Polycom® UC Software 5.5.3

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

Polycom announces the release of Polycom® Unified Communications (UC) Software, version 5.5.3. This document provides the latest information about this release.

Contents

What's New in Polycom UC Software 5.5.3

Polycom Unified Communications (UC) Software 5.5.3 is a release for Open SIP and Skype for Business deployments.This release notes provide important information on software updates, phone features, and known issues.

Polycom UC Software 5.5.3 supports the following Polycom endpoints.

Phone Support

Phone Model	Skype for Business	Open SIP
VVX 201 business media phone	✓	✓
VVX 300/301/310/311 business media phones	✓	✓
VVX 400/401/410/411 business media phones	✓	✓
VVX 500/501 business media phones	✓	✓
VVX 600/601 business media phones	✓	✓
VVX 1500 business media phone		✓
VVX D60 Wireless Handset and Base Station		✓
SoundStructure VoIP Interface	✓	✓

Polycom UC Software 5.5.3 supports the following Polycom accessories.

Accessories Support

Accessories	Skype for Business	Open SIP
VVX Camera		✓
VVX Color Expansion Module	✓	✓
VVX Paper Expansion Module		✓

If you are an existing VVX D60 user, do not use UC Software 5.4.3 Rev D, 5.4.4 Rev E, UCS 5.4.4 Rev P, or 5.5.0. If you are using any of these releases, please contact Polycom Support.

New Features and Enhancements

There are no new features in this release for software version 5.5.3. The following new fixes are available in the 5.5.3 release:

Supported Provisioning Protocols

By default, Polycom phones are shipped with FTP enabled as the provisioning protocol. You can configure the phone using the following supported provisioning protocols:

- Trivial File Transfer Protocol (TFTP)
- File Transfer Protocol (FTP)
- Hyper Text Transfer Protocol (HTTP)
- Hyper Text Transfer Protocol - Secure (HTTPS)
- File Transfer Protocol - Secure (FTPS)

  Note that when using FTPS as the provisioning protocol, use the following configuration:

  - You must set the value of the configuration parameter `log.render.file.size` to 512.
  - You must disable the Diffie Hellman key Exchange.

Phone Features and Licenses

The features and licenses required to operate the phones vary by phone model. Refer to this section to find out which phone features and licenses you require for your phone model.

The following table describes features available for each phone and indicates whether a feature license is required. 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.
- *Yes** indicates that the phone requires you to purchase a feature license from Polycom to support a feature.
- *Yes*** indicates that the phone requires you to purchase an honor-based license from Polycom to support a feature.

Phone Features and Licenses

Feature	VVX 101	VVX 201	VVX 300/ 310	VVX 301/ 311	VVX 400/ 410	VVX 401/ 411	VVX 500/ 501	VVX 600/ 601	VVX 1500	SoundStructure VoIP Interface
Asian Languages	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Conference Management	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Customizable UI Background	No	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No

Phone Features and Licenses

Feature	VVX 101	VVX 201	VVX 300/ 310	VVX 301/ 311	VVX 400/ 410	VVX 401/ 411	VVX 500/ 501	VVX 600/ 601	VVX 1500	SoundStructure VoIP Interface
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Enhanced BLF	No	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
H.323 Video	No	No	No	No	No	No	Yes	Yes	Yes*	No
Skype for Business (SfB)	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**
Server Based Call Recording	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
USB Call Recording	No	No	No	No	No	Yes	Yes	Yes	Yes	No
Voice Quality Monitoring (VQMon)	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes (Audio only)	Yes (Audio only)	Yes (Audio only)	No
XT9 Input (PinYin)	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*

*You must purchase a feature license from Polycom.

**You must purchase an honor-based license from Polycom.

Supported DHCP Sub-Options

The following table lists the individual sub-options and combination sub-options supported on VVX phones for DHCP Option 43.

DHCP Option 43 Configuration Options

Option	Result
Option 1- Subnet mask	The phone parses the value from Option 43.
Option 2 - Time offset	The phone parses the value.
Option 3 – Router	The phone parses the value.
Option 4 - Time server	The phone parses the value.
Option 6 - Domain Name Server	The phone parses the value.

DHCP Option 43 Configuration Options

Option	Result
Option 7 - Domain Log server	The phone parses the value.
Option 15 - Domain Name	The phone parses the value.
Option 42 - Network Time Protocol server	The phone parses the value.
Option 66 - TFTP Server Name	The phone parses the value.
Sub-options configured in Option 43	
Options 1, 2, 3, 4, 5, 6, 7, 15, 42, and 66	The phone parses the value.

Setting the Log Levels for Skype for Business Server

In UC Software 5.5.3 and later, you can set the log levels for Polycom phones in a Skype for Business environment on the Skype for Business Server.

To set the server-side logging levels:

1. In the command shell, enter the command `Set-CsUCPhoneConfiguration`
2. Set one of the following log levels.
   - Off
   - Low
   - Medium
   - High

The following table shows the phone log levels that correspond with the server log levels.

Corresponding Server and Phone Log Levels

	Features			Server Logging Level	
		Off	Low	Medium	High
Phone Logging Level	ICE	4	4	0	0
	TICKT	4	4	0	0
	SIP	4	4	2	0
	EC	4	4	2	2
	APP1	4	4	2	2
	SO	4	4	2	2
	AFE	5	5	2	2
	PPS	4	4	1	1
	PGUI	4	4	2	2
	BTOE	4	4	2	2
	ServiceAuth	2	2	2	2
	ServiceDevicePair	4	4	2	2
	ServiceProxy	4	2	2	2
	ServiceWad	2	2	2	2

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.5.3 release. For more information on using configuration parameters to enable or disable features, see the *Administrator Guide for Polycom UC Software for your software release* available on the Polycom Voice Support site.

Configuration File Enhancements for UC Software 5.5.3

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`, `site.cfg`	`sec.TLS.protocol.browser`	Configures the phone to control the TLS protocol used for the browser for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of the value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2 The microbrowser restarts when there is a change in the browser TLS protocol or TLS cipher settings, and the last web page displayed is not restored.	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.ldap`	Configures the phone to control the TLS protocol used for Lightweight Directory Access Protocol (LDAP) for a handshake negotiation from the phone. The security handshake from the phone starts with the highest TLS version irrespective of the value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No
`device.cfg`, `site.cfg`	`sec.TLS.protocol.sip`	Configures the phone to control the settings of the selected TLS protocol used for SIP signaling for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No

Configuration File Enhancements for UC Software 5.5.3

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	sec.TLS.protocol.sopi	Configures the phone to control the settings of the selected TLS protocol used for SOPI for a handshake negotiation from the phone. The handshake from the phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.webServer	Configures the phone to control the settings of the selected TLS protocol used for Web Server for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.xmpp	Configures the phone to use the selected protocol for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.exchangeServices	Controls the TLS Protocol used for Exchange services application and configures the phone to use the selected protocol for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No

Configuration File Enhancements for UC Software 5.5.3

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.sec.TLS.protocol.syslog	Controls the TLS Protocol used for Syslog. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No
device.cfg, site.cfg	device.sec.TLS.protocol.prov	Controls the TLS Protocol used for provisioning. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2	No
device.cfg, site.cfg	device.sec.TLS.protocol.dot1x	Controls the TLS Protocol used for 802.1x authentication. The handshake always starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_0, TLSv1_1 and TLSv1_2	No
sip-interop.cfg	tone.dtmf.rfc2833Payload_OPUS	Sets the Dual Tone Multi Frequency (DTMF) payload required to use Opus codec. 126 (default) 96 - 127	Yes
sip-interop.cfg	voIpProt.SIP.RFC3261TimerI	0 (default) - Timer I for reliable transport will be fired at five seconds. This parameter does not cause any change for unreliable transport. 1 - Timer I for reliable transport will be fired at zero seconds.	No

Configuration File Enhancements for UC Software 5.5.3

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	sec.TLS.cipherList	Specifies the cipher list for all applications except web server. (default) ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!MD5:!RC4:@STRENGTH String (Maximum of 1024 characters)	No
site.cfg	sec.TLS.webServer.cipherList	Specifies the cipher list for web server. (default) ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!AES256-SHA:!AES128-SHA:!MD5:!RC4:@STRENGTH String (Maximum of 1024 characters)	No

Release History

This following table shows the recent release history of Polycom Unified Communications (UC) Software.

Release History

Release	Release Date	Description
5.5.3	September 2017	This release includes important field fixes.
5.5.2	May 2017	This release includes the following features and field fixes: • Enterprise Directory Default Search • Registration Line Address in Status Bar • BroadWorks Anywhere EFK for Soft Keys • Hide Contact Directory and Favorites • Personal Directory • BSFT Server Based Call Logs • New Call Forwarding Icons • Updated Do Not Disturb Icon • Expanded Support for USB Headsets • Support Added for CDP in VVX D60 Base Station • ALLOW Header in 18x Provisional Responses • Improved BToE device lock
5.5.1 Rev E	February 2017	This release includes important field fixes.

Release History

Release	Release Date	Description
5.5.1	September 2016	This release adds enhancements for distribution list, QoE, device lock, Polycom BToE manual pairing, user log upload, updated UI for VVX 500 and 600, unified contact store, web sign-in for online deployments.
5.5.0	June 2016	This release introduced support for BroadSoft Executive Assistant and Flexible Seating, TR-069, the 3GPP Technical Specification, the IPV6 protocol, Off-hook Call Status control, ability to lock the web configuration utility after failed login attempts, and user interface enhancements.
5.4.3	February 2016	This release introduced the Polycom VVX D60 Wireless Handset and VVX D60 Base Station.
5.4.1	December 2015	This release includes support for the following features: • Introduced the Polycom VVX 301/311, 401/411, 501, and 601 business media phones. • Flexible line key customization for Lync (EFLK). • Master Key Identifiers (MKI). • Shared Line appearance on Lync. • BToE for Windows 10. • Smart Search for Lync ABS. • Support for simplified Chinese font on VVX 101.
5.4.0A	September 2015	This release includes support for the following features: • Microsoft Office 365 and Skype for Business Online. • Office365 and Skype for Business Provisioning and Manageability. • Time and Date Initial Setup.

Release History

Release	Release Date	Description
5.4.0	May 2015	Added support for Alcatel-Lucent CTS features including: • Advanced Conference. • Shared Call Appearance with Bridge In. • Visitor Desk Phone. This release also included support for the following features: • Barge In on Busy Lamp Field Lines. • DTMF Relay. • SIP Instance. • Comfort Noise • Opus Codec. • DNS Server Address Override. • Global Directory Synchronization. • Basic Menu Lock. • Additional features including user interface improvements and resolved known issues.
5.3.0	March 2015	Includes support for several Lync, BroadSoft, and Open SIP features.

Security Updates

Please refer to the Polycom Security Center for information about known and resolved security vulnerabilities.

Install UC Software 5.5.3

Consider the following information when installing or updating to Polycom UC Software 5.5.3.

- BToE 3.5.0 is required for use with UC Software 5.5.3. For best results, Polycom recommends deploying BToE 3.5.0 prior to deploying UCS 5.5.3. While BToE 3.5.0 is backwards compatible with previous versions of VVX firmware, Polycom does not recommend running previous versions of BToE software with UC Software 5.5.3.
- Before updating your VVX 1500 phone to UC Software 5.5.3, make sure that the phone is updated to BootBlock 3.0.4. For more information, see Technical Bulletin 695: Upgrading the Polycom VVX 1500 Business Media Phone to UC Software 5.2.0.
- If you are running UC Software 5.5.3 and the BTOE 3.4.1, then the phone doesn’t correctly follow the locking of the PC.
- For the best device lock functionality in BToE mode, upgrade the UC software and the BToE application. If the user deploys the BToE software prior to the UC Software, the older device lock functionality in BToE mode works since prior VVX changes did not expect any events from the BToE application. If the user deploys the UC Software prior to the BToE software, the device lock functionality does not work in the BToE mode. This functionality is effective only in BToE mode.

Download the Distribution Files

To download UC Software 5.5.3, 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 **5.5.3.2169 rts30**.

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 and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
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	✓
3111-48300-001.sip.ld	SIP application executable for VVX 301	x	✓
3111-46161-001.sip.ld	SIP application executable for VVX 310	x	✓
3111-48350-001.sip.ld	SIP application executable for VVX 311	x	✓
3111-46157-002.sip.ld	SIP application executable for VVX 400	x	✓
3111-48400-001.sip.ld	SIP application executable for VVX 401	x	✓
3111-46162-001.sip.ld	SIP application executable for VVX 410	x	✓
3111-48450-001.sip.ld	SIP application executable for VVX 411	x	✓
3111-44500-001.sip.ld	SIP application executable for VVX 500	x	✓
3111-48500-001.sip	SIP application executable for VVX 501	x	✓
3111-44600-001.sip.ld	SIP application executable for VVX 600	x	✓
3111-48600-001.sip	SIP application executable for VVX 601	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	✓

Understand the Combined and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
3111-17823-001.dect.ld	SIP application executable for VVX D60 Wireless Handset and Base Station	x	✓
sip.ld	Concatenated SIP application executable.	✓	x
dect.ver	Text file detailing build-identification(s) for the VVX D60.	✓	✓
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.	✓	✓
device.cfg	Configuration parameters for basic device configuration.	✓	✓
features.cfg	Configuration parameters for telephony features.	✓	✓
firewall-nat.cfg	Contains configuration parameters for telephony features.	✓	✓
H323.cfg	Configuration parameters for the H.323 signaling protocol.	✓	✓
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	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.	✓	✓
video.cfg	Configuration parameters for video connectivity.	✓	✓
video-integration.cfg	Configuration parameters for SoundStation IP 7000 and Polycom HDX system integration.	✓	✓

Understand the Combined and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
VVX-dictionary.xml	Includes native support for the following languages: • Arabic, UAE • 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	✓	✓
Welcome.wav	Startup welcome sound effect.	✓	✓
LoudRing.wav	Sample loud ringer sound effect.	✓	✓
Polycom-hold.wav	Sample ringer sound effect.	✓	✓
Warble.wav	Sample ringer sound effect.	✓	✓
polycomConfig.xsd	Master configuration file that contains the parameters and its values.	✓	✓

Resolved Issues

Resolved Issues in UC Software 5.5.3

Category	Issue Number	Description
API	EN-30131	The VVX business media phones are unable to answer a call through Push API when full-screen splash displays on the phone's user interface.
Application	EN-30101	The VVX business media phones still display the meeting status as 'Out of Office' even after the 'Out of Office' meeting status time is over.
Application	EN-30138	The VVX phones do not display Outlook Contacts when PSTN contacts are set as favorites on Skype for Business.
Application	EN-30149	The VVX business media phone does not send the XMPP status correctly after restart but works fine after reboot.
Application	EN-32455	VVX business media phone does not update Outlook Contact display name on reboot.
Application	EN-32656	VVX business media phone unable to search for active directory contacts in Outlook.
Application	EN-32358	VVX business media phone does not update the calendar events as per new time format after changing the time format from 12 hours to 24 hours and vice versa.
Application	EN-33814	Polycom BToE Connector application observes memory leak during the video playback call.
Audio	EN-30072	A one-way Rx audio is observed when answering the fifth incoming call on the VVX business media phone, while four other calls are in the ringing state.
Audio	EN-30091	When the VVX 1xx, 2xx, 3xx, and 4xx series business media phones receive close to 20 incoming calls during an active call, and the call waiting is disabled, the phone experiences an audio issue.
Browser	EN-30140	If the Diffie-Hellman (DH) key length is more than 128 bytes, client applications on the VVX business media phones takes longer time to establish a connection.
Calling	EN-30148	A Shared Line Appearance (SLA) member cannot establish a conference when other SLA members who barge-in are disconnected from the call.
Calling	EN-30163	When the configuration parameter `voIpProt.SIP.RFC3261TimerI` is set to 1, the VVX business media phone sends 481 response code to second INVITE after rejecting initial INVITE message with 488 response even when the transaction timeout is zero, as per specification for Transmission Control Protocol (TCP) transport.
Calling	EN-30150	On reboot via check-sync, the VVX business media phone becomes unregistered and remains unregistered for 2-3 minutes, and the user cannot place or receive calls.
Calling	EN-30135	A GENBAND shared line user cannot retrieve a call on hold when other users who barge-in are disconnected from the call.

Resolved Issues in UC Software 5.5.3

Category	Issue Number	Description
Calling	EN-30070	When the VVX business media phones initiate a ‘Skype for Business’ call, SfB enables video call but drops the audio.
Calling	EN-30146	The user observes that the Polycom VVX business media phone's user interface does not have an option to define the ring type for Boss calls.
Calling	EN-30096	When joining a meeting through Skype for Business client in a Better Together over Ethernet (BToE) environment, the RX audio on the VVX business media phone appears noisy.
Calling	EN-32516	If the URL dialing feature is disabled, the first two digits are missing from the call list when the user dials a URI number.
Calling	EN-30082	Call input screen is not populated to perform a call transfer in line view on the VVX 400 business media phone.
Calling	EN-30085	In a Shared Call appearance, remote resume fails when `sec.srtp` parameter is enabled.
Calling	EN-33152	After the push-to-talk (PTT) call is put on hold, the call hold reminder tone is heard sometimes later even when the phone is idle.
Calling	EN-30127	Using G729 codec on the VVX business media phones with HDvoice handset causes audio to be muffled.
Calling	EN-30162	When Flexible Line Key (FLK) is enabled, the user is unable to set ringtone for a Boss call on a VVX business media phone.
Configuration	EN-30142	Due to the wrong URL configuration in the web server, the VVX business phone causes to generate core dump files when the infinite redirect loops are found in the web server URL.
Configuration	EN-32270	Bluetooth, when enabled in the configuration, is not in effect sometimes after the Reboot/Restart/Upgrade.
Configuration	EN-30102	On VVX business media phone, `tcpIpApp.sntp.address` parameter value is ignored when `tcpIpApp.sntp.address.overrideDHCP` parameter is enabled.
Directories/ Address Books	EN-30098	The VVX business media phone displays the message "Searching..." indefinitely while performing a search on Open LDAP when the number of search results are greater than the maximum page size.
General	EN-30128	Disabling presence subscription causes a presence subscription storm on VVX business media phone when a user navigates the phone menus.
General	EN-30139	In a Skype for Business environment, the admin phone stops receiving Boss calls, after changing the password via BToE.
General	EN-28274	Session Description Protocol (SDP) clock rate for Dual Tone Multi Frequency (DTMF) does not match the clock rate for Opus codec.
Interoperability	EN-30159	In VVX 500/600 business media phones, the video stops when the server sends unsupported audio and supported video codecs in re-INVITE.

Resolved Issues in UC Software 5.5.3

Category	Issue Number	Description
Interoperability	EN-30099	The VVX business media phone remains unregistered when registering to the primary server fails.
Network	EN-30078	A possible memory leak makes SoundStructure VoIP Card unresponsive over time.
Network	EN-30073	When the parameter `softkey.feature.buddies` is enabled, 'contact' soft key has display issues on the phone's user interface.
Network	EN-30151	When Flexible Line Key (FLK) is enabled, the PSTN contact name is not synchronized and displays as a telephone number on the idle screen.
Network	EN-30084	The pairing of the base station with VVX phone is lost after the base station is powered off and then powered on again.
Network	EN-30155	The phone displays an error message on a successful blind transfer when the Session Description Protocol (SDP) origin session-id and session-version do not match with the previous INVITE request.
Security	EN-30137	This release fixes a medium-severity vulnerability (CVSSv3 score = 5.4) in the product's web user interface. An independent security researcher, Francis Alexander, has reported this vulnerability to Beyond Security's SecuriTeam Secure Disclosure program, who in turn alerted Polycom.
User Interface	EN-30126	The position of GuestIn soft key changes incorrectly when the parameter `softkey.feature.directories` is enabled or disabled.

Known Issues

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

These release notes do not provide a complete listing of all known issues that are included in the software. Issues not expected to significantly impact customers with standard voice or video conferencing environments may not be included. In addition, the information in these release notes is provided as-is at the time of release and is subject to change without notice.

Known Issues

Category	Issue ID	Release	Description	Workaround
Audio	VOIP-123096	5.5.0	VVX 600 and VVX 601 phones do not receive audio from Group Series in encrypted H.323 calls above 768k.	No workaround currently available.
Calling	VOIP-116653	5.5.0	After a Barge-in call is placed on hold, the handset still displays options to transfer and blind transfer the call.	No workaround currently available.

Known Issues

Category	Issue ID	Release	Description	Workaround
Calling	VOIP-99645	4.0.1	When you place a call on the SoundStructure VoIP Interface while there is an incoming call, the incoming call is ignored and no longer rings if the new call is ended. You can still answer the incoming call until it disconnects.	No workaround currently available.
Calling	VOIP-116259	5.5.0	In the calendar events with multiple phone numbers, the dial option does not list the numbers correctly.	No workaround currently available.
Calling	EN-33313	5.5.3	When the VVX business media phone receives a call, the speaker button light blinks twice.	No workaround currently available.
Calling	EN-34581	5.5.3	After disconnecting the call in VVX business media phone, some handsets display "The line is busy" on the screen.	No workaround currently available.
Calling	EN-33764	5.5.3	The “Do Not Disturb is Enabled” popup is still displayed on the handset after disabling Do Not Disturb (DND) option.	No workaround currently available.
Calling	EN-34488	5.5.3	When the VVX business media phone is configured with Opus codec, 2-way audio is not established between the phones.	No workaround currently available.
Expansion Module	VOIP-116348	5.5.0	On a phone with an Expansion Module connected, the first Expansion Module line is not cleared after you lock the device.	No workaround currently available.
General	VOIP-124707	5.5.2	MWI blinks on the handset, after the base station is unpaired.	No workaround currently available.
Interoperability D60 Handset	VOIP-117097	5.5.0	On a VVX phone paired with two D60 handsets, the second handset is unable to place a call after ending an intercom call with the first handset.	No workaround currently available.
Interoperability TR069	VOIP-111332	5.5.0	If you schedule a file to download from the TR069 server and then disconnect the power cord from the phone one minute before the scheduled time, the file is not downloaded when you reconnect the power cord and power the phone on again.	No workaround currently available.
Network	VOIP-116151	5.5.0	The phone incorrectly sends the “Ethernet Frame Check Sequence Incorrect” message in remote packets.	No workaround currently available.

Known Issues

Category	Issue ID	Release	Description	Workaround
Network	EN-30066	5.6.0	When the VVX business media phone is paired with VVX D60 base station via PC port and the VVX PoE is plugged in and plugged out instantly, the phone loses pairing.	Unplug the PC port of the VVX D60 Base Station to re-pair.
Network	EN-30086	5.4.7	If a VVX business media phone paired via PC port to a VVX D60 Base is powered off, the pairing is lost in sometime after the power is on.	Unplug the PC port of the VVX D60 Base Station to re-pair.
Skype for Business	VOIP-126033	5.5.2	In the VVX 201 phones, for a Skype for Business conference call, phone displays the **Hold** icon for a moment before it updates to the **Mute** icon in specific cases.	No workaround currently available.
SIP	VOIP-116412	5.5.0	Including the “&” character in a user's SIP URI prevents the user's status from changing.	No workaround currently available.
User Interface	VOIP-116471	5.5.0	When you edit a contact in the Local Directory, scrolling up does not work correctly.	No workaround currently available.
User Interface	VOIP-114345	5.5.0	The Idle Browser does not display the HTTPS:// page.	No workaround currently available.
User Interface	VOIP-116826	5.3.0	On the Favorites screen, pressing the empty third and fourth soft keys incorrectly display the Info screen.	No workaround currently available.
User Interface	VOIP-116387	5.5.0	After restarting a VVX 500 phone with an Expansion Module and a headset attached, the “Digital headset attached” message does not appear.	No workaround currently available.
User Interface	VOIP-113852	5.5.0	Pressing the back arrow from the Contact Directory takes you to the idle screen instead of to the Directories Menu.	No workaround currently available.
User Interface	VOIP-123491	5.5.2	Unregistered line displays as “Blank Line” below the **DND** or **Call Forwarding** menu on the handset.	No workaround currently available.
User Interface	VOIP-124681	5.5.2	The **New Call** soft key is displayed for fraction of a second before it transitions to the **Split** soft key after establishing the conference.	No workaround currently available.

Known Issues

Category	Issue ID	Release	Description	Workaround
User Interface	VOIP-125358	5.5.2	“Call forward Menu” name displays as “Call Transfer” (Transfert d 'appel) in French.	No workaround currently available.
Web Interface	VOIP-113193	5.5.0	The VVX D60 web interface line management page does not show the default line.	No workaround currently available.

Updates to Previous Software Releases

This section describes new features and enhancements to previous UC Software releases.

What’s New in Polycom UC Software 5.5.2

The Polycom UC Software 5.5.2 release includes the following new features and enhancements:

Enterprise Directory Default Search

You can view an initial list of contacts in the Enterprise directory. The administrator can configure the phone using the parameter `feature.broadsoftdir.showDefaultSearch` and enable the feature by setting the parameter value to 1. This allows the user to view a list of contacts even when no text is entered in the search box of the directory. To view the list of initial contacts, navigate to **Menu** > **Directories** > **Enterprise Directory**.

BroadSoft Directory Parameters

You can view an initial list of contacts in the Enterprise directory. You must enable the feature using the parameter `feature.broadsoftdir.showDefaultSearch` that allows the user to view the initial list of contacts even when no text is entered in the search box of the directory.

BroadSoft Directory Search Parameters

Parameter Template	Permitted Values
`feature.broadsoftdir.showDefaultSearch` features.cfg	Displays an initial list of contacts for an empty search box when enabled. 0 (default) – No contacts are displayed when the search box field is empty. 1 – Enables the user to view the initial list of contacts

Search the BroadSoft Directory

After the administrator configures the Enterprise directory using the `feature.broadsoftdir.showDefaultSearch` configuration parameter, you can view a list of contacts by default in the BroadSoft directory. The default contacts list is displayed even when no text is entered in the search box of the directory.

To view the initial list of contacts:

» Navigate to **Directories > BroadSoft Directory.**

The default initial list of contacts is displayed.

Registration Line Address in Status Bar

You can view the complete registration line address in the status bar of the phones screen. The administrator enables the feature by setting the parameter `lcl.status.LineInfoAtTop=1`. When enabled, the `lcl.status.LineInfoAtTopText` parameter provides the text to be displayed in the status bar. This feature is not available for Skype for Business.

Unique Line Label for Registration Lines Parameters

You can view the complete registration line address in the status bar of the phones screen. You must enable the feature using the configuration parameter `lcl.status.LineInfoAtTop`. When enabled, the `lcl.status.LineInfoAtTopText` parameter provides the text to be displayed in the status bar of the phone. You can enable the feature by setting the value of the `lcl.status.LineInfoAtTop` parameter to 1. This feature is not available for Skype for Business.

Registration Line Address Bar Parameters

Parameter Template	Permitted Values
`lcl.status.LineInfoAtTop` site.cfg	Displays the complete registration line address in the status bar of the phones screen. 0 (default) – Does not display the registration line address in the status bar of the phone's screen. 1 – The `lcl.status.LineInfoAtTopText` parameter provides the text to be displayed in the status bar of the phone.

BroadWorks Anywhere EFK for Soft Keys

Administrators can configure a soft key that enables users to navigate directly to the BroadWorks Anywhere feature menu using an Enhanced Feature Key (EFK). The user can directly view the **BroadWorks Anywhere** menu and bypass navigating to **Settings** > **Features** > **UC-One Call Settings** > **BroadWorks Anywhere**. The following EFK macro is added to support this feature:

- $**FBWSAnyWhere**$

BroadWorks Anywhere

You can configure a soft key for the BroadWorks Anywhere feature that enable users to navigate directly to the feature menu using an Enhanced Feature Key (EFK). This allows the users to bypass navigating to **Settings > Features > UC-One Call Settings > BroadWorks Anywhere**. You can configure the soft key using the following EFK macro to support this feature:

- $**FBWSAnyWhere**$

Hide Contact Directory and Favorites

Using the parameter `dir.local.UIenabled`, administrators configure the phones to hide the Contact Directory and Favorites options from all screens in the user interface on all VVX phones except the VVX 1500 phone. Moreover, the administrator can set the local directory as read only by using the `dir.local.readonly` parameter. When the `dir.local.readonly` parameter is enabled, the user is restricted to modifying the speed dials only.

Local Contact Directory Parameters

You can configure the phone to hide the Contact directory and Favorites options from all screens in the user interface on all VVX phones except the VVX 1500 phone. You can enable this feature using the `dir.local.UIenabled` configuration parameter that removes the Contact directory and Favorites options from the screen.

In addition, make sure the `dir.local.readonly` parameter is enabled to restrict the users to modify speed dials.

Hide Contacts and Favorites Options Parameters

Parameter Template	Permitted Values
`dir.local.UIenabled` features.cfg	Removes the Contact directory and Favorites option from all screens in the user interface on VVX phones except VVX 1500. 1 (default) – Displays the Contact directory and Favorites option on the phone's screen. 0 – The Contact directory and Favorites option is not displayed.

Personal Directory

You can now Add/ Delete/ Modify contacts of your choice in the Personal Directory. The administrator enables this feature by using the parameter `feature.broadsoftPersonalDir.enabled` that allows the user to select contacts from the BroadSoft Personal Directory and add to the local directory**.**

BroadSoft Personal Directory Parameters

You can configure the phone to Add/ Delete/ Modify the contacts of your choice in the Personal Directory. You must enable this feature using the parameter `feature.broadsoftPersonalDir.enabled` that allows the users to select the contacts of their choice from the BroadSoft Personal Directory and add to the local directory. When enabled, users can perform the following actions:

- Add a contact to personal directory
- Delete the selected contact from the personal directory
- Modify the details of the selected contact from the personal directory

Personal Directory Parameters

Parameter Template	Permitted Values
`feature.broadsoftPersonalDir.enabled` features.cfg	Allows the user to add/delete/modify contacts to the Personal Directory. 0 (default) – Personal Directory feature is disabled. 1 – Personal Directory feature is enabled.

BroadSoft Personal Directory

After the administrator configures the BroadSoft Personal Directory, you can do the following:

- Add a contact to personal directory
- Delete the selected contact from the personal directory
- Modify the details of the selected contact from the personal directory

BroadSoft Server-Based Call Logs

You can view the list of all call logs on tapping the **Recent** soft key on phones screen. The administrator configures the phone using the parameter `feature.broadsoft.callLogs=Basic` that allows the user to view the call logs.

Call Log Parameters

You can configure the phone to view the list of call logs when the user taps the Recent soft key on the phones screen. You can enable this feature by setting the value of the `feature.broadsoft.callLogs` parameter to Basic. When enabled, users can view the call logs from the server on the phone.

BroadSoft Server-Based Call Log Parameters

Parameter Template	Permitted Values
`feature.broadsoft.callLogs` features.cfg	Allows the user to view the call logs from the server. Basic – The BSFT server call logs feature is enabled. Disabled – The BSFT server call logs feature is disabled

New Call Forwarding Icons

The Call Forwarding icons are updated and new icons are added for all of the Call Forwarding options: all calls, no answer, and busy. When Call Forwarding is enabled on any line on a VVX phone, an icon displays in the status bar depending on which Call Forwarding option is selected. The following table includes the Call Forwarding icons added in this release.

Call Forwarding Icons

Feature	Icon
Call Forwarding – All Calls	
Call Forwarding – No Answer	
Call Forwarding – Busy	

Updated Do Not Disturb Icon

The Do Not Disturb (DND) icon was updated from  to  and displays on the line key and in the status bar when the feature is enabled.

Expanded Support for USB Headsets

Support for the following Plantronics USB Headsets with VVX 500, VVX 600, VVX 501, VVX 601, and VVX 401 phones has been added to this release:

- Blackwire C310
- Blackwire C325
- Blackwire C725
- Blackwire C325.1
- Plantronics CS520
- EncorePro HW540
- DA80 Headset Adapter

Support Added for CDP in VVX D60 Base Station

In the environment where the VVX phone and D60 Base Station is paired, Cisco Discovery Protocol (CDP) has been enabled for D60 Base Station for VLAN discovery.

Pairing the VVX D60 Base Station

You can pair the VVX D60 Base Station to a VVX business media phone using the local phone interface or the Web Configuration Utility. You can use the following methods to pair the base station with a VVX business media phone:

- PC Port Pairing

- Automatic Pairing
- Manual Pairing

The D60 Base Station can access Voice VLAN through Link-Layer Discovery Protocol (LLDP) and Cisco Discovery Protocol (CDP). CDP is supported on the D60 Base Station and is enabled on the D60 Base Station by default.

After connecting the D60 Base Station to a LAN port, allow the Base Station at least one minute to connect to the voice VLAN network and to acquire an IP address. Wait at least one minute after connecting the Base Station to a LAN port before pairing the base station with a VVX business media phone.

Configure VVX D60 Network Settings

By default, you can edit network settings for the VVX D60 Base Station. You can use the Web Configuration Utility to make changes to the Base Station's network settings.

To configure VVX D60 network settings:

1. In a web browser, enter **https://<IP address of D60 Base Station>**
2. In the Web Configuration Utility, enter the following default credentials:
   - User name: **Polycom**
   - Password: **456**
3. Navigate to **Settings > Network Settings**.

ALLOW Header in 18x Provisional Responses

Polycom phones are adding ALLOW header in the INVITE and its final and provisional responses using the parameter `voIpProt.SIP.header.allow.enable`.

Skype for Business SIP and Tel URI

In Skype for Business environments, the phone places the last dialed method for a contact either through SIP URI or Tel URI and dials via the same method the next time until it reboots to the default SIP URI.

Last Call Return Parameters

Using the `feature.broadsoft.basicCallLogs.redial.enabled` parameter, you can configure the Last Call Return feature, which allows user to redial the last number placed from any device connected to the same line or registration. When users press the redial button on any phone connected to the same line, the redial feature redials the last called number.

Last Call Return Parameters

Parameter Template	Permitted Values
`feature.broadsoft.basicCallLogs.redial.enabled` features.cfg	Allows the redial feature to retrieve the last number dialed on any device connected to same line or registration. 0 (Default) – Disables the feature to redial the last number called from any device connected to same line. 1 – Allows you to redial the last number called from any device connected to same line.

BroadSoft Directory Support

The BroadSoft directories enable users to search and view their personal, group, or enterprise contacts list. When the BroadSoft directories are integrated with Polycom UC-One Application and enabled, you can perform a search on all the directories. There are five types of BroadSoft Directories:

- **Enterprise Directory:** You can search and view Active Directory global address list of an enterprise and BroadSoft call list (Enterprise Directory and personal number lists). To access the contact information, you can query by first name, last name, and departments.
- **Group Directory:** You can view the contact information such as work, extension, and mobile numbers. Group Directory also allows you to place a call to anyone in the user's group within the enterprise.
- **Group Common Directory:** You can view the contact details such as names and phone numbers of common contacts from different groups within an enterprise.
- **Enterprise Common Directory:** You can view the contact information such as names and phone numbers of common contacts within an enterprise.
- **Personal Directory:** You can view the contact information in the user’s personal directory stored on the server. When this directory is enabled, users can select contacts from the BroadSoft Personal Directory and add to the local directory.

BroadSoft Directories

Your system administrator can configure the directory setup for either Group, Group Common, Enterprise Common or Personal Directory so that you can access BroadSoft directories.

You can do the following using BroadSoft directories:

- Search for global enterprise contacts
- View Group Directory
- View contacts in Enterprise Common Directory
- Search for contacts in your Personal Directory

View a list of Global Enterprise Directory Contacts

You can view all the global enterprise contacts or perform a search in the Enterprise directory. You can also query by first name, last name, department, and access contact information.

To view the Global Enterprise Directory Contacts

» Select **Directories > Enterprise Directory**.

View a list of Group Directory Contacts

You can view work designation, extension number, mobile number of the selected contact. You can also place a call to anyone within the user's group.

To view the Group Directory Contacts:

» Select **Directories > Group Directory.**

View a list of Group Common Directory Contacts

You can view names and contacts existing commonly in various groups within the enterprise.

To View the Group Common Directory Contacts

» Select **Directories > Group Common Directory**.

View a list of Enterprise Common Directory Contacts

You can view names and contacts existing commonly within the enterprise.

To View the Enterprise Common Directory Contacts

» Select **Directories > Enterprise Common Directory**.

View a list of Personal Directory Contacts

You can view contact details of users in personal directories stored on the BroadSoft server. You can also select and store the contacts from BroadSoft Personal Directory to your local directory.

To View the Personal Directory Contacts

» Select **Directories > Personal Directory**.

BroadSoft Anonymous Call Reject

In previous releases, the following parameter name is missing an 'l' and is incorrect: `feature.broadsoft.xsi.AnonymousCalReject.enabled`. The correct name for the parameter is: `feature.broadsoft.xsi.AnonymousCallReject.enabled`.

Call Waiting Alerts

The server-based Call Waiting feature enables the server to manage incoming calls while a user is in an active call. When the user changes the call waiting state from the phone, the phone sends a request to the server to update to the new state. You can enable this feature using the `feature.broadsoft.xsi.callWaiting.enabled` parameter.

BroadSoft Server-Based Call Waiting Parameters

The server-based call waiting feature enables the server to manage incoming calls while a user is in an active call. When the user changes the call waiting, the phone sends a request to the server to update to the new state.

Call Waiting Alerts Parameters

Parameter Template	Permitted Values
`feature.broadsoft.xsi.callWaiting.enabled` features.cfg	Allows the server to manage the incoming calls while a user is in an active call. 0 (Default) – Disables the feature to manage incoming calls by the server. 1 – Allows the server to manage the incoming calls.

Security Banner on the Web Configuration Utility

You can enable or disable the security banner on the Web Configuration Utility using this feature. When enabled, a security banner displaying the message configured, using the parameter `feature.webSecurityBanner.msg` displays on the web user interface before the user logs in. To enable or disable this feature, use the `parameter feature.webSecurityBanner.enabled`.

Security Banner Parameters

You can enable or disable the security banner on the Web Configuration Utility using `feature.webSecurityBanner.enabled`. You can configure a custom text message to be displayed on the security banner of your phone screen. To configure the text message, use the parameter `feature.webSecurityBanner.msg`.

Security Banner Parameters

Parameter Template	Permitted Values
`feature.webSecurityBanner.enabled` site.cfg	0 (default) – No security banner message is displayed on the phone's web user interface. 1 – A security banner with the configured message through `feature.webSecurityBanner.msg` is displayed on the phone's web user interface
`feature.webSecurityBanner.msg` site.cfg	Customize the text in security banner. (default) This is being displayed because the security log-on banner has been enabled and the custom text to be displayed in the security log-on banner has not been configured. 2000 characters (maximum)

Web Sign In Using Skype for Business

You can enable web sign in on the phone using the parameter `feature.webSignIn.enabled`. After the parameter is configured, you can enable or disable web sign-in for your Skype for Business profile.

Web Sign In for Skype for Business Online Parameters

You can enable web sign in on the phone using the parameter `feature.webSignIn.enabled`.

Web Sign In Parameters

Parameter Template	Permitted Values
`feature.webSignIn.enabled` features.cfg	1 (default) – In Skype for Business Base Profile, the web sign in option is displayed on the phone for the user. 0 – In Skype for Business Base Profile, the web sign in option is not displayed on the phone for the user.

Intercom Calls

In previous releases, the following parameter name includes an 's' and is incorrect: `homescreen.intercom.enable`. The correct name for the parameter is: `homeScreen.intercom.enable`.Transport Layer Security (TLS) Protocol Configuration

You can set the TLS protocol for the following applications using the specified parameters:

Browser application: `sec.TLS.protocol.browser`.

LDAP (Light weight directory access) application: `sec.TLS.protocol.ldap`

SIP application: `sec.TLS.protocol.sip`

SOPI (SIP Open Programmable Interface) application: `sec.TLS.protocol.sopi`

Web server application: `sec.TLS.protocol.webServer`

XMPP (Extensible messaging and presence protocol) application: `sec.TLS.protocol.xmpp`

Syslog application: `device.sec.TLS.protocol.syslog`

Provisioning application: `device.sec.TLS.protocol.prov`

802.1x application: `device.sec.TLS.protocol.dot1x`

TLS Protocol Configuration Parameters

You can configure the TLS Protocol for the following supported applications using specific parameters: Browser application, LDAP application, SIP application, SOPI application, Web server application, XMPP application, Syslog application, Provisioning application, and 802.1x application.

The parameters used to configure the applications are as follows.

TLS Protocol Parameters

Parameter Template	Permitted Values
`sec.TLS.protocol.browser` device.cfg, site.cfg	TLSv1_0 (default) - Handshake from phone happens with TLS 1.0 version only. SSLv2v3 - Handshake from phone starts with the highest TLS version and configured value will be the lowest TLS/SSL version used for negotiation.
`sec.TLS.protocol.ldap` device.cfg, site.cfg	The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2
`sec.TLS.protocol.sip` device.cfg, site.cfg	Controls the TLS Protocol used for SIP signaling. Configures the phone to use the selected protocol for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2

TLS Protocol Parameters

Parameter Template	Permitted Values
`sec.TLS.protocol.sopi` device.cfg, site.cfg	Controls the TLS Protocol used for SOPI. Configures the phone to use the selected protocol for a handshake negotiation from the phone for controlling the TLS protocol settings for the SOPI. The handshake from the phone starts with the highest TLS version when set to "SSLv2v3", otherwise the phone uses "TLS 1.0" TLSv1_0 (default) SSLv2v3
`sec.TLS.protocol.webServer` device.cfg, site.cfg	Controls the TLS Protocol used for Web Server. Configures the phone to use the selected protocol for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2
`sec.TLS.protocol.xmpp` device.cfg, site.cfg	Configures the phone to use the selected protocol for a handshake negotiation from the phone. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2
`device.sec.TLS.protocol.syslog` device.cfg, site.cfg	Controls the TLS Protocol used for Syslog. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2

TLS Protocol Parameters

Parameter Template	Permitted Values
`device.sec.TLS.protocol.prov` device.cfg, site.cfg	Controls the TLS Protocol used for Provisioning. TLSv1_0 (default) SSLv2v3 Handshake from the phone starts with the highest TLS version when set to "SSLv2v3", otherwise the phone uses "TLS 1.0".
`device.sec.TLS.protocol.dot1x` device.cfg, site.cfg	Controls the TLS Protocol used for 802.1x authentication. TLSv1_0 (default) Handshake from the phone will happen with "TLS 1.0" only.

Soft Key Update

In previous releases, the maximum character string the `softkey.x.action` parameter supported was 255 and is now changed to 2048 characters.

Call Waiting Menu

You can configure the phone to control the display of Call Waiting menu. After you configure the `up.callWaitingMenu.enable` parameter, the Call Waiting menu is displayed under the Preferences option on the phone.

Call Waiting Parameters

You can configure the phone using `up.callWaitingMenu.enable` parameter to allow the phone to control the display of Call Waiting menu.

Call Waiting Menu Parameters

Parameter Template	Permitted Values
`up.callWaitingMenu.enable` features.cfg	1 (default) – The phone displays Call Waiting menu in Preferences menu. 0 – The phone does not display the Call Waiting menu.

Device Lock

You can configure your phone to configure the order of display for Authorized/Emergency numbers, when device is locked in Lync profile. Using the `up.configureDeviceLockAuthList` parameter you can enable this feature.

Device Lock Parameters

You can configure your phone using the `up.configureDeviceLockAuthList` parameter to set the order of display for Authorized/Emergency numbers, when device is locked in Skype for Business profile.

Device Lock Parameters

Parameter Template	Permitted Values
`up.configureDeviceLockAuthList` features.cfg	Controls the display of Authorized list and Emergency list when the phone is locked. EmergencyNumberAtTop (default) - The E911 emergency number will be displayed followed by authorized numbers. EmergencyNumberAtBottom - The authorized numbers will be displayed followed by the E911 number. EmergencyNumberDisabled - Only the authorized numbers will be displayed.

Hide System IP Address

You can configure your phone to control the display of IP address of the phone. Using `up.hideSystemIpAddress` parameter, you can show or hide the IP address of the phone on the phones screen.

Call Control

You can configure your phone to control the ALLOW header in the 18x provisional response. Using the `voIpProt.SIP.header.allow.enable` parameter the header can either be included or excluded for `INVITE`.

Call Control Parameters

You can configure your phone using the `voIpProt.SIP.header.allow.enable` parameter to control the `ALLOW` header in the 18x provisional response.

Call Control Parameters

Parameter Template	Permitted Values
`voIpProt.SIP.header.allow.enable` sip-interop.cfg	Controls the ALLOW header in 18x provisional response. 0 (default) - The header will not be included for INVITE. 1 - The header will be included for INVITE.

Hardware and Accessories

The Polycom UC Software allows your phone to support a digital Rx boost for the handset and headset. Using the `voice.handsetHeadset.rxdg.offset` parameter, the phone offsets the RxDg range by a specified number of decibels for handset and headset.

Handset and Headset Parameters

You can configure your phone to support a digital Rx boost for the handset and headset using the `voice.handsetHeadset.rxdg.offset` parameter.

Headset and Handset Parameters

Parameter Template	Permitted Values
`voice.handsetHeadset.rxdg.offset` techsupport.cfg	This parameter allows a digital Rx boost for the handset and headset. 0 (default) 9 to -12 – Offsets the RxDg range of the handset and headset by the specified number of decibels.

Push-to-Talk for Handsfree

The Polycom UC Software allows your phone to support a digital Rx boost for Push-to-Talk. Using the `voice.handsfreePtt.rxdg.offset` parameter, the phone offsets the RxDg range by a specified number of decibels.

Push-to-Talk Parameters

You can configure the phone to support a digital Rx boost for Push-to-Talk on a handsfree mode using the `voice.handsfreePtt.rxdg.offset` parameter.

Push-to-Talk for Handsfree Parameters

Parameter Template	Permitted Values
`voice.handsfreePtt.rxdg.offset` techsupport.cfg	This parameter allows a digital Rx boost for Push To Talk. 0 (default) 9 to -12 – Offsets the RxDg range of the hands-free and hands-free Push-to-Talk (PTT) by the specified number of decibels.

Push-to-Talk for Handsfree Paging

The Polycom UC Software allows your phone to support a digital Rx boost for Push-To-Talk handsfree paging. Using the `voice.ringerPage.rxdg.offset` parameter, the phone offsets the phone can increase or decrease the volume of the ringer and hands-free page by the specified number of decibels.

Group Paging Parameters

You can configure the phone to support a digital Rx boost for Push-to-Talk handsfree paging using the `voice.ringerPage.rxdg.offset` parameter.

Push-to-Talk Parameters

Parameter Template	Permitted Values
`voice.ringerPage.rxdg.offset` techsupport.cfg	This parameter allows a digital Rx boost for Push To Talk. Use this parameter for handsfree paging Rx in high noise environments. 0 (default) 9 to -12 – Raise or lower the volume of the ringer and hands-free page by the specified number of decibels.

Executive Assistant

The Polycom UC Software allows your phone to control the display of executive soft key on the idle screen of the phone. Using `feature.BSExecutiveAssistant.execSk` parameter, you can enable this feature.

Executive Assistant Parameters

You can configure your phone to control the display of executive soft key on the idle screen of the phone using `feature.BSExecutiveAssistant.execSk` parameter.

Executive Assistant Parameters

Parameter Template	Permitted Values
`feature.BSExecutiveAssistant.execSk` features.cfg	Controls the display of Exec soft key on the idle screen of the phone. 0 (default) - The soft key will not be displayed. 1 - The soft key will be displayed.

Phone Language

You can control the language settings of the phone. Using the `device.spProfile` parameter, you can set the phone language of your choice.

Phone Language Parameters

You can configure your phone using the `device.spProfile` parameter that allows you to chose the phone language of your choice.

Phone Language Parameters

Parameter Template	Permitted Values
`device.spProfile` device.cfg	Controls the language setting of the phone. Default (default) - The default language is English. DT - The default language will be German.

Polycom CMA Account

In previous releases, the maximum value for `device.logincred.password` parameter was 255 and is now changed to 32.

Call Transfer

In the previous releases, In previous releases, the following parameter name has 'D' and is changed: `call.DefaultTransferType`. The modified name for the parameter is: `call.defaultTransferType.`

Restrict Attendant for Call Pickup

The Polycom UC Software allows you to pick up a monitored call. Using the `attendant.restrictPickup` parameter you can configure the phone to either allow the attendant to pick up or not pick up the monitored call.

Busy Lamp Field Parameters

You can configure the phone to restrict the attendant to pick up a monitored call by using `attendant.restrictPickup` parameter to enable this feature.

Busy Lamp Field Parameters

Parameter Template	Permitted Values
`attendant.restrictPickup` features.cfg	Allows picking up of a monitored call. 0 (default) - The attendant can pick up the call. 1 - The attendant cannot pick up the call.

VVX D60 Wireless Handsets

You can configure the phone to allow users to chose which lines to map to the wireless handset. Using `feature.VVXD60.allowLineMappings` parameter you can enable this feature.

Acoustic Echo Cancellation

The Polycom UC Software provides the Acoustic Echo Cancellation (AEC) feature to remove the echo of the local loudspeaker from the local microphone without removing the true near end speech. The `voice.aes.hf.duplexBalance` parameter enables this feature.

Acoustic Echo Cancellation Parameters

You can configure the Acoustic Echo Cancellation (AEC) feature to remove the echo of the local loudspeaker from the local microphone without removing the true near end speech by using the `voice.aes.hf.duplexBalance` parameter.

The AEC feature includes the following:

- Talk State Detector: Determines whether the near end user, far end user, or both are speaking.
- Linear Adaptive Filter: Adaptively estimates the loudspeaker-to-microphone echo signal and subtracts that estimate from the microphone signal.
- Non-linear Processing: Suppresses any echo remaining after the Linear Adaptive Filter.

Acoustic Echo Cancellation Parameters

Parameter Template	Permitted Values
`voice.aes.hf.duplexBalance` debug.cfg	0 Max Echo Control (default) - Balances the Acoustic Echo Suppression to maximize echo control. With this balance setting the phone hands free may be less full-duplex during continuous double-talk (near and far end users speak simultaneously). 1 - Max Full Duplex: Balances the Acoustic Echo Suppression to maximize full duplex. With this balance setting the phone hands free may be more susceptible to echo during continuous double-talk or when moving the phone or objects near the phone.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.5.2.

Configuration File Enhancements

Parameter Template	Permitted Values
`attendant.restrictPickup` features.cfg	Allows picking up of a monitored call. 0 (default) - The attendant can pick up the call. 1 - The attendant cannot pick up the call.
`call.defaultTransferType` features.cfg	Sets the transfer type the phone uses when transferring a call. Consultative (default) - Puts the call on hold while placing a new call to the other party. Blind - Transfers the call to another party.
`device.logincred.password` device.cfg, features.cfg	Specifies the length of the CMA account password. The maximum length of the string is 32 characters. Null (default) String
`device.net.etherStormFilterPpsValue` device.cfg,site.cfg	Controls packets per second for storm filter and the incoming network traffic. 0 (default) – When set to 0, the `device.net.etherStormFilterPpsValue` parameter cannot be configured. 1 – When set to 1, the `device.net.etherStormFilterPpsValue` parameter can be configured.
`device.sec.TLS.protocol.dot1x` device.cfg, site.cfg	Controls the TLS Protocol used for 802.1x authentication. TLSv1_0 (default) Handshake from the phone will happen with "TLS 1.0" only.
`device.sec.TLS.protocol.syslog` device.cfg, site.cfg	Controls the TLS Protocol used for Syslog. The handshake from phone starts with the highest TLS version irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2

Configuration File Enhancements

Parameter Template	Permitted Values
`device.sec.TLS.protocol.prov` device.cfg, site.cfg	Controls the TLS Protocol used for Provisioning. TLSv1_0 (default) SSLv2v3 Handshake from the phone starts with the highest TLS version when set to "SSLv2v3", otherwise the phone uses "TLS 1.0".
`device.spProfile` device.cfg	Controls the language setting of the phone. Default (default) - The default language is English. DT - The default language will be German.
`dir.broadsoft.regMap` features.cfg	Specifies the registration line credentials to retrieve directory information from BroadSoft UC-One directory when `dir.broadsoft.useXspCredentials=0`. 1 (default) Const_NumLineReg
`dir.corp.alt.cacheSize` features.cfg	Specifies the maximum number of entries that can be saved in the cache memory. 64 (default) 32 to 64 – The maximum number of entries that can be cached locally on the BER platform.
`dir.corp.alt.pageSize` features.cfg	Specifies the maximum number of entries requested from the corporate directory server. 16 (default) 8 to 32 – The maximum number of entries requested from the corporate directory server with each query on the BER platform.
`dir.corp.cacheSize` features.cfg	Specifies the maximum number of entries that can be saved in the cache memory. 64 (default) 32 to 64 – The maximum number of entries that can be cached locally on the BER platform.
`dir.corp.pageSize` features.cfg	Specifies the maximum number of entries requested from the corporate directory server. 16 (default) 8 to 32 – The maximum number of entries requested from the corporate directory server with each query on the BER platform
`dir.local.readonly` features.cfg	User cannot modify the speed dial when enabled. 0 (default) - Read only mode for local directory is disabled. 1 – Read only mode for local directory is enabled.

Configuration File Enhancements

Parameter Template	Permitted Values
`dir.local.UIenabled` features.cfg	Determines whether users can access Favorites and the Contact Directory from the Directories menu or the Dialpad. 1 (default) – Displays Favorites and Contact Directory options in the Directories menu, and displays the Favorites quick access menu on the Home screen of the VVX 500/501 and 600/601 phones. 0 – Hides the Contact Directory and Favorites options in the Directories menu and the Dialpad. On VVX 500/501 and 600/601, hides the Favorites quick access menu on the Home screen.
`feature.broadsoft.basicCallLogs.redial.enabled` features.cfg	Allows the redial feature to retrieve the last number dialed on any device connected to same line or registration. 0 (default) – Disables the feature to redial the last number called from any device connected to same line. 1 – Allows you to redial the last number called from any device connected to same line.
`feature.broadsoft.callLogs` features.cfg	Allows the user to view the call logs from the server. Basic – Enables the BSFT server call logs feature. Disabled – Disables the BSFT server call logs feature.
`feature.broadsoftdir.showDefaultSearch` features.cfg	0 (default) – Disables the Enterprise Directory default search feature. 1 – The Enterprise Directory default search feature allows the users to view the initial list of contacts by default.
`feature.broadsoftPersonalDir.enabled` features.cfg	Allows the user to add, delete, or modify contacts to the Personal Directory. 0 (default) – Personal Directory feature is disabled. 1 – Personal Directory feature is enabled.
`feature.BSExecutiveAssistant.execSk` features.cfg	Controls the display of Exec soft key on the idle screen of the phone. 0 (default) - The soft key will not be displayed. 1 - The soft key will be displayed.
`feature.VVXD60.allowLineMappings` features.cfg, device.cfg	Allows users to choose which lines to map to the wireless handset. 0 (default) - Only the administrator can map. 1 - Both, the user and administrator can map.

Configuration File Enhancements

Parameter Template	Permitted Values
`feature.broadsoft.xsi.callWaiting.enabled` features.cfg	Allows the server to manage the incoming calls while a user is in an active call. 0 (default) – Disables the feature to manage incoming calls by the server. 1 – Allows the server to manage the incoming calls.
`feature.webSecurityBanner.enabled` site.cfg	0 (default) – No security banner message is displayed on the phone's web user interface. 1 – A security banner with the configured message through `feature.webSecurityBanner.msg` is displayed on the phone's web user interface
`feature.webSecurityBanner.msg` site.cfg	Customize the text in security banner. (default) - Displays the default text because the security log-on banner is enabled and the custom text to be displayed in the security log-on banner is not configured. 2000 characters (maximum) 1 character (minimum)
`feature.webSignIn.enabled` features.cfg	1 (default) – In Skype for Business profile, web sign in option is displayed on the phone for the user. 0 – In Skype for Business profile, web sign in option is not displayed on the phone for the user.
`lcl.status.LineInfoAtTop` features.cfg	0 (default) – Disables the feature. 1 – Allows the registration line address to appear in the status bar.
`lcl.status.LineInfoAtTopText` features.cfg	Specifies the text to be displayed in the status bar. 0 – minimum length 14 – maximum length
`sec.TLS.protocol.browser` device.cfg, site.cfg	TLSv1_0 (default) - Handshake from phone happens with TLS 1.0 version only. SSLv2v3 - Handshake from phone starts with the highest TLS version and configured value will be the lowest TLS/SSL version used for negotiation.
`sec.TLS.protocol.ldap` device.cfg, site.cfg	The handshake from phone starts with the highest TLS version only irrespective of value configured, and the configured value is the lowest TLS/SSL version used for negotiation. TLSv1_0 (default) SSLv2v3, TLSv1_1, TLSv1_2
`sec.TLS.protocol.sip` device.cfg, site.cfg	Controls the TLS Protocol used for SIP. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2

Configuration File Enhancements

Parameter Template	Permitted Values
`sec.TLS.protocol.sopi` device.cfg, site.cfg	Controls the TLS Protocol used for SOPI. The handshake from the phone starts with the highest TLS version when set to "SSLv2v3", otherwise the phone uses "TLS 1.0" TLSv1_0 (default) SSLv2v3
`sec.TLS.protocol.xmpp` device.cfg, site.cfg	Controls the TLS Protocol used for XMPP. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2
`sec.TLS.protocol.webServer` device.cfg, site.cfg	Controls the TLS Protocol used for Web Server. TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2
`softkey.x.action` features.cfg	Controls the action or function for the custom soft key x. This value uses the same macro action string syntax as an Enhanced Feature Key. Null (default) macro action string, 2048 characters
`up.callWaitingMenu.enable` features.cfg	1 (default) – The phone displays Call Waiting menu in Preferences menu. 0 – The phone does not display the Call Waiting menu.
`up.configureDeviceLockAuthList` features.cfg	Controls the display of Authorized list and Emergency list when the phone is locked. EmergencyNumberAtTop (default) - The E911 emergency number will be displayed followed by authorized numbers. EmergencyNumberAtBottom - The authorized numbers will be displayed followed by the E911 number. EmergencyNumberDisabled - Only the authorized numbers will be displayed
`up.hideSystemIpAddress` features.cfg	Controls the display of IP address of the phone. Nowhere (default) - The IP address is displayed in the respective UI menu. Everywhere - The IP address is not displayed.

Configuration File Enhancements

Parameter Template	Permitted Values
`voice.aes.hf.duplexBalance` debug.cfg	0 Max Echo Control (default): Balances the Acoustic Echo Suppression to maximize echo control. With this balance setting the phone hands free may be less full-duplex during continuous double-talk (near and far end users speak simultaneously). 1 - Max Full Duplex: Balances the Acoustic Echo Suppression to maximize full duplex. With this balance setting the phone hands free may be more susceptible to echo during continuous double-talk or when moving the phone or objects near the phone.
`voIpProt.SIP.header.allow.enable` sip-interop.cfg	0 (default) – No ALLOW header is added in 18x response for INVITE. 1 – ALLOW header is included in 18x response for INVITE.
`voIpProt.SIP.rport` sip-interop.cfg	0 (default) – The phone does not include `rport` parameter in the SIP messages through header. 1 – The phone includes `rport` parameter in the SIP requests sent on UDP transport.
`voice.handsetHeadset.rxdg.offset` techsupport.cfg	This parameter allows a digital Rx boost for the handset and headset. 0 (default) 9 to -12 – Offsets the RxDg range of the handset and headset by the specified number of decibels.
`voice.handsfreePtt.rxdg.offset` techsupport.cfg	This parameter allows a digital Rx boost for Push To Talk. 0 (default) 9 to -12 – Offsets the RxDg range of the hands-free and hands-free Push-to-Talk (PTT) by the specified number of decibels.
`voice.ringerPage.rxdg.offset` techsupport.cfg	This parameter allows a digital Rx boost for Push To Talk. Use this parameter for handsfree paging Rx in high noise environments. 0 (default) 9 to -12 – Raise or lower the volume of the ringer and hands-free page by the specified number of decibels.

Resolved Issues

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Audio	VOIP-122319	5.5.0	Choppy audio from VVX is observed while using speaker phone in specific deployments.
Audio	VOIP-122379	5.4.5	The VVX 501 and 601 phones does not support Acoustic Fence.
Audio	VOIP-123370 VOIP-123369	5.4.1	After a factory reset, the VVX business media phone restores the VQmon feature.
Audio	VOIP-122468	5.4.5	The VVX business media phone does not enable the acoustic fence feature after a configuration update and fails to reboot.
Audio	VOIP-124956 VOIP-124959	5.5.0	In certain environments, the audio quality on the VVX phones become low when using a speaker phone.
Audio	VOIP-125195	5.5.1	Audio is intermittent and one way, and the calls drop when Office 365 is used in certain environments.
Busy lamp Field	VOIP-119502	5.4.4 5.5.0	VVX D60 handset shows offline if registered to VVX 5XX/6XX phones using `attendant.resourceList.X.address`
Busy Lamp Field	VOIP-122541	5.4.5	When a BLF line key is pressed, the VVX 400 business media phone displays incorrect remote call identity for the corresponding BLF contact.
Busy Lamp Field	VOIP-123076 VOIP-123077	5.4.5	The VVX phone displays a "URL dialing is disabled" warning message and does not initiate an outgoing call after the ongoing call is put on hold and user presses the BLF key to dial a BLF contact.
Busy Lamp Field	VOIP-123955	5.1.1	The VVX phone fails to place a call using BLF feature when the URL dialing is disabled.
BroadSoft	VOIP-123214	4.0.11 5.4.5 5.5.1	When the server sends a 403 response even before a 200 OK response during the subscription attempt for hoteling on the BroadSoft server, the GuestIn button is not displayed.
BroadSoft	VOIP-125241	5.5.1	Inbound calls do not connect to the correct extension in BroadSoft R21 environment with a specific configuration.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
BroadSoft	VOIP-125744	5.4.5 5.5.1	Logging level of the DNS log is flooding customer log server in a specific BroadSoft deployment.
BToE	VOIP-115571	5.4.2	In a BToE Scenario, VVX 5XX/6XX phones do not honor the local Exchange Outlook Web Access (OWA) Language.
BToE	VOIP-118349	5.4.2	Presence of intermittent connectivity drops of BToE during a conference call.
BToE	VOIP-117341	5.4.4	Presence of BToE pairing issues after Wi-Fi to Ethernet switchover.
BToE	VOIP-120869	5.5.1	The device lock behavior based on Lock/unlock events from paired PC in BToE mode is not honored.
BToE	VOIP-121284	5.4.4 5.4.5	When the VVX phone is paired with BToE, the call gets dropped between 55-60 seconds when the SIP URI has only numerals.
BToE	VOIP-121312	5.4.4	The subject line goes missing when the VVX phone is paired with Polycom BToE.
BToE	VOIP-123839	5.4.4	The VVX phone does not pair with the BToE application installed on a system using the default connection type as serial in PuTTY. BToE does not pair when PuTTY default connection is set to "serial".
BToE	VOIP-124312	5.4.4	In a Skype for Business meeting, the VVX phone enabled with BToE loses audio when switching to PC audio mode.
BToE	VOIP-125517	5.5.1	BToE gets unpaired from the phone when large files of size 1 GB are uploaded.
Calling	VOIP-117745	5.4.0	In a race condition where two participants on VVX500 off-hooks a call for the Response Group, a misleading missed call notification is displayed on the user which could not respond to the call.
Calling	VOIP-118484	5.5.0	No ringback tone on outbound PSTN calls from the VVX D60.
Calling	VOIP-118660	5.4.3	With exchange online, the deployment phone does not recognize the keys pressed to play the voice mails.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Calling	VOIP-118794	5.4.2	On phone with an active call that excludes incoming, outgoing, hold, or held calls, when user is into **Call List**, **Favorites**, or **Directory,** and the user presses any digit, the screen will exit and pass the dialed digit as DTMF.
Calling	VOIP-120001	5.4.2 5.4.4	The VVX 410 phone randomly stops responding.
Calling	VOIP-120716	5.5.0	VVX phones do not generate the alert tone when a ‘click-to-dial’ call is initiated from the UC-one client.
Calling	VOIP-121222	5.4.4	Inbound calls to VVX 3XX/4XX phones from BSFT Contact Center does not produce audible ring and media fails to establish.
Calling	VOIP-122974	5.5.0	Input digits are missing when the digits are pressed quickly after the transfer is initiated.
Calling	VOIP-123106	4.0.11	During a call, if the `musicOnHold.uri` is configured, the SoundPoint IP (SPIP) phone does not send the Session description protocol (SDP) information in the INVITE to Music-on-hold (MOH) server address.
Calling	VOIP-123326 VOIP-123411	4.0.4 4.0.11 5.4.5	No crypto is offered from phone during late offer transfer when `sec.srtp.offer`="1".
Calling	VOIP-123345	5.4.4 5.5.1	Dialing from the phone fails with the message "SipCallMake failed" Could you please verify if this is technically accurate?
Calling	VOIP-123408 VOIP-123612	5.4.5 5.5.1	When phone reaches 24 calls the Real-Time Protocol (RTP) port cycle does not restart from the start-port.
Calling	VOIP-123640	5.4.5	Intermittent VVX peer-to-peer calls are losing soft keys except for the END call button in some customer environments.
Calling	VOIP-123663	5.5.1	A media bypass occurs during incoming calls on a VVX phone.
Calling	VOIP-124982	5.5.1	In some cases, the SIP 302 redirect messages are not handled correctly on a VVX phone resulting in a failed call.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Calling	VOIP-125230 VOIP-116402	5.5.1	In certain environments, when an extension is dialed from the VVX phone during an ongoing conference call, the participant is not added to the call.
Configuration	VOIP-116872	5.5.0	If the Forward feature is disabled on server while updating this fix build and after upgrade, if the administrator enables forward feature on server, then user needs to clear local configuration.
Configuration	VOIP-118721	5.4.5	When we configure two lines on a VVX phone, for instance on ports 1 and 2, the NAT Keepalive message is sent for only one of the lines at a time.
Configuration	VOIP-123490	5.5.0	VVX phones with UC Version 5.5.0 22063 reboots randomly in some customer environments.
Contacts	VOIP-122327	5.4.3 5.4.4 5.4.5 5.4.6 5.5.1	In a conference call, the VVX phones fail to add a participant having account name as numeric initial while cascading to an Audio video multi-party conferencing unit (AVMCU) call.
Contacts	VOIP-122752	5.4.1	Speed dials are not displayed for a short time when logging out and back in during the change in the destination directory file.
Contacts	VOIP-122769	5.4.4 5.4.5	The VVX phones does not display the presence status for some of the users added to the Buddy list.
Directory	VOIP-119545	5.4.3 5.5.0	Exporting contacts to the phone from MAC.Directory.xml file fails intermittently.
Directory	VOIP-124268	5.5.1	The VVX phone does not support saving the entries from Call History, Corporate directory, BroadSoft Enterprise Directory, and BroadSoft Group directory to the Personal Directory.
Functionality	VOIP-117321	5.4.4	VVX 5XX/6XX phones are unable to disable `feature.LyncCCCP.enable` using UC Software 5.4.1 or later.
Functionality	VOIP-119088	5.4.4	The Polycom VVX phone fails to fallback to DNS-A record when queries to resolve the SNTP hostname by DNS-SRV fails.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Functionality	VOIP-123464	5.4.6 5.5.1	VVX 5XX/6XX phones `up.IdleViewPreferenceRemoteCalls` functionality is not working.
General	VOIP-122048	5.4.2 5.4.4	The VVX phones do not process tagged CDP packets when DHCP option 144 is configured to discover VLAN and IP
General	VOIP-120715 VOIP-123258	5.5.0	Message waiting indicator (MWI) notification accepted by VVX but not by D60.
General	VOIP-123072 VOIP-123073	5.5.0	When any key is pressed, the VVX 410 phone experiences a memory leak and the phone response becomes slow.
General	VOIP-123486	5.4.5 5.5.1	Standard Output Web UI Field Help of VVX phones states the following incorrect message - "Polycom recommends that you do not disable this setting."
General	VOIP-123788	5.5.0	In some cases, the VVX D60 does not play a ringback tone.
General	VOIP-124436	5.5.1	The VVX phones does not respond in a failover by not sending SUBSCRIBE to the line-seize event.
General	VOIP-124804	5.5.1	In certain environments, the VVX 501 phone does not respond when using the screen capture utility with an Expansion Module attached.
Hardware	VOIP-122148	5.4.4	When the Polycom VVX Expansion Module is connected to VVX 410 phone, the phone reboots continuously.
Hardware	VOIP-124149	5.5.1	After the VVX 5XX/6XX phones restart, the USB headset needs to be unplugged and re-inserted for proper functioning.
Interoperability BroadSoft	VOIP-125097	5.5.0	The VVX phone sends an additional unsubscribe message to the BroadSoft server for BroadWorks hoteling event when pressing the GuestIn or GuestOut button.
Localization	VOIP-117062	5.4.1 5.4.2 5.4.4	VVX 500/600 phones miss the translation for Skype for Business Address Book Search.
Localization	VOIP-124231	5.5.1	VVX phone does not recognize the local Exchange OWA Language for the word 'Duration'.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Localization	VOIP-122578	5.4.6	The language translation for a new call on a VVX D60 phone displays incorrectly in German, French, Italian, and Spanish.
Localization	VOIP-122782	5.4.4	The Polycom VVX Business media phones display incorrect French translation for the VVX-dictionary.xml string 2967.
Logging	VOIP-122844	5.4.3	In some user environments, RealPresence Trio and VVX business phones randomly log out.
Logging	VOIP-125482	5.4.5 5.5.1	In the application log, the VVX phone prints benign DNS major errors thereby flooding the customer log server.
Network	VOIP-122059	5.4.5 5.4.6 5.5.1	The VVX phone is not able to connect to Exchange Autodiscover.
Network	VOIP-123925	5.4.5	When the network is unavailable, the VVX phones fail to load the application.
Network	VOIP-124602	5.5.0	The VVX phone does not send a re-INVITE after receiving a 407 for INVITE or SUBSCRIBE on IPv6 when the re-registration on failover is enabled.
Networking	VOIP-119307	5.2.5 5.4.4 5.5.0	The VVX phone fails to utilize the contact header containing `maddr` parameter in the 301 response received from the outbound INVITE.
Networking	VOIP-121135	4.0,8 5.1.3 5.4.1	Lowering Maximum Transmission Unit (MTU) on VVX-1500 causes high load on SBC due to excessive fragmented packets.
Provisioning	VOIP-120514	5.5.0	VVX phones with factory default settings upgrading to 5.5.0.x through Zero Touch Provisioning (ZTP) profile requires a manual reboot to redirect to local provisioning server.
Provisioning	VOIP-121526	5.5.1	When generating Certificate Signing Request (CSR) on VVX phones, the phone sends a private key to the provisioning server.
Provisioning	VOIP-122651	5.4.4	If provisioning server address is a URL to download the sip.ld file, the VVX business media phone adds a '/' in the server name before appending '/' to the file name (sip.ld).
Provisioning	VOIP-122794	None	The `log.render.file.upload.append` parameter when set to 0 does not upload any log files to the provisioning server.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Security	VOIP-121313 VOIP-122421	5.4.1 5.4.4 5.4.5	The Secure Real-Time Protocol (SRTP) Require causes call failure when call starts with no video and then re-INVITE with video.
Security	VOIP-122553	5.4.5	The VVX business media phones fail to sign in to a Skype for Business account due to the expired GlobalSign Root R3 CA certificate.
Security	VOIP-123989	5.5.1	An open port which could significantly impact the security issues for customers who deployed VVX D60 has been discovered by a security scanner.
Shared Calls	VOIP-122655	5.4.4	An intermittent loss of reorder tone for the shared lines result in no audio alert that the call has failed.
Shared Calls	VOIP-122789	4.1.6 5.4.5 5.5.0	The VVX phones does not behave as expected when the user goes on/off-hook without dialing on a shared line.
SIP	VOIP-112492	5.5.1	In some cases, the SIP 302 redirect messages are not handled correctly on a VVX phone resulting in a failed call.
Skype for Business	VOIP-119620	5.4.1	VVX phones fail to use Extension and Pin in production environment using `dhcp.option43.override.stsUri`
Skype for Business	VOIP-120096	5.4.4	No audio path is established occasionally between Skype consumers and Skype for Business customers.
Skype for Business	VOIP-120596	5.4.4 5.5.0	Observed random reboot of the VVX phone when user registers Skype for business on-premises with a longer digit dial plan.
Skype for Business	VOIP-121292	5.4.4	In a Skype for Business conference, the call is dropped when selecting headset mode while switching to PC audio mode from the VVX phone in a BToE environment.
Skype for Business	VOIP-121564	5.4.2 5.4.4	The VVX phone reboots when large number of participants join the Skype for Business conference call.
Skype for Business	VOIP-122323	5.4.5	The VVX phones go into a reboot loop after signing into Skype for Business.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Skype for Business	VOIP-123029	5.4.3	When Trio 8800 with UC version 5.4.3.AD receives an incoming call from the Microsoft Skype for Business client, the bandwidth utilization is not displayed in the call statistics on the Bandwidth policy service monitor.
Skype for Business	VOIP-123070	5.4.5 5.5.1	Skype for Business online favorites are not displayed on the VVX 410 phone’s Home screen.
Skype for Business	VOIP-123174	5.5.1	When logging into Microsoft Skype for Business client again after changing the password in Auto Discover, the BToE or Skype for Business client does not display the popup window to enter the password.
Skype for Business	VOIP-123288	5.4.5	Customer can not hear the remote audio when the VVX500 phone calls the Skype for Business phone
Skype for Business	VOIP-123450	5.5.1	IM from User B Skype for Business client to a User A causes VVX phone to ring when User A is on DND.
Skype for Business	VOIP-124778	5.4.4 5.5.1	Internal and external URLs are interchanged for Skype for Business on prem using Auto Discover.
Skype for Business	VOIP-125793	5.4.4	The VVX phone shows incoming calls as missed when they are picked on a German Skype for Business Client.
Software Update	VOIP-124176	5.5.1	In certain environments, the VVX phone encounters a PSTN call fail after upgrading to version 5.5.1.
User Interface	VOIP-119289	5.4.1 5.4.4 5.5.0	Speed dials disappear when provisional polling is used to check for updates.
User Interface	VOIP-119540	5.4.1	MAC-Directory File exceeding 4 MB is downloaded and displayed on the VVX phone.
User Interface	VOIP-119764 VOIP-123562	5.5.0	Search in an LDAP (Lightweight Directory Access Protocol) directory of the phone displays duplicate results.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
User Interface	VOIP-121481	4.1.8 5.1.3 5.2.5 5.3.3 5.4.3 5.4.4 5.4.5 5.5.0	The VVX phone's SSL Client Hello displays incorrect time for the GMT UNIX Time.
User Interface	VOIP-122532	5.4.6	Momentary UI freeze of the VVX 501 phone while pairing with the base station, when VVX is in the call.
User Interface	VOIP-122978	5.5.0	In some environments, when the **GuestOut** soft key is pressed for flexible seating, the “Service unavailable” message is displayed on the VVX phone.
User Interface	VOIP-123637	5.5.1	If the user sets the parameter `up.callWaitingMenu.enable`, the **Call Waiting** menu below **Preferences** Menu is displayed. If this parameter is not set, the **Call Waiting** menu won't be displayed on the phone.
User Interface	VOIP-124152	5.4.5 5.5.1	Pressing the “Read” button in the voice mail menu on a VVX phone directs the user to the paging screen.
User Interface	VOIP-124672	5.5.2	If a conference call is established from the handset with other two VVX phones, and both the far end numbers are registered with maximum number of Spanish chars "abcìdúeýf123abcìdúeýf123abcìdú", then the handset conference screen displays the first caller number.
User Interface	VOIP-125154	5.5.1	Phone displays "Contact Synchronizing” continuously after registration with the microsoft.com tenant.
User Interface	VOIP-125438	5.5.1	The contact names from Favorites are not readable from Home screen or dialer with transparent line keys enabled.
User Interface	VOIP-125651	5.5.1	A pop-up to indicate expired credentials are not displayed on the phone’s interface.
Video	VOIP-118436	5.4.4	Observed a one-way video when using SRTP in few GENBAND environments.

Resolved Issues in UC Software 5.5.2

Category	Issue No.	Release	Description
Video	VOIP-122983	5.4.4 5.5.0	When joining a conference on a VVX phone using the **Join** soft key, the call gets connected to the conference server but the video is not activated.
Video	VOIP-124270	5.4.4	In a centralized conference call, when the conference initiator sends a REFER to other participant in the call, the call becomes Audio-only for the initiator.
Video	VOIP-125629	5.4.4	On VVX phones, during video calls, the video is stretched and requires the user to go full screen and back before proper video scaling is used.
VVX D60	VOIP-121844	5.4.4 5.5.0	VVX D60 when paired with VVX 5XX/6XX phone shows incorrect registration instructions on phone's Web UI.
VVX D60	VOIP-121980	5.4.5 5.5.0 5.5.1	VVX D60 shows “euer Anruf” instead of “neuer Anruf” in German Language.
VVX D60	VOIP-123008 VOIP-123879	5.5.1	When VVX D60 is paired with a VVX phone and if the SIP INFO method and RFC 2833 are enabled, the VVX phone only sends the SIP Info for DTMF.
Web Interface	VOIP-118865	4.0.10 5.2.0 5.4.1 5.4.2 5.4.3	The User Interface language falls back to the previous language after a reboot if the language selection is made through the Web UI and not the Phone UI.
Web Interface	VOIP-119498	5.4.1	Setting the `device.sntp.gmtOffset parameter` using the web configuration utility's import configuration feature does not work.
Web Interface	VOIP-124945	5.5.1	The CA certificate is not displayed on the web configuration or phone’s user interface of the VVX phone when trying to import the device settings to other phone.

What’s New in Polycom UC Software 5.5.1 Rev E

This release introduces the following new features.

Locking the Web Configuration Utility after Failed Login Attempts

You can lock access to the Web Configuration Utility after a series of failed login attempts and configure a period of time a user can attempt to log in again. Use the following parameters to configure additional security after multiple failed login attempts:

- `httpd.cfg.lockWebUI.enable`
- `httpd.cfg.lockWebUI.lockOutDuration`
- `httpd.cfg.lockWebUI.noOfInvalidAttempts`
- `httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration`

Reboot or restart of the phone will result in the reset of the following:

1. Last successful & unsuccessful Web-UI login attempt details
2. Web-UI lock state and remaining logon attempts.

Any configuration updates of the Web-UI lock parameters will result in the reset of the Web-UI lock state and remaining logon attempts.

Resolved Issues

This section lists the issues that were resolved in UC Software 5.5.1 Rev E.

Resolved Issues in UC Software 5.5.1 Rev E

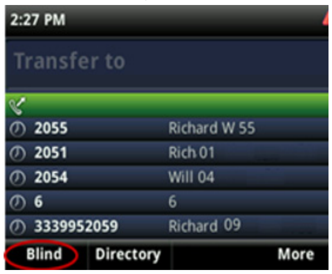
Category	Issue No.	Release	Description
Calling	VOIP-123320	5.4.5	Phone does not ring for the Busy Lamp Field (BLF) monitored line when incoming call to phone is terminated.
Configuration	VOIP-123053	5.4.2	Exchange URL was not configured and shows an error message.
Configuration	VOIP-123317 VOIP-123318 VOIP-123319	5.4.5	Phone rejects signaling from an unregistered source even with the parameter `onlySignalWithRegistered` enabled.
User Interface	VOIP-123415	5.4.5	If feature Hoteling is enabled, phone does not display the appropriate soft keys when it receives any error code from the server.
User Interface	VOIP-123416 VOIP-122577	5.5.0	When the user presses the GuestOut soft key the first time after the hoteling event SUBSCRIBE expires, the user is not getting signed out and phone displays “Service Unavailable”.
User Interface	VOIP-123511	5.4.5	The Expansion Module (EM) do not display some speed dial and Busy Lamp Field (BLF) contacts after configuration update through the phone menu when the configuration consists of more than 20 registrations and 300 speed dial contacts.

What's New in Polycom UC Software 5.5.1

This release introduces the following new features.

New Call Transfer User Interface Option

In this software version, users who transfer calls can more easily choose between **Blind** and **Consultative** transfers. On the Call Transfer screen for the user's default transfer type, the user can press **More** to access a new soft key to change to the alternate transfer type. For example, if the user's default transfer type is consultative, a **Blind** soft key is displayed.



The existing Call Transfer behavior continues to be supported. Users can press **Transfer** to initiate the default type or press and hold **Transfer** to initiate the alternate transfer type. Administrators can configure the phone to hide the **Blind**/**Consultative** soft key using new parameter `up.softkey.transferTypeOption.enabled`.

End-user Access to Ethernet and DHCP Settings

New parameter `up.basicSettings.networkConfigEnabled` lets you configure phones to allow end users to access to the Ethernet and DHCP settings through the **Basic** menu.

Distribution List

Polycom phones registered with a Microsoft server enable you to perform multiple functions with a contact distribution list:

- Search for, add, and delete a distribution list
- View a distribution list, and expand a distribution list to view all members
- View the contact card of a distribution list and of an individual member
- Conference with a distribution list
- Call an individual member of a distribution list

Distribution lists are available on the following VVX business media phones: VVX 201, VVX 300/310, VVX 301/311, VVX 400/410, VVX 401/411, VVX 500/501, VVX 600/601, and Polycom VVX Expansion Modules.

Microsoft Quality of Experience (QoE) Monitoring Server Protocol

Microsoft Quality of Experience Monitoring Server Protocol (MS-QoE) enables you to monitor the user's audio quality and troubleshoot audio problems. QoE reports contain only audio metrics and do not contain video or content sharing metrics. This feature also enables you to query the QoE status of a phone from the Web Configuration Utility.

Device Lock

You can configure phones to be protected with a lock code that enables users to access personal settings from different phones. You can configure Device Lock on the Skype for Business server or use Polycom parameters on a centralized provisioning server. If you enable Device Lock using both methods, centralized provisioning parameters take precedence. You cannot enable or disable Device Lock using the Web Configuration Utility or from the phone menu.

Polycom BToE PC Pairing

Administrators can use this feature to allow users to automatically or manually pair their VVX business media phone with their computer using the Polycom Better Together over Ethernet Connector application. Users can select the pairing mode in the Web Configuration Utility or in the Features menu on the phone. By default, BToE PC Pairing is enabled for phones registered with Skype for Business. When administrators disable BToE pairing, users cannot pair their VVX phone with their computer using BToE. In order to use this new functionality, you must install both BToE Connector App 3.4.0 and UC Software 5.5.1. For best results, Polycom recommends that you deploy BToE Connector App 3.4.0 before you deploy UC Software 5.5.1.

User Log Upload

To help troubleshoot user issues, administrators can enable or disable for users the ability to upload diagnostic logs from the phone or Web Configuration Utility and set log levels from the phone menu. This feature is available on all VVX business media phones registered with Skype for Business Server on-premises or online and with Microsoft Lync 2013 or 2010 Server.

Phone User Interface

The user interface for VVX 500 and 600 series business media phones was updated to match the theme used in the Skype for Business client. This feature is enabled by default on VVX 500/501 and 600/601 phones with the Lync/Skype Base Profile or SKU.

Unified Contact Store

Administrators can migrate users' contacts to Microsoft Exchange Server to enable synchronization when users manage contacts or contact information from an application or device, for example, the VVX business media phone, Skype for Business client, Outlook, or Outlook Web Application from a mobile device.

Web Sign-In for Online Deployments

Web Sign-in enables users to securely log in to Skype for Business from the phone using a computer web browser or mobile device. Users can sign in concurrently to a maximum of eight devices by default. When users are signed in to multiple devices and sign out from one device, users remain signed in to all other devices. This sign in option is available only for Skype for Business Online deployments.

Expanded Support for USB Headsets

Support for the following Plantronics USB Headsets with VVX 500, VVX 600, VVX 501, VVX 601, and VVX 401 phones has been added to this release:

- Blackwire C310
- Blackwire C325
- Blackwire C725
- Blackwire C325.1
- Plantronics CS520
- EncorePro HW540
- DA80 Headset Adapter

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.5.2.

Configuration File Enhancements

Parameter Template	Permitted Values
`attendant.callWaiting.enable` features.cfg	Specifies whether to use an acoustic indication for incoming attendant calls when there is already an active call on the phone. The ring type is set with parameter `attendant.callWaiting.ring.` 0 (default) – No acoustic indication is used for call waiting. 1 – An acoustic indication sounds for incoming attendant calls if there is already an active call on the phone.
`attendant.callWaiting.ring` features.cfg	Specifies the ring type used to notify the attendant when there is already an active call on the phone. This parameter is valid only if `at-tendant.callWaiting.enable` is set to 1 Silent (default) – No acoustic indication is provided. Beep – A beep tone plays when the phone is in an active call when it receives an attendant call. Ring – A ring tone specified by the parameter `at-tendant.ringType` plays when the phone is in an active call when it receives an attendant call.
`btoe.PairingMode` features.cfg	Specifies how the phone pairs with a connected computer running Better Together over Ethernet (BToE). Auto (default) - The phone pairs with the computer automatically when a computer connected to the phone's PC port is running the BToE connector application. Manual - The phone generates a six-digit pair code that must be entered in the BToE connector application running on a computer connected to the phone's PC port.

Configuration File Enhancements

Parameter Template	Permitted Values
`call.autoAnswer.playTone.enable` reg-advanced.cfg	Specifies whether the phone plays a tone when auto-answering a call. 1 (default) – Autoanswer tone is played. 0 – No auto-answer tone is played.
`call.autoAnswerMenu.enable` features.cfg	Specifies whether the Autoanswer menu displays on the phone to allow users to access the Autoanswer option. 1 (default) - Users can enable or disable the feature on the phone from the Autoanswer menu in Basic Settings. 0 – The Autoanswer menu is unavailable on the phone, and only the administrator can control the feature using configuration files.
`call.playLocalRingBackBeforeEarlyMediaArrival` sip-interop.cfg	Determines whether the phone plays a local ring-back after receiving a first provisional response from the far end. 1 (default) - The phone plays a local ringback after receiving the first provisional response from the far end. If early media is received later, the phone stops the local ringback and plays the early media. 0 - No local ringback plays, and the phone plays only the early media received.
`call.switchToLocalRingbackWithoutRTP` sip-interop.cfg	Determines whether local ringback plays in the event that early media stops. 0 (default) – No ringback plays when early media stops. 1 – The local ringback plays if no early media is received.
`device.ipv6.icmp.ignoreRedirect` device.cfg, wireless.cfg	Specifies whether Internet Control Message Protocol (ICMP) redirect messages are accepted. 0 (default) - ICMP redirect messages are accepted. 1 – ICMP redirect messages are ignored to avoid route changes.
`device.ipv6.icmp.txRateLimiting` device.cfg, wireless.cfg	Sets the maximum rate for sending ICMPv6 packets. NULL (default) 0 – 60000 ms

Configuration File Enhancements

<table>
<tr><th>Parameter Template</th><th>Permitted Values</th></tr>
<tr><td><code>device.net.etherStormFilterPpsValue</code>
device.cfg, site.cfg</td><td>Specifies the Packets per Second (PPS) value that triggers DOS storm prevention. The PPS index maps to a specific number of packets per second as shown here.
<table>
<tr><th>PPS Index</th><th>Packets per Sec.</th><th>PPS Index</th><th>Packets per Sec.</th></tr>
<tr><td>17</td><td>5887</td><td>29</td><td>19201</td></tr>
<tr><td>18</td><td>6400</td><td>30</td><td>21240</td></tr>
<tr><td>19</td><td>6911</td><td>31</td><td>23299</td></tr>
<tr><td>20</td><td>7936</td><td>32</td><td>25354</td></tr>
<tr><td>21</td><td>8960</td><td>33</td><td>27382</td></tr>
<tr><td>22</td><td>9984</td><td>34</td><td>29446</td></tr>
<tr><td>23</td><td>11008</td><td>35</td><td>31486</td></tr>
<tr><td>24</td><td>12030</td><td>36</td><td>35561</td></tr>
<tr><td>25</td><td>13054</td><td>37</td><td>39682</td></tr>
<tr><td>26</td><td>14076</td><td>38</td><td>42589</td></tr>
<tr><td>27</td><td>15105</td><td>39</td><td>56818</td></tr>
<tr><td>28</td><td>17146</td><td>40</td><td>71023</td></tr>
</table>
38 (default) – A PPS index of 38 triggers the storm filter.
17 - 40 – A PPS index between 17 and 40 triggers the storm filter.</td></tr>
<tr><td><code>device.net.etherStormFilterPpsValue.set</code>
device.cfg, site.cfg</td><td>This parameter controls whether the parameter <code>device.net.etherStormFilterPpsValue</code> is used for setting a Packets per Second (PPS) index to trigger DoS storm prevention.
0 (default) – The parameter <code>device.net.etherStormFilterPpsValue</code> is not used and storm filtering is not enabled by a specific PPS index.
1 – The parameter <code>device.net.etherStormFilterPpsValue</code> is used and storm filtering is enabled by a specific PPS index.</td></tr>
<tr><td><code>dir.corp.cacheSize</code></td><td>The maximum number of entries that can be cached locally on the phone.
8 to 256
128 (default)
Note: For VVX 101/201 phones, the permitted values are 32 to 64, and the default is 64.</td></tr>
</table>

Configuration File Enhancements

Parameter Template	Permitted Values
`dir.corp.pageSize`	The maximum number of entries requested from the corporate directory server with each query. 8 to 64 64 (default) Note: For VVX 101/201 phones, the permitted values are 8 to 32, and the default is 16.
`exchange.meeting.parseWhen` applications.cfg	Specifies whether the phone uses number searching logic when finding additional numbers in the Skype for Business calendar. NonSkypeMeeting (default) - Number-searching logic is not used to find additional numbers in Skype for Business meeting calendar entries. Always - Number-searching logic is used to find additional numbers in Skype for Business meeting calendar entries.
`exchange.reconnectOnError` application.cfg	Determines whether the phone automatically attempts to reconnect to the Exchange server after encountering an error. 1 (default) – Phone attempts to reconnect to the Exchange server after an error. 0 – Phone does not attempt to reconnect to the Exchange server after an error.
`feature.callCenterCallInformation.enable` features.cfg	Specifies whether the phone displays call center and incoming call information in a pop-up message. 1 (default) – The phone displays call center and incoming call information in a pop-up message. 0 - The phone does not display call center and incoming call information in a pop-up message.
`feature.deviceLock.enable` features.cfg	1 (Default) - Device Lock for Skype for Business is enabled. 0 - Device Lock for Skype for Business is disabled.
`feature.lync.hideSignInSignOut` features.cfg	Specifies whether the Sign In and Sign Out soft keys appear on the Home screen and phone menus. 0 (default) – The Sign In and Sign Out soft keys appear in the user interface. 1 – The Sign In and Sign Out soft keys do not appear in the user interface, and users are not able to sign in or out. Administrators can sign in and out with the Web Configuration Utility.
`feature.lync.hideSignOut` features.cfg	Specifies whether the Sign Out soft key appears on the Home screen and phone menus. 0 (default) – The Sign Out soft key appears in the user interface. 1 – The Sign Out soft key does not appear in the user interface, and users are not able to sign out. Administrators can sign in and out with the Web Configuration Utility.

Configuration File Enhancements

Parameter Template	Permitted Values
`feature.VVXD60.allowLineMappings` features.cfg, dect.cfg	Allows users to map lines on the VVX phone to a paired VVX D60 wireless handset from the Features menu. 0 (default) – Map Lines is available only on the Administrator menu. 1 – Map Lines is available on the Administrator menu and the Features menu.
`ipv6.mldVersion` device.cfg, site.cfg	Determines which version of Multicast Listener Discovery to use. 2 (default) –Multicast Listener Discovery version 2 is used. 1 –Multicast Listener Discovery version 1 is used.
`log.level.change.pec`	Set the debug log level for the Polycom Experience Cloud. 4 (default) 0-6
`net.interface.mtu6` site.cfg	Sets the Maximum Transmission Unit (MTU) value in bytes for the phone in IPv6 or dual stack mode. Note: IPv6 is qualified for an Open SIP eco-system. IPV6 is not qualified for a Skype for Business eco-system. 1500 bytes (default) 1280 – 1500 bytes
`pec.enabled`	Enables or disables Polycom Experience Cloud. 0 (default) – Disables the Polycom Experience Cloud. 1 – Enables Polycom Experience Cloud.
`pec.log.render.level`	Set the log level for Polycom Experience Cloud logs, which are uploaded to the cloud server set in the parameter `pec.server.uri.` 4 (default) 0-6
`pec.log.uploadPeriod`	Sets the period of time in minutes between Polycom Experience Cloud log uploads. 15 (default) 0 – 10080
`pec.server.uri`	Set the URI where logs for Polycom Experience Cloud are uploaded. String with 0 – 256 characters NULL (default) Note: If set to Null, https://pec.polycom.com is used by default.

Configuration File Enhancements

Parameter Template	Permitted Values
`phoneLock.authorized.x.value` features.cfg	Specifies an authorized number that can be dialed when the device is locked using a Tel URI or SIP URI. Any numbers configured for this parameter display in an Authorized Calls list. For example, `phoneLock.authorized.1.value="cwi57@cohovineyard.com".`
`qos.ethernet.tcpQosEnabled` site.cfg	Specifies whether the phone sends configured Quality of Service (QoS) priorities for SIP on TCP. 0 (default) – Phone does not send configured QoS priorities for SIP on TCP. 1 – Phone sends configured QoS priorities for SIP on TCP.
`reg.x.auth.loginCredentialType` reg-advanced.cfg	Specifies the login type and user credentials required for the phone. LoginCredentialNone (default) – Microsoft login credentials are not accepted, and users are unable to log in with Microsoft credentials. usernameAndPassword – User must enter sign-in address, user name, domain, and password in the required format in order to sign in. extensionAndPIN - User must enter extension and PIN in order to sign in.
`reg.x.auth.useLoginCredentials` reg-advanced.cfg	Configures the phone to sign in users after the phone powers up. 1 (default) - SSI Login credentials, BToE Sign in, and Web Sign types are available for authentication with the server. 0 - SSI Login credentials, BToE Sign in, and Web Sign types are not available for authentication with the server.
`reg.x.auth.usePinCredentials` reg-advanced.cfg	Configures the phone to sign users in automatically after the phone powers up. To use this sign-in method, you must enable DHCP Option 43 or enable parameter `dhcp.option43.override.stsUri`. 1 (default) - PIN authentication sign in method is available for authentication on the server. 0 (default) - PIN authentication sign in method is not available for authentication on the server.
`reg.x.lineaddress` features.cfg	Set for private lines as well for the BroadSoft call park and retrieve scenarios. If the parameter `reg.x.address` is set to an address different than the call address of the number, use parameter `reg.x.lineaddress` to enable users to park and retrieve calls. Note: If there is no value specified for `reg.x.lineAddress,reg.x.address` is used.

Configuration File Enhancements

Parameter Template	Permitted Values
`softkey.feature.simplifiedSignIn` lync.cfg	Specifies whether the Sign In and Sign Out soft keys display on the Home screen or on phone menu at **Settings** > **Features** > **Microsoft Lync** > **Sign Out.** 0 (default) - The Sign In and Sign Out soft keys display on the phone menu, but not on the Home screen. 1 - The Sign In and Sign Out soft keys display on both the Home screen and the phone menu.
`up.basicSettings.networkConfigEnabled` site.cfg	Allows you to give users access to the Ethernet and DHCP settings using the Basic menu. 0 (default) – Phone does not include Ethernet and DHCP settings on the Basic menu. 1 – Phone includes Ethernet and DHCP setting on the Basic menu.
`up.BLFDefaultLineView` features.cfg	Specifies which view is displayed when the phone receives a BLF call. 0 (default) – The phone continues to display the call view when incoming BLF call information displays. 1 – The phone displays the line view when incoming BLF call information displays.
`up.btoeDeviceLock.timeOut` features.cfg	Sets the number of seconds after which the phone locks after a period of inactivity. 10 seconds (default). 0 – 40 seconds.
`up.hideDateTimeWhenNotSet` features.cfg	Determines whether the date and time flash on the idle screen when the date and time have not been set. 0 (default) – The date and time display does not flash when the date and time have not been set. 1 – The date and time flash on the phone idle screen when the date and time have not been set.
`up.oneTouchBossAdmin` features.cfg	Enables the Boss and Admin for a line to view and pick up held calls on the boss's line by pressing the line key. 0 (default) – The user has to press and hold the line key to view and pick up held calls on the line. 1 – The user can press the line to view and pick up held calls on the line.
`up.onHookDialingEnabled` features.cfg	Specifies whether to enable on-hook dialing, which allows users to enter a number before dialing. 1 (default) – Enables on-hook dialing. 0 – Disables on-hook dialing.

Configuration File Enhancements

Parameter Template	Permitted Values
`up.softkey.transferTypeOption.enabled` site.cfg	Specifies whether a soft key is added to the Call Transfer screen to allow the user to toggle between Blind and Consultative transfers. If the soft key is not added, the user can press and hold the Transfer soft key to choose a transfer type. 1 (default) - Adds a transfer type soft key to the Call Transfer screen. 0 – No transfer type soft key is added to the Call Transfer screen.
`voIpProt.OBP.dhcpv4.option` site.cfg	Specifies the DHCPv4 option code for Outbound proxy. 120 (default) 120 - 254
`voIpProt.OBP.dhcpv4.type` site.cfg	Specifies the DHCPv4 type for Outbound proxy. 0 (default) – Request an IP address from the DHCP server as the outbound proxy type. 1 – Request a string from the DHCP server as the outbound proxy type.
`voIpProt.OBP.dhcpv6.option` site.cfg	Specifies the DHCPv6 Option code used to get the Outbound Proxy server address from the DHCP server. 21 (default) 0-254 Note: IPv6 is supported by Polycom Open SIP servers. IPv6 is not supported when using Polycom phones with Skype for Business.
`voIpProt.SIP.callinfo.precedence.overAlertinfo` sip-interop.cfg	Specifies whether the call-info header with answer-after string has precedence over alert info. 0 (default) – The call-info header with answer-after string does not have precedence over alert-info. 1 – The call-info header with answer-after string has precedence over alert-info.
`voIpProt.SIP.considerTlsDnsEntriesOnly` site.cfg	Specifies whether TLS entries are considered in the auto-discovery process. 0 (default) – TCP and TLS entries are not considered in the auto-discovery process. 1 – Only TLS entries are considered in the auto-discovery process.
`voIpProt.SIP.renewSubscribeOnTLSRefresh` sip-interop.cfg	Specifies whether to refresh BroadSoft as-feature-event subscriptions when the phone re-registers. This parameter only applies when TLS transports are in use. 0 (default) – When a registration is refreshed, the BroadSoft as-feature-event subscription is also refreshed. 1 – Does not refresh the BroadSoft as-feature-event subscription when a registration is refreshed.

[1] Change causes phone to restart or reboot.

Supported DHCP Sub-Options

The following table lists the individual sub-options and combination sub-options supported on VVX phones for DHCP Option 43:

DHCP Option 43 Configuration Options

Option	Result
Option 1- Subnet mask	The phone parses the value from Option 43.
Option 2 - Time offset	The phone parses the value.
Option 3 – Router	The phone parses the value.
Option 4 - Time server	The phone parses the value.
Option 6 - Domain Name Server	The phone parses the value.
Option 7 - Domain Log server	The phone parses the value.
Option 15 - Domain Name	The phone parses the value.
Option 42 - Network Time Protocol server	The phone parses the value.
Option 66 - TFTP Server Name	The phone parses the value.
Options 128-255	
Sub-options configured in Option 43	
Options 1, 2, 3, 4, 5, 6, 7, 15, 42, and 66	The phone parses the value.
Options 128-255	

Resolved Issues

This section lists the issues that were resolved in UC Software 5.5.1.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
Audio	VOIP-118272	5.4.2	Fixed an audio garbled issue while placing a call after incoming intercom call.
Audio	VOIP-119376	5.5.1	Intermittent audio loss and choppy audio no longer occurs during a VMR call using the Skype for Business client and a VVX phone connected using BToE.
Boss-Admin	VOIP-117416	5.5.1	The boss and delegates must be in the same Skype for Business environment, either both Online and both on-premises, to use the feature.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
Boss-Admin	VOIP-120528	5.5.1	The delegate can now successfully transfer a call using Safe Transfer during BToE Playback.
Boss-Admin	VOIP-120679	5.5.1	In a Boss-Admin scenario, the delegate’s phone now rings when the dual role is removed and a FLK enabled on it.
Boss-Admin	VOIP-120753	5.5.0	Phones in a Boss-Admin group can receive up to five incoming calls at the same time.
BroadSoft	VOIP-118504	5.5.0	The phone no longer sends an incorrect authentication header when using SIP credentials to authenticate to a BroadSoft XSP address.
BToE	VOIP-111292		The BToE Icons are now removed when you change the value of NOTIFY_ICON_EN to 0. You can find this setting at HKEY_LOCAL_MACHINE > SOFTWARE > Wow6432Node > Polycom > Polycom BToE Connector.
BToE	VOIP-114438	5.4.4	The SSH Host Key is no longer hardcoded on the phone when BToE is in use.
BToE	VOIP-115725	5.4.0	In a race condition where two participants (VVX500/BToE) off-hook a call for the Response Group, the call now connects to the first user and the other user goes to the ideal state smoothly.
BToE	VOIP-119007	5.4.4	A problem was resolved that caused the Polycom BToE Connector to log too much data at BTOE_DBG_DBG.
Busy Lamp Field	VOIP-109428		The phone correctly displays Busy Lamp Field (BLF) lines when more than 20 BLF lines are configured and the storm filter is enabled.
Busy Lamp Field	VOIP-117808	5.4.4	Busy Lamp Field (BLF) no longer fails after the first startup.
Calling	VOIP-117222	5.4.2	An issue was resolved that caused the loss of speed dial on the VVX 101 phone when `lineKey.reassignment.enabled` was enabled.
Calling	VOIP-117314		Ringer and paging, hands-free and push-to-talk (PTT), and handset and headset loudness in phones can now be increased to full scale when a low power signal is received. For more details on the configuration parameters added for this issue, see Configuration File Enhancements.
Calling	VOIP-117664	5.4.4	E911 calls no longer fail when the phone is set to a static IP address.
Calling	VOIP-117691	5.4.4	A PSTN caller can now hear the ringback tone for a Skype for Business call through a Sonus gateway to a VVX phone.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
Calling	VOIP-117747	5.4.2	An issue was resolved that caused the phone to dial the full URI if the URI contains the extension format x1234 instead of ext=1234.
Calling	VOIP-117924	5.4.3	An issue was resolved that caused the phone to lose all contacts in some specific environments.
Calling	VOIP-117951	5.4.2	Parameter `voIpProt.SIP.callinfo.precedence.overAlertinfo` was added to control whether a call-info header with answer-after string has priority over alert-info.
Calling	VOIP-118441	5.4.4	When an intercom call is manually answered during an active call, the phone now mutes the intercom call when `ringAnswerMute` is enabled.
Calling	VOIP-118609		New parameters `attendant.callWaiting.enable` and `attendant.callWaiting.ring` have been added to allow you to configure an acoustic call-waiting indication for attendant calls.
Calling	VOIP-118793		During an active call, if you navigate to the Call list, Favorites, or Directory and then press any digit, the screen will no longer exit and passes the dialed digit as a DTMF tone.
Calling	VOIP-118879 VOIP-118580	4.0.9 5.4.1	The phone now correctly shows the configured value on the server when the Call Forward Not Answered (CFNA) ring count is configured on the server.
Calling	VOIP-119043	5.4.2	When an admin terminates a call that was placed on hold by the boss, the call no longer randomly reappears as a held call on the boss' phone.
Calling	VOIP-119172 VOIP-119173	5.4.2	A problem was resolved that prevented dialing from the placed call list when the OPUS codec is enabled.
Calling	VOIP-119834	5.4.4	The phones no longer fail when forwarding a call to a number added by a user when the number has an appended domain.
Certificates	VOIP-118468	5.4.4	The VVX device certificate can now be sent via XSI when requested.
Configuration	VOIP-119497	5.4.1	Setting the `device.sntp.gmtOffset` parameter using the Web Configuration Utility's Import Configuration feature now works.
Contacts	VOIP-119544	5.4.3	The phone no longer loses contacts in some specific environments.
Directory	VOIP-116937	5.5.0	Contacts added after the maximum contact limit was reached are now displayed automatically after some contacts are deleted from the displayed 200 contacts.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
Directory	VOIP-118286	5.4.4	The phone no longer reboots during a call or when browsing the directory.
Directory	VOIP-119288	5.4.4 5.5.0	The following directory configuration changes apply to all VVX phones except VVX 1500: • Both the <MAC>-directory.xml and 000000000000-directory.xml files are downloaded at the time of a configuration update. • Download of directory files in case of checkSync will depend on parameter "`voIpProt.SIP.specialEvent.checkSync.downloadDirectory`". • Download of directory files will no longer depend on dir.local.readonly. • For the User Login feature, directory file "username-directory.xml" will be considered in place of <MAC>-directory.xml.
General	VOIP-102685 VOIP-120278	5.4.4	VVX phones no longer take up to 3 minutes to restart an application after receiving NOTIFY check-sync.
General	VOIP-116674	5.5.0	The phone no longer uploads a core dump during a restart.
General	VOIP-116715		New parameter `call.autoAnswer.playTone.enable` has been added to allow you to specify whether to plan an auto-answer tone.
General	VOIP-116924		The VVX phones now complete blind transfers with the Competella switch board.
General	VOIP-117262	5.4.4	An issue was resolved that prevented presence from failing over to the new server.
General	VOIP-117293 VOIP-117294 VOIP-115022	5.4.3	The parameter `reg.x.lineaddress` can now be used to specify the line extension used for parking either private or shared lines. If the registration number specified by `reg.x.address` is different from the actual line address, configuring this parameter is required for using call park and retrieve.
General	VOIP-117505	5.5.0	The parameter `device.Services.VoiceService.x.VoiceProfile.x.SIP.RegisterRetryInterval` was removed from the TR-069 map.
General	VOIP-117527	5.5.1	Polycom phones do not support special characters ({ or }) in usernames.
General	VOIP-117678	5.5.1	The MWI tone is not played on Jabra GN9120 EHS headsets in on-hook mode as the headset discards this tone.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
General	VOIP-117690	5.4.4	The phone now rejects the SIP INVITE and SIP NOTIFY on reception of SIP 400 Bad request.
General	VOIP-117777	5.4.5	The following email address pattern is supported: {noformat} "[a-zA-Z0-9\\+\\._\\%\\-]{1,256}" "\\@" "[a-zA-Z0-9][a-zA-Z0-9\\-]{0,64}" "(" "\\." "[a-zA-Z0-9][a-zA-Z0-9\\-]{0,25}" ")+" {noformat}
General	VOIP-117819	5.4.2	The phone no longer locks up or reboots while placing a PSTN call at time registration or re-registration.
General	VOIP-117951		New parameter `voIpProt.SIP.callinfo.precedence.overAlertinfo` was added to control whether a call-info header with answer-after string has priority over alert-info.
General	VOIP-117989	4.0.11	Removed CA bundle(ca10.crt) for Web Server profile.
General	VOIP-118056	5.5.1	The phone no longer signs out and signs in automatically when it is left idle for more than 3 days.
General	VOIP-118108	5.4.4	An issue was resolved that caused the phone to reboot and created core dumps on incoming calls in some specific environments.
General	VOIP-118399	5.5.0	A problem was resolved that caused the phone to reboot when reading a large configuration file.
General	VOIP-118623 VOIP-118624	5.4.4	Audio loss no longer occurs on a VVX phone for around 8 seconds for call center calls through Anywhere365.
General	VOIP-118770 VOIP-118771	5.4.3	The phone now sends media attributes in SDP immediately when the Hold button is pressed.
General	VOIP-118791 VOIP-118795	5.4.2	VVX 101 and VVX 201 phones no longer remove Transfer and Hold when the parameter `softkey.feature.basicCallManagement.redundant` is set to 0.
General	VOIP-118873 VOIP-120659	5.4.3	A spelling error was corrected in parameter `feature.VVXD60.allowLineMappings`. The parameter name is no longer spelled with three p's.
General	VOIP-118902	5.4.1	The VVX phone no longer fails to use Extension & PIN in a production environment when using `dhcp.option43.override.stsUri`.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
General	VOIP-119045	5.5.1	The phone no longer goes into a reboot loop when failover to DVD from LLDP/CDP.
General	VOIP-119250		Parameter `voIpProt.SIP.renewSubscribeOnTLSRefresh` was added for as-feature events. The parameter refreshes SUBSCRIBE together with re-REGISTER when TLS breaks.
General	VOIP-119894	5.4.2	An issue was resolved that prevented the phone from passing LLDP packets to a laptop connected to the phone's PC port.
General	VOIP-120146	5.4.4	Modified the parameter values for parameters `dir.corp.cacheSize` and `dir.corp.pageSize` to accommodate for VVX 101/201 phones.
General	VOIP-120234	5.5.0	VVX phones with factory default settings no longer require a manual reboot to redirect the phone to the local provisioning server when upgrading to 5.5.0.x using Zero Touch Provisioning.
General	VOIP-120277	5.4.4	The phone no longer takes 3 minutes to restart after receiving NOTIFY check-sync.
General	VOIP-120301	5.4.2	The phones no longer resync or download software during server maintenance.
General	VOIP-120512 VOIP-120511	5.5.0	VVX phones no longer upload core dumps when subscribing to attendant.uri in an Asterisk 13 environment.
General	VOIP-120609 VOIP-120616	5.4.2	Polycom devices no longer unintentionally resync and download new software during customer maintenance of their Edge system when resync is not applied to the phone.
General	VOIP-120673	5.5.0	VVX phones with default factory settings no longer require a manual reboot when upgrading to UC Software 5.5.0 or later.
Hardware	VOIP-117879 VOIP-118639	5.4.2	Ringer and paging loudness have been improved for better usability.
Language	VOIP-121138 VOIP-121165	5.4.4 5.5.0	"Wachtstand" for holding calls is now displayed properly in the call appearance window when the language is set to `lcl.ml.lang="Dutch_Netherlands"`
Languages	VOIP-121160 VOIP-121166	4.0.11 5.4.4 5.5.0 5.5.1	Sort in Spanish is now "Ordenar" instead of "Arreglar"
Logging	VOIP-119134	5.4.4	An issue was resolved that caused D60 wireless handset to print spurious benign EVENT 4 messages in the Phone Log.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
Microbrowser	VOIP-117597	5.3.0	The microbrowser now restarts automatically when RAM usage is over 40% for more than 30 minutes. When this happens, the last web page displayed is not restored.
Microsoft	VOIP-118250	5.4.4	An issue was resolved that caused the phone to reboot during large Skype for Business conference calls
Microsoft	VOIP-118294	5.5.1	The phone is creating a roster view with multiple self participants for every Meet Now. This is a Skype for business server issue.
Microsoft	VOIP-118301	5.5.1	The phone doesn't join an active call when paired with the Skype for Business client while the client is already in the call. This is a Skype for Business client issue.
Microsoft	VOIP-118908	5.5.1	Exchange Calendar now works and syncs in 8 mins after re-registering after the network was down and the user is signed into Skype for Business on the phone via device pairing.
Microsoft	VOIP-119471	5.4.4	Local conference calling is now available in Skype for Business environments.
Microsoft	VOIP-119732	5.5.1	Entering the pound key (#) for a DTMF command is not working when an unanswered call is routed to Voicemail. This is a Skype for Business server issue.
Microsoft	VOIP-119798	5.4.4	The phone no longer reboots when attempting to join an internal Skype for Business meeting.
Microsoft	VOIP-120720	5.5.1	The Skype for Business Device Lock feature is now mutually exclusive with the Phone Lock feature.
Network	VOIP-113342		The number of daily network requests sent by the VVX system to the Exchange server has been reduced.
Network	VOIP-113993		New parameter `qos.ethernet.tcpQosEnabled` allows you to configure the phone to send configured QoS priorities for SIP on TCP.
Network	VOIP-117142		A computer connected to the PC port on the phone now experiences throughput at the speed of the LAN.
Network	VOIP-117472		The phone now fails over correctly when re-registration on failover is enabled and failover fallback mode is set to Registration.
Network	VOIP-118275	5.4.4	An issue was resolved that caused the VVX system to create a new TLS socket when a call was canceled shortly after it was dialed, causing the phone to lose registration.
Network	VOIP-118723	5.4.4	An issue was resolved that caused the NAT Keepalive message to be sent for only one registered line when the phone was configured for two lines.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
Network	VOIP-119127 VOIP-119128	5.4.1 5.4.2	VVX Phones upon receiving INVITE with multiple diversion headers now displays the first diversion header when parameter `voIpProt.SIP.header.diversion.list.useFirst` is enabled.
Network	VOIP-119152 VOIP-119153	5.4.2	An issue with the duration timer used for failback to the primary server has been resolved.
Network	VOIP-119166	5.4.4	DNS queries are now sent shortly after the phone is powered on. This no longer results in NTP failures and the message "Time/date not set" does not display on the phone.
Network	VOIP-119287	5.4.4	The phones now correctly utilize contact headers that contain the addr parameter in a 301 response to an outbound INVITE.
Network	VOIP-120270 VOIP-123203 VOIP-123204	5.5.0	The phone now registers with IPv4 when using dual stack IPv4IPv6.
Power Management	VOIP-119928	5.4.1	VVX 501/601 phones can now power a VVX Camera and a VVX Color Expansion module connected to the phone over IEEE 802.af source.
Registration	VOIP-119322	5.5.0	Phones configured for PIN authentication no longer unregister during system maintenance.
Server	VOIP-120069	5.5.1	Switching a Skype for Business call between a Jabra Evolve 40 head-set and the VVX phone is inconsistent because of a server related is-sue.
Software Update	VOIP-118208	5.4.4	The phones now correctly request DHCP Option 144 after a firmware upgrade.
Software Update	VOIP-118657 VOIP-118994	5.4.3 5.4.4	An issue was resolved which caused phones to not to show the PIN Authentication option after upgrading software in certain environments.
User Interface	VOIP-112418		On systems using the Busy Lamp Field (BLF), the caller's name and number now scroll across the screen if the information is wider than the available screen space.
User Interface	VOIP-112625		The warning icon is no longer displayed on VVX phones after the administrator password is changed.
User Interface	VOIP-113621		New parameter `feature.lync.hideSignInSignOut` was added to allow you to hide the Sign In and Sign Out buttons on the user interface.
User Interface	VOIP-114845		Labels now split correctly in the user interface when alignment is set to Right or None. When text alignment is set to Left, labels may not correctly split.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
User Interface	VOIP-115572	5.4.2	In Skype for Business environments, the message displayed upon sign-in is now properly displayed.
User Interface	VOIP-115884		Outbound calls from Multiple Appearance Directory Number (MADN) lines now correctly display the called number.
User Interface	VOIP-116825	5.5.0	On the Favorites screen, pressing the empty third and fourth soft keys incorrectly displays the Info screen.
User Interface	VOIP-116859		When Multiple Appearance Directory Number (MADN) is enabled, outbound calls now display the called number.
User Interface	VOIP-116943		Performance of the VVX 1500 phone has been improved to resolve problems that caused sluggishness on the phone's user interface.
User Interface	VOIP-117392		VVX phones now correctly display the caller ID for outbound calls from MADN lines.
User Interface	VOIP-117522		The phone now correctly display codes from 128 to 159 (€,ƒ„…†‡ˆ‰Š‹ŒŽ ’’""•—-˜™š›œžŸ).
User Interface	VOIP-118285	5.4.2	An issue was resolved that caused LDAP query search results to fail to display all the fields if `dir.corp.attribute.x.label` values contain non-ASCII characters.
User Interface	VOIP-118792 VOIP-118796	5.4.4	An issue was resolved that caused an incorrect destination to display in the call list when registrations are configured with multiple line keys.
User Interface	VOIP-119073		New parameter `up.basicSettings.networkConfigEnabled` was added to allow users to access Ethernet and DHCP settings on the Basic menu.
User Interface	VOIP-119308	5.3.0	The Home screen no longer shows a blue background for a long period of time when a user presses the Back key from Recent list on a VVX 1500.
User Interface	VOIP-119433 VOIP-119434	5.3.0	An issue was resolved that caused the VVX 1500 Home screen to show a blue background when you pressed the Back key while viewing the phone list.
Video	VOIP-118435	5.4.4	The phone now handles the video call over SRTP or TLS, and the two-way video call works without any issue over TLS/SRTP.
VVX D60	VOIP-113704 VOIP-113521 VOIP-114469		The wireless handset occasionally displays as out of range and the signal strength is not shown for a few seconds before displaying again.

Resolved Issues in UC Software 5.5.1

Category	Issue No.	Release	Description
VVX D60	VOIP-120061 VOIP-120344	5.4.4	VVX D60 wireless handsets no longer fail when the dial string includes a pound (#) sign, which was causing feature access codes dialed on the wireless handset to fail.
VVX D60	VOIP-120297 VOIP-120351	5.4.4	VVX D60 wireless handsets now handle in-band DTMF properly when RFC 2833 is unavailable.
Web Configuration Utility	VOIP-118714 VOIP-118840	5.4.1	The phone now applies device certificates and device parameters from the phone's web interface.
Web Interface	VOIP-110074		You can now upload the device configuration files using the phone’s web interface.

What’s New in Polycom UC Software 5.5.0

Polycom UC Software 5.5.0 includes the features and functionality of previous releases and includes the following new features.

Note: For information on using parameters to configure features, see the UC Software *Administrator’s Guide* at Polycom Support.

BroadSoft Executive-Assistant

BroadSoft Executive-Assistant is a feature on the BroadWorks R20 and later server that enables a system administrator to assign users as executives or assistants for private or shared lines.

Executives can use call filtering to send calls directly to an assistant’s phone to answer. Executives and assistants can also use screening to allow the executive’s phone to display the incoming call notification for all filtered calls, allowing the executive to decide whether to accept the call or allow an assistant to manage the call on their behalf. The feature also allows an assistant to initiate a call on behalf of an executive. In this case, the receiving party sees the call as coming from the executive, and for an executive to barge in (silently or otherwise) to a call that the assistant is managing on their behalf.

Administrators can configure this feature using the following parameters:

- `feature.BSExecutiveAssistant.enabled`
- `feature.BSExecutiveAssistant.regIndex`
- `feature.BSExecutiveAssistant.userRole`

This feature is not supported on the SoundStructure VoIP Interface.

Support for TR-069

Polycom phones can now be remotely configured and managed by provisioning systems that support the TR-069 (Technical Report 069) technical specification.

Support for 3GPP Technical Specification

For phones deployed in an IP Multimedia Subsystem (IMS) environment, Polycom introduces support for a subset of the 3rd Generation Partnership Project technical specifications (3GPP TS) as defined by standard RFCs and the 3GPP TS specifications 24.229, 24.615, and 24.629.

This release adds the following IMS feature enhancements:

- The call waiting ringback tone plays to inform you that the call is waiting at the far end.
- The SIP Response Code 199 (defined in RFC-6228) is supported.

The Path extension header field in the SIP Register request message allows accumulating and transmitting the list of proxies between a user agent and Registrar server. The administrator can configure the parameter `reg.x.path` to enable or disable support for this header field for a specific line registration.

The caller phone can support the p-early-media SIP header that determines whether the caller phone should play a network-provided media or its own media as a ringback tone. The administrator can configure the parameter `voIpProt.SIP.header.pEarlyMedia.support` to enable or disable support for this header field on the caller phone.

The VQMon messages that are generated by the phone can contain service route information in SIP route headers. The administrator can configure the parameter `voice.qualityMonitoring.processServiceRoute.enable` to enable or disable this header field on the VQMon messages generated by a phone device.

In a NAT network, a phone may need to send keep-alive messages to maintain the IP addresses mapping in the NAT table. The parameters `nat.keepalive.udp.payload` and `nat.keepalive.tcp.payload` are introduced to configure a customizable string as the payload of the UDP and TCP keep-alive messages.

BroadSoft Flexible Seating

You can configure host phones to allow users to log in to their registered phone line remotely. After the user logs in, the user's configurations are replicated to the host phone. The user's registered phone line is then active on both the primary phone and the host phone.

This feature is not supported on the SoundStructure VoIP Interface.

Support for IPv6 Protocol

The VVX Business Media Phones now supports IPv6 in the Open SIP environment, as well as IPv4 and dual stack (IPv4/IPv6) modes.

Off-Hook Screen View and In-Call Status Display

You can configure the default user interface for dialer screen events on the Polycom VVX 500 and 600 series business media phones. For example, you can configure the Dialer view or the Lines screen as the default screen that is displayed when the line goes off hook. You can also configure active call information to show in the Active Call screen or in the status bar on the Lines screen. You can configure the user interface using the following parameters:

- `up.OffHookLineView.enabled`

- `up.LineViewCallStatus.enabled`
- `up.LineViewCallStatus.timeout`

Microbrowser Support for VVX 201 Business Media Phone

The VVX 201 business media phone now supports a microbrowser. However, due to the smaller screen size, the VVX 201 microbrowser behavior and display differ in appearance from other VVX phone models. Note that the VVX 101 business media phone does not support a microbrowser.

Locking the Web Configuration Utility after Failed Login Attempts

You can lock access to the Web Configuration Utility after a series of failed login attempts and configure a period of time a user can attempt to log in again. Use the following parameters to configure additional security after multiple failed login attempts:

- `httpd.cfg.lockWebUI.enable`
- `httpd.cfg.lockWebUI.lockOutDuration`
- `httpd.cfg.lockWebUI.noOfInvalidAttempts`
- `httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration`

Off-Hook Idle Browser

Typically, the microbrowser only appears when the phone is idle and not in a call. On VVX 500 and 600 series business media phones, you can use the parameter `up.OffHookIdleBrowserView.enabled` to enable the microbrowser and associated soft keys to continue to display on the phone when the phone goes off-hook. When enabled, the microbrowser continues to display until the user enters a number.

User Profile Login Enhancement

User profile authentication can now be performed on the provisioning server instead of on the phone for improved security.

BroadWorks Call Decline

For shared lines in a BroadSoft BroadWorks environment, you can set the parameter `call.shared.reject` to 1 to enable users to reject calls on the shared line. When a user rejects a call to the shared line, the call is rejected on all phones registered with the shared line.

User Interface Themes

Users can now choose from two user interface themes for the VVX 500 and 600 series business media phones: Classic (default) or Modern. The Modern theme is new for this release and includes a new color scheme and icons. Users can select a theme from the Basic settings menu on the phone, or administrators can configure a theme using the following configuration parameter:

- `device.theme`

Minimum Ringer Volume

You can now configure a minimum ringer volume using new parameter `up.ringer.minimumVolume`. This parameter defines how many volume steps are accessible below the maximum level.

Password Protection for Editing Contacts Directory

You can now configure the system to require a password to edit the Contacts Directory.

Configuration File Enhancements

Changing the following configuration parameters no longer causes a restart or reboot when you change the value:

- `attendant.reg`
- `attendant.uri`
- `attendant.behaviors.display.spontaneousCallAppearances.normal`
- `attendant.behaviors.display.spontaneousCallAppearances.automata`
- `attendant.behaviors.display.remoteCallerID.normal`
- `attendant.behaviors.display.remoteCallerID.automata`
- `attendant.resourceList.x.callAddress`
- `attendant.resourceList.x.address`
- `attendant.resourceList.x.label`
- `attendant.resourceList.x.type`
- `attendant.resourceList.x.proceedingIsRecipient`
- `attendant.resourceList.x.requestSilentBargeIn`
- `attendant.resourceList.x.bargeInMode`

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.5.0.

Configuration File Enhancements

Parameter Template	Permitted Values
`call.shared.preferCallInfoCID` sip-interop.cfg	Specify whether Caller ID information is displayed. 0 (default) – Caller ID received from 200OK is ignored if NOTIFY message includes display information. 1 – Caller ID received from 200OK is displayed if NOTIFY message includes display information.
`call.shared.reject` sip-interop.cfg	For shared line calls on the BroadWorks server. 0 – The phone displays a Reject soft key to reject an incoming call to a shared line. 1 – The Reject soft key does not display.

Configuration File Enhancements

Parameter Template	Permitted Values
`call.urlNumberModeToggling` site.cfg	Determines whether the phone uses Number mode or URL mode when a URL call is initiated. 0 (default) – URL mode is used for URL calls. 1 – Number mode is used for URL calls.
`device.dhcp.bootSrvUseOpt` device.cfg	Specifies the source for the boot server address for the phone. It can take values from 0 to 9. In IPv6 mode, the following values are applicable: • 4 - The phone uses the boot server configured through the Server menu. • 5 - The phone uses the boot server option provided through DHCPv6. In Dual Stack Mode (IPv4/IPv6 mode), the following values are applicable: • 6 - The phone uses the boot server configured through the Server menu. • 7 - The phone gets the boot server details from DHCPv6 option or the Option 66 on DHCP server. • 8 - The phone gets the boot server details through DHCPv6 or through the custom option configured on DHCP server for the provisioning. • 9 - The phone gets the boot server from DHCPv6 option or custom option or option 66 configured on DHCP server for the provisioning.
`device.feature.tr069.enabled` tr069.cfg	0 (default) – Disables the TR-069 feature. 1 – Enables the TR-069 feature.
`device.ipv6.icmp.echoReplies` device.cfg, wireless.cfg	0 (default) 1
`device.ipv6.icmp.genDestUnreachable` device.cfg, wireless.cfg	0 (default) 1
`device.ipv6.icmp.ignoreRedirect.set` device.cfg	0 (default) 1
`device.ipv6.icmp.txRateLimiting` device.cfg	0 6000 (default)
`device.net.ipStack` device.cfg, site.cfg	Specify a valid global IPv6 unicast address for the phone. Null (default)

Configuration File Enhancements

Parameter Template	Permitted Values
`device.net.ipv6AddrDisc` device.cfg, site.cfg	Specify whether the IPv6 address and related parameters for the phone are obtained from DHCPv6 or SLAAC or statically configured for the phone. 1 (Default) -IPv6 global address and options are configured from DHCPv6. 2 - IPv6 global address is configured using prefixes received from Router Advertisements (RA) and options are configured from stateless DHCPv6. 0 - IPv6 global address and options must be configured manually.
`device.net.ipv6Address` device.cfg, site.cfg	Specify a valid global IPv6 unicast address for the phone. Null (default)
`device.net.ipv6Gateway` device.cfg, site.cfg	Specify theIPv6 address of the default gateway for the phone. Null (default)
`device.net.ipv6LinkAddress` device.cfg, site.cfg	Specifies a valid Link Local IPv6 address for the phone. Null (default)
`device.net.ipv6PrivacyExtension` device.cfg, site.cfg	Configure whether or not the IPv6 global and link local addresses are in 64-bit Extended Unique Identifier (EUI-64) format. 0 (Default) - IPv6 global and link local addresses are in EUI-64 format. 1 - Global and link local IPv6 addresses are not in EUI-64 format. Instead, the last 48 bits for the IPv6 address are generated randomly.
`device.net.ipv6ULAAddress` device.cfg, site.cfg	Specifies a valid Unique Local IPv6 address (ULA) for the phone. Null (default)
`device.net.preferredNetwork` device.cfg, site.cfg	Specify IPv4 or IPv6 as the preferred network in a Dual Stack mode. 1 (default) - Specifies IPv6 as a preferred network. 0 - Specifies IPv4 as a preferred network.
`device.theme` device.cfg	Modern (default) - The phone uses the Modern theme. Classic - The phone uses the Classic theme.
`device.theme.set` device.cfg	1 (Default) - The phone supports both the Classic and Modern theme. The device.theme parameter specifies which theme to use. 0 - The phone supports only Modern theme.
`device.tr069.acs.password` tr069.cfg	Sets the TR-069 ACS server password used to authenticate the phone. Null (default) String (256 maximum characters)

Configuration File Enhancements

Parameter Template	Permitted Values
`device.tr069.acs.url` tr069.cfg	Sets the URL for the TR-069 ACS server. Null (default) URL (256 maximum characters)
`device.tr069.acs.username` tr069.cfg	Sets the TR-069 ACS server username used to authenticate the phone. PlcmSpip (default) String (256 maximum characters)
`device.tr069.cpe.password` tr069.cfg	Specifies the TR-069 CPE password, which authenticates a connection request from the ACS server. Null (default) String (256 maximum characters)
`device.tr069.cpe.username` tr069.cfg	Specifies the TR-069 CPE user name, which authenticates a connection request from the ACS server. PlcmSpip (default) String (256 maximum characters)
`device.tr069.periodicInform.enabled` tr069.cfg	Indicates whether the CPE must periodically send CPE information to ACS using the **Inform** method call. 0 (default) - Periodic **Inform** call is disabled. 1 - Periodic **Inform** call is enabled.
`device.tr069.periodicInform.interval` tr069.cfg	Specifies the time interval in seconds in which the CPE must attempt to connect with the ACS to send CPE information if set to TRUE. 18000 (default) 0 to 36000
`device.tr069.upgradesManaged.enabled` tr069.cfg	Indicates whether the ACS manages image upgrades for the phone or not. 0 (default) – The phone uses ACS or provisioning server for upgrade. 1 - The phone upgrades only from the ACS server.
`dir.local.passwordProtected` features.cfg	Specifies whether you are prompted for an Admin or User password when adding, editing, or deleting contacts in the Contact Directory. 0 (default) – No password prompt is displayed and pressing and holding the Line-key displays the Add or Edit menu. 1 – You are prompted for your Admin or User password while adding, editing, or deleting contacts in the Contact Directory.
`feature.BSExecutiveAssistant.enabled` features.cfg	0 (default) - Disables the BroadSoft Executive-Assistant feature. 1 - Enables the BroadSoft Executive-Assistant feature.

Configuration File Enhancements

Parameter Template	Permitted Values
`feature.BSExecutiveAssistant.regIndex` features.cfg	Specifies the registered line assigned to the Executive or Assistant for the BroadSoft Executive-Assistant feature. 1 (default) to 255 - The registered line for the Executive or Assistant. Note that a line icon for the role specified by the parameter `feature.BSExecutiveAssistant.userRole` displays even if you do not assign an Executive-Assistant service to a line in the BroadSoft Web Portal. Ensure that the services assigned to the line match the user role.
`feature.BSExecutiveAssistant.userRole` features.cfg	Specifies whether the phone is set to an Executive or an Assistant role. Note that an Executive and an Assistant line cannot be set on the same phone. ExecutiveRole (default) - Sets the registered line as an Executive line. AssistantRole - Sets the registered line as an Assistant line.
`feature.logUpload.enabled` features.cfg	1 (default) - Enable log uploads for Skype for Business. 0 - Disable log uploads for Skype for Business.
`hoteling.reg` features.cfg	1 (default) - Specifies the phone line on the host phone which hosts the guest line.
`httpd.cfg.lockWebUI.enable` site.cfg	Specifies whether web configuration login lock is ena-bled. 1 (default) – Enable the Web Configuration Login Lock feature. 0 - Disable the Web Configuration Login Lock feature.
`httpd.cfg.lockWebUI.lockOutDuration` site.cfg	Specifies how long the user is locked out of the Web Configuration Utility. 60 seconds (default) - The period of time during which the user is locked out of the Web Configuration Utility. The user can try logging in again after this time. 60 - 300 seconds
`httpd.cfg.lockWebUI.noOfInvalidAttempts` site.cfg	Specifies the number of failed login attempts after which the user is locked out of the Web Configuration Utility. 5 (default) 3 - 20
`httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration` site.cfg	Specifies time period during which the user must log in successfully to avoid being locked out of the Web Configuration Utility. The user can try logging in again after the lock-out duration set by `httpd.cfg.lockWebUI.lockOutDuration`. 60 seconds (default) 60 - 300 seconds

Configuration File Enhancements

Parameter Template	Permitted Values
`lcl.ml.lang.japanese.font.enabled`[1] site.cfg	Specifies whether the Japanese Kanji font is enabled. This parameter applies to VVX 400, 401, 410, 411, 500, 501, 600, 601, and 1500. 0 (default) – The phone does not use Japanese Kanji character font. 1 - The phone displays Japanese Kanji character font.
`log.level.change.tr069` tr069.cfg	Sets the log levels for the TR-069 feature. 4 (default) 0 - 6
`nat.keepalive.tcp.payload` sip-interop.cfg	Sets a customizable string as the payload of a TCP keep-alive message. Note that the string value cannot be blank. CRLFCRLFCRLFCRLFCRLFCRLFCRLFCRLF(default) string
`nat.keepalive.udp.payload` sip-interop.cfg	Sets a customizable string as the payload of a UDP keep-alive message. CRLFCRLF (default) String Blank (for empty payload)
`prov.login.localPassword.hashed` site.cfg	Specifies whether the phone generates a custom digest hash to encrypt the user password. 0 (default) – The phone does not generate a custom digest hash to encrypt the user password. You must store the user password in `prov.login.localPassword`. 1 – The phone generates a custom digest hash to encrypt the user password and store it.
`prov.login.password.encodingMode` site.cfg	Configures the default Encoding mode for the text in the password field on the User Login screen. 123 (default) Abc ABC Abc
`prov.login.useProvAuth` site.cfg	Specifies whether phones use server authentication. 0 (default) – The phones do not user server authentication. 1 – The phones use server authentication.
`prov.login.userId.encodingMode` site.cfg	Configures the default Encoding mode for the text in the User ID field on the User Login screen. abc (default) ABC Abc 123

Configuration File Enhancements

Parameter Template	Permitted Values
`reg.x.header.pEarlymedia.support` reg-advanced.cfg	Specifies whether the line supports the p-early-media header. 0 (Default) – The p-early-media header is not supported on the specified line registration. 1 – The p-early-media header is supported by the specified line registration.
`reg.x.insertOBPAddressInRoute` reg-basic.cfg	Specifies whether the outbound proxy address for the phone is added in the route header. If added, the out-bound proxy address is added as the top most route header. 0 – The outbound proxy address is not added to the route header. 1 (default) – The outbound proxy address is added as the top-most route header.
`reg.x.regevent` reg-advanced.cfg	Allows you to subscribe a specific phone line to registration event notifications from the SIP server, along with related information. When enabled, this parameter overrides the `voIpProt.SIP.regevent` parameter, which allows global level configuration for the phone device. 0 (default) – The phone is not subscribed to notifications for the specific phone line. 1 – The phone is subscribed to notifications for the specific phone line.
`reg.x.rejectNDUBInvite` reg-advanced.cfg	Specifies whether the phone accepts a call for a particular registration in case of a Network Determined User Busy (NDUB) event advertised by the SIP server. 0 (default) – Phone rejects the call with a 603 Decline response code. 1 – Phone accepts the call.

Configuration File Enhancements

Parameter Template	Permitted Values
`reg.x.server.y.specialInterop` reg-advanced.cfg	Specifies the server-specific feature set supported by the line registration. VVX 101 = Standard GENBAND GENBAND-A2 ALU-CTS VVX 201 = Standard GENBAND GENBAND-A2 ALU-CTS ocs2007r2 lync2010 All other phones = Standard GENBAND GENBAND-A2 ALU-CTS ocs2007r2 lync2010 lcs2005
`sec.TLS.LDAP.strictCertCommonNameValidation` site.cfg	Specifies whether the server certificate common name must be validated during an LDAP or LDAPS connection over TLS. 1 (default) – Requires validation of server certificate common name during LDAP or LDAPS connection over TLS. 0 – Does not require validation of server certificate common name during LDAP or LDAPS connection over TLS.
`sec.TLS.profile.webServer.cipherSuiteDefault` site.cfg	Specifies whether the phone uses the default cipher suite for the web server profile. 1 (default) – Uses the default cipher suite for the web server profile. 0 – Uses the custom cipher suite for the web server profile.
`sec.TLS.profile.x.cipherSuite` site.cfg, wireless.cfg	Specifies which cipher suite the phone uses for the TLS Application Profile. Null (default) 1 – 8 – Choose the cipher suite for the TLS Application Profile.
`sec.TLS.profile.x.cipherSuiteDefault` site.cfg, wireless.cfg	Specifies the default cipher suite for the TLS Application Profile. 1 (default) – Use the default cipher suite. 0 – Use the custom cipher suite for the TLS Application Profile.

Configuration File Enhancements

Parameter Template	Permitted Values
`sec.TLS.webServer.cipherList` site.cfg	Specifies the cipher list for a web server profile. The format for the cipher list uses OpenSSL syntax found at http://www.openssl.org/docs/apps/ciphers.html. RSA:!EXP:!LOW:!NULL:!MD5:!RC4:@STRENGTH (default) String
`up.deviceLock.createLockTimeout` features.cfg	Specifies the timeout in minutes for the Create Lock Code prompt after Device Lock is enabled. 0 (default) – The Create Lock Code prompt does not time out. 1 – 3 minutes
`up.deviceLock.signOutOnIncorrectAttempts` features.cfg	Configures phone behavior after six unsuccessful unlock attempts for Device Lock. 0 (default) – After six unsuccessful unlock attempts, phone prompts the user to wait 60 seconds before trying again. 1 – Signs the user out after six unsuccessful unlock attempts.
`up.LineViewCallStatus.enabled` features.cfg	Specifies the Active Call Screen or Line Screen as default user interface for a call. 0 (default) – Active Call Screen is set as default user interface for an active call. Any incoming or outgoing call triggers the Active Call Screen. 1 – Line Screen is set as default user interface for an active call. For a call, the phone remains in Line Screen and the active call details show in the status ribbon bar.
`up.LineViewCallStatusTimeout` features.cfg	Specifies the number of seconds the Active Call screen displays before returning to the Line screen. This parameter is applicable when the Line Screen is set as default user interface for any call. 10 seconds (default) 2-9 seconds
`up.OffHookIdleBrowser-View.enabled` features.cfg	Enables the microbrowser and associated soft keys to continue to display on the phone when the phone goes off-hook. 0 (Default) – The idle browser does not display on screen after the phone goes off-hook. 1 – The idle browser continues to display on screen after the phone goes off-hook.
`up.OffHookLineView.enabled` features.cfg	Specifies the default user interface displayed when the phone goes off-hook. 0 (default) – Home Screen displays when the phone goes off-hook. 1 – Line Screen displays when the phone goes off-hook.

Configuration File Enhancements

Parameter Template	Permitted Values
`up.ringer.minimumVolume` site.cfg	Configure the minimum ringer volume. This parameter defines how many volume steps are accessible below the maximum level. 16 (default) – The full 16 steps of volume range are accessible. 1-15 0 – Ring volume is not adjustable by the user and the phone uses maximum ring volume. Upon bootup, the volume is set to ½ the number of configured steps below the maximum (16). So, if the parameter is set to 8, on bootup, the ringer volume is set to 4 steps below maximum.
`voice.cn.hs.attn` site.cfg	Sets the attenuation of the inserted comfort noise in dB, where smaller values insert louder noise. The default value 30 is quite loud. This parameter is used only when voice.cn.hs.enable is set to 1. 30 dB (default) 3 – 90 dB
`voice.cn.hs.enable` site.cfg	Specifies whether Comfort Noise (CN) is added to the transmit path of the handset. This feature should only be used when users at the far end perceive that the phone has gone "dead" when the near-end user stops talking. 0 (default) – No Comfort Noise is added. 1 – Comfort Noise is added to the handset.
`voice.plcCnEnable` site.cfg	Specifies whether the existing G.711 Appendix 1 Packet Loss Concealment (PLC) process is augmented by adding Comfort Noise (CN) during an extended loss. This prevents the synthesized concealment audio from decaying to silence. 0 (default) – No Comfort Noise is added. 1 – Comfort Noise is added.
`voice.plcCnGain` site.cfg	Specifies the gain applied to the synthesized Packet Loss Concealment (PLC) comfort noise in dB. Adjusting the PLC CN gain may be useful when interoperating with endpoints whose background noise is not well matched to the CN synthesis algorithm. This parameter is used only used when `voice.plcCnEnable` is 1. 0 (default) -20 – 20 dB
`voice.qoe.event.lossrate.threshold.bad` features.cfg	Defines the threshold for the network loss rate. Total packets lost for an interval/total packets expected for the interval *256 as stated in RFC-2611, section 4.7.1. 38 (default) - Approximately a 15% packet loss. 0 to 100

Configuration File Enhancements

Parameter Template	Permitted Values
`voice.qoe.event.lossrate.threshold.poor` features.cfg	Defines the threshold for the network loss rate. Total packets lost for an interval/total packets expected for the interval *256 as stated in RFC-2611, section 4.7.1. 25 ms (default) - Approximately a 10% packet loss. 0 to 100
`voice.qoe.event.networkmos.threshold.bad` features.cfg	Defines the threshold for Network MOS using the average of MOS-LQO wideband, as specified by [ITUP.800.1] section 2.1.2, based on the audio codec used and the observed packet loss and inter-arrival packet jitter. 19 (default) - MOS score of 1.9 10 - 50 - MOS score between 1 - 5 networkMOS > 2.9 signifies good quality networkMOS > 2.9 < 1.9 signifies poor quality networkMOS < 1.9 signifies bad quality
`voice.qoe.event.networkmos.threshold.poor` features.cfg	Defines the threshold for Network MOS using the aver-age of MOS-LQO wideband, as specified by [ITUP.800.1] section 2.1.2, based on the audio codec used and the observed packet loss and inter-arrival packet jitter. 29 (default) - MOS score of 2.9 10 - 50 - MOS score between 1 - 5 networkMOS > 2.9 signifies good quality networkMOS > 2.9 < 1.9 signifies poor quality networkMOS < 1.9 signifies bad quality
`voice.qualityMonitoring.processServiceRoute.enable` features.cfg	Specifies whether the SIP route headers for the VQMon messages generated by the phone contain service route information. 0 (default) – The VQMon messages generated by the phone do not contain service route information in SIP route headers. 1 – The VQMon messages generated by phone, contain service route information, if available, in SIP route headers.

Configuration File Enhancements

Parameter Template	Permitted Values
`voIpProt.server.x.specialInterop` sip-interop.cfg	Specifies the server-specific feature set supported for all line registrations. VVX 101 = Standard GENBAND GENBAND-A2 ALU-CTS VVX 201 = Standard GENBAND GENBAND-A2 ALU-CTS ocs2007r2 lync2010 All other phones = Standard GENBAND GENBAND-A2 ALU-CTS ocs2007r2 lync2010 lcs2005
`voipProt.SIP.anat.enabled` sip-interop.cfg	Enables or disables Alternative Network Address Types (ANAT). 0 (default) - ANAT is disabled. 1 - ANAT is enabled.
`voIpProt.SIP.header.pEarlyMedia.support` sip-interop.cfg	Specifies whether the caller phone supports the p-early-media header. 0 (Default) – The p-early-media header is not supported by the caller phone. 1 – The p-early-media header is supported by the caller phone.
`voIpProt.SIP.IMS.enable` sip-interop.cfg	Configures support on the phone device for IMS features that are introduced in UC Software 5.5.0 or later. This parameter is applicable for all registered or unregistered SIP lines on the phone. 0 (Default) – Phone cannot support IMS features that are introduced in UC Software 5.5.0 or later. 1 – Phone supports IMS features that are introduced in UC Software 5.5.0 or later.
`voIpProt.SIP.looseContact` sip-interop.cfg	Configures addition of the ephemeral port parameter to the contact header. 0 (default) - The ephemeral port is added to the contact header in TLS case. 1 – The port parameter is not added to the contact header or SIP messages.

Configuration File Enhancements

Parameter Template	Permitted Values
`voIpProt.SIP.regevent` reg-advanced.cfg	Configures subscription of all phone lines on a phone to registration event notifications from the SIP server along with related information. When enabled, this parameter configuration is overridden by the `reg.x.regevent` parameter, which is configuration for a specific phone line. 0 (default) – The phone is not subscribed to notifications for all phone lines. 1 – The phone is subscribed to notifications for all phone lines.
`voIpProt.SIP.rejectNDUBInvite` reg-advanced.cfg	Specifies whether the phone accepts a call for all registrations in case of a Network Determined User Busy (NDUB) event advertised by the SIP server. 0 (default) – Phone rejects the call with a 603 Decline response code. 1 – Phone accepts the call.
`voIpProt.SIP.specialEvent.checkSync.downloadCallList` site.cfg	Specifies whether the phone downloads the current user's call list when a check-sync event NOTIFY message is received from the server. 0 (default) – Call list is not downloaded after receiving a check-sync event in the NOTIFY message. 1 – Call list is not downloaded after receiving a check-sync event in the NOTIFY message.
`voIpProt.SIP.supportFor199` sip-interop.cfg	Specifies whether the phone supports the 199 response code. For details, see the RFC-6228, Session Initiation Protocol (SIP) Response Code for Indication of Terminated Dialog. 0 (default) – The phone does not support 199 response code. 1 – The phone supports the 199 response code.

[1] Change causes the phone to restart or reboot.

Supported DHCP Sub-Options

The following table lists the individual sub-options and combination sub-options supported on VVX phones for DHCP Option 43:

DHCP Option 43 Configuration Options

Option	Result
Option 1- Subnet mask	The phone parses the value from Option 43.
Option 2 - Time offset	The phone parses the value.
Option 3 – Router	The phone parses the value.
Option 4 - Time server	The phone parses the value.

DHCP Option 43 Configuration Options

Option	Result
Option 6 - Domain Name Server	The phone parses the value.
Option 7 - Domain Log server	The phone parses the value.
Option 15 - Domain Name	The phone parses the value.
Option 42 - Network Time Protocol server	The phone parses the value.
Option 66 - TFTP Server Name	The phone parses the value.
Options 128-255	
Sub-options configured in Option 43	
Options 1, 2, 3, 4, 5, 6, 7, 15, 42, and 66	The phone parses the value.
Options 128-255	

Resolved Issues

This section lists the issues that were resolved in UC Software 5.5.0.

Resolved Issues in UC Software 5.5.0

Category	Issue No.	Release	Description
Audio	VOIP-116379	5.4.1	Using Plantronics Voyager Legend UC no longer causes any abrupt call drops on VVX phones.
Audio	VOIP-113375		An issue was resolved that caused audio interruption on the Plantronics headset when a fourth caller tries to join a local three-way conference and then cancels.
Audio	VOIP-111806		Using the 3CX call park feature with the TCP trunk no longer causes one-way audio and no longer prevents unparking a call.
Audio	VOIP-105505		A problem was resolved that caused audio drop when an attended transfer is triggered with the Competella Attendant Console.
Audio	VOIP-112844	5.3.1	When you enable the soft key using ESK, the user can access and launch the browser by pressing the soft key configured for the micro browser.
Busy Lamp Field	VOIP-115996		On phones with call waiting disabled, Busy Lamp Field activity no longer causes call waiting tones to be played.
Busy Lamp Field	VOIP-112438	5.3.1	A problem was resolved that caused Busy Lamp Field activity to trigger call waiting tones on phones where call waiting was disabled.
Calling	VOIP-115446		Blind transfer with SLA line and with `exposeAutoHold` enabled is now working as expected

Resolved Issues in UC Software 5.5.0

Category	Issue No.	Release	Description
Calling	VOIP-115285		The parameter `call.shared.preferCallInfoCID` was added to enable configuring whether Caller ID information is displayed.
Calling	VOIP-114466		A Polycom VVX phone configured to use Simultaneous Ring Personal no longer rings for an incoming call when Do Not Disturb is enabled.
Calling	VOIP-114287		An issue was resolved that caused an incoming click-to-dial call to play an incorrect tone.
Calling	VOIP-113925		Transferring an internal call between Polycom VVX phones when using the NUANCE dial-by-voice system now works as expected.
Calling	VOIP-113922		Joining a PSTN user to conference call now works on Skype for Business Online.
Calling	VOIP-113478	5.4.1	When the SoundStructure VoIP Interface is in a call, sending a "set voip_send VoIP Out" command to the SoundStructure no longer causes the call to disconnect. Pressing a digit on a Polycom Touch Control paired with the SoundStructure during a call now works correctly.
Calling	VOIP-112886		Line seize behavior for accessing voicemail using Enhanced Feature Keys (EFK) has been improved.
Calling	VOIP-111991		Parameter `call.urlNumberModeToggling` now allows you to specify whether the phone uses Number mode or URL mode when a URL call is initiated.
Calling	VOIP-109593		The parameter `call.urlNumberModeToggling` was added to resolve a problem with URL dialing.
Calling	VOIP-109311		The phone now correctly sends "user=phone" in the invite message when a user enters a number that ends with "#" or "*".
Calling	VOIP-107290		The phone now ignores any unrecognized parameters included in check-sync messages.
Calling	VOIP-115425		New parameter `call.shared.reject` was added to allow you to configure phones to display a Reject soft key for calls on a shared line.
Calling	VOIP-116273	4.0.9	Phones now use the contact URI and TELURI in the request line of BYE message, so calls end correctly when the `reg.1.telUri` is enabled or disabled.
Calling	VOIP-116228 VOIP-119773	5.4.2	Using blind transfer for calls to Exchange Auto attendant in a Skype for Business Online environment now works correctly.
Calling	VOIP-116207	5.4.2	An issue was resolved that caused a core dump after pressing Transfer and the extension.

Resolved Issues in UC Software 5.5.0

Category	Issue No.	Release	Description
Calling	VOIP-112294	5.4.0	An issue was resolved that was caused when an Inbound call was transferred, conferenced, and then transferred again.
Calling	VOIP-111987	4.0.8	The phone now uses the blind transfer behavior from the Enhanced Feature Key (EFK) soft keys and sends a HOLD message before the REFER message.
Contact Directory	VOIP-110651		Phones no longer reboot if a contact is selected and dialed within two seconds of receiving the first results in a Corporate Directory search.
Contact Directory	VOIP-110199		VVX 1500 integration with RPRM has been improved for IP, H323, E164, and Annex-O Phonebook storing and dialing.
Contacts	VOIP-115334		Parameter `voIpProt.SIP.looseContact` was added to control whether an ephemeral port is added to the contact header in a TLS environment.
Directory	VOIP-113115	5.4.2	The Contact Directory is now uploaded when sent check-sync;upload=directory is set.
General	VOIP-109359		EFK configured for dialing a number from shared Line 1 now works as expected, allowing users to dial out from Line 1.
General	VOIP-114622		The default User ID encoding mode for parameters `prov.login.userId.encodingMode` and `prov.login.password.encodingMode` was changed to abc/ASCII.
General	VOIP-113119	5.4.2	A problem was resolved that caused the Boss phone to reboot in a Boss-Admin situation where phones were running version 5.4.X software.
General	VOIP-111603	5.4.0	If the top of the route list's transport is UDP, phone now checks if it set by default or from the record route header and uses the same default transport mechanism for acknowledgment.
General	VOIP-111357	5.4.1	If the phone receives a 407 from the BYE message, it now responds adding the proxy-authorization header with credentials.
General	VOIP-110472	5.2.1	VVX Keys are now optimized for responsiveness, speed, and stability even after a long period of uptime until phone is rebooted
General	VOIP-110017	5.4.1	DHCP stability issues on the VVX 310 phone have been resolved.

Resolved Issues in UC Software 5.5.0

Category	Issue No.	Release	Description
General	VOIP-113036		Several security configuration parameters were added to configure the phone to prompt users for SIP credentials at login. These credentials are then used for all SIP authorization. These parameters include: `prov.login.useProvAuth`, `voIpProt.SIP.specialEvent.checkSync.downloadCallList`, `prov.login.userId.encodingMode`, and `prov.login.password.encodingMode`.
Interoperability BroadSoft	VOIP-113154		Users can now search the BroadSoft directory using either the first name or the last name.
Interoperability BroadSoft	VOIP-109598	5.4.0	The star (*) and pound (#) symbols now display in the search field in the BroadSoft Directory.
Interoperability GENBAND	VOIP-115465	5.4.0	When saving a GENBAND Global Address Book (GAB) to the phone contact list, the contact's phone number is now saved correctly.
Interoperability GENBAND	VOIP-113314	5.4.1	A buddy's presence status is now updated on the Home screen when the parameter `voIpProt.SIP.presence.nortelShortMode` is set to True and the parameter `dir.local.serverFeatureControl.method` is set to GENBANDSOPI.
Interoperability GENBAND	VOIP-109599	5.4.1	On VVX phones, users can now watch buddies set in the GENBAND Personal Address Book when the parameter `feature.presence.enabled` is set to 1.
Interoperability Microsoft	VOIP-111382		Improvements have been made for Outlook calendar event synchronization.
Interoperability Microsoft	VOIP-113865	5.4.2	Stability issues in certain Lync or Skype for Business environments have been addressed.
Interoperability Microsoft	VOIP-111093	5.4.1	VVX Phones on Office 365 are now able to redial a number previously dialed using a Lync client pinned contact.
Interoperability Skype for Business	VOIP-115263	5.4.2	When you use the Boss-Admin for Skype for Business feature, only the Boss now gets the notification email regarding the admin's activity on Boss Number.
Microbrowser	VOIP-110527	5.3.1	The microbrowser now correctly displays the local time when the phone is set to the Lync profile.
Network	VOIP-108242		An issue was resolved that prevented the VVX phones from synchronizing after an interruption in Exchange connectivity.
Network	VOIP-111998		Enabling SSLv3 on the LDAP server and disabling SSLv3 on the phone no longer causes issues on the phone.

Resolved Issues in UC Software 5.5.0

Category	Issue No.	Release	Description
Network	VOIP-115990	4.0.8	A problem was resolved that stopped NAT keep-alive messages when the provisioning server applies a firmware upgrade.
Network	VOIP-113928	5.4.1	VVX phones with edge registrations using an Audio codes gateway now negotiate ICE correctly.
Registration	VOIP-115741	5.4.2	An issue was resolved that caused the phone to unregister when ending an invalid URI call in a Lync environment.
Registration	VOIP-113016	5.4.1	When unregistered or powered off, the phone now correctly sends a notification event to unsubscribe from presence. When it registers, the phone now sends a notification event to subscribe for presence.
Reporting	VOIP-112424	5.4.0	The phone now displays the correct NLR and MOSQ when the phone generates DTMF in the generated VQMoN reports.
Reporting	VOIP-111764	5.4.1	Accurate overall Mean Opinion Scores (MOSs) are now created when there are several Synchronization Source range allocation (SSRC) changes that could occur as a result of codec changes. The phone will trigger a VQMon report as soon as an SSRC change is reported by DSP.
Reporting	VOIP-110308	5.4.0	The phone now displays the correct NLR and MOSQ when the phone generates DTMF in the generated VQMoN reports.
Security	VOIP-115481		The password used to authenticate to the GENBAND server (set with parameter `dir.corp.alt.password`) is now hidden in the configuration export.
Security	VOIP-110213		Multiple Denial of Service vulnerabilities in OpenSSL have been resolved.
Security	VOIP-109345		You can now use the parameter `dir.local.passwordProtected` to specify whether users are prompted for an Admin or User password when adding, editing, or deleting contacts from the Contact Directory.
Security	VOIP-113463		Parameters `sec.TLS.profile.webServer.cipherSuiteDefault` and `sec.TLS.webServer.cipherList` were added to allow configuration of the cipher suites for the web server profile.
Software Update	VOIP-113590		The user now remains signed in on the phone after upgrading the software.
Software Update	VOIP-113298	5.4.2	A problem was resolved that caused a problem when upgrading the phone from the Polycom hosted server on the phone's web interface.
User Interface	VOIP-115821		Parameter lcl.ml.lang.japanese.font.enabled was added to enable Japanese Kanji characters to display correctly.

Resolved Issues in UC Software 5.5.0

Category	Issue No.	Release	Description
User Interface	VOIP-115524		When you enter the special character code &#201 in the web interface, it now gets replaced with the Unicode replacement character. The font used on Polycom VVX 3.x.x, 2.x.x, and 1.x.x phones does not support special characters with numbers greater than 255, so these phones replace the special characters with a blank space.
User Interface	VOIP-115523		On the Polycom VVX Expansion Module, the labels are now correctly split when Text Alignment is set to Right or None.
User Interface	VOIP-114955		Call Control management soft keys now appear when initiating a conference call on the VVX phones when URL dialing is disabled.
User Interface	VOIP-114845		Labels now split correctly in the user interface when alignment is set to Right or None. When text alignment is set to Left, labels may not correctly split.
User Interface	VOIP-112884		When you enable the soft key using Enhanced Feature Key (EFK), the user can access and launch the browser by pressing the soft key configured for the micro browser.
User Interface	VOIP-112421		After paging, the user's presence now returns to Available as expected.
User Interface	VOIP-115653	5.4.2	Polycom VVX 601 phones now display the correct time for GMT -6 and Eastern time zones.
User Interface	VOIP-99845	5.4.0	A problem with the display of the Simultaneous Ring Personal field label has been resolved.
User Interface	VOIP-114143	5.3.0	An issue has been resolved that caused the phone to display "Unknown" when the caller's number is available.
User Interface	VOIP-102718	5.3.0	The VVX phone now consistently displays the Encoding soft key on the Single Sign In menu.
User Interface	VOIP-113916	5.4.0	The UC-One presence status and message now display correctly when the VVX presence status is updated.
User Interface	VOIP-109649	5.4.0	The VVX 600 phone now displays the Park soft key when the phone has a single registered line with one call per line configured.
VQMon	VOIP-110308		The phone now displays the correct NLR and MOSQ when the phone generates DTMF in the generated VQMoN reports.
Web Interface	VOIP-115031		Enabling or disabling the phone's web server no longer causes it to switch to using the DNS static cache entry instead of using a network DNS query to resolve the provisioning server FQDN.
Web Interface	VOIP-112342	5.4.1	The phone's Web Configuration Utility now correctly displays the selected Time Zone field.

Get Help

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

For information about Polycom partner solutions, see Polycom Global Strategic Partner Solutions.

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