



Polycom[®] UC Software 5.6.1

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

Polycom announces the release of Polycom[®] Unified Communications (UC) Software, version 5.6.1. This document provides the latest information about this release.

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What's New in Polycom UC Software 5.6.1

Polycom UC Software 5.6.1 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.

Polycom UC Software 5.6.1 supports the following Polycom endpoints.

Phone Support

Phone Model	Skype for Business	Open SIP
VVX 101 business media phone		✓
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 5.6.1 supports the following Polycom accessories.

Accessories Support

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

New Features and Enhancements

There are no new features in UC Software 5.6.1.

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.

Phones shipped beginning May 15, 2017 include a free VQMon license.

VVX Business Media Phone Series 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
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Enhanced BLF	No	No	Yes	Yes	Yes	Yes	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
H.323 Video	No	No	No	No	No	No	Yes	Yes
Server-Based Call Recording	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
USB Call Recording	No	No	No	No	No	Yes	Yes	Yes
VQMon	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes (Audio only)	Yes (Audio only)

Supported DHCP Sub-Options

The following table lists the individual sub-options and combination sub-options supported on VVX business media 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.

DHCP Option 43 Configuration Options

Option	Result
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	

Server Logging Level for Skype for Business Server

In UC Software 5.6.0 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
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

Corresponding Server and Phone Log Levels

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 5.6.1 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 5.6.1

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	tone.dtmf.rfc2833Payload _OPUS	Sets the Dual Tone Multi Frequency (DTMF) payload required to use the Opus codec. 126 (default) 96 - 127	Yes
sip-interop.cfg	voIpProt.SIP.RFC3261Time rI	0 (default) - Sets Timer I for reliable transport to fire at 5 seconds. This parameter does not cause any change for unreliable transport. 1 -Sets Timer I to fire at 0 seconds.	No
site.cfg	voice.txPacketFilter	Allows changing the frequency response of narrowband Tx. 0 (default) - Tx filtering is not performed. 1 - Enables narrowband 300 Hz flat Tx high pass filter. 2 - Enables narrowband 300 Hz pre-emphasis Tx high pass filter.	No
site.cfg	voice.rxPacketFilter	Allows changing the frequency response of narrowband Rx. 0 (default) - Rx filtering is not performed. 1 - Enables narrowband 300 Hz flat Rx high pass filter. 2 - Enables narrowband 300 Hz pre-emphasis Rx high pass filter with loudness boost.	No

Release History

This following table lists the release history of Polycom UC Software.

Release History

Release	Release Date	Features
5.6.1	December 2017	This release includes important field fixes.
5.6.0	July 2017	<p>This release includes support for the following features:</p> <p>Open SIP Enhancements</p> <p>This release introduced support for FIPS 140-2 Compliance Support, Two-Way Active Measurement Protocol support, Caller ID Display from the SIP Invite, BroadSoft Server-Based Call Waiting, Call Line Identification, BroadSoft Server-Based Redial, Remote Party Disconnect Alert Tone, Support for Siren 7 Audio Codec.</p> <p>The BroadSoft Directory now includes the following new directories:</p> <ul style="list-style-type: none"> • Group and Group Common Directory • Enterprise Common Directory • Personal Directory <p>Skype for Business Enhancements</p> <p>This release introduced support for Dial Plan Normalization, Multiple Emergency Number Dial Plan, Skype for Business User Interface Enhancements, Skype for Business Conference Enhancements, Device Lock Enhancements, Profile Picture on Device Lock Screen, Secure Single Sign-On With Third-Party Supporting Solutions, Safe Transfer for Boss-Admin Enhancements, and Busy Options for Incoming Calls.</p> <p>VVX D60 Enhancements</p> <p>This release introduced support for FLK Support in VVX Business Media Phone with VVX D60, Maximum Number of Handset Registrations, VVX D60 Base Station Pairing Permissions, Mac Address Pairing with VVX D60 Base Station, Continuous Attempt to Re-pair with a VVX D60 Base Station, and VVX D60 Call Handoff.</p>
5.5.3	September 2017	This release includes important field fixes.
5.5.2	May 2017	<p>This release includes the following features and field fixes:</p> <ul style="list-style-type: none"> • 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

Release History

Release	Release Date	Features
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: <ul style="list-style-type: none"> • Introduced the Polycom VVX 301/311, 401/411, 501, and 601 business media phones • Flexible line key customization for Skype for Business (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: <ul style="list-style-type: none"> • Microsoft Office 365 and Skype for Business Online • Office365 and Skype for Business Provisioning and Manageability • Time and Date Initial Setup
5.4.0	May 2015	Added support for Alcatel-Lucent CTS features including <ul style="list-style-type: none"> • Advanced Conference • Shared Call Appearance with Bridge In • Visitor Desk Phone This release includes support for the following features: <ul style="list-style-type: none"> • 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 5.6.1

Consider the following information when installing or updating to Polycom UC Software 5.6.1.

- The Polycom BToE Connector 3.6.0 application is required for use with UC Software 5.6.1. For best results, Polycom recommends deploying the BToE Connector 3.6.0 application prior to deploying UC Software 5.6.1. While BToE 3.6.0 is backwards compatible with previous versions of VVX firmware, Polycom does not recommend running previous versions of BToE software with UC Software 5.6.1.
- Before updating your VVX 1500 phone to UC Software 5.6.1, 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 5.6.1 and the BToE Connector 3.6.0, then the phone doesn't lock with the computer.

Download the Distribution Files

To download UC Software 5.6.1, 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.id 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.id and resource files is **5.6.1.1740 rts27**.

Understand the Combined and Split ZIP Files

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

Understand the Combined ZIP and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
3111-40250-001.sip.ld	SIP application executable for VVX 101	No	Yes
3111-40450-001.sip.ld	SIP application executable for VVX 201	No	Yes
3111-46135-002.sip.ld	SIP application executable for VVX 300	No	Yes
3111-48300-001.sip.ld	SIP application executable for VVX 301	No	Yes
3111-46161-001.sip.ld	SIP application executable for VVX 310	No	Yes
3111-48350-001.sip.ld	SIP application executable for VVX 311	No	Yes
3111-46157-002.sip.ld	SIP application executable for VVX 400	No	Yes
3111-48400-001.sip.ld	SIP application executable for VVX 401	No	Yes
3111-46162-001.sip.ld	SIP application executable for VVX 410	No	Yes
3111-48450-001.sip.ld	SIP application executable for VVX 411	No	Yes
3111-44500-001.sip.ld	SIP application executable for VVX 500	No	Yes
3111-48500-001.sip	SIP application executable for VVX 501	No	Yes
3111-44600-001.sip.ld	SIP application executable for VVX 600	No	Yes
3111-48600-001.sip	SIP application executable for VVX 601	No	Yes
2345-17960-001.sip.ld	SIP application executable for VVX 1500	No	Yes
3111-33215-001.sip.ld	SIP application executable for SoundStructure VoIP Interface	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-identification(s) for the VVX D60	Yes	Yes
sip.ver	Text file detailing build-identification(s) 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

Understand the Combined ZIP and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
features.cfg	Configuration parameters for telephony features	Yes	Yes
firewall-nat.cfg	Configuration parameters for firewall related features	Yes	Yes
H323.cfg	Configuration parameters for the H.323 signaling protocol	Yes	Yes
lync.cfg	Contains Lync/SfB specific configuration parameters	Yes	Yes
pstn.cfg	Contains parameters for PSTN use	Yes	Yes
reg-advanced.cfg	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 that are set for each site	Yes	Yes
video.cfg	Configuration parameters for video connectivity	Yes	Yes
video-integration.cfg	Configuration parameters for Polycom HDX system integration	Yes	Yes

Understand the Combined ZIP and Split ZIP Files

Distributed Files	File Purpose and Application	Combined	Split
VVX-dictionary.xml	Includes native support for the following languages: <ul style="list-style-type: none"> • 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 	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

Resolved Issues

The following table lists resolved issues in UC Software 5.6.1.

Resolved Issues

Category	Issue ID	Description
API	EN-32378	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-56309	The VVX business media phones do not send the XMPP status correctly after restart but works fine after reboot.
Application	EN-41830	The VVX business media phones do not update Outlook Contact display name on reboot.
Application	EN-41787	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-41831	The 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.
Audio	EN-33579	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-32349	When the VVX 100, 200, 300, and 400 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.
Audio	EN-56174	A delay in audio is observed when the incoming call is answered using the D60 handset.
Calling	EN-32397	In a Shared Call appearance, remote resume fails when <code>sec.srtp</code> parameter is enabled.
Calling	EN-32421	When the configuration parameter <code>voIpProt.SIP.RFC3261TimerI</code> is set to 1, the VVX business media phone sends a 481 response code to a second INVITE after rejecting initial INVITE message with a 488 response even when the transaction timeout is zero, as per specification for Transmission Control Protocol (TCP) transport.
Calling	EN-35868	Using G729 codec on the VVX business media phones with HDvoice handset causes audio to be muffled.
Calling	EN-54766	The VVX business media phone freezes during the call when the digits are pressed quickly.
Calling	EN-41306	The VVX business media phones does not end or reject the incoming call after the handset is put on-hook quickly.
Calling	EN-50024	When you simultaneously keep receiving a PTT call during an active call, the call hold reminder tone continues to play even when the call is disconnected and phone is in idle state.

Resolved Issues

Category	Issue ID	Description
Calling	EN-43129	In a Skype for Business environment, the VVX business media phone does not display the Shared Line Appearance (SLA) remote notifications approximately after 30 minutes.
Calling	EN-54671	When the VVX business media phone is locked and the configuration parameter <code>DisableHandsetOnLockedMachine</code> is set to 2, the phone does not allow outgoing calls over the Skype for Business client.
Calling	EN-56178	Call input screen is not populated to perform a call transfer in line view on VVX 101, 201, 300 series and 400 series business media phones.
Calling	EN-44597	Blind transfer fails on a D60 handset if the call is answered on the twinned line.
Calling	EN-34647	The D60 assigned lines becomes usable when the D60 handset is twinned with a shared line of VVX business media phone which has Flexible Line Key (FLK) enabled.
Calling	EN-54742	VVX D60 does not dial a voicemail number that contains alphanumeric text.
Calling	EN-58346	During an incoming call, the VVX business media phones play two short tones instead of the configured ringtone.
Call Management	EN-43871	The VVX business media phone reboots if the server does not maintain unique dialog ID for different BLF calls.
Configuration	EN-50724	Bluetooth, when enabled, is not in effect sometimes after the Reboot/Restart/Upgrade.
Configuration	EN-50634	On the Polycom Trio and VVX business media phones, when the AutoDiscover option in the web server is disabled and the exchange server URL is blank, the configuration parameter <code>exchange.server.url</code> is not removed and is saved as a blank value.
Configuration	EN-43083	When the configuration parameter <code>saf.x</code> is set to value <code>"/</code> , the phone keeps downloading the application repeatedly on reboot.
Directories/ Address Books	EN-32450	The VVX business media phone displays the message "More entries found, refine search" when the number of entries displayed by Open LDAP server is equal to the value of parameter <code>dir.corp.pageSize</code> or <code>dir.corp.PageSize-1</code> .
General	EN-28216	Session Description Protocol (SDP) clock rate for Dual Tone Multi Frequency (DTMF) does not match the clock rate for Opus codec.
Interoperability	EN-32493	In VVX 500/600 business media phones, the video stops when the server sends unsupported audio and supported video codecs in re-INVITE.
Localization	EN-33818	VVX business media phone displays the "Forward" menu name incorrectly in German language.
Network	EN-32485	When the VVX business media phone is set to the forced 1000FD mode and is connected to a Cisco switch having Auto-Negotiation disabled, the LAN connection in the phone goes down.

Resolved Issues

Category	Issue ID	Description
Network	EN-58239	PC port becomes inactive when the PC having Intel NIC cards is configured from Auto to 100FD mode.
Security	EN-33163	The VVX business media phone shows the password as plain text in the phone information menu when the password is provided as a part of server address.
Security	EN-57841	The VVX business media phone's microbrowser does not send Server Name Indication (SNI) in TLS negotiation.
Security	EN-54675	The VVX business media phone re-registers and gets unregistered immediately when changing the transport protocol from a non-secure mode to secure mode along with TLS profile configuration.
Security	EN-42794	The VVX business media phone fails to authenticate to Microsoft Exchange when the Active Directory domain UPN does not match the SIP domain name.
User Interface	EN-33760	If the URL dialing feature is disabled, the first two digits are missing from the call list when the user dials a URI number.
User Interface	EN-32873	The position of the GuestIn soft key changes incorrectly when the parameter <code>softkey.feature.directories</code> is enabled or disabled.
User Interface	EN-35239	VVX business media phone incorrectly displays the newly added Outlook Contacts in Korean language.
User Interface	VOIP-121446	The VVX 201 business media phone fails to display the phone number or line address in the status bar when <code>lcl.status.LineInfoAtTop</code> parameter is set to 1.
User Interface	EN-32658	When changing the language from French for France (fr-fr) to French for Canada (fr-ca) or vice versa, the VVX business media phone requires a reboot to display the translated words on the phone's user interface.
User Interface	EN-35546	VVX 101,201,300,301,310,311 business media phone displays the "Enabled" and "Disabled" text incorrectly on the phone's user interface upon changing to Korean language.
User Interface	EN-41330	The VVX business media phone displays corrupt characters for date/time display at the status bar when the preferred language is set as Irish or Spanish.
User Interface	EN-41782	When the parameter <code>softkey.feature.buddies</code> is enabled, 'Contact' soft key has display issues on the phone's user interface.
User Interface	EN-35418	The VVX business media phone continues to display the missed calls list even after viewing the missed call numbers in the call history.

Known Issues

The following table lists all known issues and suggested workarounds for UC Software 5.6.1.



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	VOIP-127458	5.5.2	The VVX business media phone takes 4 to 5 seconds to establish an audio when answering the Skype for Business response group call.	No workaround currently available.
Calling	VOIP-126993	5.6.0	In a Skype for Business environment, when a participant using VVX business media phone creates a conference call during an active point-to-point call with the PSTN user, the PSTN user fails to join the conference and the call does not rollback to point-to-point call.	No workaround currently available.
Network	EN-60890	5.6.1	Host/PC having Intel NIC cards shows one-way low throughput when connected to the PC port that is set to 100FD mode.	No workaround currently available.
Registration	VOIP-126447	5.6.0	When you change the value for the maximum handsets registered to the paired VVX D60 Base Station during a software upgrade, the registered handsets are not removed.	Do not change the value for handset registration during software upgrade.
Registration	VOIP-127453	5.6.0	When the TLS version for VVX business media phone is changed using Web Configuration Utility, the phone de-registers from the server.	No workaround currently available.
User Interface	VOIP-127544	5.6.0	The security classification is not displayed on the VVX business media phone when an active call is handoff to VVX business media phone from VVX D60 wireless handset.	No workaround currently available.

Updates to Previous Software Releases

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

What's New in Polycom UC Software 5.6.0

Polycom UC Software 5.6.0 includes the following new features and enhancements.

- [Open SIP Enhancements](#)
- [Phone Features and Licenses](#)
- [VVX D60 Enhancements](#)

Open SIP Enhancements

This section include new features and enhancements for Open SIP deployment.

BroadSoft Directory Support

The BroadSoft Directories now support the following new directories along with the existing Enterprise Directory for the 5.6.0 release:

- **Group Directory:** This directory enables you to view the contact details such as work, extension, and mobile numbers of contacts. You can place a call to anyone in the user's group within the enterprise.
- **Group Common Directory:** This directory enables you to view the contact details such as names and phone numbers of common contacts from various groups within an enterprise.
- **Enterprise Common Directory:** This directory enables you to view the contact details such as names and phone numbers of common contacts within an enterprise.
- **Personal Directory:** This directory enables you to view the contact details such as names and phone numbers of the contacts 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.

You can enable Group and Group Common directories using `feature.broadsoftGroupDir.enabled` parameter.

Similarly, you can enable Enterprise and Enterprise Common directories using `feature.broadsoftdir.enabled` parameter.

The Personal directory can be enabled using `feature.broadsoftPersonalDir.enabled` parameter.

The parameters that configure these features are listed in [Phone Features and Licenses](#).

BroadSoft Server-Based Call Waiting

The server-based Call Waiting feature enables the server to manage incoming calls while a user is in an active call. If the user changes the call waiting state when logged in from another device or client using the same account, 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.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

BroadSoft Server-Based Redial

Polycom UC Software allows you to configure the BroadSoft Server-Based Redial feature, allowing users to redial the last number dialed from any device connected to the same line or registration.

You can configure the BroadSoft Server-Based Redial feature using `feature.broadsoft.basicCallLogs.redial.enabled` parameter.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

BroadSoft Executive-Assistant Menu Enhancements

You can simplify the BroadSoft Executive-Assistant menu by using the following parameters:

- Add or remove the pick call and barge-in soft keys from the executive actions menu, using the parameter `feature.BSExecutiveAssistant.SimplifiedAssistant.enabled`.
- Add or remove the call screening option from the executive settings menu, using the parameter `feature.BSExecutiveAssistant.SimplifiedExec.enabled`

The parameters that configure this feature are listed in [Phone Features and Licenses](#).

Automatic Call Distribution Menu

You can view or hide the menu items on the Automatic Call Distribution (ACD) menus. You can configure the phone using the parameter `acd.simplifiedAgentStateControl` to hide or display the ACD soft keys such as **ASignIN** or **ASignOut**, and **Available**.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

TLS Protocol Configuration for Supported Applications

Polycom UC Software allows you to configure the TLS Protocol for the following supported applications:

- Browser application
- LDAP application
- SIP application
- SOPI application
- Web server application
- XMPP application
- Exchange services application
- Syslog application
- Provisioning application
- 802.1x application

The parameters that configure these applications are listed in [Phone Features and Licenses](#).

Two-Way Active Measurement Protocol Support

Polycom UC Software supports Two-Way Active Measurement Protocol (TWAMP), which is RFC 5357 compliant, to check network performance by measuring the round-trip time between two devices using TWAMP protocols. TWAMP defines the following protocols:

- TWAMP Control protocol, which uses TCP.
- TWAMP Test protocol, which uses UDP.

TWAMP includes the following limitations:

- TWAMP Control and Test protocols only support unauthenticated mode
- A maximum of 10 clients can establish a connection with the server
- The server is limited to handle a maximum of 10 sessions per client

You can configure your phone to enable the two-way active measurement protocol feature using `feature.twamp.enabled` parameter.

The parameters that configure this feature are listed in [Phone Features and Licenses](#).

FIPS 140-2 Compliance Support

Polycom UC Software allows your phone to work seamlessly when using Federal Information Processing Standard (FIPS 140-2) compliant cryptography. You can configure your phone to use FIPS 140-2 compliant cryptography with the `device.sec.TLS.FIPS.enabled` configuration parameter.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

Caller ID Display from the SIP Invite

This feature enables you to specify which SIP request and response messages to use to retrieve caller ID information using the following parameters:

- `voIpProt.SIP.CID.request.sourceSipMessage`
- `voIpProt.SIP.CID.response.sourceSipMessage`

The parameters that configure this feature are listed in [Phone Features and Licenses](#).

Call Line Identification

You can set the priority order to display the caller and callee's identity. The following is the priority order to display the connected line identity: P-Asserted-Identity,Remote-Party-ID,From.

You can set the priority order to display the caller's identity for incoming calls using the `voIpProt.SIP.caller.SourcePreference` parameter.

Similarly, using the `voIpProt.SIP.callee.SourcePreference` parameter, you can set the priority order to display the callee's identity for outgoing calls.

The parameters that configure this feature are listed in see [Phone Features and Licenses](#).

Remote Party Disconnect Alert Tone

When you configure the `call.remoteDisconnect.toneType` parameter, the phone plays a tone to alert the user when the remote party or network disconnects in an active call.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

Additional Open SIP Enhancements in this Release

Some of the existing parameters include the following modifications for the 5.6.0 release:

- **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`.

- **Soft Keys Maximum Characters**

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

- **Intercom Calls**

In previous releases, the following parameter name included an 's' and was incorrect: `homescreen.intercom.enable`. The correct name for the parameter is: `homeScreen.intercom.enable`.

- **Call Transfer**

In the previous releases, the following parameter name was incorrectly spelled using a 'D' and has been corrected: `call.DefaultTransferType`. The corrected name for the parameter is: `call.defaultTransferType`.

Skype for Business Enhancements

This section include new features and enhancements for Skype for Business deployment.

Dial Plan Normalization

This Skype for Business feature enables you to enable and configure dial plans on the Skype for Business server or on your provisioning server.

You can configure dial plan normalization on your provisioning server using `reg.1.applyServerDigitMapLocally`.

For more information on regular expressions used on the Skype for Business server, see [.NET Framework Regular Expressions](#) on Microsoft Developer Network.

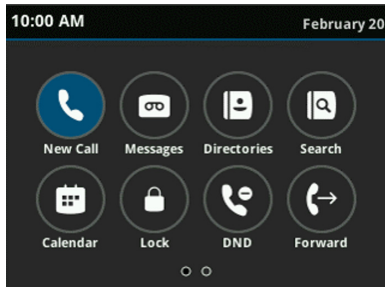
Multiple Emergency Number Dial Plan

You can configure multiple emergency numbers on the Skype for Business server when registering Polycom devices with Skype for Business. When you correctly configure the multiple emergency numbers on the Skype for Business server, users can make calls to the emergency numbers from the Skype for Business client or from a phone, even when the phone is locked.

Skype for Business User Interface Enhancements

The user interface for VVX 400, 500, and 600 series business media phones was updated to match the theme used in the Skype for Business 2016 client. This feature is enabled by default on supported phones with the Skype Base Profile or SKU.

The following figure shows the updated user interface on the Home screen of a VVX 500 series business media phone.




Skype for Business Conference on VVX Business Media Phones

Polycom UC Software provides a one-touch Join button on the phone's screen. This button allows you to join a Skype for Business conference in a federated environment, even if you haven't configured Transport Neutral Encapsulation Format (TNEF).

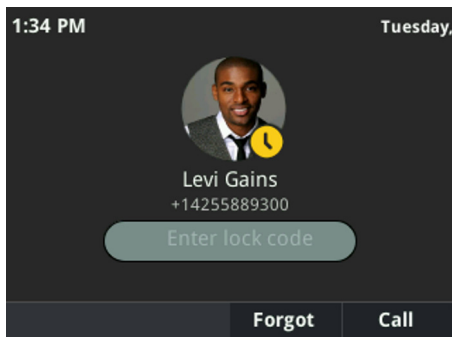
Device Lock Enhancements

The Device Lock feature includes the following modifications and enhancements for the 5.6.0 release:

- The user's sign-in name, registered number, and profile photo display on the Lock screen.
- The Unlock icon  was removed from the Lock screen, and users can enter their lock code directly on the Lock screen.
- In BToE mode, the phone no longer locks or unlocks automatically when the status of connected and paired computer changes between active and inactive. The phone only locks or unlocks when the computer locks or unlocks.
- The user's presence status string was removed from the Lock screen.
- The parameter `feature.deviceLock.enable` no longer requires a reboot.
- Users can enable or disable the Device Lock feature from the Settings menu on the phone.
- The procedures for unlocking the phone and resetting the lock code were updated.
- Administrators can now use the in-band provisioning parameter `DisableHandsetOnLockedMachine` to determine whether users must enter their lock code to answer incoming calls.

Profile Photo on Device Lock Screen

When a user is signed in to their Skype for Business account on the VVX 400, 500, and 600 series business media phones, the user's Microsoft Exchange or public website photo displays on the Lock screen, as shown next.



This feature is enabled by default when the Microsoft Exchange Service is enabled. Profile photos set using Active Directory are not supported and do not display on the phone.

Secure Single Sign-On with Third-Party Supporting Solutions

Polycom VVX business media phones currently support third-party single sign-on solutions such as Okta and Ping Federate. These single sign-on solutions provide secure access to cloud-based applications such as Microsoft Exchange and Skype for Business.

Safe Transfer for Boss-Admin Enhancements

The following enhancements were added for the Safe Transfer feature in a Boss-Admin environment:

- The Resume soft key displays only when a transferred call is returned to a delegate's phone.
- Placing the handset in the cradle places the transferred call on hold and does not end the transferred call.
- A tone plays on the delegate's phone after the call is returned to the delegate.
- If the boss does not answer the transferred call, the call is automatically resumed on the caller's phone.

Busy Options for Incoming Calls

Polycom phones support Busy Options to help you manage incoming calls when you are already on another call. With this feature, you can forward an incoming call to voicemail or reject the call with a 'busy' notification.

Support for Siren 7 Audio Codec

The following VVX business media phones support the Siren 7 audio codec:

- VVX 300/301/310/311
- VVX 400/401/410/411
- VVX 500/501
- VVX 600/601
- SoundStructure VoIP Interface

The VVX 101, 201, and 1500 business media phones do not support Siren 7.

The Siren 7 audio codec is disabled by default.

VVX D60 Enhancements

This section include new features and enhancements for VVX D60.

FLK Support in VVX Business Media Phone with VVX D60

Polycom UC software now supports flexible line key (FLK) on the VVX business media phone when the VVX D60 Base Station is paired with the VVX business media phone. No additional configuration is required for this functionality to work.

Maximum Number of Handset Registrations

You can control the number of handsets that can be registered to the VVX D60 Base Station. A minimum of one and a maximum of five handsets can be configured for a VVX D60 Base Station. You can configure the handset registration feature using `VVXD60.handset.maxCount` parameter.

The number of handsets that can be registered to a VVX D60 Base Station when paired to VVX 101 or 201 business media phone is limited to one.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

VVX D60 Base Station Pairing Permissions

Administrators can control whether users are allowed to pair a base station only, unpair a base station only, or pair and unpair a base station with a VVX business media phone using `VVXD60.allowPairing` parameter.

The parameter that configures this feature is listed in see [Phone Features and Licenses](#).

VVX Business Media Phone and VVX D60 Base Station Pairing

Polycom UC Software allows you to pair the VVX D60 Base Station with the VVX business media phone by updating the configuration parameter `VVXD60.base.mac` or by manually setting the MAC address on VVX business media phone. Ensure that the VVX D60 Base Station and the VVX business media phone are on the same subnet.

The parameter that configures this feature is listed in [Phone Features and Licenses](#).

Continuous Attempt to Re-pair with a VVX D60 Base Station

When a VVX business media phone unpairs from a base station due to a power outage or the base station is powered off, the phone will continuously attempt to re-pair with the base station. Users must get unpair permissions from an administrator to cancel the re-pairing attempt or the administrator can cancel the re-pairing attempt from the Administrator menu on the phone.

After powering on, the VVX D60 Base Station may take up to 60 seconds to re-pair with the VVX business media phone.

VVX D60 Call Handoff

When VVX D60 wireless handsets are mapped to a twinned line on the host VVX business media phone, the handoff feature of a call answered on the VVX D60 handset or the host VVX business media phone is available. This feature is available for local shared lines (VVX+DECT lines).

VVX D60 Wireless Handset Enhancements

The VVX D60 Wireless Handset user interface includes the following modifications for the 5.6.0 release:

- The VVX D60 Wireless Handset displays the **DND** menu only when the `feature.doNotDisturb.enable` parameter value is set to 1.
- The VVX D60 Wireless Handset displays the **Call Forwarding** menu only when the `feature.forward.enable` parameter value is set to 1.
- The VVX D60 Wireless Handset displays the **Call Park** option for a call only when the `feature.callPark.enabled` parameter value is set to 1.

- The VVX D60 Wireless Handset displays the **BargeIn** option only when the `reg.x.bargeInEnabled` parameter value is set to 1.

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.6.0.

Configuration File Enhancements

Parameter Template	Permitted Values
<code>acd.simplifiedAgentStateControl</code> <code>features.cfg</code>	0 (default) - Displays menu items. 1 - Hides ASignIN and associated soft keys. Also hides Menu items under Menu > Settings > Feature > ACD .
<code>call.remoteDisconnect.toneType</code> <code>features.cfg</code>	Select the disconnect ringtone to play when the remote party ends the call. Silent (default) messageWaiting, instantMessage, remoteHoldNotification, localHoldNotification, positiveConfirm, negativeConfirm, welcome, misc1, misc2, misc3, misc4, misc5, misc6, misc7, custom1, custom2, custom3, custom4, custom5, custom6, custom7, custom8, custom9, custom10
<code>reg.1.applyServerDigitMapLocally</code> <code>sip-interop.cfg</code>	1 (default) - Enable dial plan normalization. 0 - Disable dial plan normalization.
<code>dialplan.1.lyncDigitmap.timeOut</code> <code>site.cfg</code>	Choose the dial plan timeout method for static and off hook dial scenarios. 4 (default) 0 - 100
<code>dialplan.x.Additionale911dialstrings</code> <code>site.cfg</code>	Use this parameter to configure additional emergency dial strings. NULL (default) string
<code>dialplan.TranslationInAutoComp</code> <code>site.cfg</code>	1 (default) - The translated string displays in the autocomplete list. 0 - The translated string does not display in the autocomplete list.
<code>device.sec.TLS.FIPS.enabled</code> <code>site.cfg</code>	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.

(continued)Configuration File Enhancements

Parameter Template	Permitted Values
<code>sec.TLS.protocol.browser</code> <code>device.cfg, site.cfg</code>	<p>Controls the TLS Protocol used for Browser 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.</p> <p>TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2</p> <p>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.</p>
<code>sec.TLS.protocol.ldap</code> <code>device.cfg, site.cfg</code>	<p>The security 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.</p> <p>TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2</p>
<code>sec.TLS.protocol.sip</code> <code>device.cfg, site.cfg</code>	<p>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.</p> <p>TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2</p>
<code>sec.TLS.protocol.sopi</code> <code>device.cfg, site.cfg</code>	<p>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 when set to "SSLv2v3", otherwise the phone uses TLS 1.0.</p> <p>TLSv1_0 (default) SSLv2v3</p>
<code>sec.TLS.protocol.webServer</code> <code>device.cfg, site.cfg</code>	<p>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.</p> <p>TLSv1_0 (default) SSLv2v3, TLSv1_1 and TLSv1_2</p>

(continued)Configuration File Enhancements

Parameter Template	Permitted Values
<code>sec.TLS.protocol.xmpp</code> <code>device.cfg, site.cfg</code>	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
<code>sec.TLS.protocol.exchangeServices</code> <code>device.cfg, site.cfg</code>	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
<code>device.sec.TLS.protocol.syslog</code> <code>device.cfg, site.cfg</code>	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
<code>device.sec.TLS.protocol.prov</code> <code>device.cfg, site.cfg</code>	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".
<code>device.sec.TLS.protocol.dot1x</code> <code>device.cfg, site.cfg</code>	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
<code>feature.BSExecutiveAssistant.SimplifiedAssistant.enabled</code> <code>features.cfg</code>	0 (default) - Remove the pick call and barge-in soft keys from executive actions menu. 1 - Add the pick call and barge-in soft keys from executive actions menu.

(continued)Configuration File Enhancements

Parameter Template	Permitted Values
feature.BSExecutiveAssistant.SimplifiedExec.enabled features.cfg	0 (default) - Remove the call screening option from the executive settings menu. 1 - Add the call screening option from the executive settings menu.
feature.broadsoft.basicCallLogs.redial.enabled features.cfg	0 (default) - Disable the redial of last dialed number. 1 - Enable the redial of last dialed number.
feature.broadsoft.xsiDirSrchAllFields.enabled features.cfg	0 (default) – Search every field for the string entered in the basic search field to maximize results. 1– Search only the name field for the string entered into the basic search field.
feature.broadsoftdir.showDefaultSearch features.cfg	0 (default) – No contacts are displayed when the search box field is empty. 1 – Enables the user to view the initial list of contacts for an empty search box.
feature.broadsoft.xsi.callWaiting.enabled features.cfg	0 (Default) – Does not allow to manage the incoming calls during an active call. 1 – Allows to manage the incoming calls during and active call.
feature.broadsoftGroupDir.enabled features.cfg	0 (default) - Disable the BroadSoft Group Directory. 1 - Enable the BroadSoft Group Directory.
feature.broadsoftdir.enabled features.cfg	Simple search on Enterprise and Enterprise Common directories. 0 (Default) – Disables the Enterprise and Enterprise Common directories. 1 – Enables the Enterprise and Enterprise Common directories.
feature.broadsoftPersonalDir.enabled features.cfg	0 (Default) – Disables the Personal Directory. 1 – Enables the Personal Directory.
feature.deviceLock.enable features.cfg	0 (default) - Disable the device lock feature. 1 - Enable the device lock feature.
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.
VVXD60.base.mac new.cfg, dect.cfg	Configure the base MAC address from the provisioning server. NULL (default) string, maximum 12 characters

(continued)Configuration File Enhancements

Parameter Template	Permitted Values
<code>VVXD60.handset.maxCount</code> <code>new.cfg, dect.cfg</code>	Specify the maximum number of handsets that the VVX supports while pairing D60. 5 (default) 1 - 5
<code>feature.VVXD60.allowPairing</code> <code>new.cfg, features.cfg, dect.cfg</code>	None (default) - Users are not allowed to pair or unpair a base station from the VVX phone. Pairing - Users are allowed to pair the base station with the VVX phone, but unpairing is not allowed. Unpairing - Users are allowed to unpair the base station from the phone, but pairing is not allowed. Both - Users are allowed to pair and unpair the base station with the VVX phone.
<code>feature.twamp.enabled</code> <code>site.cfg</code>	0 (default) - Disable TWAMP protocol support. 1 - Enable TWAMP protocol support.
<code>twamp.port.udp.PortRangeEnd</code> <code>site.cfg</code>	Set the TWAMP UDP session max port range value. 60000 (default) 1024 - 65486
<code>twamp.port.udp.PortRangeStart</code> <code>site.cfg</code>	Set the TWAMP UDP session start port range value. 40000 (default) 1024 - 65485
<code>twamp.udp.maxSession</code> <code>site.cfg</code>	Set the maximum UDP session supported by TWAMP. 1 (default) 1 - 10
<code>log.level.change.twamp</code> <code>techsupport.cfg</code>	Set the TWAMP log level. 4 (default) 0 - 6
<code>log.level.change.nisvc</code> <code>techsupport.cfg</code>	The log level configuration for the ni service module. 2 (default) 0 - 6
<code>log.level.change.rssvc</code> <code>techsupport.cfg</code>	The log level configuration for resource service module. 2 (default) 0 - 6
<code>tone.localDtmf.onTime</code> <code>sip-interop.cfg</code>	Set the time in milliseconds DTMF tones play locally when the phone plays out a DTML tone sequence automatically. The time you set is also the minimum time the tone plays when manually dialing. 50 ms (default) 1 - 65535 ms

(continued)Configuration File Enhancements

Parameter Template	Permitted Values
<code>voIpProt.SIP.CID.request.sourceSipMessage</code> <code>sip-interop.cfg</code>	Specify from which SIP request message to retrieve caller ID information from. The values for this parameter depend on the preference you configure using <code>voIpProt.SIP.CID.sourcePreference</code> . NULL (default) - Remote party caller ID information from INVITE is used. INVITE PRACK ACK This parameter does not apply to shared lines.
<code>voIpProt.SIP.CID.response.sourceSipMessage</code> <code>sip-interop.cfg</code>	Specify which SIP response message to retrieve caller ID information from. The values for this parameter depend on the preference you configure using <code>voIpProt.SIP.CID.sourcePreference</code> . NULL (default) - Remote party caller ID information from INVITE is used. 100, 180, 183, 200 This parameter does not apply to shared lines.
<code>voIpProt.SIP.callee.sourcePreference</code> <code>sip-interop.cfg</code>	Specify the priority order for the sources of callee's ID information. The headers can be in any order. NULL (default) - Caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order. One of the following strings: From,P-Asserted-Identity Remote-Party-ID P-Asserted-Identity From,Remote-Party-ID
<code>voIpProt.SIP.caller.sourcePreference</code> <code>sip-interop.cfg</code>	Specify the priority order for caller's ID information. The headers can be in any order. NULL (default) - Caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order. One of the following strings: From,P-Asserted-Identity Remote-Party-ID P-Asserted-Identity From,Remote-Party-ID
<code>voIpProt.SIP.header.allow.enable</code> <code>sip-interop.cfg</code>	0 (default) - Do not add the ALLOW header in 18x provisional responses. 1 - Add the ALLOW header in 18x provisional responses.

(continued) Configuration File Enhancements

Parameter Template	Permitted Values
<code>dir.local.UIenabled</code> <code>features.cfg</code>	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 business media phones. 0 – Hides the Contact Directory and Favorites options in the Directories menu and the Dialpad. On VVX 500/501 and 600/601 business media phones, hides the Favorites quick access menu on the Home screen.
<code>reg.x.srtp.simplifiedBestEffort</code> <code>sip-interop.cfg</code>	0 (default) - SRTP is not supported. 1 - Negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported. This per-registration parameter overrides <code>sec.srtp.simplifiedBestEffort</code> .
<code>sec.srtp.simplifiedBestEffort</code> <code>sip-interop.cfg</code>	0 (default) - SRTP is not supported. 1 - Negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported.
<code>tone.dtmf.onTime</code> <code>sip-interop.cfg</code>	Set the time in milliseconds DTMF tones play on the network when DTMF tones play automatically. The time you set is also the minimum time the tone plays when manually dialing. 50 ms (default) 1 - 65535 ms
<code>voIpProt.SIP.assuredService.enable</code> <code>features.cfg</code>	0 (default) - Disable the Assured Service SIP feature. 1 - Enable the Assured Service SIP feature.
<code>voIpProt.SIP.assuredService.namespace</code> <code>features.cfg</code>	The namespace scheme used in SIP signaling. UCRdsn (default) dsn, drsn, UCRdsn, UCRdrsn, custom
<code>voIpProt.SIP.assuredService.precedenceThreshold</code> <code>features.cfg</code>	The minimum call priority required for a call to be considered as a precedence call. 2 (default) 1 - 10

Resolved Issues

The following table lists resolved issues in UC Software 5.6.0

Resolved Issues

Category	Issue ID	Description
API	VOIP-124359	When the base profile is set to Generic on a VVX 500 business media phone, the user is unable to dial numbers on the phone when Rest API is enabled.
Audio	VOIP-126069	In a BToE environment, the Rx audio on VVX business media phones is noisy when joining a meeting from a Skype for Business client.
Audio	VOIP-125774	When you place a call from Skype for Business consumer to a VVX business media phone registered with Skype for Business server, there is no audio from Skype for Business consumer's end.
Audio	VOIP-125338 VOIP-125195	Audio is intermittent, one way, and the calls drop when Office 365 is used in certain environments.
Audio	VOIP-123213	The dial tones on the VVX D60 handsets do not match the North American telephone dual tone standards, having a low pitch at 350Hz and 440Hz.
Boss-Admin	VOIP-126186	In a Skype for Business environment, the VVX 601 business media phone does not allow changing the ringtone type for Boss-Admin when flexible line key feature is enabled.
BToE	VOIP-125436	In Skype for Business environment, when the user password is changed using BToE, the VVX business media phone does not receive calls from Boss.
Calling	VOIP-127145	When switching between the headset and speaker phone using the button on VVX business media phone while connecting to a conference, the call is dropped.
Calling	VOIP-127014	The VVX business media phones get unregistered from the server and is unable to place or receive call for sometime after the phone reboots through check-sync method.
Calling	VOIP-126572	When the VVX D60 wireless handset initiates an outgoing PSTN call, ringback tone is not received.
Calling	VOIP-126538	When the VVX 410 business media phones registered to Skype for Business server are connected in a conference call, the phone restarts or stops responding.
Calling	VOIP-126508	The phones registered to Skype for Business server, having the presence feature disabled, results in presence subscription storm as the user navigates through call list and directories menu when placing or receiving calls.
Calling	VOIP-126241	In a Skype for Business call, when the VVX 400/411 series business media phones receive an audio call from Skype for Business client, and the Skype for Business client enables video, the call becomes audio only.

Resolved Issues

Category	Issue ID	Description
Calling	VOIP-125892 VOIP-116402	In certain environments, when an extension is dialed from the VVX business media phone during an ongoing conference call, the participant is not added to the call.
Calling	VOIP-125621 VOIP-124982	In some cases, the SIP 302 redirect messages are not handled correctly on a VVX business media phone, resulting in a failed call.
Calling	VOIP-121497 VOIP-121133	Lowering Maximum Transmission Unit (MTU) on VVX 1500 business media phone causes high load on session border controller (SBC) due to excessive fragmented packets.
Contact Directory	VOIP-125804	When you add Outlook Contacts that are set as Skype for Business Favorites, the contact's name is not displayed on the VVX business media phone.
Interop Broadsoft	VOIP-125890 VOIP-125097	The VVX business media phone sends an additional unsubscribe message to the BroadSoft server for BroadWorks hoteling event when pressing the GuestIn or GuestOut button.
Interop BroadSoft	VOIP-125747 VOIP-125744	In a specific BroadSoft deployment, the logging level of DNS log floods the customer log server.
Localization	VOIP-126176	The VVX 500/600 business media phones show no translation for Skype for Business Address Book Search.
Logging	VOIP-124387	In a Boss-Admin setup, Polycom VVX 3xx business media phones slow down due to excessive logging.
Network	VOIP-126460	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.
Network	VOIP-126147	The phone sends a REFER request before sending the INVITE to the conference server when the participants who barged-in to the call gets dropped.
Network	VOIP-126151	When an INVITE with no Session Description Protocol (SDP) is sent to the phone on hold, the phone sends a 200OK response with <code>sendrecv</code> instead of <code>sendonly</code> .
Provisioning	VOIP-127323	Polycom VVX 410 business media phone constantly crashes and reboots when using File Transfer Protocol Secure (FTPS) for provisioning server and the configuration logs for sip=0, btoe=0.
Registration	VOIP-127309	When the registration to primary server fails, the VVX business media phones remains unregistered.
Registration	VOIP-125204	In some environments, the VVX business media phone fails to register on the IPv6 network and generates an incorrect message via header.
Security	VOIP-126724	The VVX 500/600 business media phone fails to log in when registered to Skype for Business server using the built-in certificate.

Resolved Issues

Category	Issue ID	Description
Security	VOIP-127483	The Pin-Auth fails when VVX business media phones registered to Skype for Business Server on-premises using commercial certificates.
Security	VOIP-125514	The VVX business media phones negotiate with TLS 1.0 and does not provide an option to disable the TLS 1.0.
Security	VOIP-126480	The VVX business media phone's web server allows the use of unsecured encryption ciphers.
Software Update	VOIP-113255	During software upgrade from 5.2.x to 5.4.x, the VVX 500 and 600 series business media phones create core dumps.
User Interface	VOIP-121437	The VVX business media phone shows incoming calls as missed when they are picked on a German Skype for Business Client.
User Interface	VOIP-126509	The Polycom business media phones do not search for a contact name that starts with an umlaut character.
User Interface	VOIP-126726	The phone does not display contacts after failing to recover the root CA certificate placed in the configuration file.
User Interface	VOIP-126872	In certain environments, the VVX business media phone's caller ID information includes a phone context (Phone-Context = CloudConnectorDefaultProfile) during an incoming call.
User Interface	VOIP-127344	When performing an LDAP search in an Open LDAP directory, the VVX business media phone continues to display "Searching..." if the number of entries is greater than maximum page size.
User Interface	VOIP-127076	When searching for an exchange contact having umlaut character on VVX 500/600 business media phones, the phone displays an '?' entry for umlaut character in the search result.
User Interface	VOIP-127074	On a VVX business media phone, the highlighted position of a menu item is not retained in Flexible Line Key (FLK) customization screen when using *,# for navigation.
User Interface	VOIP-126700	Polycom VVX D60 wireless handset displays incorrect French translation.
User Interface	VOIP-126266	When a contact having Umlaut character is assigned to a key, the VVX 500/600 series business media phones display number instead of contact name.
User Interface	VOIP-126162	Polycom SoundStation IP 5000 phone has a memory leak issue that results in a medium-security vulnerability (CVSSv3 score = 5.4) in the Web Configuration Utility.
User Interface	VOIP-125000 VOIP-123634 VOIP-120843	The VVX 500 business media phone does not display the day, month, and date in single line.
User Interface	VOIP-126415	The VVX 601 business media phone registered with Skype for Business server online displays the PSTN contacts as telephone number instead of contact name when using flexible line keys.

Resolved Issues

Category	Issue ID	Description
User Interface	VOIP-126229	The VVX 600 business media phone registered with Skype for Business and Exchange server displays an overdue reminder pop-up message at the start time of the meeting even after dismissing the first reminder pop-up.
User Interface	VOIP-126145	In a Skype for Business environment, the phone continues to display the contacts synchronization message after registering with the Microsoft Exchange.
User Interface	VOIP-126023	The ring type option for Boss Lines on the VVX business media phone is not displayed when "Show Only Registration" is enabled using <code>up.regOnPhone</code> parameter.
User Interface	VOIP-125593	The web application launched using a soft key on the VVX phone consumes more time (~15 seconds) to load.
User Interface	VOIP-125447 VOIP-124945	The CA certificate is not displayed on the VVX business media phone's user interface or the Web Configuration Utility page when trying to import the device settings to another phone.
User Interface	VOIP-125538	The VVX D60 Handsets fails to display the lower-case letters when performing a simple search in the Corporate directory.
User Interface	VOIP-124864	The VVX 300 series business media phones do not display some Polish characters.
User Interface	VOIP-123461	The VVX business media phone's SSL Client Hello displays incorrect time for the GMT UNIX Time.
User Interface	VOIP-118087	The VVX 300 series business media phones do not display '+' even when the dial plan is configured to show '+' on call display.
User Interface	VOIP-116769	The VVX business media phone does not display the line key label properly on the top left corner of the Home screen.
Video	VOIP-120458	The video quality becomes poor after receiving an incoming call from RealPresence Trio system on the VVX business media phone.
Video	VOIP-125630	On the VVX business media phones, the video content is stretched initially in a video call and requires the user to go full screen and back for proper video scaling.

Get Help

For more information about installing, configuring, and administering Polycom products, refer to Documents and Downloads at [Polycom Support](#).

To find all Polycom partner solutions, see [Polycom Global Strategic Partner Solutions](#).

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