Poly UC Software 5.9.5

Applies to Polycom VVX Business Media Phones, Poly VVX Business IP Phones, and Polycom SoundStructure VoIP Interface Phones

Poly announces the release of Polycom Unified Communications (UC) Software, version 5.9.5. This document provides the latest information about this release.

Contents

What's New

Poly Unified Communications (UC) Software 5.9.5 is a release for OpenSIP and Skype for Business deployments. These release notes provide important information on software updates, phone features, and known issues.

Important Note About Use of UC Software in Skype for Business Deployments

Customers in Skype for Business deployments should only use software releases that Microsoft has qualified or maintenance releases built on a qualified release.

Microsoft qualification may fall behind Poly Generally Available dates for UC Software, hence we request customers to check whether a release has been qualified before deploying new software. For all latest Microsoft qualified releases, visit the Polycom UC Software for Skype for Business Deployments page for a list of qualified releases.

New Features and Enhancements

Poly UC Software 5.9.5 includes the following new feature and enhancements:

- Expanded Interoperability Testing
- Dual-Tone Multi-Frequency Tones Parameter
- Session Traversal Utilities for NAT (STUN) Parameters
- FIPS 140-2 Compliance Support

Expanded Interoperability Testing

Poly expanded its interoperability testing of the Poly Unified Communications (UC) Software 5.9.5 to include:

- Avaya Aura Session Manager
- Avaya Aura Communication Manager

Note: Poly supports SIP telephony feature interoperability with Avaya Aura Communication Manager and Avaya Aura Session Manager following published standards including IETF Requests for Comments (RFCs) and Internet drafts last validated by Avaya in March 2019 contingent on Avaya allowing SIP compliant 3rd-party endpoints to register and interoperate with their call platforms.

- Aura Application Server 5300
- Avaya IP Phone 1140E
- Cisco Unified Communications Manager (CUCM)

Note: Poly supports SIP telephony feature interoperability with Cisco Unified Communications Manager (CUCM) following published standards including IETC RFCs and contingent on CUCM allowing SIP compliant 3rd-party endpoints to register and interoperate with their call platforms.

- Poly headsets (Blackwire and Encore Pro families)

See Products Tested with This Release.

Dual-Tone Multi-Frequency Tones Parameter

Poly UC Software 5.9.5 introduces a new parameter for Dual-Tone Multi-Frequency (DTMF) to publish the DTMF frequency on the Opus codec.

`tone.dtmf.rfc2833.SupportOpusClockRate`

1 – (default) Publishes the Telephone-event DTMF frequency as 48000 Hz along with 8000 Hz on Opus codec.

0 - Publishes the Telephone-event DTMF frequency as 8000 Hz on Opus codec.

Session Traversal Utilities for NAT (STUN) Parameters

Poly UC Software 5.9.5 supports Session Traversal Utilities for NAT (STUN), a network protocol used in NAT traversal for real-time IP communications, such as voice, video, and messaging.

This section lists parameters that configure Simple Traversal of UDP though NAT (STUN).

`nat.refresh.interval`

Set the time interval for the phone to send STUN binding indications to keep the NAT port open and the phone reachable.

30 seconds (default) - The phone sends STUN binding indications for every 30 seconds to keep the NAT port open and the phone reachable.

0 second - Disable STUN Binding indication to refresh NAT bindings.

3600 seconds

`nat.device.pollInterval`

Set the time interval for the phone to send STUN binding request to the STUN server to detect whether NAT device is rebooted.

60 seconds (default) - The phone sends the STUN binding requests to the STUN server for every 60 seconds. If NAT IP address or the port details in the STUN binding response don’t match with the previous binding response, the phone automatically restarts.

0 - The phone doesn’t check whether NAT device is rebooted. If NAT device is rebooted and the NAT IP address or the port is changed, the phone doesn’t receive any incoming messages as the IP address and port details published in SIP register message don’t match. You need to restart the phone manually to make the changes effective. Poly recommends not to set the value as 0 second.

900 seconds

FIPS 140-2 Compliance Support

The Federal Information Processing Standard (FIPS 140-2) compliance is a cryptographic function. This feature is now supported on VVX 501 and 601 business media phones.

The following parameter enables or disables the FIPS 140-2 feature.

`device.sec.TLS.FIPS.enabled`

0 (default) - Does not allow the phone to use the FIPS-compliant cryptography feature.

1 - Allows the phone to use the FIPS-compliant cryptography feature.

Configuration File Enhancements

The following list provides configuration file enhancements that include new or changed parameters for the Poly UC Software 5.9.5 release. For more information on using configuration parameters to enable or disable features, see the latest *Polycom UC Software Administrator Guide* for your release, available at Latest Polycom UC Software Release**.**

`reg.x.secureTransportRequiresSrtp`

0 (default) – Doesn't allow the phone to dynamically overwrite the configured values of `reg.x.srtp.offer` parameter and `reg.x.srtp.require` parameter based on the NAPTR response for per line registration.

1 – Allows the phone to dynamically overwrite the configured values of `reg.x.srtp.offer` parameter and `reg.x.srtp.require` parameter based on the NAPTR response for per line registration to enable SRTP only.

`dns.cache.dynamicRestore.enable`

1 – Allows the phone to restore the expired cache entries to a specified TTL when the DNS server isn't reachable.

0 (default) – Doesn't allow the phone to restore the expired cache entries to a specified TTL when the DNS server isn't reachable.

`dns.cache.dynamicRestore.ttl`

Specify a TTL value to restore the expired cache entries when the DNS server isn't reachable.

120 (default)

90 to 600 seconds.

Security Updates

See the Polycom Security Center for information about known and resolved security vulnerabilities.

Installation

Consider the following information when installing or updating to Poly UC Software 5.9.5:

Note:	Poly recommends setting the Contrast value to 25 on the VVX 3XX phone when you upgrade to Poly UC Software 5.9.5.

- You must use BToE 3.8.0 or later with UC Software 5.9.5. Poly recommends that you upgrade existing BToE and PDC desktop applications to the latest version starting with Poly UC Software 5.9.5. If you update the BToE or PDC application without updating to UC Software 5.9.5 or vice versa, the applications don't work with Polycom phones.

  For more information, see *Updating to the Latest Versions of the Polycom Better Together over Ethernet (BToE)* and *Polycom Desktop Connector (PDC) Desktop Applications (EA 318).*
- Before updating your VVX 1500 business media phone to UC Software 5.9.5, make sure you first update the phone to BootBlock 3.0.4.

For more information see *Upgrading the Polycom VVX 1500 Business Media Phone to UC Software 5.2.0.*

Download the Distribution Files

To download UC Software 5.9.5, 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. This enables 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 Error! Reference source not found. table. If you're provisioning your phones centrally using configuration files, download the corresponding resource file and extract the configuration files to the provisioning server. Make sure that you maintain the folder hierarchy in the ZIP file.

The current build ID for the sip.ld and resource files is **UCS 5.9.5.0614 rts10.**

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 business media phones.	No	Yes
3111-40450-001.sip.ld	SIP application executable for VVX 201 business media phones.	No	Yes
3111-46135-002.sip.ld	SIP application executable for VVX 300 business media phones.	No	Yes
3111-48300-001.sip.ld	SIP application executable for VVX 301 business media phones.	No	Yes
3111-46161-001.sip.ld	SIP application executable for VVX 310 business media phones.	No	Yes
3111-48350-001.sip.ld	SIP application executable for VVX 311 business media phones.	No	Yes
3111-46157-002.sip.ld	SIP application executable for VVX 400 business media phones.	No	Yes
3111-48400-001.sip.ld	SIP application executable for VVX 401 business media phones.	No	Yes

Distributed Files	*File Purpose and Application*	*Combined*	*Split*
3111-46162-001.sip.ld	SIP application executable for VVX 410 business media phones.	No	Yes
3111-48450-001.sip.ld	SIP application executable for VVX 411 business media phones.	No	Yes
3111-44500-001.sip.ld	SIP application executable for VVX 500 business media phones.	No	Yes
3111-48500-001.sip	SIP application executable for VVX 501 business media phones.	No	Yes
3111-44600-001.sip.ld	SIP application executable for VVX 600 business media phones.	No	Yes
3111-48600-001.sip	SIP application executable for VVX 601 business media phones.	No	Yes
2345-17960-001.sip.ld	SIP application executable for VVX 1500 business media phones.	No	Yes
3111-48810-001.sip.ld	SIP application executable for VVX 150	No	Yes
3111-48820-001.sip.ld	SIP application executable for VVX 250	No	Yes
3111-48830-001.sip.ld	SIP application executable for VVX 350	No	Yes
3111-48840-001.sip.ld	SIP application executable for VVX 450	No	Yes
3111-33215-001.sip.ld	SIP application executable for SoundStructure VoIP Interface phones.	No	Yes
3111-17823-001.dect.ld	SIP application executable for VVX D60 Wireless Handset and Base Station.	No	Yes
sip.ld	Concatenated SIP application executable.	Yes	No
dect.ver	Text file detailing build-identifications for the VVX D60 handset.	Yes	Yes
sip.ver	Text file detailing build-identifications for the release.	Yes	Yes
000000000000.cfg	Master configuration template file.	Yes	Yes
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.	Yes	Yes
applications.cfg	Configuration parameters for microbrowser and browser applications.	Yes	Yes
device.cfg	Configuration parameters for basic device configuration.	Yes	Yes
features.cfg	Configuration parameters for telephony features.	Yes	Yes

Distributed Files	*File Purpose and Application*	*Combined*	*Split*
firewall-nat.cfg	Contains configuration parameters for telephony features.	Yes	Yes
H323.cfg	Configuration parameters for the H.323 signaling protocol.	Yes	Yes
lync.cfg	Contains Lync/Skype for Business-specific configuration parameters.	Yes	Yes
pstn.cfg	Contains parameters for Public Switched Telephone Network (PSTN) use.	Yes	Yes
reg-advanced.cfg	Contains configuration parameters for the line and call registration and advanced phone feature settings.	Yes	Yes
reg-basic.cfg	Configuration parameters for line and call registration and basic phone settings.	Yes	Yes
region.cfg	Configuration parameters for regional and localization settings such as time and date and language.	Yes	Yes
sip-basic.cfg	Configuration parameters for the VoIP server and softswitch registration.	Yes	Yes
sip-interop.cfg	Configuration parameters for the VoIP server, Softswitch registration, and interoperability configuration.	Yes	Yes
site.cfg	Configuration parameters set for each site.	Yes	Yes
video.cfg	Configuration parameters for video connectivity.	Yes	Yes
video-integration.cfg	Configuration parameters for Polycom SoundStation IP 7000 conference phone and Polycom HDX system integration.	Yes	Yes
Welcome.wav	Startup welcome sound effect.	Yes	Yes
LoudRing.wav	Sample loud ringer sound effect.	Yes	Yes
Polycom-hold.wav	Sample ringer sound effect.	Yes	Yes
Warble.wav	Sample ringer sound effect.	Yes	Yes
polycomConfig.xsd	Master configuration file that contains the parameters and its values.	Yes	Yes

Version History

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

Version History

Release	*Release Date*	*Features*
5.9.5	January 2020	This release includes important field fixes and support for the following features: • Expanded Interoperability Testing • Introduction of new parameters to Sesstion Traversal Utilities for NAT • New parameter for Dual-Tone Multi-Frequency Tones for OPUS codec
5.9.4	September 2019	This release includes important field fixes and support for the following features: • Third-Party Application ID Implementation on Skype for Business Phones • Sign In Remotely Using Web Sign-In for Skype for Business
6.1.0	August 2019	This release includes important field fixes and support for following features: • Reverse Name Lookup for OpenSIP • Call Park Reminder Tone • Microsoft Exchange Calendar using OAuth support • Enhanced IPv6 ICMP Management • Session Management on Web Configuration Utility • Macro for Enhanced Feature Keys Functional Improvements • Support for Plantronics Headsets • Software Upgrade Resiliency • STUN Parameters • New Language support • Polycom Acoustic Fence Support • Data Protection Menu • Call and Hold Timer Configuration • DTMF Improvements for Opus Codec
5.9.3	July 2019	This release includes the following new and enhancement features: • DHCP IP Address Cache • TLS Support for BToE • Polycom Cloud Connector • Enhancement to Wi-Fi Settings
5.8.4	May 2019	This release includes enhancements to Wi-Fi settings and other important security fixes.

Release	Release Date	Features
6.0.0	April 2019	This release includes important field fixes and support for the following features: • Guest Soft Key Customization • Plantronics Headset Settings • TLS Support for BToE • Improved Flexible Line Key Assignments for Static BLFs and Enhanced Feature Keys • Font Size Customization • Enhanced Feature Keys – BLF Support • Advanced uaCSTA • VVX Pagination • Key System Emulation • DHCP IP Address Cache
5.9.2	March 2019	This release includes important field fixes.
5.6.5	March 2019	This release includes important field fixes.
5.8.3	January 2019	This release includes important field fixes.
5.9.1	January 2019	This release includes enhancement to VLAN ID and Wi-Fi dongle support.
5.9.0	December 2018	This release includes important field fixes and support for the following features: • Session Traversal Utilities for NAT (STUN) • Device Analytics Support for PDMS-SP • Multilevel Precedence and Preemption (MLPP) for Assured Services - Session Initiation Protocol (AS-SIP) • Support for H.264 Packetization Mode • Enhanced Busy Lamp Field (BLF) • Busy Lamp Field Hold Alerting • Busy Lamp Field (BLF) Spontaneous Call Appearance on Per BLF Basis • Enhanced Feature Key Macro Actions • Retrieve Logs from Support Information Package Page in the Web Configuration Utility • Simple Certificate Enrollment Protocol • Privacy for Call Logs and Contacts • Enhancement to Wireless Network Connectivity • Call Hold Timer • GZIP Encoding of SIP INFO Messages • Enhanced Boss-Admin for VVX phones • Web Sign In for Skype for Business On-Premise Deployment

Products Tested with This Release

Poly products are tested extensively with a wide range of products. You can view a list of the products that have been tested for compatibility with this release.

Poly strives to support any system that is standards-compliant and investigates reports of Poly systems that are not interoperable with other vendor systems. Note that the following list is not a complete inventory of compatible equipment, but the products that have been tested with this release.

Note: Poly recommends that you upgrade all of your Poly systems with the latest software versions. Any compatibility issues may already have been addressed by software updates. See the Current Polycom Interoperability Matrix.

Products Tested with this Project

Product	***Tested Versions***
Server	
CUCM	12.0.1
Avaya Aura System Manager	8.1.0.0.733078
Avaya Aura Session Manager	8.1.0.0.810007
Avaya Aura Communications Manager	8.1.0.0.890
Avaya Aura Media Server	8.0.0.169
Avaya AS5300	15.1.15.1
Endpoint	
Poly Trio 8300	5.9.2.7527
Poly Trio 8500	5.9.2.7527
Poly Trio 8800	5.9.2.7527
Polycom VVX 150	5.9.5.0614
Polycom VVX 411	5.9.5.0614
Polycom VVX 450	5.9.5.0614
Polycom VVX 601	5.9.5.0614
Avaya IP Phone 1140E	SIP1140e.04.04.38.00
Headset Model/Name	
Model B4210/Voyager 4210	
B7225/Blackwire 7225	

Product	*Tested Versions*
BW3320/Blackwire 3320	
C5200/Blackwire 5200	
C5210/Blackwire 5210	
HW510/Encore Pro 510	
HW520/Encore Pro 520	

Resolved Issues

The following table lists the resolved issues in UC Software 5.9.5.

Resolved Issues in UC Software 5.9.5

Category	*Issue No.*	*Description*
General	EN-142634	The contrast ratio of the selected menu on the VVX 3XX phone is improved.

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 and 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	EN-159441	5.9.5	When the VVX **feature.callPark** option is enabled, the screen flashes and goes back to a non-call state, after a random number call is parked, then later selected to park again.	None.
Audio	EN-113674	5.9.0	When you place a call from a VVX D60 handset to a remote phone, there may be choppy audio.	None.
Device Management	EN-137080	5.9.2	VVX phones show a “Cannot charge device” warning message for USB when no USB device is connected.	None.

Category	Issue ID	Release	Description	Workaround
Device Management	EN-146754	5.9.4	When PLT c3200 model headset is used in BToE mode and VVX phone is in playback call, the call can't be mute from Skype for Business client.	Mute from the headset
General	EN-108973	5.9.0	When you execute a packet capture command from the cloud, libPcap occasionally doesn't start the packet capture.	Resend the start packet capture command.
Network	EN-146412	5.9.4	When payload format isn't received in SDP of 200 OK and 183, phone sends DynamicRTP-Type-127 for DTMF	Configure `tone.dtmf.rfc2833Control` to "0"

Constraints and Limitations

This section identifies known constraints and limitations of this release as it interoperates with other platforms or devices.

Successful Connection in an SRTP Environment

For endpoints to successfully connect in a Securte Real-time Transport Protocol (SRTP)-enabled environment, endpoint must be able to negotiate an SRTP offer. That means, endpoints must have at a minimum Offer SRTP set to Yes. If the endpoints are not so configured, blind transfers will fail.

Some Call Flow Failures with Avaya Aura 7.1

After upgrading to Avaya Aura 7.1, failures in some call flows occur when connected SIP devices change URIs with "sips:" indicators to "sip:" on an intermediate route and via headers, which is not allowed as per SIP RFC 3261.

Headset Volume Change and VVX Phone Display not in Synch

The volume up and down settings on some headsets (Blackwire 5200 series and Encore Pro HW500 series) are not always synchronize with the VVX phones LCD display.

Poly VVX Phone Status Always Green

When in an Avaya Aura Session Manage 5300 environment, Poly VVX phones registered as buddy to an Avaya 1140 endpoint always show green status when the Avaya 1140 endpoint is in a call.

Language Support

The VVX phone user interface 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

Previous Software Releases

This section describes the new features and enhancements to previous UC Software 5.9.*x* releases.

What's New in UC Software 5.9.4

Polycom Unified Communications (UC) Software 5.9.4 is a release for OpenSIP and Skype for Business deployments. These release notes provide important information on software updates, phone features, and known issues.

New Features and Enhancements

Polycom UC Software 5.9.4 includes the following new feature and enhancements:

- Third-Party Application ID Implementation on Skype for Business Phones

- Sign In Remotely Using Web Sign-In for Skype for Business

Third-Party Application ID Implementation on Skype for Business Phones

Microsoft announced the end of support for the existing Azure application ID currently used by third-party device vendors that use the OAuth 2.0 authorization protocol. To retain authentication to Microsoft services and the ability to sign in to Skype for Business, you must update the firmware to implement a third-party application ID on all impacted certified Skype for Business phones by January 15, 2020.

The following deployments include Skype for Business IP phones certified under Microsoft's 3rd Party IP Phones (3PIP):

- Skype for Business Online
- Skype for Business On-Premises Hybrid (with modern authentication deployed)

Polycom adheres to the Microsoft announcement and has implemented the third-party application ID.

To implement the third-party application on Skype for Business phones:

1. Go to the consent URL and sign in with tenant administrator credentials.
2. Click **Accept** to grant permission to allow the Poly ID application to access the required resource's permissions.

   Note: You must go to the consent URL and accept permission before you upgrade to the new firmware.
3. Upgrade the impacted phones to the new firmware version.

System Constraints and Limitations

This section provides information on contraints and limitations when installing UCS 5.9.4 in your environment.

- Authentication of Microsoft Exchange 2013 Services
- Downgrade Behavior from Polycom UC Software 5.9.4

Authentication of Microsoft Exchange 2013 Services

Polycom UC Software 5.9.4 contains embedded Poly application ID code, which can cause the authentication for Exchange services with Hybrid Modern Authentication (HMA) enabled to fail with OAuth 2.0 on VVX phones.

Downgrade Behavior from Polycom UC Software 5.9.4

Downgrading from UC Software 5.9.4 to any earlier UC Software version may result in sign-in failure. To continue using Skype for Business on VVX phone successfully, re-sign in to Skype for Business.

Sign In Remotely Using Web Sign-In for Skype for Business

You can sign in to Skype for Business remotely using the system web interface.

To sign in remotely using the web sign-in for Skype for Business:

1. Enter your phone's IP address into a web browser on your computer.
2. Select **Admin/User** as the login type, enter the admin/user password (the default is `456/123`), and click **Submit**.
3. Select **Settings > Skype for Business Sign In**.
4. Select **Web Sign in** from **Authentication Type**.
5. Select **Sign In**.
   A URL and a sign-in code display.
6. Enter the URL into the web browser on your computer.
7. Enter the sign-in code and select **Continue**.
8. Enter your Skype for Business login information.
   A confirmation message displays when the phone successfully signs in to Skype for Business.

Configuration File Enhancements

The following table list provides configuration file enhancements that include new or changed parameters for the Polycom UC Software 5.9.4 release. For more information on using configuration parameters to enable or disable features, see the latest *Polycom UC Software Administrator Guide* for your release, available at Latest Polycom UC Software Release.

`feature.remoteWebSignIn.enabled`

1 (default) – Enables the web sign-in option on the system web interface

0 – Disables the web sign-in option on the system web interface

`voIpProt.SIP.forkedRespRecommendedCseq.enable`

1 (default) - Generates the RFC compliance CSeq number

0 - Generates the call-specific CSeq number

`lync.E911.notificationUri.expansion.enabled`

0 (default) - Enables the expansion of distribution lists received as part of a notification URI

1 – Disables the expansion of distribution lists received as part of a notification URI

`lync.E911.notificationUri.maxUrls`

30 (default) - Sets the limit for the number of URLs in the notification URI.

Resolved Issues

The following table lists the resolved issues in UC Software 5.9.4.

Resolved Issues in UC Software 5.9.4

Category	*Issue No.*	*Description*
Application	EN-144175	In VVX phones, buffer overflow occurs in BToE messages.
Audio	EN-104885	During a Skype call, the audio is lost from the far end when the network firewall uses port overlapping.
Calendaring	EN-144252	VVX phones fail to join a conference when the meeting subject contains special characters.
Call Management	EN-140715	When you connect a VVX phone to the HELD server, the phone must respond with a valid certificate after receiving the request message from the server.
Calling	EN-143749	When a local conference is initiated on a Shared Call Appearance (SCA) phone, a remote SCA phone shows missed calls.
Calling	EN-135932	When dialing to an anonymous user from a call list, the phone sends the acknowledgement with spaces in the SIP request-URI, causing network congestion.
Calling	EN-142967	After a blind transfer, calls from a VVX D60 handset fail and users can’t resume the call.
Calling	EN-123763	When a call is initiated from a VVX phone, the phone sends DTMF using dynamic payload type 127, even if the far end phone doesn't have DTMF payload support.
Calling	EN-140387	VVX phones delay a call transfer by 20 seconds when Microsoft direct team routing setup is used for O365.
Calling	EN-142635	VVX phones fail to place a call on hold when using a call recording service.
Calling	EN-139948	VVX phones don’t send notifications to a Distribution List (DL) group during E911 calls.
Calling	EN-146076	VVX phones increment CSeq numbers per transaction, causing phones to receive `500` as a response for `BYE`.
Calling	EN-146251 EN-137437	VVX phones don't show incoming busy lamp field call alerts when you set the `reg.1.callsPerLineKey` parameter to 1 and the other call is in an offering state.
Certificate Management	EN-134009	Sometimes, SCEP fails to install the intermediate certificate on the phone.
Device Management	EN-143719	Sometimes, VVX phones fail to complete a consultative transfer using the **Transfer** hard key when a call is answered through the speaker or a headset.

Category	*Issue No.*	*Description*
Device Management	EN-139755	Sometimes, VVX phones fail to complete a consultative transfer.
Device Management	EN-136149	When a USB Wi-Fi wireless network adapter is connected to a VVX phone, the phone displays an "Unsupported USB device attached" error message.
Device Management	EN-137493	When you pair a VVX phone with VVX D60 handset and click either the **DND** or **Call Forward** menus on the handset, a "Not supported" message displays.
Device Management	EN-138253	VVX phones fail to transmit audio from a headset if joined to a call through an external Skype for Business meeting in PC Audio/Playback mode.
General	EN-138982	VVX phones crash after receiving an SDP attribute as "a=recvonly" in the initial OFFER.
General	EN-137073	The `$FServerACDAgentAvailable$` macro softkey doesn't work when the agent is on call.
General	EN-139956	Sometimes, the calendaring service fails while using OAuth with proxy.
General	EN-136492 EN-145579	Secure Real-Time Transport Protocol (SRTP) configured parameter values don't display on the VVX phone local interface and system web interface.
General	EN-122349	When VVX phones receive a call from a saved contact, the contact name doesn't display.
General	EN-135594	VVX phones don't allow you to set the ring count value to 1 when server-based call forward is enabled.
General	EN-140942	VVX phones continuously play DTMF tones when you press the customized EFK softkey and answer a call through the speaker.
General	EN-143760	VVX phones don't use an external conference server for shared lines if a 202 response is received instead of a 200 ok response for a line-seize SUBSRIBE message.
General	EN-141219	VVX phones fail to send DTMF digits through SIP-info messages configured in the EFK soft key.
General	EN-149494	VVX phones fail to perform hold operation after installing SFB July 2019 Cumulative Server Update.
General	EN-147430	VVX phones on UCS 5.9.3 fails to onboard to PDMS-SP
Logs	EN-137310	BToE logs keep increasing in size until the application is restarted.

Category	*Issue No.*	*Description*
Network	EN-138180	VVX phones frequently fail if the DNS server changes the order of DNS records for the registrar or outbound proxy server.
Network	EN-140698	VVX phones fail to add SDP while generating Session Refresh INVITES.
Network	EN-136432	VVX phones send the local IP in SDP connections when connected to a 3CX stun server.
Network	EN-145159	When TLS is turned on VVX phone goes to unregistered state for a short period and then recovers back to registered state.
Security	EN-134733	When TLS-DSK authentication is disabled, VVX phones delay displaying Address Book Service (ABS) search results.
User Interface	EN-138501	After adding a localized Russian soft key label for “Parking” to VVX phones, the touchscreen and hard keys don’t respond properly.
User Interface	EN-142092	Sometimes, VVX 601 business media phone screens freeze and fail to respond.
User Interface	EN-140021	Sometimes, VVX phone using BroadSoft Receptionist thin client don’t work properly.
Video	EN-139753	VVX phones fail to publish video capabilities in Presence status.

What’s New in Polycom UC Software 5.9.3

Polycom Unified Communications (UC) Software 5.9.3 is a release for OpenSIP and Skype for Business deployments. These release notes provide important information on software updates, phone features, and known issues.

New Features and Enhancements

Polycom UC Software 5.9.3 includes the following new feature and enhancements:

- DHCP IP Address Cache
- TLS Support for BToE
- Polycom Cloud Connector
- Enhancement to Wi-Fi Settings

DHCP IP Address Cache

Polycom UC Software supports the Dynamic Host Configuration Protocol (DHCP) IP address cache to retain IP addresses on VVX phones. When you enable the IP address caching feature, there aren't any service interruptions even if the IP address lease time expires and the DHCP server doesn't respond.

TLS Support for BToE

Polycom BToE application supports the Transport Layer Security (TLS) protocol to authenticate VVX phones with the following features:

- Polycom UC Software uses TLS protocol v1.2 to authenticate VVX phones using BToE application v4.0.0.
- The TLS protocol takes precedence over the SSH protocol.
- If TLS connection fails between the VVX phone and the Polycom BToE Connector application, then the connection falls back to SSH.

Polycom Cloud Connector

Polycom UC Software introduces the Polycom Cloud Connector to send device analytics to Polycom Cloud Services. You must enable the `device.da.enabled` parameter to send device analytics to Polycom Cloud Services.

Enhancement to Wi-Fi Settings

Polycom VVX phones now display Wi-Fi settings only when you connect a Wi-Fi dongle. Wi-Fi dongle doesn't support on Expansion modules USB port.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for the Polycom UC Software 5.9.3 release. For more information on using configuration parameters to enable or disable features, see the latest *Polycom UC Software Administrator Guide* for your release, available at Latest Polycom UC Software Release**.**

Configuration File Enhancements for UC Software 5.9.3

Parameter	*Permitted Values*	*Restart or Reboot*
`device.net.cachedIPAddress`	0 (default) – IP addresses aren't cached. 1 – If the phone doesn't receive a DHCP response, the phone uses the last assigned IP address, provided one is cached already. A DHCP discover message is retried every `device.net.cachedIPAddressRetryTime` second.	No

Parameter	*Permitted Values*	*Restart or Reboot*
`device.net.cachedIPAddressRetryTime`	Specify the time in seconds to send new DHCP to discover messages when using the cached IP address. This parameter is only applicable when `device.net.cachedIPAddress` is enabled. 3600 (default) Seconds 300 – 7200 Seconds	No
`feature.pcc.enabled`	0 (default) - Disable the Polycom Cloud Connector. 1 - Enable the Polycom Cloud Connector.	Yes
`pcc.url`	Set the URL for the Polycom Cloud Connector interface. https://api-global.plcm.cloud/globaldirectory (default) 0 - 256	Yes
`pcc.accountCode`	Enter the Polycom Cloud Connector account code to connect your device with a Polycom Cloud Services account. Null (default) 0 - 256	Yes
`da.organizationID`	Define the organization ID of the device. Null (default) 0 – 256	Yes
`da.roomId`	Define the room ID of the device. Empty (default) 0 – 256	Yes
`da.siteId`	Define the site ID of the device. Null (default) 0 – 256	Yes
`voIpProt.SIP.ignoreEntityHost`	0 (default) – Doesn't ignore the host part of the entity received in the XML body of NOTIFY for a dialog event. 1 - Ignores the host part of the entity received in the XML body of NOTIFY for a dialog event.	No
`voIpProt.H323.p2pURLDialingThroughGK`	0 (default) - VVX phones don't route the H.323 URL dialing calls through the gatekeeper. 1 - VVX phones route the H.323 URL dialing calls through the gatekeeper.	No

Parameter	*Permitted Values*	*Restart or Reboot*
`up.offHookSpeedDialShortcut.enable`	1 (default) - VVX phones display the speed dial shortcut for one or two digits followed by # in the off-hook state. 0 – VVX phones don't display the speed dial shortcut for one or two digits followed by # in the off-hook state.	No
`feature.wifi.visibilityinmenus.enable`	1 (default) – Enables the Wi-Fi settings on VVX phones when the dongle is connected. 0 – Disables the Wi-Fi settings on VVX phones when the dongle is connected.	No

Resolved Issues

The following table lists the resolved issues in UC Software 5.9.3.

Resolved Issues in UC Software 5.9.3

Category	*Issue No.*	*Description*
Audio	EN-130973	VVX phones experience a one-way audio issue after a call transfer when you set the parameter `voIpProt.SIP.allowTransferOnProceeding` to 2.
Audio	EN-126223	Failed consultative call audio transfers to the speaker instead of headset or handset even if you disable handsfree mode.
Audio	EN-123326	There's an audio delay on the VVX phone when you perform a consultative call transfer.
Calling	EN-129003	You cannot answer incoming calls on VVX phones using the speaker, headset, or handset if you enable the Call Alert feature on a BLF monitored line.
Calling	EN-122145	VVX phones are unable to make H.323 URL calls to an external number.
Calling	EN-121474	VVX phones fail to dial the number immediately when dialed one or two digits followed by #.
Calling	EN-113951	You cannot make outgoing calls from a VVX business media phone when the monitored BLF alerts and `reg.1.callsPerLineKey` parameter is set to 1.
Calling	EN-133622	VVX phones reboot while processing the error response for conference calls.
Calling	EN-136762	VVX phone trims the remote number extension while using the BLF key for pickup calls.

Category	*Issue No.*	*Description*
Device Management	EN-115124	When you pair a D60 handset to a VVX phone and receive a call through Automatic Call Distribution (ACD), the first call rings on both the VVX phone and the D60 handset but the second call rings only on the VVX phone.
General	EN-132676	Polycom VVX 601 business media phones reboot while processing Presence state.
General	EN-128391 EN-126330	VVX phones lose dial plan information during inband refresh.
General	EN-128257	Busy Lamp Field (BLF) line calls don't work if the entity URI received in BLF NOTIFY isn't same as the REGISTRAR server configured.
General	EN-134063	VVX phones don't send BLF Subscribe after failover.
General	EN-137361	VVX phone application responses may become sluggish due to a memory leak.
General	EN-132896	When the 2nd BLF NOTIFY message is received, VVX phone dials an incorrect number using the legacy pickup method.
General	EN-142416	Sometimes, VVX 601 business media phone screen freezes and fails to respond.
Hardware	EN-133730	VVX 500 and 600 series business media phones fail to disable the Wi-Fi settings even if you detach Wi-Fi dongle.
Network	EN-129938	VVX phones can't receive an IP address if the voice and data Virtual Local Area Networks (VLANs) are configured as equal to VLAN IDs.
Network	EN-123349	VVX phones stop sending the Buddy Watch's subscribe after the phone fails to connect to a new proxy server.
Network	EN-130443	VVX phones use the parameter `device.sntp.serverName` even though the parameter `tcpIpApp.sntp.address.overrideDHCP` is enabled along with parameter `tcpIpApp.sntp.address`.
Network	EN-124142	VVX phone is signed in but doesn't receive calls for 10-15 mins when there is a network interruption.
Network	EN-132893	Sometimes VVX phones fail to receive the Location Information Server (LIS) from the server.
Security	EN-132606	Fixed security vulnerabilities.
Security	EN-131056	When SHA 1 cipher is configured with TLS v1.2, TLS-DSK authentication fails.

Category	*Issue No.*	*Description*
User Interface	EN-94424	When you change the BroadSoft directory to Enterprise directory strings, the corresponding translation of the Enterprise directory updates the support languages on the VVX D60 handset.

What's New in Polycom UC Software 5.9.2

Polycom Unified Communications (UC) Software 5.9.2 is a release for OpenSIP and Skype for Business deployments. These release notes provide important information on software updates, phone features, and known issues.

New Features and Enhancements

Polycom UC Software 5.9.2 includes the following enhancement.

Enhancement to OpenSSL Upgrade

OpenSSL is upgraded from version 1.0.2j to version 1.0.2q for Polycom VVX phones.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for the Polycom UC Software 5.9.2 release. For more information on using configuration parameters to enable or disable features, see the latest *Polycom UC Software Administrator Guide* for your release, available at Latest Polycom UC Software Release**.**

Configuration File Enhancements for UC Software 5.9.2

Parameter	**Permitted Values**	**Restart or Reboot**
`video.codecPref.XH264UC`	Sets the Microsoft H.264 UC video codec preference priority. 0 (default) 0 - 7	No
`video.codecPref.XUlpFecUC`	Sets the forward error correction (FEC) codec priority. 0 (default) 0 - 7	No

Resolved Issues

The following table lists the resolved issues in UC Software 5.9.2.

Resolved Issues in UC Software 5.9.2

Category	*Issue No.*	*Description*
Application	EN-125282	Polycom Skype for Business phone doesn't refresh the access token as per the expiry value that the exchange server receives.
Application	EN-124638	VVX 500 and 600 series business media phones don't display the **Favorites** option in the **Directories** menu when you enable the BroadSoft directory.
Application	EN-120423	DUT doesn't update presence information properly.
Application	EN-112876	VVX phones lose assigned keys after restarting.
Application	EN-110172	VVX phones don't display application endpoint status correctly when you add it as a favorite contact.
Application	EN-110042	VVX 500 and 600 series business media crashed while receiving a call from Redcom soft client with H264 along with video resolution configured to Auto mode.
Audio	EN-120425	Response group calls drop when phones receive an early reinvite from the response group server.
Audio	EN-123326	There's an audio delay on the VVX phone when you perform a consultative transfer.
Audio	EN-98474	When VVX phones are in BToE playback mode, the audio streamed to the far end is choppy.
Audio	EN-124862	During MOH in Silk negotiated call misalignment of host and DSP version cause static noise and one-way audio.
Calling	EN-115626	When you initiate a call when paired with Skype for Business in BToE mode, a termination changes delay on DUT.
Calling	EN-119506	VVX phones don't update missed call entries in the database list.
Calling	EN-117876	VVX phone is logging the hunt group incoming call as a missed call while answering and ending the call by other VVX phone sharing the same registered lines.
Calling	EN-122620	During an active call, when the phone hits the idle timeout threshold, the phone display dims for 2 seconds and then brightens again.
Calling	EN-128889	VVX business media phone stores voicemail URI as a default dial string, due to which the subsequent calls made using short press are stored in the voicemail without ringing on the far end.

Category	*Issue No.*	*Description*
Conference Management	EN-123340	Calls drop when the policy on the server sets `AllowIPAudio` to false.
Configuration	EN-120429	The **Directory** softkey displays in a Skype for Business profile even though the profile doesn't support it.
Configuration	EN-113385	All VVX phones accept the `up.accessibilityFeatures` parameter, but the parameter only applies to VVX 1500 business media phones.
Call Management	EN-114345	VVX 150 business IP phones don't disable the **Transfer** and **Hold** softkey during an active call when you set the parameter `softkey.feature.basicCallManagement.redundant` to 0.
Call Management	EN-123815	Call forward soft key displays even if the `feature.forward.enable` parameter is set to 0.
General	EN-124307	When VVX phones receive forwarded calls from an external number, the phones display the last four digits of the forwarding number received in the **Via** field.
General	EN-115808	Web Proxy Auto-Discovery (WPAD) fails to extract a domain from a specific URL.
General	EN-113522	The Field Help and parameter name are missing in the software upgrade section of WEB UI for VVX business media phones.
General	EN-117611	VVX phones change to the default language after a guest user signs in.
General	EN-114672	When refreshing the registration, VVX phones become unregistered when they try to directly query SRV instead of NAPTR due to a transport parameter in the contact header of the 200OK message.
General	EN-122406	After you complete a consultative transfer, VVX phones display the number "10" when you press the star (*) key when the phone is in an idle state.
General	EN-124590	VVX phone doesn't allow to change the hook switch mode to Plantronics from regular mode upon software upgrade to UCS 5.9.0 or onwards.
General	EN-120233	VVX phones display SIP uri instead of call queue name while receiving a call from Skype for Business call queue.
General	EN-119699	Users are intermittently signed out from VVX business media phone registered to Skype for Business server when using Web Configuration Utility to sign-in.

Category	*Issue No.*	*Description*
Logs	EN-123961	Fixed security vulnerabilities.
Messages	EN-117332	VVX phones restrict the dial number URI length to 64 characters.
Network	EN-120743	VVX phones fail to charge devices using the rear USB port.
Network	EN-120428	VVX phones have a network speed of 10 Mbit instead of 1Gbps for a PC port.
Network	EN-105978	VVX phones fail to receive the Location Information Server (LIS) from the server.
Network	EN-120000	VVX business media phones don't handle authentication error code correctly.
Network	EN-114075	VVX phones fail to make an outgoing call and receive a 488 error when you configure only SVC codec for the INVITE request.
Network	EN-126173	VVX phones failover and send two SUBSCRIBE messages for BLF after receiving the 503 response for the INVITE message.
Network	EN-123868	VVX phones don't send DNS refresh queries and REGISTER messages after a few failovers due to an improper DNS resolution sequence.
Network	EN-102890	Metaswitch registration lines go unregistered due to DNS resolution failure.
Network	EN-113273	VVX phones display the call log date as of February 7, 2036, while SNTP server responds with the value 0x00000000 or 0xFFFFFFFF.
Peripherals	EN-116813	VVX phones using USB headsets drop an outgoing call when they receive a simultaneous call and the other phone with simultaneous ringing answers the simultaneous call.
Provisioning	EN-120427	Server settings override local parameter values because of in-band provisioning.
Reports	EN-120165	Polycom devices registered with Skype for Business publish different QoE server results than those reported on the QoE site.
Reports	EN-117331	Polycom devices registered with Skype for Business sometimes report inaccurate QoE statistics.
Software Updates	EN-122718	Upgrading the VVX D60 handset software fails when you assign the base station with a xxx.xxx.xxx.xxx IP address.

What's New in Polycom UC Software 5.9.1

Polycom UC Software 5.9.1 includes the following enhancements:

- Enhancement to VLAN ID
- Wi-Fi Dongle Support

Enhancement to VLAN ID

After installing a new software package on VVX phones and changing the Virtual Local Area Networks (VLANs), the following are behavioral changes in VVX phones:

Updater

The phone doesn't reboot and starts the DHCP sequence on new VLAN to get the new IP address.

Application

- The phone doesn't restart and triggers the DHCP sequence on discovering a valid VLAN ID from an invalid VLAN ID to get the new IP address.
- Phone restarts and triggers the DHCP sequence on discovering any VLAN ID from an invalid VLAN ID to get the new IP address.

Wi-Fi Dongle Support

Polycom VVX phones now support Edimax USB Wi-Fi Dongle (EW-8711UTC) along with the Polycom Wi-Fi wireless network adapter.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for the Polycom UC Software 5.9.1 release. For more information on using configuration parameters to enable or disable features, see the latest *Polycom UC Software Administrator Guide* for your release, available on Latest Polycom UC Software Release**.**

Configuration File Enhancements for UC Software 5.9.1

Parameter	Permitted Values	Restart or Reboot
`video.profile.H264.packetizationMode0.payloadType`	Specifies the RTP payload format type for H264/90000 packetization Mode 0 MIME type. 109 (default) 96 to 127	Yes

Parameter	Permitted Values	Restart or Reboot
`video.profile.H264.payloadType`	Specifies the RTP payload format type for H264/90000 Mode 1 MIME type. 99 (default) 96 to 127	Yes

Resolved Issues

The following table lists the resolved issues in UC Software 5.9.1.

Resolved Issues in UC Software 5.9.1

Category	*Issue No.*	*Description*
General	EN-111322	Rest API management returns an incorrect value for the variable packetsExpected.
Network	EN-113463	VVX phones fail to get the IP address on correct VLAN after a network switch reboot.
Reports	EN-116609 EN-116846	Incorrect packet loss and Mean Opinion Score (MOS) is reported during SIP publish.

What's New in Polycom UC Software 5.9.0

Polycom Unified Communications (UC) Software 5.9.0 is a release for Open SIP and Skype for Business deployments. These release notes provide important information on software updates, phone features, and known issues.

Future Feature Releases for VVX Business Media Phones

With the end of the sale of some models of VVX business media phones, the following phone models will no longer receive new features:

- VVX 300/310
- VVX 400/410
- VVX 500
- VVX 600
- VVX 1500

As a result, UC Software 5.9.0 will be the last release to contain significant feature development for these models. Future major feature releases will not include images to support these models. In line with the Polycom End of Life policy, Polycom will continue to provide bug fixes in maintenance and patch releases on the UC Software 5.9.x stream. For details, see the Polycom End of Life Policy.

The following VVX phone models will continue to receive feature releases:

- VVX 101/201
- VVX 150
- VVX 250
- VVX 301/311
- VVX 350
- VVX 401/411
- VVX 450
- VVX 501
- VVX 601

Plantronics headsets are compatible with VVX phones. To know compatibility list, refer Plantronics Compatibility Guide.

Polycom UC Software 5.9.0 includes the following new features and enhancements:

- Session Traversal Utilities for NAT (STUN)
- Device Analytics Support for PDMS-SP
- Multilevel Precedence and Preemption (MLPP) for Assured Services - Session Initiation Protocol (AS-SIP) On Shared Lines
- Support for H.264 Packetization Mode 1 and H.264 Constrained Baseline Profile
- Enhanced Busy Lamp Field (BLF)
- Busy Lamp Field Hold Alerting
- Busy Lamp Field(BLF) Spontaneous Call Appearance on Per BLF Basis
- Enhanced Feature Key Macro Actions
- Retrieve Logs from Support Information Package Page in the Web Configuration Utility
- Simple Certificate Enrollment Protocol
- Privacy for Call Logs and Contacts
- Enhancement to Wireless Network Connectivity
- Call Hold Timer
- GZIP Encoding of SIP INFO Messages
- Enhanced Boss-Admin for VVX Phones
- Web Sign In for Skype for Business On-Premise Deployment

Session Traversal Utilities for NAT (STUN)

Polycom UC Software supports Session Traversal Utilities for NAT (STUN), a network protocol used in NAT traversal for real-time IP communications, such as voice, video, and messaging. STUN service is provided using UDP. STUN using TCP or TLS is not available.

You can configure the phone to act as a STUN client to send a request to the STUN server to discover the public IP and port(s). You can also configure the phone to send keep-alive messages to refresh NAT bindings.

Device Analytics Support for PDMS-SP

Polycom introduces device analytics to enable and configure your phone to provide details on many aspects of the phone's system and usages such as network stats, feature usage, memory and CPU, SIP service state, and connected peripherals.

A Polycom Cloud Services account is required to access these analytics. For more information, refer https://console.plcm.cloud.

Multilevel Precedence and Preemption (MLPP) for Assured Services - Session Initiation Protocol (AS-SIP) On Shared Lines

Multilevel Precedence and Preemption (MLPP) enables you to configure a precedence level for outgoing calls. Polycom implements MLPP in accordance with the standards set by Assured Services for Session Initiation Protocol (AS-SIP).

Higher precedence calls preempt – and thereby end - active calls with a lower precedence level. When an active call is preempted, the phone plays a preemption tone and displays a preemption screen. Polycom now provides the capability for shared lines when operating in a Ribbon environment.

Support for H.264 Packetization Mode 1 and H.264 Constrained Baseline Profile

VVX business media phones support H.264 Packetization Mode 1 for incoming and outgoing video calls. Packetization Mode 1 enables high resolution video by allowing media packets to be fragmented during transport.

VVX business media phones also support H.264 constrained base Profile which enables to send and receive calls to IR94 capable devices.

Enhanced Busy Lamp Field (BLF)

VVX phones supporting Broadworks Enhanced Busy Lamp Field (BLF) are now able to enable and configure new short and long key press behaviors when touching the line key for a monitored BLF user. New actions are 1-touch blind transfer or 1-touch park and retrieve.

A new preferences option in the basic settings is available when configured allowing the phone user to choose their preferred default key press action.

Busy Lamp Field Hold Alerting

VVX phones now support the Busy Lamp Field (BLF) sip.rendering state. For call control platforms that also support or allow this dialog attribute to be used. VVX phones can add display information when the monitored user places a call on hold with accompanying changes to LED behavior, key press action precedence, and an optional ringtone.

Busy Lamp Field (BLF) Spontaneous Call Appearance on Per BLF Basis

VVX phone supports Spontaneous Call Appearance property for an incoming call and incoming ringtone per Busy Lamp Field (BLF).

Enhanced Feature Key Macro Actions

A new macro $Tconsult$ is added to execute the consultative transfer functionality irrespective of default transfer type (Consultative/Blind) set on the phone.

Retrieve Logs from Support Information Package Page in the Web Configuration Utility

You can export the Support Information Package (.tar file) using Web Configuration Utility.

The support information package includes the following log files:

- . pbu file
- app log file
- boot log file
- audit log file

Simple Certificate Enrollment Protocol

The Simple Certificate Enrollment Protocol (SCEP) is a protocol that enables you to automatically enroll devices to retrieve new digital certificates or re-enroll to renew expired or expiring certificates.

This feature applies to all phones except VVX 1500 business media phones.

Privacy for Call Logs and Contacts

Your call logs and contacts you save are stored on the phone and a server. You can clear your personal history of stored call logs and contacts from the phone. You can also restrict the phone from uploading your call logs and contacts to the server.

Enhancement to Wireless Network Connectivity

You can configure the phone so that users can view the Wi-Fi menu under Basic settings. This allows users to also add a Wi-Fi network manually. Additionally, you can also configure the phone to display the Wi-Fi icon on the phone’s status bar and home screen.

You can manually add a new wireless network on VVX phone from the **Basic** menu.

Call Hold Timer

Polycom VVX phones will now display the timer when an active call is put on hold. The active call timer will resume when the user retrieves the call.

`up.holdTimerDisplay.enable` parameter is used to enable or disable the hold timer display on the VVX phone.

GZIP Encoding of SIP INFO Messages

Polycom VVX phone sends notifications for various activities to the server in gzip format saving network bandwidth.

You can configure this feature by `voIpProt.SIP.gzipEncoding.enable` parameter that is set to 0 by default.

Enhanced Boss-Admin for VVX Phones

When using Skype for Business you can configure Enhanced Boss-Admin feature on allowing users to add and edit delegates from the phone's user interface using the contacts list.

Users can also set Call Forward or Simultaneous Call Ringing option from the phone's user interface. When Enhanced Boss-Admin delegation occurs, you can view the delegate key icon on the phone's screen.

Web Sign In for Skype for Business On-Premise Deployment

Polycom UC Software 5.9.0 allows users to sign in to Skype for Business client on VVX phones. Web Sign In enables users to securely log in to Skype for Business from the phone using a computer web browser or mobile device browser. 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.

> Web Sign In for Skype for Business server support requires Hybrid Modern Authentication (HMA) to be enabled. To use the capability of HMA with Skype for Business On-premise AD should be federated with Azure AD. For more information about Hybrid Modern Authentication (HMA), refer Hybrid Modern Authentication for Skype for Business.

Common Area phone (CAP) feature is not supported for Web Sign In for Skype for Business On-Premises.

> Polycom VVX 250, 350, and 450 business IP phones support on-premises deployments only.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for the Polycom UC Software 5.9.0 release. For more information on using configuration parameters to enable or disable features, see the latest *Polycom UC Software Administrator Guide* for your release, available on Latest Polycom UC Software Release.

Configuration File Enhancements for UC Software 5.9.0

Parameter	*Permitted Values*	*Restart or Reboot*
`feature.nat.stun.enabled`	0 (default) - Disable the STUN feature on the phones. 1 - Enable the STUN feature on all registered Open SIP lines on a phone.	Yes
`nat.stun.server`	Enter a STUN server IP address or domain name. Null (default)	Yes
`nat.stun.port`	Set the server port number for all Open SIP-registered phones. 3478 (default)1 to 65535	Yes
`reg.x.nat.traversal.mode`	Enable or disable NAT traversal mode with STUN for signaling and media on the basis of the phone-level STUN feature. Auto (default) - Apply NAT configuration to both media and signalling per registration. Disabled - The phone does not use STUN for NAT traversal for this registration. For example, if `feature.nat.stun.enabled` is set to 1, and `reg.x.nat.traversal.mode` is set as Auto, the STUN feature is enabled for signaling and media for the registered line.	No
`nat.refresh.interval`	Set the time interval for the phone to send NAT keep-alive packets that keep the NAT port open and the phone reachable. 30 seconds (default) - The phone sends NAT keep-alive packets that keep the NAT port open and the phone reachable. 5 - 3600 seconds	No
`calendar.monthView.enabled`	0 (default) - Disables the **Month View** soft key. 1 - Enables the **Month View** soft key.	No
`device.da.enabled.set`	0 (default) - Do not use the `device.da.enabled` value. 1 - Use the `device.da.enabled` value.	No
`device.da.enabled`	0 (default) – Disable the device analytics feature. 1 – Enable the device analytics feature.	Yes

Parameter	*Permitted Values*	*Restart or Reboot*
`feature.obitalk.enabled`	0 (default) - Disable the connection to the OBiTalk cloud. 1 - Enable the connection to the OBiTalk cloud.	Yes
`obitalk.accountcode`	Specifies the account code provided to the service provider during registration. Null (default) String (maximum of 256 characters).	Yes
`da.supported.services`	Specify the Device Analytics service to enable. all (default) Comma seperated list of below strings need to be configured (maximum of 2048 characters) sdi ni service tsid pcap log config core vqmon cdr uptimeanalytics hardwareanalytics uianalytics blf sca restart reboot resettofactory	Yes
`attendant.callAction`	Specify the call action behavior for an Active call. Dial-Pick up (default) – An active call goes on hold and dials to monitor line when you short press the monitored line keys Blind – Blind transfer an active call on the monitored line keys Park – Parks an active call on the monitored line keys. If there is already a parked call on a monitored line then it will retrieve the parked call.	No

Parameter	*Permitted Values*	*Restart or Reboot*
`attendant.callActionMenu.enabled`	This parameter is configured to get the Attendant Call Action menu on the phone when dynamic BLF is configured on the phone. 0 (default) – **Attendant Call Action** menu will not appear on the phone. 1 - **Attendant Call Action** menu will appear on the phone.	No
`attendant.displayHoldState.enable=1`	Specifies the control of the display on the phone for BLF hold state. 0 (default) –The phone displays a busy state. 1 – The phone displays a hold state. Note: This parameter is only applicable to static BLF.	No
`attendant.resourceList.x.hold.ringer`	The ringtone that plays on the phone when BLF is in a hold state. The parameter depends on the value set for the parameter `attendant.displayHoldState.enabled`. If the parameter `attendant.displayHoldState.enable` is set to 1, use the parameter `attendant.resourceList.x.hold.ringer` Triplet (default) – Specifies the ringtone name for the parameter ringer11. Ringtone for BLF Hold should play for only 10 sec.	No
`ind.pattern.blfHold.step.1.state`	0 – Turns off the LED indicator for BLF Hold. 1 (default) – Turns on the LED indicator for BLF Hold.	Yes
`ind.pattern.blfHold.step.1.duration`	Specify the duration of the LED indicator for the pattern when BLF is in a hold state. 1000 (default) 0- 32767	Yes
`ind.pattern.blfHold.step.1.color`	Set the color of the LED indicator for the pattern when BLF is in a hold state. Red (default) – LED indicator turns to red when the BLF is in a hold state. Green – LED indicator turns green when the BLF is in a hold state.	Yes

Parameter	*Permitted Values*	*Restart or Reboot*
`ind.pattern.blfHold.step.2.state`	0 (default) – Turns off the LED indicator for BLF Hold. 1– Turns on the LED indicator for BLF Hold.	Yes
`ind.pattern.blfHold.step.2.duration`	Specify the duration of the LED indicator for the pattern when BLF is in a hold state. 1000 (default) 0 – 32767	Yes
`ind.pattern.blfHold.step.2.color`	Set the color of the LED indicator for the pattern when BLF is in a hold state. Red (default) – LED indicator turns to red when the BLF is in a hold state. Green – LED indicator turns green when the BLF is in a hold state.	Yes
`attendant.resourceList.x.display.spontaneousCallAppearances`	This parameter is applicable to Static BLF. Specifies spontaneous call appearance property for an incoming call. This parameter will override the phone level configuration parameters `attendant.behaviors.display.spontaneousCallAppearances.normal` and `attendant.behaviors.display.spontaneousCallAppearances.automata` to show or hide the call appearance property for BLF incoming call based on the resource type. Auto (default) – This value will use phone-level configuration depending on the BLF resource type. Show – This value will override the phone-level configuration and show the call appearance. Hide – This value will override the phone-level configuration and hide the call appearance. Note: Existing BLF ringtone will not stop if new BLF call comes.	No
`attendant.resourceList.x.ringType`	This parameter is applicable to Static BLF. Specifies incoming ringtone for each static BLF line defaultAll (default) – Specifies the ringtone type ring for the ringtone name. ringer1 - ringer 24. If no ringtone is configured for any static BLF line, then phone level incoming ringtone defined with `attendant.ringType` parameter will be played.	No

Parameter	*Permitted Values*	*Restart or Reboot*
`SCEP.CAFingerprint`	Configure the CA certificate fingerprint to confirm the authenticity of the CA response during enrollment. null (default) 0 - 255 characters	No
`SCEP.certPoll.retryCount`	Specify the number of times to poll the SCEP server when the SCEP server returns a Certificate Enrollment Response Message with the pkiStatus set to 'pending'. 12 (default) 1 – 24	No
`SCEP.certPoll.retryInterval`	Specify the number of seconds to wait between poll attempts when the SCEP server returns a Certificate Enrollment Response Message with the pkiStatus set to 'pending'. 300 seconds (default) 300 - 3600 seconds	No
`SCEP.certRenewalRetryInterval`	Specify the time interval to retry certificate renewal. 86400 seconds (default) 28800 - 259200 seconds	No
`SCEP.certRenewalThreshold`	Specify the percentage of the certificate validity interval to initiate a renewal. 80 (default) 50 – 100	No
`SCEP.challengePassword`	Specify the challenge password to send with the Certificate Signing Request (CSR) when requesting a certificate. null (default) 0 - 255 characters	No
`SCEP.csr.commonName`	Specify the common name to use for CSR generation. null (default) 0 – 64	No
`SCEP.csr.country`	Specify the country name to use for CSR generation. null (default) 0 – 2	No

Parameter	*Permitted Values*	*Restart or Reboot*
`SCEP.csr.email`	Specify the email address to use for CSR generation. null (default) 0 – 64	No
`SCEP.csr.organization`	Specify the organization name to use for CSR generation. null (default) 0 – 64	No
`SCEP.csr.state`	Specify the state name to use for CSR generation. null (default) 0 - 128 characters	No
`SCEP.enable`	0 (default) - Disable the SCEP feature. 1 - Enable the SCEP feature.	No
`SCEP.enrollment.retryCount`	Specify the number of times to retry the enrolment process in the case of enrolment failure. 12 (default) 1 – 24	No
`SCEP.enrollment.retryInterval`	Specify the time interval to retry the enrolment process. 300 seconds (default) 300 - 3600 seconds	No
`SCEP.http.password`	Specify the password that authenticates with the SCEP server. null (default) string, max 255 characters	No
`SCEP.http.username`	Specify the user name that authenticates with the SCEP server. null (default) string, max 255 characters	No
`SCEP.url`	Specify the URL of the SCEP server. null (default) 0 - 255 characters	No
`feature.wifi.basicmenu.enable`	1 (default) – The phone displays the Wi-Fi menu under **Basic** settings. 0 – The phone does not display the Wi-Fi menu.	No

Parameter	*Permitted Values*	*Restart or Reboot*
`status.wifi.icon.enable`	1 (default) – Display the Wi-Fi icon on the status bar of the phone's screen. Users can access Wi-Fi settings by selecting the Wi-Fi icon. 0 – Does not display the Wi-Fi icon on the status bar.	No
`homeScreen.wifi.enable`	1 (default) – Display the Wi-Fi icon on the phone's Home screen. 0 – Does not display the Wi-Fi icon on the phone's Home screen.	No
`video.profile.H264.packetizationMode`	Set to control the H.264 encoding and decoding capabilities on supported VVX business media phones. 0 (default) - Supports Single NAL unit mode. For Incoming calls: • If the remote endpoint supports only Non-Interleaved mode, the VVX business media phones reject the video with m line 0. • If the remote endpoint supports Single NAL Unit mode, then the VVX business media phones answer the incoming call with Single NAL mode. For Outgoing calls: • In all outgoing calls, the VVX business media phones send packetization-mode=0 in the offer. 1 - Supports both Single NAL Unit mode and Non-Interleaved mode. For Incoming calls: • The VVX business media phone answers both Single NAL Unit mode and Non-Interleaved mode. For Outgoing calls: • The VVX business media phones send packetization-mode=0 and packetization-mode=1 in the offer.	No
`up.enhancedbossadmin`	0 (default) - Disables the Enhanced Boss-Admin feature. 1- Enables the Enhanced Boss-Admin feature.	Yes
`up.phoneBootStatusPopupEnabled`	1 (default) - Phone displays the Popups after reboot. 0 – Phone does not display any popup after reboot.	No

Parameter	Permitted Values	Restart or Reboot
`voIpProt.SIP.gzipEncoding.enable`	0 (default) – Disable the Gzip encoding. Notifications will not be sent to server in gzip format. 1 – Enable the Gzip encoding. Notifications will be sent to the server in gzip format	No
`up.holdTimerDisplay.enable`	0 (default) – Hold Timer will not display. 1 – Hold Timer will display	No

Security Updates

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

The following table contains the security updates in UC Software 5.9.0.

Security Updates

Category	Issue ID	Release	Description
Security	EN-103735	5.9.0	Don't use basic authentication on VVX phones due to security issues.

Resolved Issues

The following table lists the resolved issues in UC Software 5.9.0

Resolved Issues

Category	Issue ID	Description
Application	EN-112576	The phone crashed during a reboot.
Application	EN-104339	VVX business media phone reboots while processing Reverse Name Lookup (RNL).
Application	EN-104422	Once you disable KeepAlive you cannot enable until you restart the phone.
Audio	EN-105156	Audio quality is bad when using the handset.
Audio	EN-111048	When audio and video is streamed from VVX phones to far end through BTOE application, the audio is choppy due to a shortage of CPU cycles.
Audio	EN-29792	In a Lync environment, the far-end occasionally hears distorted music for Music on Hold (MOH) when the call is established via TCP.
Audio	EN-94188	In a GENBAND environment, Music on Hold (MOH) doesn't play to the far end because the phone doesn't send the full codec list when holding the call.

Category	*Issue ID*	*Description*
Audio	EN-88372	There is a 5-second audio delay on the VVX phone when you resume the remote held call on the shared phone.
Calling	EN-105254	When you set the `feature.persistentMute.enabled` parameter to 1, the active call audio splits to the speaker along with the handset after a PTT/Paging call.
Calling	EN-104607	DUT is sending binding request to STUN during the hold and resume calls.
Calling	EN-99796	When making a point-to-point call between a VVX 1500 business media phone and an HDX system with media encryption enabled, the VVX 1500 business media phone doesn’t receive audio.
Calling	EN-97739	In an intercom call between the two VVX D60 handsets, when the first handset answers an incoming call, the second VVX D60 handset is unable to resume the on-hold intercom call.
Calling	EN-97737	In a certain environment, the VVX business media phone is not able to retrieve the parked call using the BLF line key.
Calling	EN-95752	When server-based Automatic Call Distribution feature is enabled on the phone and the agent is in a call, pressing the EFK configured Unavailable soft key doesn’t change the agent's state.
Calling	EN-97753	When you configure the Busy Lamp Field feature on the VVX business media phones and search for a directory via the directory soft-key, the phone returns to the Home screen.
Calling	EN-101429	VVX phone is unable to resume the held call during call transfer when off-hooked.
Calling	EN-97737	In a certain environment, the VVX business media phone is not able to retrieve the parked call using the BLF line key.
Call Management	EN-110627	VVX phones failed to accept the finalized peer-reflexive ICE candidates published by the far end.
Call Management	EN-108200	The VVX phone doesn’t pick up the Busy Lamp Field (BLF) incoming call when the line key is pressed while the phone is off-hook.
Certificate Management	EN-107069	DUT is disclosing certificate private key and sending insecure HTTP message.
Configuration	EN-104784	VVX phones fail to open the Web Configuration Utility in the browser in Chinese due to the insufficient size of the buffer to hold the Chinese Language file. The buffer size was increased to 256MB to hold large files.
Configuration	EN-101931	The SRTP status shows incorrect details on VVX 411 and 501 business media phones.
Configuration	EN-96691	Default value of parameter `callLists.logConsultationCalls` should be changed to 1.

Category	*Issue ID*	*Description*
Configuration	EN-76181	The `dialplan.routing.emergency.x.value` parameter considers value only up to 255 characters.
Configuration	EN-111457	VVX phone displays Speed Dial Keys configuration even if the phone doesn't support the feature. This is applicable to all VVX business media phones except the VVX 1500 business media phones.
Configuration	EN-98530	The SwitchingFunctionDevices event sends incorrect registration line details when registration address contains domain name along with registration number.
Directories/Address Books	EN-109289	The VVX expansion modules don't refresh the speed dial list after a favorite is added or removed from the phone keys.
Directories/Address Books	EN-98753	When you configure the Busy Lamp Field feature on the VVX business media phones and search for a directory via the directory soft-key, the phone returns to the Home screen.
General	EN-109713	Polycom terminates NOTIFY line to seize Bridged Line Appearance (BLA) dialog event after it gets a 500-internal error with a retry-after 3 seconds from UAS. The parameter `voIpProt.SIP.blaGlareHonorRetryAfter` was introduced to control this issue.
General	EN-108308	A 3-second timer is applied to 911 calls on VVX phones when "[2-9]11" is not at the beginning of the digit map.
General	EN-101084	Removed `up.onHookDialingEnabled` parameter from the code as this parameter is applicable only for Spectralink wireless phones.
General	EN-95586	VVX business media phone continues to display the Message Waiting Indicator (MWI) after reboot / de-registration even though no voicemail is available on phone.
General	EN-107207 EN-92897	Intermittently, a VVX phone doesn't register with the backup server after an outage.
General	EN-92078	VVX business media phones receive a 400-missing contact error from session border controllers (SBC) even when sending the contact header in a 200 OK for NOTIFY.
General	EN-76384	When entering the PIN code to a conference bridge, some digit plays a long tone as if it gets stuck.
General	EN-97274	VVX business media phones intermittently display LDAP error on corporate directory screen during search operations at times of peak load.
General	EN-94806	Polycom UC software upgrade fails on VVX phones while upgrading from UC software version 5.4.5 and 5.5.1 due to "Bad Image Checksum" error.
General	EN-103385	In Skype for Business environment, RGB color for VVX phones is low in UC Software 5.7.x version.

Category	Issue ID	Description
Interoperability	EN-98487	When processing SIP URLs for a Record-Route header, VVX business media phones incorrectly prepend SIP in the URL.
Localization	EN-111652	VVX phones are not translating the Inactive string to other languages other than default language.
Localization	EN-105378	DUT sending En-Us as default language when inviting far-end PSTN users into a conference.
Logs	EN-95299	In a Skype for Business environment, the username and password for the Skype for Business account display in plain text in the **app** logs and sys logs.
Network	EN-107375	In a Skype for Business environment, the description is missing in Field Help for the `server.log.setting.enabled` parameter.
Network	EN-104641	VVX phones fail to get an IP address when they receive two server identifiers (Option 54) in a DHCP offer.
Network	EN-102127	VVX business media phones use an incorrect TLS version, so TLS v1.2 connections fail.
Network	EN-98838	The INVITE for Session Description Protocol (SDP) in a conference call does not include SAVP and cryptographic line details.
Network	EN-92987	When you set the `feature.EWSAutodiscover.enabled` parameter to 1, VVX business media phones don't send the WPAD PAC file request.
Network	EN-112091	DUT fails to upgrade in a remote network using NTLM authentication.
Network	EN-102142	Phone authentication fails for the first time when using the TLS method with Polycom certificates for 802.1x authentication.
Provisioning	EN-93405	After a factory reset, while provisioning, the VVX business media phone accepts the parameter value of `voIpProt.SIP.assuredService.namespace` as "ets" even though configured as "dsn".
Provisioning	EN-101878	VVX business media phones restrict uploading Certificate Signing Request (CSR) and CSV files to the provisioning server root directory which causes RealPresence Resource Manager provisioning server failure.
Shared Lines	EN-104744	In VVX business media phones, the call is being answered on the headset when trying to pick up the held Multiple Appearance Directory Number (MADN) call on the remote shared device using the handset.
Security	EN-102441	When you enable the Device lock feature for the guest user, the phone asks to set a PIN lock.
Security	EN-95756	VVX business media phones don't support the NTLM v2 mechanism with the down-level format.

Category	*Issue ID*	*Description*
User Interface	EN-104782	VVX business media phones don't send any error response when the phone is in an idle state and receives an INVITE with replace header.
User Interface	EN-104591	Headset volume is increasing while switching between headset and handset.
User Interface	EN-101057	The VVX business media phone web UI help text for the `msg.bypassInstantMessage` parameter displays incorrect information.
User Interface	EN-100103	While using handsfree, headset or lifting handset to initiate a call, the idle display screensaver continues to stay on VVX business IP phone's screen.
User Interface	EN-99550	The brightness/ contrast level on Expansion Modules (EMs) connected to the VVX 601 business media phone is lower when compared to EMs connected to the VVX 600 business media phone.
User Interface	EN-93448	VVX business media phone is unable to display Unicode text having Armenian characters.
User Interface	EN-93031	While inviting a participant to the conference, the Reverse Name Lookup fails to continue with the next source when the display name is not received in SIP signaling.
User Interface	EN-92639	When "P-Asserted-Identity" in a 200 OK response doesn't have a display name, the phone doesn't display the caller ID details for an incoming call during call pick-up.
User Interface	EN-79960	When you connect an unsupported USB device to a VVX business IP phone, the phone doesn't display the rear USB port's power alert pop-up.
User Interface	EN-97749	On VVX business media phones, the call center queue status notification menu doesn't close automatically after 30 seconds.
User Interface	EN-97248	While playing video on an idle microbrowser, VVX 501 and 601 business media phones don't display incoming calls.
User Interface	EN-96709	During failover/ failback, the phone resets the SIP Open Programmable Interface (SOPI) subscription and phone loses speed dial and directory.
General	EN-100838	VVX business media phone does not increment the Cseq value in the PRACK message in call forking.
User Interface	EN-100910	EagleEye Mini Camera does not publish privacy report which causes VVX phones to show the wrong LED.

Related Poly and Partner Resources

See the following sites for information related to this release.

- The Polycom Support Site is the entry point to online product, service, and solution support information including Licensing & Product Registration, Self-Service, Account Management, Product-Related Legal Notices, and Documents & Software downloads.
- The Polycom Document Library provides support documentation for active products, services, and solutions. The documentation displays in responsive HTML5 format so that you can easily access and view installation, configuration, or administration content from any online device.
- The Polycom Community provides access to the latest developer and support information. Create an account to access Poly support personnel and participate in developer and support forums. You can find the latest information on hardware, software, and partner solutions topics, share ideas, and solve problems with your colleagues.
- The Polycom Partner Network are industry leaders who natively integrate the Poly standards-based RealPresence Platform with their customers' current UC infrastructures, making it easy for you to communicate face-to-face with the applications and devices you use every day.
- The Polycom Collaboration Services help your business succeed and get the most out of your investment through the benefits of collaboration.

GETTING HELP

For more information about installing, configuring, and administering Poly/Polycom products or services, go to Polycom Support.

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