt of a `checksync NOTIFY` message. The content of the Global Directory file will be combined with the content of the Personal Directory file for display and use on the phone. Any changes to either the Global or Personal Directory files are reflected in the directory on the phone. When merging the two files, the Personal directory will always take precedence. Thus, if a user modifies a contact from the Global Directory, the contact is saved in the Personal Directory file, so when the files are next uploaded, the contact from the Global Directory is ignored and the Personal Directory version is used instead.

Using the new parameter `voIpProt.SIP.specialEvent.checkSync.downloadDirectory`, administrators can configure the phone to download the updated directory files upon receipt of a `checksync NOTIFY` message. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates.

Basic Menu Lock

This feature enables administrators to lock the Basic menu under Settings and prevent users from customizing the phones on the network. If enabled, user will require a password to access Basic settings. This feature is disabled by default.

The parameter `up.basicSettingsPasswordEnabled` is new for this feature.

Additional Features in This Release

The following feature enhancements were made for this UC Software 5.4.0 release:

- Added configuration parameters to enable or disable the display of icons on the Home screen. See the section Configuration File Enhancements for a list of parameters for the Home screen icons.

- Added a Redial soft key and Home screen icon, and added parameters to configure the display of the soft key and icon. See the section Configuration File Enhancements for the parameter for this soft key.
- Removed the New Call, Redial, Forward, MyStatus, and Contacts soft keys from the phone when a user is not signed into Lync.
- Replaced the X and Exit soft keys with the Back soft key consistently throughout all screens on the VVX 500 and 600 phones.
- Replaced the Reject Waiting Call menu option with Call Waiting menu option, and replaced the parameter `call.rejectOnNoCallWaiting` with the `call.callWaiting.enabled` parameter.
- Added the MOSCQ and MOSLQ scores for the Voice Quality Monitoring (VQMon) feature to Media Statistics screen on the phone.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.4.0 for release.

Note: Using configuration parameters to enable features

For more information on using configuration parameters to enable or disable features, see the *Administrator Guide for Polycom UC Software 5.4* available on the Polycom Voice Support site.

Configuration File Enhancements

Parameter	*Permitted Values*	*Default*
attendant.resourceList.x.bargeInMode	**All, Normal, Listen, Whisper**	**None**
Enables Barge In feature for phones registered with the Alcatel-Lucent CTS serve, and chooses the default Barge In mode. **Note**: The default value for this parameter is empty. If no value is entered, the Barge In feature is disabled.		
attendant.resourceList.x.requestSilentBargeIn	**0 or 1**	**0**
Plays a tone when a contact barges in on a call. If 0, a tone plays when a contact barges in on a call. If 1, no tone is played when a contact barges in on a call.		
bossLine.xAdminURI	**string**	**null**
Specify the URI of a Boss contact you set a ring type for using `bossLine.x.RingType`.		
bossLine.x.RingType	**default, ringer1 to ringer24**	**ringer2**
Specify a ring type for a Boss contact.		

Parameter	*Permitted Values*	*Default*
call.callWaiting.enabled	**0 or 1**	**1**
Enables or disables incoming calls during active calls. If set to 1, incoming calls during an active call are allowed and display on the Incoming Call screen. If set to 0, incoming calls are rejected, the phone plays a busy tone, and the Incoming Call screen does not display.		
dhcp.option43.override.stsUri	**URI**	**null**
If enabled, the phone displays the PIN Authentication menu. If disabled and DHCP Option 43 is not used, the phone does not display the PIN Auth menu and the PIN Auth menu in the Web Configuration Utility is not available.		
feature.advancedConference.enabled	**0 or 1**	**0**
Enables and disables advanced conferences and conference controls for ALU. If enabled, conference controls are displayed during conferences. If disabled, conference controls are not displayed during conferences and the Roster continues to display.		
feature.scap.HoldRequestUriUserPart	**string**	**SCAP-Hold**
Specifies the Hold request for Shared Call Appearance calls to the ALU server. This value must match the value configured on ALU server for SCA hold request.		
feature.scap.defCallTypeExclusive	**0 or 1**	**0**
Controls the default behavior of a Shared Call Appearance call. By default, an outgoing call from the call group is private. After the call is answered, the user needs to press Share soft key to make the call public so that other people on the line can bridge in to the call.		
feature.VDP.enabled	**0 or 1**	**0**
If 1, VDP is enabled and the phone displays the Visitor Login soft key. If 0, VDP is disabled and the phone does not display the Visitor Login soft key.		
homeScreen.calendar.enable	**0 or 1**	**1**
Enables or disables the display of the Calendar icon on the Home screen.		
homeScreen.directories.enable	**0 or 1**	**1**
Enables or disables the display of the Directories menu icon on the Home screen.		
homeScreen.features.enable	**0 or 1**	**1**
Enables or disables the display of the Features menu icon on the Home screen.		
homeScreen.messages.enable	**0 or 1**	**1**
Enables or disables the display of the Messages menu icon on the Home screen.		
homeScreen.newCall.enable	**0 or 1**	**1**
Enables or disables the display of the New Call icon on the Home screen.		
homeScreen.redial.enable	**0 or 1**	**1**
Enables or disables the display of the Redial icon on the Home screen.		

Parameter	*Permitted Values*	*Default*
homeScreen.settings.enable	**0 or 1**	**1**
Enables or disables the display of the Settings menu icon on the Home screen.		
homeScreen.status.enable	**0 or 1**	**1**
Enables or disables the display of the Status menu icon on the Home screen.		
prov.vdp.accessCode.login	**string**	***771**
Specify the VDP login service access code.		
prov.vdp.accessCode.logout	**string**	***772**
Specify the VDP logout service access code.		
reg.x.advancedConference.maxParticipants	**3-25**	**3**
Specifies the maximum number of participants allowed in a push to conference.		
reg.x.advancedConference.subscribeForConfEvents	**0 or 1**	**1**
Enables the conference participants to receive notifications for conference events.		
reg.x.advancedConference.subscribeForConfEventsOnCCPE	**0 or 1**	**1**
Enables the conference host to receive notifications for conference events.		
reg.x.advancedConference.pushToConference	**0 or 1**	**0**
Enables and disables the push to conference functionality for advanced conferences for ALU. If enabled, users can select multiple contacts when initiating a conference and during an active conference. **Note**: The values for this parameter must match with the value configured on the server.		
reg.x.bridgeInEnabled	**0 or 1**	**0**
Enables or disables the Bridge In feature.		
reg.x.gruu	**0 or 1**	**0**
Specify if the phone sends sip.instance in the REGISTER request.		
softkey.feature.redial	**0 or 1**	**0**
Enables or disables the display of the Redial soft key on the Home screen. **Note**: The parameter `feature.enhancedFeatureKeys.enabled` must be set to 1 first to configure this feature, and the parameter `efk.softkey.alignleft` must be set to 1 to move enabled soft keys into the positions of disabled soft keys.		
tcpIpApp.dns.address.overrideDHCP	**0 or 1**	**0**
When set to 0, a DNS address is requested from the DHCP server. When set to 1, a DNS primary and secondary address are set using the parameters `tcpIpApp.dns.server` and `tcpIpApp.dns.altServer`.		

Parameter	*Permitted Values*	*Default*
tcpIpApp.dns.domain.overrideDHCP	**0 or 1**	**0**
When set to 0, a domain name is retrieved from the DHCP server, if one is available. When set to 1, the DNS domain name is set using the parameter `tcpIpApp.dns.domain`.		
tcpIpApp.port.rtp.mediaPortRangeEnd	**Default, 1024 to 65485**	**2269**
Choose the maximum supported end range of audio ports.		
tcpIpApp.port.rtp.videoPortRangeEnd	**Default, 1024 to 65535**	**2319**
Choose the maximum supported end range of video ports.		
Up.SLA.ringType	**default, ringer1 to ringer24**	**ringer2**
Specify a ring type for an SLA line.		
up.basicSettingsPasswordEnabled	**0 or 1**	**0**
If enabled, a password is required for access the Basic settings menu on the phone. If set to 1, the Basic menu requires a password to access. If set to 0, no password is required to access the Basic settings menu.		
voice.CNControl	**0 or 1**	**0**
Publishes support for Comfort Noise in the SDP body of the INVITE message and includes the supported comfort noise payloads in the media line for audio. If set to 1, either the payload type 13 for 8 KHz sample rate audio codec is sent for Comfort Noise, or the dynamic payload type for 16 KHz audio codecs are sent in the SDP body.		
voice.CN16KPayload	**96 to 127**	**122**
Alters the dynamic payload type used for Comfort Noise RTP packets.		
voIpProt.SIP.specialEvent.checkSync.downloadDirectory	**0 or 1**	**1**
When set to 1, the phone downloads the directory file along with any software and configuration updates. When set to 0, the phone only downloads software and configuration updates.		
voIpProt.SIP.dtmfViaSignaling.rfc2976	**0 or 1**	**0**
Enables and disables DTMF relays for active SIP calls. Not supported for H.323 calls.		
voIpProt.server.1.specialInterop	**standard, lcs2005, ocs2007r2, lync2010, GENBAND, GENBAD-A2, or ALU-CTS**	**standard**
Enables server-specific features. Set to ALU-CTS to enable Alcatel-Lucent features.		

Install UC Software 5.4.0

Consider the following installation and update information when using Polycom UC Software 5.4.0.

Caution: Updating VVX 1500 to UC Software 5.4.0

Before updating your VVX 1500 phone to UC Software 5.4.0, make sure that the phone is upgraded to BootBlock 3.0.4. See *Technical Bulletin 695: Upgrading the Polycom VVX 1500 Business Media Phone to UC Software 5.2.0* for more information.

Download the Distribution Files

To download UC Software 5.4.0, you can choose the combined UC Software package or the split UC Software package, both in ZIP file format. The combined version contains all files for all phone models. The split software package is smaller, downloads more quickly, and contains sip.ld files for each phone model, enabling you to choose provisioning software for your phone model and maintain software versions for each model in the same root directory.

For general use, Polycom recommends using the split resource file that corresponds to the phone models for your deployment. To match the correct UC software resource file to your phone model, see the table Understand the Combined ZIP and Split ZIP 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 5.4.0.5841**.

Understand the Combined and Split ZIP Files

To understand the files distributed in the combined ZIP file, refer to the following table.

Understand the Combined ZIP and Split ZIP Files

Distributed Files	*File Purpose and Application*	*Combined ZIP*	*Split SIP*
3111-40250-001.sip.ld	SIP application executable for VVX 101	x	✓
3111-40450-001.sip.ld	SIP application executable for VVX 201	x	✓
3111-46135-002.sip.ld	SIP application executable for VVX 300	x	✓
3111-46161-001.sip.ld	SIP application executable for VVX 310	x	✓
3111-46157-002.sip.ld	SIP application executable for VVX 400	x	✓
3111-46162-001.sip.ld	SIP application executable for VVX 410	x	✓
3111-44500-001.sip.ld	SIP application executable for VVX 500	x	✓

Distributed Files	*File Purpose and Application*	*Combined ZIP*	*Split SIP*
3111-44600-001.sip.ld	SIP application executable for VVX 600	x	✓
2345-17960-001.sip.ld	SIP application executable for VVX 1500	x	✓
3111-33215-001.sip.ld	SIP application executable for SoundStructure VoIP Interface	x	✓
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 (zeroes) with the MAC address of the phone and remove the ~ (tilde) from the file name	✓	✓
applications.cfg	Configuration parameters for microbrowser and browser applications	✓	✓
features.cfg	Configuration parameters for telephony features	✓	✓
firewall-nat.cfg	Contains configuration parameters for telephony features	x	✓
H323.cfg	Configuration parameters for the H.323 signaling protocol	✓	
lync.cfg	Contains Lync specific configuration parameters	x	✓
pstn.cfg	Contains parameters for PSTN use	x	✓
reg-advanced.cfg	Contains configuration parameters for the line and call registration and advanced phone feature settings	✓	✓
reg-basic.cfg	Configuration parameters for line and call registration and basic phone settings	✓	✓
region.cfg	Configuration parameters for regional and localization settings such as time and date and language	✓	✓
sip-basic.cfg	Configuration parameters for the VoIP server and softswitch registration	✓	✓
sip-interop.cfg	Configuration parameters for the VoIP server, softswitch registration, and interoperability configuration	✓	✓
site.cfg	Configuration parameters that are set for each site	✓	✓

Distributed Files	*File Purpose and Application*	*Combined ZIP*	*Split SIP*
video.cfg	Configuration parameters for video connectivity	✓	x
video-integration.cfg	Configuration parameters for SoundStation IP 7000 and Polycom HDX system integration	✓	x
VVX-dictionary.xml	Includes native support for the following languages: • Chinese, Traditional • Chinese, Simplified • Danish, Denmark • Dutch, Netherlands • English, Canada • English, United Kingdom • English, United States • French, Canada • French, France • German, Germany • Italian, Italy • Japanese, Japan • Korean, Korea • Norwegian, Norway • Polish, Poland • Portuguese, Brazil • Russian, Russia • Slovenian, Slovenia • Spanish, Spain • Swedish, Sweden • Arabic, UAE	✓	✓
Welcome.wav	Startup welcome sound effect	✓	✓
LoudRing.wav	Sample loud ringer sound effect	✓	✓
Warble.wav	Sample ringer sound effect	✓	✓

Known Issues

The section lists the known issues and suggested workarounds for this release and previous releases.

Known Issues and Suggested Workarounds for UC Software 5.4.0

Category	*Issue No.*	*Release*	*Description*	*Workaround*
BroadSoft	VOIP-95821	5.4.0	In a BroadSoft BroadWorks environment with Barge In enabled, the number of the transferred party is not displayed after the call is transferred and answered.	No workaround is currently available.
BTOE	VOIP-101207	5.4.0	With BTOE enabled and paired, an existing participant is not added to a Lync conference call after rejoining the call immediately for the second time.	No workaround is currently available.
Functionality	VOIP-101656	5.4.0	Meetings initiated from a remote federated location do not display in the Recent Calls list on the phone.	No workaround is currently available.
Functionality	VOIP-101940	5.4.0	For VVX 1500 phones in an Alcatel-Lucent-CTS environment, the phone reboots continuously after restarting when the provisioning server is unavailable or an invalid IP address of the provisioning server is entered on the phone.	Check the status of the provisioning server. The phone will recover when the server is reachable.
Functionality	VOIP-102249	5.4.0	The message “Old password is incorrect when you change the user password more than once without logging out first.	Log out of the phone, change the password, log into the phone with the new password.
Lync	VOIP-100154	5.4.0	In an outage situation in a Lync environment, remote notifications are not sent by the server to users of a shared line, and they are not aware of active calls on the Shared Line Appearance group.	No workaround is currently available.
Lync	VOIP-102237	5.4.0	In a Lync CCCP call, you cannot add a mobile number to a federated Skype meeting using your phone.	A mobile number to the meeting using the Lync client.
Microsoft	VOIP-100998	5.4.0	You cannot add or delete contacts on the phone when Exchange Web Service is enabled and a user is singed into the phone with SSI credentials.	No workaround is currently available.
User Interface	VOIP-101095	5.4.0	The Private Ring Type option displays on the phone when a phone is not registered to Lync or the Lync user is does not have a private line.	No workaround is currently available.

Category	Issue No.	Release	Description	Workaround
User Interface	VOIP-101249	5.4.0	The SIP submenu displays under the Call Server Configuration menu on the phone when a phone only supports SIP.	No workaround is currently available.
User Interface	VOIP-102245	5.4.0	When you press the Back icon on the Contact Directory screen, the Lines screen displays instead of the Directories screen.	No workaround is currently available.

Server Related Known Issues and Suggested Workarounds for UC Software in BroadSoft and Lync Deployments

Category	Issue No.	Release	Description	Workaround
Functionality	VOIP-90875	4.0.5	In a shared call scenario, the user who barged into the conversation is unable to start recording when the primary user started and stopped the call recording and when the phones are configured in “on demand” mode (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91274	5.1.0	The end-to-end video transmission pauses when the user starts or stops the call recording and the phone is configured in “on demand” mode (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91286	5.1.0	The phone fails to start call recording on a held call when configured in “on demand” mode (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91287	5.1.0	Recording is resumed automatically when a phone transfers the call to the 3rd party (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91393	5.1.0	Call is recorded only on a single phone if the "Start" recording is pressed on two phones at the same time (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91440	5.1.0	The phone fails to record PSTN/GSM calls as the server is sending the record: off attribute instead of the record: on attribute (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91465	5.1.0	In a shred call scenario, audio is dropped when the multiple video enabled destinations barge in to a call (BroadSoft R20 server).	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-91560	5.1.0	Server is not sending "recordpref: off" during the SCA hold-resume scenario when the recording is stopped (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91607	5.1.0	Centralized conference fails sometimes when the recording mode is enabled on the phone (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-91781/ 91768/91634	5.1.0	A few issues are observed in BroadSoft call recording during call transfer scenarios and shuffle recording at both ends (BroadSoft R20 server).	No workaround is currently available.
Functionality	VOIP-92163	5.1.0	Video-enabled phones are unable to blind transfer the barge-in enabled conference call (BroadSoft R20 server issue).	No workaround is currently available.
Lync	VOIP-90516	5.0.1	In the Lync Boss-Admin scenario, phones fail to connect to the call when administrators of both the parties are trying to pick up the held calls of their respective bosses (Lync Server).	No workaround is currently available.
Lync	VOIP-90534	5.0.1	In the Lync Boss-Admin scenario, administrators are unable to pick up the held boss call simultaneously at the same time (Lync Server).	No workaround is currently available.
Lync	VOIP-90700	5.0.1	In the Lync Boss-Admin scenario, boss is not showing up on the remote call notification when the administrator has maximum “on-behalf-of” calls on hold (Lync Server).	No workaround is currently available.
Lync	VOIP-91925	5.1.0	In a Lync Boss-Admin scenario, there is no remote active notification on Boss or Admin when the phone is registered with a secondary server in case of outage (Lync Server).	No workaround is currently available.
Lync	VOIP-91926	5.1.1	The phone gets unregistered during a data center outage while the administrator is on a federation call (Lync Server).	No workaround is currently available.
Lync	VOIP-91972	5.0.2	In a Lync Boss-Admin scenario, Boss-Admin indications do not work after failover/failback (Lync Server).	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Lync	VOIP-92034	5.0.1	In case of a data center outage, the boss is unable to pick up on-behalf-of calls made by the administrator (Lync Server).	No workaround is currently available.
User Interface	VOIP-91441	5.1.0	The phone displays a misleading "Call Recording Stopped" message when the user starts call recording if the simultaneous ring feature is enabled and the phone is configured in "On Demand" mode. (BroadSoft R20 server)	No workaround is currently available.

Known Issues and Suggested Workarounds for Previous Releases

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Microsoft	VOIP-95271	5.1.2	RealPresence Group Series systems registered to Polycom DMA solution might not disconnect properly from Lync calls after the call is ended on a VVX business media phone.	No workaround is currently available.
Audio	VOIP-92271	5.2.0	A dial tone mixed with page audio is played from the handset and chassis until the dial tone expires when a user switches the termination and the page is in progress.	No workaround is currently available.
Audio	VOIP-94973	5.2.0	In a Lync Meet now conference, noise is still heard for all participants if one party is muted and a Lync client, a VVX phone, and a SoundStructure VoIP Interface is on a call	No workaround is currently available.
Audio	VOIP-97698	5.3.0	When a call is placed to a VVX phone from a mobile Lync client on an Android phone, and the call is transferred from the VVX phone to another mobile Lync client on an Android phone, audio is heard on one of the Android phones only.	No workaround is currently available.
BroadSoft	VOIP-99060	5.3.0	In a BroadSoft environment, when Do Not Disturb (DND) is enabled for a line and the line becomes unregistered due to server unavailability, DND is enabled for all registered lines.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
BroadSoft	VOIP-99158	5.3.0	When contacts are added as favorites in the BroadSoft UC-One client, the contacts display in the UC-One Contacts group but not on the Lines screen.	No workaround is currently available.
BToE	VOIP-86478 VOIP-88053	5.0.0.	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.	No workaround is currently available.
BToE	VOIP-86901	5.0.0.	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.	No workaround is currently available.
BToE	VOIP-87292	5.0.0.	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.	Try to configure the location information manually on the phone.
BToE	VOIP-87552	5.0.0	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.	Ensure that the phone displays the correct date and time before connecting to the PC.
BToE	VOIP-87785	5.0.0	In a BToE scenario, issues arise sometimes when the call is answered using the phone and content sharing is enabled using the Lync client.	No workaround is currently available.
BToE	VOIP-87908	5.0.0	The Polycom BTOE Connector application does not work if the computer is running in IPv6 mode.	No workaround is currently available.
BToE	VOIP-88034	5.0.0	The Polycom BTOE Connector application is not supported on a Windows XP platform.	No workaround is currently available.
BToE	VOIP-88062	5.0.0	In a BToE scenario, the phone does not always fetch the call when BToE pairing is initiated during an active call on Lync client.	No workaround is currently available.
BToE	VOIP-88233	5.0.0	Running the Polycom BTOE Connector application on your computer decreases the media volume on YouTube videos in the web browser.	On your computer, in the Start menu, select Control Panel > Hardware and Sound > Sound -> Communications, and select Do Nothing

Category	*Issue No.*	*Release*	*Description*	*Workaround*
BToE	VOIP-88252	5.0.0	Launching the Polycom BTOE Connector application on your computer while a media file is playing on the Windows Media Player will pause the media player.	On your computer, in the Start menu, select Control Panel > Hardware and Sound > Sound > Communications, and select Do Nothing
BToE	VOIP-88749 VOIP-89308	5.0.1	You need administrator privileges to install the Polycom BTOE Connector application.	No workaround is currently available.
BToE	VOIP-89543		In a BToE scenario, the phone displays the message “Successfully Paired”, and is unusable when the phone is already signed-in and connected to the Lync client of a different user.	No workaround is currently available.
BToE	VOIP-93272	5.2.0	In a BToE scenario, the phone displays a “BToE unpaired” pop-up instead of a “Successfully Un-paired” pop-up after a PC port link is unplugged.	No workaround is currently available.
Configuration	VOIP-48905		The jitter parameter is not correctly computed on the SoundStation IP 6000/7000 as per RFC3550.	No workaround is currently available.
Configuration	VOIP-61091	SIP 3.3.0	The configuration parameter `tcpIpApp.port.rtp.forceSend` set to 1024 works only for the SoundStation IP 6000, 7000 and VVX 1500. It does not work correctly for SoundPoint IP phones.	No workaround is currently available.
Configuration	VOIP-70728	4.0.2	Software Upgrade does not work if `<partnumber>.xml` file is not specified as a part of upgrade.custom.server.url configuration value.	Ensure the `part-number.xml` file is part of the `upgrade.custom.serverurl` configuration value.
Configuration	VOIP-72898	4.0.0	Hard key external URL mapping requires EFK enabled on the SoundPoint IP 650.	Enable EFK using configuration files.
Configuration	VOIP-75195	4.0.1.	The Hold, Transfer, and Conference soft keys do not display when the parameter `softkey.feature.basicCallManagement.redundant` is set to 0 (applies to SoundStation Duo).	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Configuration	VOIP-77039	4.0.2	When PTT is enabled, sender name/ID, updated through the parameter `reg.x.displayname`, does not update during the PPT call.	No workaround is currently available.
Configuration	VOIP-77076		When the XT9 input mode is enabled, the phone displays unmatched UIMA-focused items in the first position during XT9 (PinYin) input.	No workaround is currently available.
Configuration	VOIP-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.	Register the phone manually with the correct credentials.
Configuration	VOIP-98825	5.3.0	In an Enhanced Call Park scenario, the phone reboots when the user logs in and out with a parameter misconfigured when a call is parked against the number.	Reboot the phone.
Configuration	VOIP-98992	5.3.0	When two lines are registered with the same BLF line, the phone does not display the configured BLF line key when the flexible line key feature is enabled and the parameter `attendant.resourceList.x.address` is not in sequential order.	No workaround is currently available.
Expansion Modules	VOIP-99188	5.3.0	Sometimes when a line key for a favorite is pressed on the VVX Expansion Module, the phone displays the Contact Information instead of placing a call to the contact.	No workaround is currently available.
Functionality	VOIP-37175		If configuration files are used to set the SNTP server address, date validity checking on CA certificates are ignored for HTTPS provisioning.	Set the SNTP server address through the phone UI or use DHCP to inform the phone of the SNTP server address.
Functionality	VOIP-46997		Camera brightness adjustment does not work between levels 3 to 6 on the VVX 1500.	No workaround is currently available.
Functionality	VOIP-54027		The receiving phone does not re-invite with a new key at the half-life of the key life-time.	Ensure that both ends use the same key life time so that the sending phone initiates a key re-negotiation.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-54028	SIP 3.2.2	Key changes do not function correctly when multiple crypto suites are enabled.	Configure a single crypto suite on the phone.
Functionality	VOIP-54799	SIP 3.2.2	The VVX 1500 transmits H.264 QCIF video to Tandberg MXPs in H.323 calls.	Set the video bit rate on the VVX 1500 to 512 Kbps to avoid the issue.
Functionality	VOIP-66251		British Telecom Caller ID type is not correctly supported (applies to SoundStation Duo).	No workaround is currently available.
Functionality	VOIP-68815	4.0.0	The phone does not send a CallState=CallConference notification when a conference is established (applies to all SoundPoint IP and SpectraLink 84xx).	No workaround is currently available.
Functionality	VOIP-69502	3.3.1	The confirm Click-to-dial text does not appear on the SoundPoint IP 331 phone when SNTP fails.	Configure SNTP.
Functionality	VOIP-69552	3.3.1	The music on hold (MOH) call dialog does not get terminated when there is an update from the MOH server.	End the call to restore normal state.
Functionality	VOIP-69735	4.4.0	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 and 1500).	No workaround is currently available.
Functionality	VOIP-71800		Users cannot change the user password in the Web Configuration Utility.	Change the user password on the phone.
Functionality	VOIP-72082	4.0.0	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).	No workaround is currently available.
Functionality	VOIP-72211		An explicitly trusted Intermediate CA fails TLS verification when it is the issuer of a server certificate.	No workaround is currently available.
Functionality	VOIP-72299	3.3.1.	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.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-72387	3.3.2.	After pressing the Transfer soft key, the remote BLA line does not show remote hold status when call.shared.exposeAutoHolds is set to 1.	No workaround is currently available.
Functionality	VOIP-72677	3.3.2.	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).	No workaround is currently available.
Functionality	VOIP-73015	4.0.0	The Life Size Team 220 incorrectly remains in a connecting state when there is a call from VVX 1500 over H323.	No workaround is currently available.
Functionality	VOIP-75049	4.0.1B	When using on-hook dialing, the SoundStructure VoIP Interface will not indicate certain call states through the `voip_call_appearance_state` parameter message. These states include `Dialtone`, `Setup`, and `Overlap`.	No workaround is currently available.
Functionality	VOIP-75157	3.3.2.	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.	No workaround is currently available.
Functionality	VOIP-75427	4.0.1	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).	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-75614	4.0.1B	After a SoundStructure and/or SoundStructure VoIP Interface reboot, the `voip_line_state` parameter message may return "line_not_registered" for one or more lines even though the line is successfully registered (applies to SoundStructure VoIP Interface).	Send the following commands to the SoundStructure for each unregistered line: set voip_line "VoIP Out" <line-number> set phone_connect "VoIP Out" 0 For instance, if line 3 is showing as unregistered, but it normally was registered, then send the following two commands: set voip_line "VoIP Out" 3 set phone_connect "VoIP Out" 0
Functionality	VOIP-75661		The multi-key combination shortcuts for uploading logs and rebooting the phone sometimes do not work (applies to VVX 500).	No workaround is currently available.
Functionality	VOIP-75671	4.0.1	When parking a call from the Favorites menu, the call park input dialog (where users enter a park extension) disappears (applies to VVX 500).	No workaround is currently available.
Functionality	VOIP-75898	4.0.1	Pressing the App hard key on the phone and trying to dial the highlighted/focused SIP/Tel URI does not work with the micro browser (applies to VVX 1500 and VVX 500).	No workaround is currently available.
Functionality	VOIP-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.	Use two stars (**).
Functionality	VOIP-76881		On a shared call, the reorder tone is not played to the user when a Resume attempt fails.	No workaround is currently available.
Functionality	VOIP-76977	4.0.1	Adding a new registration line changes the BLF-monitored lines label from first/last name to its extension number.	Reboot the phone.
Functionality	VOIP-79634	4.0.4.	During paging, the receiving phone displays the MAC address of the sender instead of the caller ID.	Restart the phone.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-81272	4.1.0	When the held call is transferred to a CX600 phone, the call is established as a one-way call on the far end.	Hold and resume the call on the CX600 to establish a two-way call.
Functionality	VOIP-81315	4.1.0	The call logs of the first user are available on the phone when a new user logs in without signing out the first user.	No workaround is currently available.
Functionality	VOIP-82873 VOIP-82877	4.1.2	The phone fails to update its presence state when trying to dial the emergency call number 911.	No workaround is currently available.
Functionality	VOIP-83782	4.1.6	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.	Reboot or restart the phone.
Functionality	VOIP-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).	No workaround is currently available.
Functionality	VOIP-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.	Use any other bit rate except 768 Kbps, for example, 384, 512, and 1024
Functionality	VOIP-84125		The phone cannot switch the call mode from audio-video to audio only in SIP protocol when auto-routing is enabled and the parameter `feature.audioVideoToggle.enabled` is set to 1 (applies to VVX 500 and VVX 600).	Select the SIP protocol manually from the protocol menu to switch the phone from video mode to audio only mode.
Functionality	VOIP-84289	Updater 5.1.2	When the EDGE server is down, the phone takes slightly longer to establish a call with CX 3000 within the same organization.	No workaround is currently available.
Functionality	VOIP-84774	4.1.4	Calls display in the Call Logs menu according to the logging time.	No workaround is currently available.
Functionality	VOIP-84795	4.1.4	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).	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-85606	4.3.1	Setting the DND presence state from the "UC-One Application" or "My status" menu doesn't set the local DND to ON.	No workaround is currently available.
Functionality	VOIP-86172		Adding, deleting, or editing the BroadSoft directory contact from the phone is not available.	No workaround is currently available.
Functionality	VOIP-87847	5.0.0	The phone currently plays the same sound for reboot, restart, and calendar notification.	No workaround is currently available.
Functionality	VOIP-88029	5.0.0	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).	Delete the contacts with a time delay of 3 to 4 seconds.
Functionality	VOIP-88174	5.0.0	Creating a mixed environment using UC Software 5.0.0 and previous Lync-supported software versions for Lync Boss-Admin is not supported.	No workaround is currently available.
Functionality	VOIP-88182	5.0.0	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.	No workaround is currently available.
Functionality	VOIP-88276	5.0.0	In a Lync Boss-Admin 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.	No workaround is currently available.
Functionality	VOIP-88278	5.0.0	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.	No workaround is currently available.
Functionality	VOIP-88290	5.0.0	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.	Press the DND soft key to disable DND.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-88308	5.0.0	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 micro browser using a headset (applies to VVX 500).	No workaround is currently available.
Functionality	VOIP-91637	5.1.0	In a Lync environment, the message "Logon information needed", displays after the user is registered.	No workaround is currently available.
Functionality	VOIP-92271		A dial tone mixed with page audio is played from the handset and chassis until the dial tone gets expired.	No workaround is currently available.
Functionality	VOIP-92271	5.1.0	In a group paging scenario, if the phone receives a page while it is off hook, the phone plays a dial tone mixed with the Group Paging audio from the chassis and handset.	No workaround is currently available.
Functionality	VOIP-92304	5.1.0	Editing the first characters of the SIP URI in the recent dialed contact with more than 30 characters is currently unavailable.	No workaround is currently available.
Functionality	VOIP-92459	5.1.0	The phone number is appended to the first name when the first name is a combination of Arabic and English in the corporate directory.	No workaround is currently available.
Functionality	VOIP-92642		An irregular ring back tone is heard when VVX600 is registered with corporate Lync server.	No workaround is currently available.
Functionality	VOIP-92681	5.1.0	In a centralized conferencing scenario, the call's appearance is changed to the video call layout after multiple instances of holding and resuming calls.	No workaround is currently available.
Functionality	VOIP-97439	5.3.0	In a Lync environment with the same user registered on multiple endpoints, when the phone receives calls from PSTN numbers, the phone fails to update received and missed calls in Recent Calls lists.	No workaround is currently available.
Functionality	VOIP-99568	5.3.0	After disabling the top and rear USB ports, USB charging devices, like mobile phones, are charging when the line label includes special characters and the length is 256 characters.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Functionality	VOIP-99645	4.0.1B	If there is a new call started on the SoundStructure VoIP Interface while there is an incoming call, the incoming call is ignored. This means that the incoming call will no longer ring even if the new call is ended. The incoming call will still be available for answering until it disconnects.	No workaround is currently available.
Hardware	VOIP-74120		Plantronics Audio 646 DSP USB headset volume control does not work (applies to VVX 500).	Adjust the volume using the volume keys on the phone.
Hardware	VOIP-89018	5.0.1	Some voice echo issues when the Plantronics EHS headset is used.	No workaround is currently available.
Hardware	VOIP-92326	5.1.0	The phone is unable to answer the second call with Plantronics Savor M1100 Bluetooth headset when the first call is placed on hold.	No workaround is currently available.
Hardware	VOIP-92333	5.1.0	The Plantronics Voyager PRO UC v2 USB headset is unable to answer the second call while another call is in progress.	No workaround is currently available.
Headset	VOIP-97099	5.3.0	When using a Sennheiser USB headset, you cannot adjust the ringtone or call audio volume using the controls on the headset.	No workaround is currently available.
Lync	VOIP-75591		In the Lync environment, when the user logs out, the phone does not logout all the user login credential-dependent applications.	No workaround is currently available.
Lync	VOIP-75778		Using Microsoft Lync, if a user dials an invalid extension, the entry is sometimes not logged in the Placed Calls call list.	No workaround is currently available.
Lync	VOIP-80212	4.1.0	In a Lync environment, when the corporate directory and parameter `dir.corp.sortcontrol` are enabled, the contact search does not fetch any contacts.	Set the parameter `dir.corp.sortcontrol`=0.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Lync	VOIP-82043	4.1.0.	When a Lync profile is used along with the boot server, 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.	Use VI editor or Edit Plus editor.
Lync	VOIP-82302	4.1.0	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.	Perform a consultative transfer.
Lync	VOIP-84598	4.1.4	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.	Reboot the phone after the second user signs in.
Lync	VOIP-84692	4.1.4	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).	No workaround is currently available.
Lync	VOIP-87129	5.0.0.	The network administrator or user has to manually set the base profile of the phone to Lync before establishing a BToE connection.	No workaround is currently available.
Lync	VOIP-87342	5.0.0	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.	No workaround is currently available
Lync	VOIP-87655	5.0.0	In a Lync environment, the phone displays the complete SIP URI for outgoing PSTN calls.	No workaround is currently available.
Lync	VOIP-87814	5.0.0	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.	No workaround is currently available.
Lync	VOIP-88254	5.0.0	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.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Lync	VOIP-88643	5.0.1	In a Lync Boss-Admin 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.	No workaround is currently available.
Lync	VOIP-88678	5.0.1	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.	No workaround is currently available.
Lync	VOIP-92310	5.1.0	In a Lync share line appearance scenario, the far end phone displays the phone’s extension and the message that delegates are ringing instead of the display name and the message that delegates are ringing when the boss phone is set to forward all calls to the Delegate.	No workaround is currently available.
Lync	VOIP-92642	5.1.0	In the Lync corporate network, a choppy ring back tone is heard (applies to VVX 600).	No workaround is currently available.
Lync	VOIP-93775	5.2.0	When BToE is enabled, the phone crashes when the privacy mode for contacts is changed to Blocked and 200 contacts are added to the Lync client at the same time.	Keep fewer than 200 contacts.
Lync	VOIP-94171	5.2.0	The phone doesn’t have an option to set the presence status to Off Work from its UI although the same can be done from the Lync 2013 client.	Set the **Off Work** status via the Lync client**.**
Lync	VOIP-94402	5.2.0	The phone loses synchronization with the server if multiple contacts are removed from the communicator simultaneously.	Do not delete multiple contacts at one time in the Lync client.
Lync	VOIP-95205	5.3.0	An unauthorized response is received when a presenter who is muted as a part of an audience tries to unmute his or her microphone.	No workaround is currently available.
Lync	VOIP-95684	5.2.0	When using Lync 2010, in a Boss-Admin scenario, the hold call fails when a Boss or delegate tries to answer it.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Lync	VOIP-96916	5.3.0	On VVX phones, Lync reverse name lookup does not work with Lync Address Book Search or Outlook contacts.	No workaround is currently available.
Lync	VOIP-97352	5.3.0	The phone is unable to register users when PIN authentication credentials are entered using configuration files.	No workaround is currently available.
Lync	VOIP-97460	5.3.0	In a Lync Boss-Admin scenario with BToE enabled, the phone does not properly display caller ID information for incoming boss calls from PSTN endpoints.	No workaround is currently available.
Lync	VOIP-97919	5.3.0	You cannot answer two or more incoming video calls on the phone during BToE audio playback.	No workaround is currently available.
Lync	VOIP-98533	5.3.0	In a Lync environment with Boss-Admin enabled, when the parameter `lineKey.reassignment.enabled` is set to 1, the delegate's line does not display on the boss's phone.	No workaround is currently available.
Lync	VOIP-98889	5.3.0	In a Lync Boss-Admin scenario, when the boss phone is BToE-enabled and playing audio through audio playback mode and an incoming call for the boss line is answered on a delegate's phone, the boss does not receive a notification that the call was answered by a delegate.	No workaround is currently available.
Lync	VOIP-99160	5.3.0	In a Lync scenario, the phone does not display the time of the voice mail for older voicemails, but displays the day and week.	No workaround is currently available.
Lync	VOIP-99190	5.3.0	The phone is not always updating Lync favorites on the phone when a favorite is added and deleted in the Lync client.	No workaround is currently available.
Networking	VOIP-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.	Use the correct subnet mask.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Networking	VOIP-52142		Video connections with the Counter Path 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.	Use a different codec or use another version of Eyebeam client.
Networking	VOIP-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).	Set the call for transcoding on the MGC.
Networking	VOIP-54976	SIP 3.2.2	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.	Configure system for encryption required.
Networking	VOIP-54977	SIP 3.2.2	H.264 calls to a Tandberg Edge95 MXP device using a Tandberg Gateway result in lip sync issues on the VVX 1500.	No workaround is currently available.
Networking	VOIP-62482		Server certificate Serial Number is checked against the host name if the outbound proxy is configured.	No workaround is currently available.
Networking	VOIP-63527	SIP 3.3.1	The phone sends out INVITE and CANCELS messages if no provisional response is received.	No workaround is currently available.
Networking	VOIP-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).	Use Cisco ACS server 5.1 or higher.
Networking	VOIP-78340	4.0.0	Sending several MWI NOTIFY messages within a few seconds of each other might cause the phone to reset.	Avoid sending multiple MWI messages close together.
Networking	VOIP-83101		In a federated environment, when the UDP traffic is blocked on the firewall, the phone might fail to connect the calls.	No workaround is currently available.
Networking	VOIP-91966	5.1.0	When the SSI Domain and DHCP Option 15 domains are the same, the DNS query is sent with the domain values concatenated.	No workaround is currently available.
Networking	VOIP-92678	5.1.0	The phone is unable to re-register after receiving 430 flow failed message from the server	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Networking	VOIP-94488	5.2.0	The phone network starts before the phone displays an Application started; message due to which the early dialogue is missing (applies to VVX 1500).	No workaround is currently available.
Polycom Desktop Connector	VOIP-70480	4.1.0	When the phone uses the Polycom Desktop Connector, the keyboard arrow keys do not support active and inactive call navigation (applies to VVX 500).	No workaround is currently available.
Provisioning	VOIP-99408	4.0.1B	After a factory reset of the SoundStructure VoIP Interface, a voip_prov_serv_address status command returns a non-empty provisioning server address. A SoundStructure client will receive the following message: val voip_prov_serv_address "VoIP In" "https://PlcmSpIp:PlcmSpIp@ztp.polycom.com" .	No workaround is currently available
Security	VOIP-82212	4.1.0	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).	No workaround is currently available.
Server	VOIP-98581	5.3.0	When an incoming call is answered in a Lync Mediation Server 2010 environment, it takes six seconds for the Transfer and Hold soft keys to display on VVX 400 and VVX 600 phones.	Enable media bypass on the Lync Mediation Server and the media gateway.
Software	VOIP-52141		During software upgrades to daisy-chained SoundStation IP 7000 phones, the upgrades sometimes stop.	Press any key on the phone to continue the upgrade.
UI?UX	VOIP-79735	4.1.0	Changing the language of the phone from German to any language other than English results in a display of diacritic letters (applies to VVX 500 and SoundPoint 331).	Change the language to English first.
User Experience	VOIP-97370	5.3.0	The LED Message Waiting Indicator flashes while you mark messages as read or unread on the phone.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
User Interface	VOIP-37273		If the custom idle display and idle browser features are both enabled the phone UI displays incorrectly.	Do not set `ind.idleDisplay.enabled` to 1 and enable the Idle Browser at the same time.
User Interface	VOIP-37984		Enabling the idle bit-map on SoundPoint IP 330 and 320 phones causes the Line Key labels and dialed digits to be invisible.	Do not use the idle bit-map on 330/320 phones; instead, set ind.idleDisplay.enabled=0.
User Interface	VOIP-59812	SIP 3.3.0	Blind transfer to a URL is not successful on the SoundStation IP 7000. Eventually, the URL soft key becomes unavailable.	No workaround is currently available.
User Interface	VOIP-62387	SIP 3.3.1	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	Reset the phone.
User Interface	VOIP-71386	4.1.0	Soft key URIs does not function when the phone is in the Enter Number screen (applies to VVX 1500).	No workaround is currently available.
User Interface	VOIP-74533	SIP 3.2.5	A phone configured with a Synergy call server displays the incorrect caller ID on the UI for an incoming call (applies to VVX 1500).	No workaround is currently available.
User Interface	VOIP-75229	SIP 3.2.7	A phone configured with a Synergy call server displays the local conference UI when establishing a centralized conference using the Join soft key.	No workaround is currently available.
User Interface	VOIP-75759	4.0.1	Numeric data entered using the dial pad on the phone browser cannot be deleted on the dial pad.	Use the virtual keyboard.
User Interface	VOIP-75869	4.0.1	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.	Exit and re-enter the directory.
User Interface	VOIP-76522	4.0.2	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.	The call center administrator can set the call center name.
User Interface	VOIP-76753	4.0.1	Removing a BLF line from the server causes the speed dial icon to disappear.	Restart or reboot the phone.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
User Interface	VOIP-78232	4.0.2	During a remote conference pickup on a shared line, the phone does not display the call appearance and call indicator.	No workaround is currently available.
User Interface	VOIP-80227	4.0.3	The phone does not display the saved name of the contact in the local contact directory.	Use the full URI while adding the contacts in the local contact directory.
User Interface	VOIP-82401	4.1.2	The call order widget disappears on the phone screen after scrolling through five of the maximum number of calls (24).	No workaround is currently available.
User Interface	VOIP-83157		The phone does not display the protocol field for the local contacts.	No workaround is currently available.
User Interface	VOIP-83330		In a call center scenario, an incoming call during a guest sign-in displays some non-functional soft keys.	No workaround is currently available.
User Interface	VOIP-83378	4.1.3G	When the SoundStructure VoIP Interface has multiple lines registered, and the following commands are sent to the SoundStructure: set voip_line "VoIP Out" 1 set voip_line "VoIP Out" 2 set phone_connect "VoIP Out" 0 then most times, the phone remains off-hook and the dial tone is still heard. The order of the first two commands does not matter and there can be more than two lines registered and this issue will be seen.	To put the phone on-hook, send the following commands to the SoundStructure: set phone_connect "VoIP Out" 1 set phone_connect "VoIP Out" 0 set phone_connect "VoIP Out" 0
User Interface	VOIP-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.	Enable the call forward feature on the phone.
User Interface	VOIP-83887 VOIP-83889	4.1.3	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.	Use H.263 video codec with a bit rate greater than 1500 Kbps.
User Interface	VOIP-84061	4.1.3	In a call center scenario, the phone does not display the call center information on the default screen when the VVX Camera is attached.	Press the call center info soft key to retrieve the call center information.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
User Interface	VOIP-84103		When the user tries to navigate back from the diagnostics menu, a colored screen appears on the phone (applies to VVX 300/310).	No workaround is currently available.
User Interface	VOIP-88618	5.0.1	The line label is not displayed properly when you set a long user name mixed with numbers when the language is set to Arabic.	No workaround is currently available.
User Interface	VOIP-89082	5.0.1	The call list icon on the phone is not displayed when the message “DND when locked” displays and the phone is set in a locked state.	No workaround is currently available.
User Interface	VOIP-89132	5.0.1	The display name on the phone is truncated during a video call when the language is set to Arabic on the phone.	No workaround is currently available.
User Interface	VOIP-92679	5.1.0	The phone is displaying “All Contacts” instead of “Other contacts” in the Contacts menu under Groups.	No workaround is currently available.
User Interface	VOIP-93172	5.1.1	Observed that Dial and Add Contact soft keys are not getting displayed after performing the CMA search (applies to VVX-1500)	Try dialing using hard Keys.
User Interface	VOIP-93272		The phone does not display a “Successfully Un-paired” pop up after PC port link is unplugged.	No workaround is currently available.
User Interface	VOIP-93944	5.2.0	The phone’s user interface response is slow when the log level is not set to the default level for all the modules.	Keep the log levels at standard except when necessary.
User Interface	VOIP-94352	5.2.0	The phone list of the total number of calls disappears occasionally when scrolling through the calls on the phone interface.	No workaround is currently available.
User Interface	VOIP-95943	5.2.1	When the Barge-In feature is enabled only on phone and not on the server, the phone UI does not display any soft keys in the filtered view.	No workaround is currently available.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
User Interface	VOIP-98341	5.3.0	The SoundStructure VoIP Interface log will report "Failed to get mic mute state, failure 2" when there is no active call. This error can be ignored if there is no active call when it occurred.. The SoundStructure "VoIP Out" channel mute state can be changed when there is no active call, but the SoundStructure VoIP Interface may not be muted or unmuted as intended.	Do not set the SoundStructure VoIP Out channel mute state while there is no active call. There will still be the failure reported in the log, but that can be ignored if there is no active call.
User Interface	VOIP-99136	5.3.0	If a call is placed five minutes after an upgrade and before the call lists are synchronized, a message stating that the call list is synchronizing displays on the phone.	No workaround is currently available.
User Interface	VOIP-99237	5.3.0	The phone displays the latest message if multiple messages were displayed at the same time while the phone was starting.	No workaround is currently available.
User Interface	VOIP-99250	5.3.0	Changing the mode from number to URI while editing call entries in the Recent Calls list using the onscreen keyboard may bring back the deleted text on VVX 500 and 600 phones and CX5500 systems.	No workaround is currently available.
User Interface	VOIP-99450	4.0.1B	If on-hook dialing is used on the SoundStructure VoIP Interface and a number that is not in the dialplan is dialed and sent, then the call will end, and the following message may not be received: val phone_connect "VoIP Out" 0. This can cause the control system to display as though the SoundStructure VoIP Interface is still off-hook or in a call.	To get out of the state in the description, send the following command two times: `set phone_connect "VoIP Out" 0`. Do not use on-hook dialing for numbers that are not in the dialplan.
Web Configuration Utility	VOIP-97032 VOIP-97029	5.3.0	When you upload a background image in the Web Configuration Utility, you cannot upload the same image as the background for the phone and the VVX Color Expansion Module at separate times.	Delete the image, and re-upload the image for both the phone and the expansion module at the same time.

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Web Configuration Utility	VOIP-99441	5.3.0	Switching between the user and administrator credentials on the Web Configuration Utility may not work.	Clear your browsing data and recent history. In your web browser, navigate to History, and delete cookies, saved passwords, and cache.

Resolved Issues

The section lists the issues that were resolved in this release.

Resolved Issues

Category	*Issue No.*	*Release*	*Description*
Calling	VOIP-103284	5.4.0	Phone now successfully plays and sends the DTMF for “0” and “*” digits without any issue
BroadSoft	VOIP-100600	5.2.0	The GuestIn for BroadWorks Hoteling host no longer causes intermittent issues on VVX phones.
BToE	VOIP-100103	5.2.0	The phone now remains connected to BToE after placing a call to a PSTN number established through the Clarity Connect system.
BToE	VOIP-87338	5.0.0	In a BToE scenario and while installing the Polycom BTOE Connector application, computers no longer request a reboot.
Calling	VOIP-101875	5.3.0	In a conference call, the first PSTN user no longer hears DTMF tones when the conference initiator adds another contact to the conference.
Calling	VOIP-99376	4.1.7	The phone now successfully answers a call that was transferred from another user after holding and resuming the call multiple times.
Configuration	VOIP_99665	5.2.0	Multiple Enhanced Feature Keys for transferring calls now work accordingly.
Configuration	VOIP-100622	4.0.7, 4.1.6	The phone no longer forwards to a SIP URI and no longer shows the URI entry when SIP URI dialing is disabled.
Configuration	VOIP-101244	4.1.8, 5.3.0	The phone now compares and displays messages based on origin IP when the parameter `onlySignalWithRegistered` is set to 0.

Category	*Issue No.*	*Release*	*Description*
Configuration	VOIP-101842	5.3.0	Removed the Spectralink parameters `messaging.quicknotes.x` and `messaging.maxImMessages` from the cfgParamDef.xml file.
Configuration	VOIP-101873	5.3.0	The phone no longer goes into a 503 response loop when `TCPpreferred` is set to TCP first and alternates between UDP.
Configuration	VOIP-102006	5.3.0	The phone now picks up BLF calls when the parameter `CallsPerLineKey` is set to 1.
Contact Directory	VOIP-101589	4.0.6, 5.2.0	VVX 1500 phones with RealPresence Resource Manager can now store Guest Book entries in the Contact Directory.
Contacts	VOIP-99458	5.3.0	When you select All Contacts on the Groups screen, contacts in the Other Contacts group no longer display instead of all the contacts on the phone.
Directory	VOIP-102293	5.3.0	The phone now displays the correct number in the UC-One Directory when there is an extension for the number and no longer attaches the extension to the phone number.
Expansion Module	VOIP-101662	5.2.2	Stability and performance have been improved for VVX Expansion Modules.
Expansion Module	VOIP-99533	5.2.0	Any issues regarding the line keys or handling calls on the VVX Expansion Modules no longer occurs.
Functionality	VOIP-100095	5.2.0, 5.3.0	The phone now correctly displays the call center information.
Functionality	VOIP-100141	4.0.7, 5.2.0	The phone now properly handles MWI NOTIFY messages. Note that some issues with freezing or restarting may occur very rarely.
Functionality	VOIP-100423	5.2.0	The phone now sends the media attribute as inactive and no longer causes any for Music on Hold issues on AudioCode gateways.
Functionality	VOIP-100630	5.2.0	The phone now properly handles session expirations.

Category	*Issue No.*	*Release*	*Description*
Functionality	VOIP-101938	4.0.5, 5.2.0, 5.3.0	Network information, including the phone's IP address and subnet mask, now properly display on the phone when performing serial commands.
Functionality	VOIP-102077	4.0.8	The phone now properly handles the Group Paging subscription options.
Functionality	VOIP-102174	5.0.1	Mismatch file system versions due to a flash corruption no longer occurs, and the phone starts successfully without displaying the Fix soft key.
Functionality	VOIP-99699	5.2.2	The Intercom alert info header now functions properly as per RFC 3261.
Functionality	VOIP-99718	5.2.0	The phone now sends BYE to the same route value that it received in the 200 OK of INVITE.
Functionality	VOIP-99937	5.1.1, 5.1.2, 5.2.0	Calls no longer fail on phones registered with Lync Server where the media port range is set to 65300 or higher. The formula to determine the TURN port has been updated to resolve this.
GENBAND	VOIP-101182	4.0.5	In a GENBAND environment, the phone now displays the correct caller ID after resuming a held MADN call.
Group Paging/PTT	VOIP-97407	5.2.0	The phone no longer freezes due to heavy broadcast messages.
Headset	VOIP-100305	5.2.0	When a phone is in a call and using a connected Bluetooth headset to handle calls, all additional calls are also handled using the headset instead of the speakerphone.
Lync	VOIP-100848	5.2.2	The phone now successfully registers the user to Lync 2013 server without user intervention after upgrading from a Survivable Branch Appliance or Server.

Category	*Issue No.*	*Release*	*Description*
Lync	VOIP-101383	5.2.0	In some specific environments, the phone now properly displays the caller ID for incoming calls.
Lync	VOIP-101731	5.3.0	When the Exchange Web Service is enabled, the phone no longer displays unwanted exchange related error messages when a Lync user signs into the phone using Pin Authentication.
Lync	VOIP-101874	5.3.0	All applicable soft keys now display when a call is forwarded from the Lync client to a PSTN line.
Lync	VOIP-102383	5.3.0	In some networks, the phones no longer sign out of Lync randomly.
Lync	VOIP-97048	5.1.2, 5.1.3, 5.2.0	In some Lync environments, the phone no longer displays duplicate contacts in the Lync directory.
Lync	VOIP-98823	5.3.0	Call log entries in the Outlook Conversation History folder now show a contact's display name for calls made on phones with BToE enabled or disabled.
Microsoft	VOIP-98849	5.3.0	The Exchange Autodiscovery feature is now supported.
PTSN	VOIP-101113	5.3.0	In a location-based routing environment, issues regarding PSTN no longer occurs.
Shared Calls	VOIP-100633	5.2.2, 5.3.0	Using a shared appearance on multiple phones, a remote held call can now be resumed successfully by the shared users on other phones.
Shared Calls	VOIP-99280	5.3.0	The filtered view for a shared line now displays held calls when a user performs a long press on the line key with multiple SLA calls on hold
Software Update	VOIP-100442	5.2.2, 5.3.0	The phone now updates correctly after an Updater upgrade when you connect the phone using the manual DHCP option 128 VLAN setting.

Category	*Issue No.*	*Release*	*Description*
Software Update	VOIP-102311	5.4.0	Automatic software upgrade no longer fails when using a customer server (CX5500 only).
Software Update	VOIP-99650	5.2.0	The phone now retrieves the upgrade or downgrade software that matches the revision ID on the XML file.
UC-One	VOIP-101989	5.3.0	In an UC-One scenario, the phone now correctly updates UC-One contacts and groups entered using the UC-One client when the language is set to Spanish.
UC-One	VOIP-102295	5.3.0	There are no longer any issues when calling an UC-One contact from a Shared Call Appearance line when URL dialing is enabled.
User Experience	VOIP-94299	5.2.0	When the phone receives multiple incoming calls while the phone is on the Transfer screen, and one of the incoming calls disconnects, the incoming call icon is no longer removed.
User Interface	VOIP-99362	5.2.0	User can now set the Power Saving duration on the phone in a range of 1 - 24 hours.
Video	VOIP-101939	5.1.2	After a call is held and resumed, SRTP no longer generates a new key on the phone when the parameter `sec.srtp.answerWithNewKey` is set to 0 for VVX video-enabled phones.
Video	VOIP-99417	5.2.0	The Audio and Video soft keys no longer display when video is disabled and the parameter feature.audioVideoToggle.enabled is set to 0.
Video	VOIP-99488	5.3.0	If the camera shutter is closed before the phone is restarted, the Video Mute icon now displays on the far-end during a video call after the phone is restarted.

Category	*Issue No.*	*Release*	*Description*
Web Configuration Utility	VOIP-92019	5.0.1	When registering a phone with another user account after it was registered using PIN authentication, the Extension and PIN fields in the Web Configuration Utility now clear.
Web Configuration Utility	VOIP-101631	5.3.0	The Web Configuration Utility now displays the Additional Preferences menu under the Preferences menu when the base profile is set to Lync.
Web Configuration Utility	VOIP-101733	5.3.0	The Web Configuration Utility now displays only the Enforced by Server settings under the Call Diversion menu.
Web Configuration Utility	VOIP-98315	5.3.0	When you upload a ringtone in the Web Configuration Utility and select that ringtone as the default ringtone, the correct ringtone is now selected and played.

Get Help

For more information about installing, configuring, and administering Polycom products, refer to Documents and Downloads at Polycom Support.

For additional information about the Polycom VVX Business Media Phones, the VVX Camera, the VVX Expansion Modules, and SoundStructure VoIP Interface, view the following support pages:

- Polycom VVX 300 and 310
- Polycom VVX 400 and 410
- Polycom VVX 500
- Polycom VVX 600
- Polycom VVX 1500
- Polycom VVX Camera
- Polycom VVX Expansion Modules
- Polycom SoundStructure

You can view the following types of documents on each product page:

- **User Documents**:
  - *Quick Tips* A quick reference on how to use the phone's most basic features.
- **Setup and Maintenance Documents**:
  - *Quick Start Guide* This guide describes the contents of your package, how to assemble the phone or accessory, and how to connect the phone to the network. The quick start guide is included in your phone package.
  - *Wallmount Instructions* This document provides detailed instructions for mounting your phone on the wall. To install your phone on the wall, you need the optional wallmount package, which includes the wallmount instructions.
  - *Administrator Guide* This guide provides detailed information about setting up your network and configuring phone features.
- **Feature Descriptions and Technical Notifications** These documents describe workarounds to existing issues and provide expanded descriptions and examples for phone settings and features. You can find these documents on the Polycom Profiled UC Software Features and Polycom Engineering Advisories and Technical Notifications support pages.

The Polycom Community

The Polycom Community gives you access to the latest developer and support information. Participate in discussion forums to share ideas and solve problems with your colleagues. To register with the Polycom Community, create a Polycom online account. When logged in, you can access Polycom support personnel and participate in developer and support forums to find the latest information on hardware, software, and partner solutions topics.

Copyright and Trademark Information

6001 America Center Drive

San Jose, CA 95002

USA

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.

Polycom | Support

Visit the *Polycom Support Center* for End User License Agreements, software downloads, product documents, product licenses, troubleshooting tips, service requests, and more.