RELEASE NOTES Software 5.5.1 | November 2016 | 3725-42645-002A

Polycom® UC Software 5.5.1

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

Contents

What's New in Polycom UC Software 5.5.1

Polycom Unified Communications (UC) Software 5.5.1 is a release for Open SIP and Skype for Business deployments.

Polycom UC Software 5.5.1 for Skype for Business supports the following Polycom endpoints:

- VVX 201 business media phones
- 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
- SoundStructure VoIP Interface

Polycom UC Software 5.5.1 for Skype for Business supports the following Polycom accessories:

- VVX Expansion Module

Polycom UC Software 5.5.1 for Open SIP environments supports the following Polycom endpoints:

- VVX 101/201 business media phones
- 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 phones
- SoundStructure VoIP Interface

Polycom UC Software 5.5.1 for Open SIP environments supports the following Polycom accessories:

- VVX Camera
- VVX Expansion Module
- VVX D60 Wireless Handset and Base Station

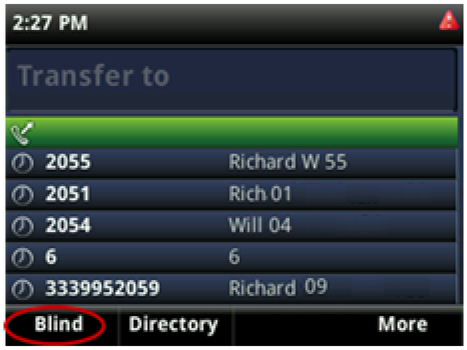
These release notes provide important information on software updates, phone features, and known issues.

What's New in Polycom UC Software 5.5.1

This release introduces the following new features.

New Call Transfer User Interface Option

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



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

End-user Access to Ethernet and DHCP Settings

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

Distribution List

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

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

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

Microsoft Quality of Experience (QoE) Monitoring Server Protocol

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

Device Lock

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

Polycom BToE PC Pairing

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

User Log Upload

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

Phone User Interface

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

Unified Contact Store

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

Web Sign-In for Online Deployments

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

Expanded Support for USB Headsets

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

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

Configuration File Enhancements

The following table lists configuration file enhancements that include new or changed parameters for this Polycom UC Software 5.5.1.

Configuration File Enhancements

Parameter ***Template***	***Permitted Values***
`attendant.callWaiting.enable` features.cfg	Specifies whether to use an acoustic indication for incoming attendant calls when there is already an active call on the phone. The ring type is set with parameter `attendant.callWaiting.ring`. 0 (default) – No acoustic indication is used for call waiting. 1 – An acoustic indication sounds for incoming attendant calls if there is already an active call on the phone.
`attendant.callWaiting.ring` features.cfg	Specifies the ring type used to notify the attendant when there is already an active call on the phone. This parameter is valid only if `attendant.callWaiting.enable` is set to `1` Silent (default) – No acoustic indication is provided. Beep – A beep tone plays when the phone is in an active call when it receives an attendant call. Ring – A ring tone specified by the parameter `attendant.ringType` plays when the phone is in an active call when it receives an attendant call.

Parameter ***Template***	***Permitted Values***
`btoe.PairingMode` features.cfg	Specifies how the phone pairs with a connected computer running Better Together over Ethernet (BToE). Auto (default) - The phone pairs with the computer automatically when a computer connected to the phone's PC port is running the BToE connector application. Manual - The phone generates a six-digit pair code that must be entered in the BToE connector application running on a computer connected to the phone's PC port.
`call.autoAnswer.playTone.enable` reg-advanced.cfg	Specifies whether the phone plays a tone when auto-answering a call. 1 (default) – Auto-answer tone is played. 0 – No auto-answer tone is played.
`call.autoAnswerMenu.enable` features.cfg	Specifies whether the Autoanswer menu displays on the phone to allow users to access the Autoanswer option. 1 (default) - Users can enable or disable the feature on the phone from the Autoanswer menu in Basic Settings. 0 – The Autoanswer menu is unavailable on the phone, and only the administrator can control the feature using configuration files.
`call.playLocalRingBackBeforeEarlyMediaArrival` sip-interop.cfg	Determines whether the phone plays a local ringback after receiving a first provisional response from the far end. 1 (default) - The phone plays a local ringback after receiving the first provisional response from the far end. If early media is received later, the phone stops the local ringback and plays the early media. 0 - No local ringback plays, and the phone plays only the early media received.
`call.switchToLocalRingbackWithoutRTP` sip-interop.cfg	Determines whether local ringback plays in the event that early media stops. 0 (default) – No ringback plays when early media stops. 1 – The local ringback plays if no early media is received.

<table>
<tr><th>Parameter
Template</th><th>Permitted Values</th></tr>
<tr><td><code>device.ipv6.icmp.ignoreRedirect</code>
device.cfg, wireless.cfg</td><td>Specifies whether Internet Control Message Protocol (ICMP) redirect messages are accepted.
0 (default) - ICMP redirect messages are accepted.
1 – ICMP redirect messages are ignored to avoid route changes.</td></tr>
<tr><td><code>device.ipv6.icmp.txRateLimiting</code>
device.cfg, wireless.cfg</td><td>Sets the maximum rate for sending ICMPv6 packets.
NULL (default)
0 – 60000 ms</td></tr>
<tr><td><code>device.net.etherStormFilterPpsValue</code>
device.cfg, site.cfg</td><td>Specifies the Packets per Second (PPS) value that triggers DOS storm prevention. The PPS index maps to a specific number of packets per second as shown here.
<table>
<tr><th>PPS Index</th><th>Packets per Sec.</th><th>PPS Index</th><th>Packets per Sec.</th></tr>
<tr><td>17</td><td>5887</td><td>29</td><td>19201</td></tr>
<tr><td>18</td><td>6400</td><td>30</td><td>21240</td></tr>
<tr><td>19</td><td>6911</td><td>31</td><td>23299</td></tr>
<tr><td>20</td><td>7936</td><td>32</td><td>25354</td></tr>
<tr><td>21</td><td>8960</td><td>33</td><td>27382</td></tr>
<tr><td>22</td><td>9984</td><td>34</td><td>29446</td></tr>
<tr><td>23</td><td>11008</td><td>35</td><td>31486</td></tr>
<tr><td>24</td><td>12030</td><td>36</td><td>35561</td></tr>
<tr><td>25</td><td>13054</td><td>37</td><td>39682</td></tr>
<tr><td>26</td><td>14076</td><td>38</td><td>42589</td></tr>
<tr><td>27</td><td>15105</td><td>39</td><td>56818</td></tr>
<tr><td>28</td><td>17146</td><td>40</td><td>71023</td></tr>
</table>
38 (default) – A PPS index of 38 triggers the storm filter.
17 - 40 – A PPS index between 17 and 40 triggers the storm filter.</td></tr>
<tr><td><code>device.net.etherStormFilterPpsValue.set</code>
device.cfg, site.cfg</td><td>This parameter controls whether the parameter <code>device.net.etherStormFilterPpsValue</code> is used for setting a Packets per Second (PPS) index to trigger DoS storm prevention.
0 (default) – The parameter <code>device.net.etherStormFilterPpsValue</code> is not used and storm filtering is not enabled by a specific PPS index.
1 – The parameter <code>device.net.etherStormFilterPpsValue</code> is used and storm filtering is enabled by a specific PPS index.</td></tr>
</table>

Parameter ***Template***	***Permitted Values***
`dir.corp.cacheSize`	The maximum number of entries that can be cached locally on the phone. 8 to 256 128 (default) **Note**: For VVX 101/201 phones, the permitted values are 32 to 64, and the default is 64.
`dir.corp.pageSize`	The maximum number of entries requested from the corporate directory server with each query. 8 to 64 64 (default) **Note**: For VVX 101/201 phones, the permitted values are 8 to 32, and the default is 16.
`exchange.meeting.parseWhen` applications.cfg	Specifies whether the phone uses number searching logic when finding additional numbers in the Skype for Business calendar. NonSkypeMeeting (default) - Number-searching logic is not used to find additional numbers in Skype for Business meeting calendar entries. Always - Number-searching logic is used to find additional numbers in Skype for Business meeting calendar entries.
`exchange.reconnectOnError` application.cfg	Determines whether the phone automatically attempts to reconnect to the Exchange server after encountering an error. 1 (default) – Phone attempts to reconnect to the Exchange server after an error. 0 – Phone does not attempt to reconnect to the Exchange server after an error.
`feature.callCenterCallInformation.enable` features.cfg	Specifies whether the phone displays call center and incoming call information in a pop-up message. 1 (default) – The phone displays call center and incoming call information in a pop-up message. 0 - The phone does not display call center and incoming call information in a pop-up message.
`feature.deviceLock.enable` features.cfg	1 (Default) - Device Lock for Skype for Business is enabled. 0 - Device Lock for Skype for Business is disabled.

Parameter *Template*	*Permitted Values*
`feature.lync.hideSignInSignOut` features.cfg	Specifies whether the Sign In and Sign Out soft keys appear on the Home screen and phone menus. 0 (default) – The Sign In and Sign Out soft keys appear in the user interface. 1 – The Sign In and Sign Out soft keys do not appear in the user interface, and users are not able to sign in or out. Administrators can sign in and out with the Web Configuration Utility.
`feature.lync.hideSignOut` features.cfg	Specifies whether the Sign Out soft key appears on the Home screen and phone menus. 0 (default) – The Sign Out soft key appears in the user interface. 1 – The Sign Out soft key does not appear in the user interface, and users are not able to sign out. Administrators can sign in and out with the Web Configuration Utility.
`feature.VVXD60.allowLineMappings` features.cfg, dect.cfg	Allows users to map lines on the VVX phone to a paired VVX D60 wireless handset from the Features menu. 0 (default) – Map Lines is available only on the Administrator menu. 1 – Map Lines is available on the Administrator menu and the Features menu.
`ipv6.mldVersion` device.cfg, site.cfg	Determines which version of Multicast Listener Discovery to use. 2 (default) –Multicast Listener Discovery version 2 is used. 1 –Multicast Listener Discovery version 1 is used.
`log.level.change.pec`	Set the debug log level for the Polycom Experience Cloud. 0-6 4 (default)
`net.interface.mtu6` site.cfg	Sets the Maximum Transmission Unit (MTU) value in bytes for the phone in IPv6 or dual stack mode. Note: IPv6 is qualified for an Open SIP eco-system. IPV6 is not qualified for a Skype for Business eco-system. 1500 bytes (default) 1280 – 1500 bytes

Parameter *Template*	*Permitted Values*
`pec.enabled`	Enables or disables Polycom Experience Cloud. 0 (default) – Disables the Polycom Experience Cloud. 1 – Enables Polycom Experience Cloud.
`pec.log.render.level`	Set the log level for Polycom Experience Cloud logs, which are uploaded to the cloud server set in the parameter `pec.server.uri`. 0-6 4 (default)
`pec.log.uploadPeriod`	Sets the period of time in minutes between Polycom Experience Cloud log uploads. 0 – 10080 15 (default)
`pec.server.uri`	Set the URI where logs for Polycom Experience Cloud are uploaded. String with 0 – 256 characters NULL (default) Note: If set to Null, https://pec.polycom.com is used by default.
`phoneLock.authorized.x.value` features.cfg	Specifies an authorized number that can be dialed when the device is locked using a Tel URI or SIP URI. Any numbers configured for this parameter display in an Authorized Calls list. For example, `phoneLock.authorized.1.value=”cwi57@cohovineyard.com”.`
`qos.ethernet.tcpQosEnabled` site.cfg	Specifies whether the phone sends configured Quality of Service (QoS) priorities for SIP on TCP. 0 (default) – Phone does not send configured QoS priorities for SIP on TCP. 1 – Phone sends configured QoS priorities for SIP on TCP.
`reg.x.auth.loginCredentialType` reg-advanced.cfg	Specifies the login type and user credentials required for the phone. LoginCredentialNone (default) – Microsoft login credentials are not accepted, and users are unable to log in with Microsoft credentials. usernameAndPassword – User must enter sign-in address, user name, domain, and password in the required format in order to sign in. extensionAndPIN - User must enter extension and PIN in order to sign in.

Parameter *Template*	*Permitted Values*
`reg.x.auth.useLoginCredentials` reg-advanced.cfg	Configures the phone to sign in users after the phone powers up. 1 (default) - SSI Login credentials, BToE Sign in, and Web Sign types are available for authentication with the server. 0 - SSI Login credentials, BToE Sign in, and Web Sign types are not available for authentication with the server.
`reg.x.auth.usePinCredentials` reg-advanced.cfg	Configures the phone to sign users in automatically after the phone powers up. To use this sign-in method, you must enable DHCP Option 43 or enable parameter `dhcp.option43.override.stsUri`. 1 (default) - PIN authentication sign in method is available for authentication on the server. 0 (default) - PIN authentication sign in method is not available for authentication on the server.
`reg.x.lineaddress` features.cfg	Set for private lines as well for the BroadSoft call park and retrieve scenarios. If the parameter `reg.x.address` is set to an address different than the call address of the number, use parameter `reg.x.lineaddress` to enable users to park and retrieve calls. **Note:** If there is no value specified for `reg.x.lineAddress`, `reg.x.address` is used.
`softkey.feature.simplifiedSignIn` lync.cfg	Specifies whether the Sign In and Sign Out soft keys display on the Home screen or on phone menu at Settings > Features > Microsoft Lync > Sign Out. 0 (default) - The Sign In and Sign Out soft keys display on the phone menu, but not on the Home screen. 1 - The Sign In and Sign Out soft keys display on both the Home screen and the phone menu.
`up.basicSettings.networkConfigEnabled` site.cfg	Allows you to give users access to the Ethernet and DHCP settings using the Basic menu. 0 (default) – Phone does not include Ethernet and DHCP settings on the Basic menu. 1 – Phone includes Ethernet and DHCP setting on the Basic menu.

Parameter ***Template***	***Permitted Values***
`up.BLFDefaultLineView` features.cfg	Specifies which view is displayed when the phone receives a BLF call. 0 (default) – The phone continues to display the call view when incoming BLF call information displays. 1 – The phone displays the line view when incoming BLF call information displays.
`up.btoeDeviceLock.timeOut` features.cfg	Sets the number of seconds after which the phone locks after a period of inactivity. 10 seconds (default) 0 – 40 seconds
`up.hideDateTimeWhenNotSet` features.cfg	Determines whether the date and time flash on the idle screen when the date and time have not been set. 0 (default) – The date and time display does not flash when the date and time have not been set. 1 – The date and time flash on the phone idle screen when the date and time have not been set.
`up.oneTouchBossAdmin` features.cfg	Enables the Boss and Admin for a line to view and pick up held calls on the boss's line by pressing the line key. 0 (default) – The user has to press and hold the line key to view and pick up held calls on the line. 1 – The user can press the line to view and pick up held calls on the line.
`up.onHookDialingEnabled` features.cfg	Specifies whether to enable on-hook dialing, which allows users to enter a number before dialing. 1 (default) – Enables on-hook dialing. 0 – Disables on-hook dialing.
`up.softkey.transferTypeOption.enabled` site.cfg	Specifies whether a soft key is added to the Call Transfer screen to allow the user to toggle between Blind and Consultative transfers. If the soft key is not added, the user can press and hold the Transfer soft key to choose a transfer type. 1 (default) - Adds a transfer type soft key to the Call Transfer screen. 0 – No transfer type soft key is added to the Call Transfer screen.

Parameter *Template*	*Permitted Values*
`voIpProt.OBP.dhcpv4.option` site.cfg	Specifies the DHCPv4 option code for Outbound proxy. 120 (default) 120 - 254
`voIpProt.OBP.dhcpv4.type` site.cfg	Specifies the DHCPv4 type for Outbound proxy. 0 (default) – Request an IP address from the DHCP server as the outbound proxy type. 1 – Request a string from the DHCP server as the outbound proxy type.
`voIpProt.OBP.dhcpv6.option` site.cfg	Specifies the DHCPv6 Option code used to get the Outbound Proxy server address from the DHCP server. 21 (default) 0-254 Note: IPv6 is supported by Polycom Open SIP servers. IPv6 is not supported when using Polycom phones with Skype for Business.
`voIpProt.SIP.callinfo.precedence.overAlertinfo` sip-interop.cfg	Specifies whether the call-info header with answer-after string has precedence over alert info. 0 (default) – The call-info header with answer-after string does not have precedence over alert-info. 1 – The call-info header with answer-after string has precedence over alert-info.
`voIpProt.SIP.considerTlsDnsEntriesOnly` site.cfg	Specifies whether TLS entries are considered in the auto-discovery process. 0 (default) – TCP and TLS entries are not considered in the auto-discovery process. 1 – Only TLS entries are considered in the auto-discovery process.
`voIpProt.SIP.renewSubscribeOnTLSRefresh` sip-interop.cfg	Specifies whether to refresh BroadSoft as-feature-event subscriptions when the phone re-registers. This parameter only applies when TLS transports are in use. 0 (default) – When a registration is refreshed, the BroadSoft as-feature-event subscription is also refreshed. 1 – Does not refresh the BroadSoft as-feature-event subscription when a registration is refreshed.

[1] Change causes phone to restart or reboot.

Phone Features and Licenses

The features and licenses required to operate the phones vary by 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, and *Yes** indicates that the phone requires you to purchase a feature license from Polycom to support a feature.

VVX Series Features and Licenses

Feature	*VVX 101*	*VVX 201*	*VVX 300/ 310*	*VVX 301/ 311*	*VVX 400/ 410*	*VVX401/ 411*	*VVX 500/ 501*	*VVX 600/ 601*	*VVX 1500*	*SoundStructure VoIP Interface*
Asian Languages	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Conference Management	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Customizable UI Background	No	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
Enhanced BLF	No	No	Yes	Yes	Yes	Yes	Yes	Yes	No	No
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
H.323 Video	No	No	No	No	No	No	Yes	Yes	Yes	No
Server-Based Call Recording	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
USB Call Recording	No	No	No	No	No	Yes	Yes	Yes	Yes	No
VQMon	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes (Audio only)	Yes (Audio only)	Yes (Audio only)	No

- Requires purchasing a feature license from Polycom.

Supported DHCP Sub-Options

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

DHCP Option 43 Configuration Options

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

Server Logging Levels for Skype for Business Server

In UC Software 5.5.1 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**

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

Release History

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

Release History

Release	*Release Date*	*Description*
5.5.1	September 2016	This release adds enhancements for distribution list, QoE, device lock, Polycom BToE manual pairing, user log upload, updated UI for VVX 500 and 600, unified contact store, web sign-in for online deployments.
5.5.0	June 2016	This release introduced support for BroadSoft Executive Assistant and Flexible Seating, TR-069, the 3GPP Technical Specification, the IPV6 protocol, Off-hook Call Status control, ability to lock the web configuration utility after failed login attempts, and user interface enhancements.
5.4.3	February 2016	This release introduced the Polycom VVX D60 Wireless Handset and VVX D60 Base Station.

Release	*Release Date*	*Description*
5.4.1	December 2015	This release includes support for the following features: • Introduced the Polycom VVX 301/311, 401/411, 501, and 601 business media phones. • Flexible line key customization for Lync (EFLK) • Master Key Identifiers (MKI) • Shared Line appearance on Lync • BToE for Windows 10 • Smart Search for Lync ABS • Support for simplified Chinese font on VVX 101
5.4.0A	September 2015	This release includes support for the following features: • Microsoft Office 365 and Skype for Business Online • Office365 and Skype for Business Provisioning and Manageability • Time and Date Initial Setup
5.4.0	May 2015	Added support for Alcatel-Lucent CTS features including • Advanced Conference • Shared Call Appearance with Bridge In • Visitor Desk Phone This release also included support for the following features: • Barge In on Busy Lamp Field Lines • DTMF Relay • SIP Instance • Comfort Noise • Opus Codec • DNS Server Address Override • Global Directory Synchronization • Basic Menu Lock • Additional features including user interface improvements and resolved known issues.
5.3.0	March 2015	Includes support for several Lync, BroadSoft, and Open SIP features.

Install UC Software 5.5.1

Consider the following information when installing or updating to Polycom UC Software 5.5.1.

- BToE 3.4.0 is required for use with UC Software 5.5.1. For best results, Polycom recommends deploying BToE 3.4.0 prior to deploying UCS 5.5.1. While BToE 3.4.0 is backwards compatible with previous versions of VVX firmware, Polycom does not recommend running previous versions of BToE software with UC Software 5.5.1.
- Before updating your VVX 1500 phone to UC Software 5.5.1, 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.

Download the Distribution Files

To download UC Software 5.5.1, 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.ld files for each phone model, enabling you to choose provisioning software for your phone model and maintain software versions for each model in the same root directory.

For general use, Polycom recommends using the split resource file that corresponds to the phone models for your deployment. To match the correct UC software resource file to your phone model, see the table Understand the Combined ZIP and Split ZIP Files. If you are provisioning your phones centrally using configuration files, download the corresponding resource file and extract the configuration files to the provisioning server, maintaining the folder hierarchy in the ZIP file.

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	x	✓
3111-40450-001.sip.ld	SIP application executable for VVX 201	x	✓
3111-46135-002.sip.ld	SIP application executable for VVX 300	x	✓
3111-48300-001.sip.ld	SIP application executable for VVX 301	x	✓
3111-46161-001.sip.ld	SIP application executable for VVX 310	x	✓
3111-48350-001.sip.ld	SIP application executable for VVX 311	x	✓
3111-46157-002.sip.ld	SIP application executable for VVX 400	x	✓
3111-48400-001.sip.ld	SIP application executable for VVX 401	x	✓

Distributed Files	*File Purpose and Application*	*Combined*	*Split*
3111-46162-001.sip.ld	SIP application executable for VVX 410	x	✓
3111-48450-001.sip.ld	SIP application executable for VVX 411	x	✓
3111-44500-001.sip.ld	SIP application executable for VVX 500	x	✓
3111-48500-001.sip	SIP application executable for VVX 501	x	✓
3111-44600-001.sip.ld	SIP application executable for VVX 600	x	✓
3111-48600-001.sip	SIP application executable for VVX 601	x	✓
2345-17960-001.sip.ld	SIP application executable for VVX 1500	x	✓
3111-33215-001.sip.ld	SIP application executable for SoundStructure VoIP Interface	x	✓
3111-17823-001.dect.ld	SIP application executable for VVX D60 Wireless Handset and Base Station	x	✓
sip.ld	Concatenated SIP application executable	✓	x
dect.ver	Text file detailing build-identification(s) for the VVX D60	✓	✓
sip.ver	Text file detailing build-identification(s) for the release	✓	✓
000000000000.cfg	Master configuration template file	✓	✓
000000000000-directory~.xml	Local contact directory template file. To apply for each phone, replace the (zeroes) with the MAC address of the phone and remove the ~ (tilde) from the file name	✓	✓
applications.cfg	Configuration parameters for microbrowser and browser applications	✓	✓
device.cfg	Configuration parameters for basic device configuration	✓	✓
features.cfg	Configuration parameters for telephony features	✓	✓
firewall-nat.cfg	Contains configuration parameters for telephony features	✓	✓
H323.cfg	Configuration parameters for the H.323 signaling protocol	✓	✓
lync.cfg	Contains Lync specific configuration parameters	✓	✓
pstn.cfg	Contains parameters for PSTN use	✓	✓

Distributed Files	*File Purpose and Application*	*Combined*	*Split*
reg-advanced.cfg	Contains configuration parameters for the line and call registration and advanced phone feature settings	✓	✓
reg-basic.cfg	Configuration parameters for line and call registration and basic phone settings	✓	✓
region.cfg	Configuration parameters for regional and localization settings such as time and date and language	✓	✓
sip-basic.cfg	Configuration parameters for the VoIP server and softswitch registration	✓	✓
sip-interop.cfg	Configuration parameters for the VoIP server, softswitch registration, and interoperability configuration	✓	✓
site.cfg	Configuration parameters that are set for each site	✓	✓
video.cfg	Configuration parameters for video connectivity	✓	✓
video-integration.cfg	Configuration parameters for SoundStation IP 7000 and Polycom HDX system integration	✓	✓
VVX-dictionary.xml	Includes native support for the following languages: • Arabic, UAE • Chinese, Traditional • Chinese, Simplified • Danish, Denmark • Dutch, Netherlands • English, Canada • English, United Kingdom • English, United States • French, Canada • French, France • German, Germany • Italian, Italy • Japanese, Japan • Korean, Korea • Norwegian, Norway • Polish, Poland • Portuguese, Brazil • Russian, Russia • Slovenian, Slovenia • Spanish, Spain	✓	✓

Distributed Files	*File Purpose and Application*	*Combined*	*Split*
	• Swedish, Sweden		
Welcome.wav	Startup welcome sound effect	✓	✓
LoudRing.wav	Sample loud ringer sound effect	✓	✓
Polycom-hold.wav	Sample ringer sound effect	✓	✓
Warble.wav	Sample ringer sound effect	✓	✓
polycomConfig.xsd	Master configuration file that contains the parameters and its values	✓	✓

Resolved Issues

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

Resolved Issues in UC Software 5.5.1

Category	*Issue No.*	*Release*	*Description*
Audio	VOIP-118272	5.4.2	Fixed an audio garbled issue while placing a call after incoming intercom call.
Audio	VOIP-119376	5.5.1	Intermittent audio loss and choppy audio no longer occurs during a VMR call using the Skype for Business client and a VVX phone connected using BToE.
Boss-Admin	VOIP-117416	5.5.1	The boss and delegates must be in the same Skype for Business environment, either both Online and both on-premises, to use the feature.
Boss-Admin	VOIP-120528	5.5.1	The delegate can now successfully transfer a call using Safe Transfer during BToE Playback.
Boss-Admin	VOIP-120679	5.5.1	In a Boss-Admin scenario, the delegate’s phone now rings when the dual role is removed and a FLK enabled on it.
Boss-Admin	VOIP-120753	5.5.0	Phones in a Boss-Admin group can receive up to five incoming calls at the same time.
BroadSoft	VOIP-118504	5.5.0	The phone no longer sends an incorrect authentication header when using SIP credentials to authenticate to a BroadSoft XSP address.
BToE	VOIP-111292		The BToE Icons are now removed when you change the value of NOTIFY_ICON_EN to 0. You can find this setting at to HKEY_LOCAL_MACHINE > SOFTWARE > Wow6432Node > Polycom > Polycom BToE Connector.
BToE	VOIP-114438	5.4.4	The SSH Host Key is no longer hardcoded on the phone when BToE is in use.
BToE	VOIP-115725	5.4.0	In a race condition where two participants (VVX500/BToE) off-hook a call for the Response Group, the call now connects to the first user and the other user goes to the ideal state smoothly.
BToE	VOIP-119007	5.4.4	A problem was resolved that caused the Polycom BToE Connector to log too much data at BTOE_DBG_DBG.
Busy Lamp Field	VOIP-109428		The phone correctly displays Busy Lamp Field (BLF) lines when more than 20 BLF lines are configured and the storm filter is enabled.
Busy Lamp Field	VOIP-117808	5.4.4	Busy Lamp Field (BLF) no longer fails after the first startup.
Calling	VOIP-117222	5.4.2	An issue was resolved that caused the loss of speed dial on the VVX 101 phone when lineKey.reassignment.enabled was enabled.

Category	*Issue No.*	*Release*	*Description*
Calling	VOIP-117314		Ringer and paging, hands-free and push-to-talk (PTT), and handset and headset loudness in phones can now be increased to full scale when a low power signal is received. For more details on the configuration parameters added for this issue, see Configuration File Enhancements.
Calling	VOIP-117664	5.4.4	E911 calls no longer fail when the phone is set to a static IP address.
Calling	VOIP-117691	5.4.4	A PSTN caller can now hear the ringback tone for a Skype for Business call through a Sonus gateway to a VVX phone.
Calling	VOIP-117747	5.4.2	An issue was resolved that caused the phone to dial the full URI if the URI contains the extension format x1234 instead of ext=1234.
Calling	VOIP-117924	5.4.3	An issue was resolved that caused the phone to lose all contacts in some specific environments.
Calling	VOIP-117951	5.4.2	Parameter voIpProt.SIP.callinfo.precedence.overAlertinfo was added to control whether a call-info header with answer-after string has priority over alert-info.
Calling	VOIP-118441	5.4.4	When an intercom call is manually answered during an active call, the phone now mutes the intercom call when ringAnswerMute is enabled.
Calling	VOIP-118609		New parameters attendant.callWaiting.enable and attendant.callWaiting.ring have been added to allow you to configure an acoustic call-waiting indication for attendant calls.
Calling	VOIP-118793		During an active call, if you navigate to the Call list, Favorites, or Directory and then press any digit, the screen will no longer exits and passes the dialed digit as a DTMF tone.
Calling	VOIP-118879	5.4.1	The phone now correctly shows the configured value on the server when the Call Forward Not Answered (CFNA) ring count is configured on the server.
Calling	VOIP-119043	5.4.2	When an admin terminates a call that was placed on hold by the boss, the call no longer randomly reappears as a held call on the boss' phone.
Calling	VOIP-119172	5.4.2	A problem was resolved that prevented dialing from the placed call list when the OPUS codec is enabled.
Calling	VOIP-119834	5.4.4	The phones no longer fail when forwarding a call to a number added by a user when the number has an appended domain.
Certificates	VOIP-118468	5.4.4	The VVX device certificate can now be sent via XSI when requested.
Configuration	VOIP-119497	5.4.1	Setting the device.sntp.gmtOffset parameter using the Web Configuration Utility's Import Configuration feature now works.
Contacts	VOIP-119544	5.4.3	The phone no longer loses contacts in some specific environments.

Category	*Issue No.*	*Release*	*Description*
Directory	VOIP-116937	5.5.0	Contacts added after the maximum contact limit was reached are now displayed automatically after some contacts are deleted from the displayed 200 contacts.
Directory	VOIP-118286	5.4.4	The phone no longer reboots during a call or when browsing the directory.
Directory	VOIP-119288	5.4.4 5.5.0	The following directory configuration changes apply to all VVX phones except VVX 1500: • Both the <MAC>-directory.xml and 000000000000-directory.xml files are downloaded at the time of a configuration update. • Download of directory files in case of checkSync will depend on parameter “voIpProt.SIP.specialEvent.checkSync.downloadDirectory”. • Download of directory files will no longer depend on dir.local.readonly. • For the User Login feature, directory file “username-directory.xml” will be considered in place of <MAC>-directory.xml.
General	VOIP-102685	5.4.4	VVX phones no longer take up to 3 minutes to restart an application after receiving NOTIFY check-sync.
General	VOIP-116674	5.5.0	The phone no longer uploads a core dump during a restart.
General	VOIP-116715		New parameter call.autoAnswer.playTone.enable has been added to allow you to specify whether to plan an auto-answer tone.
General	VOIP-116924		The VVX phones now complete blind transfers with the Competella switch board.
General	VOIP-117262	5.4.4	An issue was resolved that prevented presence from failing over to the new server.
General	VOIP-117293 VOIP-117294 VOIP-115022	5.4.3	The parameter reg.x.lineaddress can now be used to specify the line extension used for parking either private or shared lines. If the registration number specified by reg.x.address is different from the actual line address, configuring this parameter is required for using call park and retrieve.
General	VOIP-117505	5.5.0	The parameter device.Services.VoiceService.x.VoiceProfile.x.SIP.RegisterRetryInterval was removed from the TR-069 map.
General	VOIP-117527	5.5.1	Polycom phones do not support special characters ({ or }) in usernames.
General	VOIP-117678	5.5.1	The MWI tone is not played on Jabra GN9120 EHS headsets in on-hook mode as the headset discards this tone.
General	VOIP-117690	5.4.4	The phone now rejects the SIP INVITE and SIP NOTIFY on reception of SIP 400 Bad request.

Category	*Issue No.*	*Release*	*Description*
General	VOIP-117777	5.4.5	The following email address pattern is supported: {noformat} "[a-zA-Z0-9\\+\\._\\%\\-]{1,256}" "\\@" "[a-zA-Z0-9][a-zA-Z0-9\\-]{0,64}" "(" "\\." "[a-zA-Z0-9][a-zA-Z0-9\\-]{0,25}" ")+" {noformat}
General	VOIP-117819	5.4.2	The phone no longer locks up or reboots while placing a PSTN call at time registration or re-registration.
General	VOIP-117951		New parameter voIpProt.SIP.callinfo.precedence.overAlertinfo was added to control whether a call-info header with answer-after string has priority over alert-info.
General	VOIP-117989	4.0.11	Removed CA bundle(ca10.crt) for Web Server profile.
General	VOIP-118056	5.5.1	The phone no longer signs out and signs in automatically when it is left idle for more than 3 days.
General	VOIP-118108	5.4.4	An issue was resolved that caused the phone to reboot and created core dumps on incoming calls in some specific environments.
General	VOIP-118399	5.5.0	A problem was resolved that caused the phone to reboot when reading a large configuration file.
General	VOIP-118623 VOIP-118624	5.4.4	Audio loss no longer occurs on a VVX phone for around 8 seconds for call center calls through Anywhere365.
General	VOIP-118770	5.4.3	The phone now sends media attributes in SDP immediately when the Hold button is pressed.
General	VOIP-118791		VVX 101 and VVX 201 phones no longer remove Transfer and Hold when the parameter softkey.feature.basicCallManagement.redundant is set to 0.
General	VOIP-118873 VOIP-120659	5.4.3	A spelling error was corrected in parameter feature.VVXD60.allowLineMappings. The parameter name is no longer spelled with three p’s.
General	VOIP-118902	5.4.1	The VVX phone no longer fails to use Extension & PIN in a production environment when using dhcp.option43.override.stsUri.
General	VOIP-119045	5.5.1	The phone no longer goes into a reboot loop when failover to DVD from LLDP/CDP.

Category	*Issue No.*	*Release*	*Description*
General	VOIP-119250		Parameter voIpProt.SIP.renewSubscribeOnTLSRefresh was added for as-feature events. The parameter refreshes SUBSCRIBE together with re-REGISTER when TLS breaks.
General	VOIP-119894	5.4.2	An issue was resolved that prevented the phone from passing LLDP packets to a laptop connected to the phone's PC port.
General	VOIP-120146	5.4.4	Modified the parameter values for parameters dir.corp.cacheSize and dir.corp.pageSize to accommodate for VVX 101/201 phones.
General	VOIP-120234	5.5.0	VVX phones with factory default settings no longer require a manual reboot to redirect the phone to the local provisioning server when upgrading to 5.5.0.x using Zero Touch Provisioning.
General	VOIP-120277	5.4.4	The phone no longer takes 3 minutes to restart after receiving NOTIFY check-sync.
General	VOIP-120301	5.4.2	The phones no longer resync or download software during server maintenance.
General	VOIP-120512	5.5.0	VVX phones no longer upload core dumps when subscribing to attendant.uri in an Asterisk 13 environment.
General	VOIP-120609	5.4.2	Polycom devices no longer unintentionally resync and download new software during customer maintenance of their Edge system when resync is not applied to the phone.
General	VOIP-120673	5.5.0	VVX phones with default factory settings no longer require a manual reboot when upgrading to UC Software 5.5.0 or later.
Hardware	VOIP-117879	5.4.2	Ringer and paging loudness has been improved for better usability.
Language	VOIP-121165	5.4.4	"Wachtstand" for holding calls is now displayed properly in the call appearance window when the language is set to lcl.ml.lang="Dutch_Netherlands"
Languages	VOIP-121166	5.4.4	Sort in Spanish is now "Ordenar" instead of "Arreglar"
Logging	VOIP-119134	5.4.4	An issue was resolved that caused D60 wireless handset to print spurious benign EVENT 4 messages in the Phone Log.
Microbrowser	VOIP-117597	5.3.0	The microbrowser now restarts automatically when RAM usage is over 40% for more than 30 minutes. When this happens, the last web page displayed is not restored.
Microsoft	VOIP-118250	5.4.4	An issue was resolved that caused the phone to reboot during large Skype for Business conference calls.
Microsoft	VOIP-118294	5.5.1	The phone is creating a roster view with multiple self participants for every Meet Now. This is a Skype for business server issue.
Microsoft	VOIP-118301	5.5.1	The phone doesn't join an active call when paired with the Skype for Business client while the client is already in the call. This is a Skype for Business client issue.

Category	*Issue No.*	*Release*	*Description*
Microsoft	VOIP-118908	5.5.1	Exchange Calendar now works and syncs in 8 mins after re-registering after the network was down and the user is signed into Skype for Business on the phone via device pairing.
Microsoft	VOIP-119471	5.4.4	Local conference calling is now available in Skype for Business environments.
Microsoft	VOIP-119732	5.5.1	Entering the pound key (#) for a DTMF command is not working when an unanswered call is routed to Voicemail. This is a Skype for Business server issue.
Microsoft	VOIP-119798	5.4.4	The phone no longer reboots when attempting to join an internal Skype for Business meeting.
Microsoft	VOIP-120720	5.5.1	The Skype for Business Device Lock feature is now mutually exclusive with the Phone Lock feature.
Network	VOIP-113342		The number of daily network requests sent by the VVX system to the Exchange server has been reduced.
Network	VOIP-113993		New parameter qos.ethernet.tcpQosEnabled allows you to configure the phone to send configured QoS priorities for SIP on TCP.
Network	VOIP-117142		A computer connected to the PC port on the phone now experiences throughput at the speed of the LAN.
Network	VOIP-117472		The phone now fails over correctly when re-registration on failover is enabled and failover fallback mode is set to Registration.
Network	VOIP-118275	5.4.4	An issue was resolved that caused the VVX system to create a new TLS socket when a call was canceled shortly after it was dialed, causing the phone to lose registration.
Network	VOIP-118723	5.4.4	An issue was resolved that caused the NAT Keepalive message to be sent for only one registered line when the phone was configured for two lines.
Network	VOIP-119127	5.4.2	VVX Phones upon receiving INVITE with multiple diversion headers now displays the first diversion header when parameter voIpProt.SIP.header.diversion.list.useFirst is enabled.
Network	VOIP-119152	5.4.2	An issue with the duration timer used for failback to the primary server has been resolved.
Network	VOIP-119166	5.4.4	DNS queries are now sent shortly after the phone is powered on. This no longer results in NTP failures and the message "Time/date not set" does not display on the phone.
Network	VOIP-119287	5.4.4	The phones now correctly utilize contact headers that contain the addr parameter in a 301 response to an outbound INVITE.
Network	VOIP-120270	5.5.0	The phone now registers with IPv4 when using dual stack IPv4IPv6.

Category	*Issue No.*	*Release*	*Description*
Power Management	VOIP-119928	5.4.1	VVX 501/601 phones can now power a VVX Camera and a VVX Color Expansion module connected to the phone over IEEE 802.af source.
Registration	VOIP-119322	5.5.0	Phones configured for PIN authentication no longer unregister during system maintenance.
Server	VOIP-120069	5.5.1	Switching a Skype for Business call between a Jabra Evolve 40 headset and the VVX phone is inconsistent because of a server related issue.
Software Update	VOIP-118208	5.4.4	The phones now correctly request DHCP Option 144 after a firmware upgrade.
Software Update	VOIP-118994	5.4.4	An issue was resolved which caused phones to not to show the PIN Authentication option after upgrading software in certain environments.
User Interface	VOIP-112418		On systems using the Busy Lamp Field (BLF), the caller's name and number now scroll across the screen if the information is wider than the available screen space.
User Interface	VOIP-112625		The warning icon is no longer displayed on VVX phones after the administrator password is changed.
User Interface	VOIP-113621		New parameter feature.lync.hideSignInSignOut was added to allow you to hide the Sign In and Sign Out buttons on the user interface.
User Interface	VOIP-114845		Labels now split correctly in the user interface when alignment is set to Right or None. When text alignment is set to Left, labels may not correctly split.
User Interface	VOIP-115572	5.4.2	In Skype for Business environments, the message displayed upon sign-in is now properly displayed.
User Interface	VOIP-115884		Outbound calls from Multiple Appearance Directory Number (MADN) lines now correctly display the called number.
User Interface	VOIP-116825	5.5.0	On the Favorites screen, pressing the empty third and fourth soft keys incorrectly displays the Info screen.
User Interface	VOIP-116859		When Multiple Appearance Directory Number (MADN) is enabled, outbound calls now display the called number.
User Interface	VOIP-116943		Performance of the VVX 1500 phone has been improved to resolve problems that caused sluggishness on the phone's user interface.
User Interface	VOIP-117392		VVX phones now correctly display the caller ID for outbound calls from MADN lines.
User Interface	VOIP-117522		The phone now correctly display codes from 128 to 159 (€,ƒ„…†‡ˆ‰Š‹ŒŽ ’’""•—-˜™š›œžŸ).

Category	*Issue No.*	*Release*	*Description*
User Interface	VOIP-118285	5.4.2	An issue was resolved that caused LDAP query search results to fail to display all the fields if dir.corp.attribute.x.label values contain non-ASCII characters.
User Interface	VOIP-118792	5.4.4	An issue was resolved that caused an incorrect destination to display in the call list when registrations are configured with multiple line keys.
User Interface	VOIP-119073		New parameter up.basicSettings.networkConfigEnabled was added to allow users to access Ethernet and DHCP settings on the Basic menu.
User Interface	VOIP-119308	5.3.0	The Home screen no longer shows a blue background for a long period of time when a user presses the Back key from Recent list on a VVX 1500.
User Interface	VOIP-119433	5.3.0	An issue was resolved that caused the VVX 1500 Home screen to show a blue background when you pressed the Back key while viewing the phone list.
Video	VOIP-118435	5.4.4	The phone now handles the video call over SRTP or TLS, and the two-way video call works without any issue over TLS/SRTP.
VVX D60	VOIP-113704 VOIP-113521 VOIP-114469		The wireless handset occasionally displays as out of range and the signal strength is not shown for a few seconds before displaying again.
VVX D60	VOIP-120061	5.4.4	VVX D60 wireless handsets no longer fail when the dial string includes a pound (#) sign, which was causing feature access codes dialed on the wireless handset to fail.
VVX D60	VOIP-120297	5.4.4	VVX D60 wireless handsets now handle in-band DTMF properly when RFC 2833 is unavailable.
Web Configuration Utility	VOIP-118714 VOIP-118840	5.4.1	The phone now applies device certificates and device parameters from the phone's web interface.
Web Interface	VOIP-110074		You can now upload the device configuration files using the phone’s web interface.

Known Issues

The following table lists the known issues and suggested workarounds for UC Software 5.5.1.

Known Issues

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Busy Lamp Field	VOIP-114129	5.4.2	Busy Lamp Field contacts are not consistently updated on the Expansion Module.	
Calling	VOIP-116653	5.5.0	After a Barge-in call is placed on hold, the handset still displays options to Transfer and Blind Transfer the call.	
Calling	VOIP-116417	5.5.0	The VVX D60 wireless handset displays both parties of the conference call even though one of the parties has disconnected from the call.	
Calling	VOIP-99645	4.0.1B	When you place a call on the SoundStructure VoIP Interface while there is an incoming call, the incoming call is ignored and no longer rings if the new call is ended. You can still answer the incoming call until it disconnects.	
Calling	VOIP-116259	5.5.0	In Calendar Events with multiple phone numbers, the Dial Option does not list the numbers correctly.	
Expansion Module	VOIP-116348	5.5.0	On a phone with an Expansion Module connected, the first Expansion Module line is not cleared after you lock the device.	
Expansion Module	VOIP-116348	5.5.0	On a phone with an Expansion Module connected, the first Expansion Module line is not cleared after you lock the device.	
Hardware	VOIP-116899	5.5.0	A VVX phone in an active call using the Plantronics Blackwire C420-M USB headset is not able to answer an incoming call.	
Interoperability D60 Handset	VOIP-117097	5.5.0	On a VVX phone paired with two D60 handsets, the second handset is unable to place a call after ending an intercom call with the first handset.	

Category	*Issue No.*	*Release*	*Description*	*Workaround*
Interoperability TR069	VOIP-111332	5.5.0	If you schedule a file to download from the TR069 server and then disconnect the power cord from the phone one minute before the scheduled time, the file is not downloaded when you reconnect the power cord and power the phone on again.	
Network	VOIP-116151	5.5.0	The phone incorrectly sends the “Ethernet Frame Check Sequence Incorrect" message in remote packets.	
Registration	VOIP-115965	5.5.0	If you change the base station name on the VVX system and then unregister the D60 handset, the new base station name does not display on the handset.	Unregister the handset and then register it with the base station again.
SIP	VOIP-116412	5.5.0	Including the "&" character in a user's SIP URI prevents the user's status from changing.	
User Interface	VOIP-116471	5.5.0	When you edit a contact in the Local Directory, scrolling up does not work correctly.	
User Interface	VOIP-116895	5.5.0	The Back Soft key is seen on the microbrowser home screen.	
User Interface	VOIP-116353	5.5.0	On the VVX D60 wireless handset, the Silence key is incorrectly displayed for a waiting call.	
User Interface	VOIP-116211	5.5.0	The fonts in the user interface display incorrectly in Arabic for long names on the Expansion Module.	
User Interface	VOIP-114345	5.5.0	The Idle Browser does not display the HTTPS:// page.	
User Interface	VOIP-115472	5.4.4	Missed calls notifications do not disappear on the VVX D60 phone's main display.	
User Interface	VOIP-116826	5.3.0	On the Favorites screen, pressing the empty third and fourth soft keys incorrectly displays the Info screen.	
User Interface	VOIP-117145	5.5.0	In a call between the VVX phone, its paired handset, and another VVX phone, the handset incorrectly displays details about both phones after one of the phones drops from the call.	

Category	*Issue No.*	*Release*	*Description*	*Workaround*
User Interface	VOIP-116387	5.5.0	After restarting a VVX 500 phone with an expansion module and a headset attached, the “Digital headset attached” message does not appear.	
User Interface	VOIP-113852	5.5.0	Pressing the back arrow from the Contact Directory takes you to the idle screen instead of to the Directories Menu.	
User Interface	VOIP-114800	5.4.3	On the VVX D60 wireless handset name screen, the Back soft key displays instead of the Delete soft key after you enter a space.	
Web Interface	VOIP-113192	5.5.0	In the VVX system web interface Handset Settings, a mapped line is incorrectly listed twice.	
Web Interface	VOIP-113193	5.5.0	The VVX D60 web interface line management page does not show the default line.	

Updates to Previous Software Releases

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

What's New in Polycom UC Software 5.5.0

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

Note: Using configuration parameters to enable features

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

BroadSoft Executive-Assistant

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

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

Administrators can configure this feature using the following parameters:

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

This feature is not supported on the SoundStructure VoIP Interface.

Support for TR-069

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

Support for 3GPP Technical Specification

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

This release adds the following IMS feature enhancements:

- The call waiting ringback tone plays to inform you that the call is waiting at the far end.
- The SIP Response Code 199 (defined in RFC 6228) is supported.
- The Path extension header field in the SIP Register request message allows accumulating and transmitting the list of proxies between a user agent and Registrar server. The administrator can configure the parameter `reg.x.path` to enable or disable support for this header field for a specific line registration.
- The caller phone can support the p-early-media SIP header that determines whether the caller phone should play a network-provided media or its own media as a ringback tone. The administrator can configure the parameter `voIpProt.SIP.header.pEarlyMedia.support` to enable or disable support for this header field on the caller phone.
- The VQMon messages that are generated by the phone can contain service route information in SIP route headers. The administrator can configure the parameter `voice.qualityMonitoring.processServiceRoute.enable` to enable or disable this header field on the VQMon messages generated by a phone device.
- In a NAT network, a phone may need to send keep-alive messages to maintain the IP addresses mapping in the NAT table. The parameters `nat.keepalive.udp.payload` and `nat.keepalive.tcp.payload` are introduced to configure a customizable string as the payload of the UDP and TCP keep-alive messages.

BroadSoft Flexible Seating

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

This feature is not supported on the SoundStructure VoIP Interface.

Support for IPv6 Protocol

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

Off-Hook Screen View and In-Call Status Display

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

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

Microbrowser Support for VVX 201 Business Media Phone

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

Locking the Web Configuration Utility after Failed Login Attempts

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

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

Off-Hook Idle Browser

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

User Profile Login Enhancement

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

BroadWorks Call Decline

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

User Interface Themes

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

- `device.theme`

Minimum Ringer Volume

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

Password Protection for Editing Contacts Directory

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

Configuration File Enhancements

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

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

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

Configuration File Enhancements

Parameter ***Template***	***Permitted Values***
`call.shared.preferCallInfoCID` sip-interop.cfg	Specify whether Caller ID information is displayed. 0 (default) – Caller ID received from 200OK is ignored if NOTIFY message includes display information. 1 – Caller ID received from 200OK is displayed if NOTIFY message includes display information.

Parameter Template	Permitted Values
`call.shared.reject` sip-interop.cfg	For shared line calls on the BroadWorks server. 0 – The phone displays a Reject soft key to reject an incoming call to a shared line. 1 – The Reject soft key does not display.
`call.urlNumberModeToggling` site.cfg	Determines whether the phone uses Number mode or URL mode when a URL call is initiated. 0 (default) – URL mode is used for URL calls. 1 – Number mode is used for URL calls.
`device.dhcp.bootSrvUseOpt` device.cfg	Specifies the source for the boot server address for the phone. It can take values from 0 to 9. In IPv6 mode, the following values are applicable: • 4 - The phone uses the boot server configured through the Server menu. • 5 - The phone uses the boot server option provided through DHCPv6. In Dual Stack Mode (IPv4/IPv6 mode), the following values are applicable: • 6 - The phone uses the boot server configured through the Server menu. • 7 - The phone gets the boot server details from DHCPv6 option or the Option 66 on DHCP server. • 8 - The phone gets the boot server details through DHCPv6 or through the custom option configured on DHCP server for the provisioning. • 9 - The phone gets the boot server from DHCPv6 option or custom option or option 66 configured on DHCP server for the provisioning.
`device.feature.tr069.enabled` tr069.cfg	0 (default) – Disables the TR-069 feature. 1 – Enables the TR-069 feature.
`device.ipv6.icmp.echoReplies` device.cfg, wireless.cfg	0 (default) 1
`device.ipv6.icmp.genDestUnreachable` device.cfg, wireless.cfg	0 (default) 1
`device.ipv6.icmp.ignoreRedirect.set` device.cfg	0 (default) 1
`device.ipv6.icmp.txRateLimiting` device.cfg	0 6000 (default)

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

Parameter *Template*	*Permitted Values*
`device.theme.set` device.cfg	1 (Default) - The phone supports both the Classic and Modern theme. The `device.theme` parameter specifies which theme to use. 0 - The phone supports only Modern theme.
`device.tr069.acs.password` tr069.cfg	Sets the TR-069 ACS server password used to authenticate the phone. Null (default) String (256 maximum characters)
`device.tr069.acs.url` tr069.cfg	Sets the URL for the TR-069 ACS server. Null (default) URL (256 maximum characters)
`device.tr069.acs.username` tr069.cfg	Sets the TR-069 ACS server username used to authenticate the phone. PlcmSpip (default) String (256 maximum characters)
`device.tr069.cpe.password` tr069.cfg	Specifies the TR-069 CPE password, which authenticates a connection request from the ACS server. Null (default) String (256 maximum characters)
`device.tr069.cpe.username` tr069.cfg	Specifies the TR-069 CPE user name, which authenticates a connection request from the ACS server. PlcmSpip (default) String (256 maximum characters)
`device.tr069.periodicInform.enabled` tr069.cfg	Indicates whether the CPE must periodically send CPE information to ACS using the **Inform** method call. 0 (default) - Periodic **Inform** call is disabled. 1 - Periodic **Inform** call is enabled.
`device.tr069.periodicInform.interval` tr069.cfg	Specifies the time interval in seconds in which the CPE must attempt to connect with the ACS to send CPE information if `set to` TRUE. 18000 (default) 0 to 36000

Parameter Template	Permitted Values
`device.tr069.upgradesManaged.enabled` tr069.cfg	Indicates whether the ACS manages image upgrades for the phone or not. 0 (default) – The phone uses ACS or provisioning server for upgrade. 1 - The phone upgrades only from the ACS server.
`dir.local.passwordProtected` features.cfg	Specifies whether you are prompted for an Admin or User password when adding, editing, or deleting contacts in the Contact Directory. 0 (default) – No password prompt is displayed and pressing and holding the Line-key displays the Add or Edit menu. 1 – You are prompted for your Admin or User password while adding, editing, or deleting contacts in the Contact Directory.
`feature.BSExecutiveAssistant.enabled` features.cfg	0 (default) - Disables the BroadSoft Executive-Assistant feature. 1 - Enables the BroadSoft Executive-Assistant feature
`feature.BSExecutiveAssistant.regIndex` features.cfg	Specifies the registered line assigned to the Executive or Assistant for the BroadSoft Executive-Assistant feature. 1 (default) to 255 - The registered line for the Executive or Assistant. Note that a line icon for the role specified by the parameter `feature.BSExecutiveAssistant.userRole` displays even if you do not assign an Executive-Assistant service to a line in the BroadSoft Web Portal. Ensure that the services assigned to the line match the user role.
`feature.BSExecutiveAssistant.userRole` features.cfg	Specifies whether the phone is set to an Executive or an Assistant role. Note that an Executive and an Assistant line cannot be set on the same phone. ExecutiveRole (default) - Sets the registered line as an Executive line AssistantRole - Sets the registered line as an Assistant line
`feature.logUpload.enabled` features.cfg	1 (default) - Enable log uploads for Skype for Business. 0 - Disable log uploads for Skype for Business.
`hoteling.reg` features.cfg	1 (default) - Specifies the phone line on the host phone which hosts the guest line.

Parameter *Template*	*Permitted Values*
`httpd.cfg.lockWebUI.enable` site.cfg	Specifies whether web configuration login lock is enabled. 1 (default) – Enable the Web Configuration Login Lock feature. 0 - Disable the Web Configuration Login Lock feature.
`httpd.cfg.lockWebUI.lockOutDuration` site.cfg	Specifies how long the user is locked out of the Web Configuration Utility. 60 seconds (default) - The period of time during which the user is locked out of the Web Configuration Utility. The user can try logging in again after this time. 60 - 300 seconds
`httpd.cfg.lockWebUI.noOfInvalidAttempts` site.cfg	Specifies the number of failed login attempts after which the user is locked out of the Web Configuration Utility. 5 (default) 3 - 20
`httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration` site.cfg	Specifies time period during which the user must log in successfully to avoid being locked out of the Web Configuration Utility. The user can try logging in again after the lock-out duration set by `httpd.cfg.lockWebUI.lockOutDuration`. 60 seconds (default) 60 - 300 seconds
`lcl.ml.lang.japanese.font.enabled`[1] site.cfg	Specifies whether the Japanese Kanji font is enabled. This parameter applies to VVX 400, 401, 410, 411, 500, 501, 600, 601, and 1500. 0 (default) – The phone does not use Japanese Kanji character font. 1 - The phone displays Japanese Kanji character font.
`log.level.change.tr069` tr069.cfg	Sets the log levels for the TR-069 feature. 4 (default) 0 - 6
`nat.keepalive.tcp.payload` sip-interop.cfg	Sets a customizable string as the payload of a TCP keep-alive message. Note that the string value cannot be blank. CRLFCRLFCRLFCRLFCRLFCRLFCRLFCRLF(default) string

Parameter *Template*	*Permitted Values*
`nat.keepalive.udp.payload` sip-interop.cfg	Sets a customizable string as the payload of a UDP keep-alive message. CRLFCRLF (default) String Blank (for empty payload)
`prov.login.localPassword.hashed` site.cfg	Specifies whether the phone generates a custom digest hash to encrypt the user password. 0 (default) – The phone does not generate a custom digest hash to encrypt the user password. You must store the user password in `prov.login.localPassword`. 1 – The phone generates a custom digest hash to encrypt the user password and store it.
`prov.login.password.encodingMode` site.cfg	Configures the default Encoding mode for the text in the password field on the User Login screen. 123 (default) Abc ABC Abc
`prov.login.useProvAuth` site.cfg	Specifies whether phones use server authentication. 0 (default) – The phones do not user server authentication. 1 – The phones use server authentication.
`prov.login.userId.encodingMode` site.cfg	Configures the default Encoding mode for the text in the User ID field on the User Login screen. abc (default) ABC Abc 123
`reg.x.header.pEarlymedia.support` reg-advanced.cfg	Specifies whether the line supports the p-early-media header. 0 (Default) – The p-early-media header is not supported on the specified line registration. 1 – The p-early-media header is supported by the specified line registration.

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

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

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

Parameter Template	Permitted Values
`up.OffHookIdleBrowserView.enabled` features.cfg	Enables the microbrowser and associated soft keys to continue to display on the phone when the phone goes off-hook. 0 (Default) – The idle browser does not display on screen after the phone goes off-hook. 1 – The idle browser continues to display on screen after the phone goes off-hook.
`up.OffHookLineView.enabled` features.cfg	Specifies the default user interface displayed when the phone goes off-hook. 0 (default) – Home Screen displays when the phone goes off-hook. 1 – Line Screen displays when the phone goes off-hook.
`up.ringer.minimumVolume` site.cfg	Configure the minimum ringer volume. This parameter defines how many volume steps are accessible below the maximum level. 16 (default) – The full 16 steps of volume range are accessible. 1-15 0 – Ring volume is not adjustable by the user and the phone uses maximum ring volume. Upon bootup, the volume is set to ½ the number of configured steps below the maximum (16). So, if the parameter is set to 8, on bootup, the ringer volume is set to 4 steps below maximum.
`voice.cn.hs.attn` site.cfg	Sets the attenuation of the inserted comfort noise in dB, where smaller values insert louder noise. The default value 30 is quite loud. This parameter is used only when voice.cn.hs.enable is set to 1. 30 dB (default) 3 – 90 dB
`voice.cn.hs.enable` site.cfg	Specifies whether Comfort Noise (CN) is added to the transmit path of the handset. This feature should only be used when users at the far end perceive that the phone has gone "dead" when the near-end user stops talking. 0 (default) – No Comfort Noise is added. 1 – Comfort Noise is added to the handset.

Parameter *Template*	*Permitted Values*
`voice.plcCnEnable` site.cfg	Specifies whether the existing G.711 Appendix 1 Packet Loss Concealment (PLC) process is augmented by adding Comfort Noise (CN) during an extended loss. This prevents the synthesized concealment audio from decaying to silence. 0 (default) – No Comfort Noise is added. 1 – Comfort Noise is added.
`voice.plcCnGain` site.cfg	Specifies the gain applied to the synthesized Packet Loss Concealment (PLC) comfort noise in dB. Adjusting the PLC CN gain may be useful when interoperating with endpoints whose background noise is not well matched to the CN synthesis algorithm. This parameter is used only used when `voice.plcCnEnable` is 1. 0 (default) -20 – 20 dB
`voice.qoe.event.lossrate.threshold.bad` features.cfg	Defines the threshold for the network loss rate. Total packets lost for an interval/total packets expected for the interval *256 as stated in RFC 2611, section 4.7.1. 38 (default) - Approximately a 15% packet loss. 0 to 100
`voice.qoe.event.lossrate.threshold.poor` features.cfg	Defines the threshold for the network loss rate. Total packets lost for an interval/total packets expected for the interval *256 as stated in RFC 2611, section 4.7.1. 25 ms (default) - Approximately a 10% packet loss. 0 to 100
`voice.qoe.event.networkmos.threshold.bad` features.cfg	Defines the threshold for Network MOS using the average of MOS-LQO wideband, as specified by [ITUP.800.1] section 2.1.2, based on the audio codec used and the observed packet loss and inter-arrival packet jitter. 19 (default) - MOS score of 1.9 10 - 50 - MOS score between 1 - 5 networkMOS > 2.9 signifies good quality networkMOS > 2.9 < 1.9 signifies poor quality networkMOS < 1.9 signifies bad quality

Parameter *Template*	*Permitted Values*
`voice.qoe.event.networkmos.threshold.poor` features.cfg	Defines the threshold for Network MOS using the average of MOS-LQO wideband, as specified by [ITUP.800.1] section 2.1.2, based on the audio codec used and the observed packet loss and inter-arrival packet jitter. 29 (default) - MOS score of 2.9 10 - 50 - MOS score between 1 - 5 networkMOS > 2.9 signifies good quality networkMOS > 2.9 < 1.9 signifies poor quality networkMOS < 1.9 signifies bad quality
`voice.qualityMonitoring.processServiceRoute.enable` features.cfg	Specifies whether the SIP route headers for the VQMon messages generated by the phone contain service route information. 0 (default) – The VQMon messages generated by the phone do not contain service route information in SIP route headers. 1 – The VQMon messages generated by phone, contain service route information, if available, in SIP route headers.
`voIpProt.server.x.specialInterop` sip-interop.cfg	Specifies the server-specific feature set supported for all line registrations. VVX 101 = Standard GENBAND GENBAND-A2 ALU-CTS VVX 201 = Standard GENBAND GENBAND-A2 ALU-CTS ocs2007r2 lync2010 All other phones = Standard GENBAND GENBAND-A2 ALU-CTS ocs2007r2 lync2010 lcs2005

Parameter ***Template***	***Permitted Values***
`voipProt.SIP.anat.enabled` sip-interop.cfg	Enables or disables Alternative Network Address Types (ANAT). 0 (default) - ANAT is disabled. 1 - ANAT is enabled.
`voIpProt.SIP.header.pEarlyMedia.support` sip-interop.cfg	Specifies whether the caller phone supports the p-early-media header. 0 (Default) – The p-early-media header is not supported by the caller phone. 1 – The p-early-media header is supported by the caller phone.
`voIpProt.SIP.IMS.enable` sip-interop.cfg	Configures support on the phone device for IMS features that are introduced in UC Software 5.5.0 or later. This parameter is applicable for all registered or unregistered SIP lines on the phone. 0 (Default) – Phone cannot support IMS features that are introduced in UC Software 5.5.0 or later. 1 – Phone supports IMS features that are introduced in UC Software 5.5.0 or later.
`voIpProt.SIP.looseContact` sip-interop.cfg	Configures addition of the ephemeral port parameter to the contact header. 0 (default) - The ephemeral port is added to the contact header in TLS case. 1 – The port parameter is not added to the contact header or SIP messages.
`voIpProt.SIP.regevent` reg-advanced.cfg	Configures subscription of all phone lines on a phone to registration event notifications from the SIP server along with related information. When enabled, this parameter configuration is overridden by the `reg.x.regevent` parameter, which is configuration for a specific phone line. 0 (default) – The phone is not subscribed to notifications for all phone lines. 1 – The phone is subscribed to notifications for all phone lines.
`voIpProt.SIP.rejectNDUBInvite` reg-advanced.cfg	Specifies whether the phone accepts a call for all registrations in case of a Network Determined User Busy (NDUB) event advertised by the SIP server. 0 (default) – Phone rejects the call with a 603 Decline response code. 1 – Phone accepts the call.

Parameter Template	Permitted Values
voIpProt.SIP.specialEvent.checkSync.downloadCallList site.cfg	Specifies whether the phone downloads the current user's call list when a check-sync event NOTIFY message is received from the server. 0 (default) – Call list is not downloaded after receiving a check-sync event in the NOTIFY message. 1 – Call list is not downloaded after receiving a check-sync event in the NOTIFY message.
voIpProt.SIP.supportFor199 sip-interop.cfg	Specifies whether the phone supports the 199 response code. For details, see the RFC 6228, Session Initiation Protocol (SIP) Response Code for Indication of Terminated Dialog. 0 (default) – The phone does not support 199 response code. 1 – The phone supports the 199 response code.

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

Supported DHCP Sub-Options

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

DHCP Option 43 Configuration Options

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

Resolved Issues

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

Resolved Issues in UC Software 5.5.0

Category	*Issue No.*	*Release*	*Description*
Audio	VOIP-116379	5.4.1	Using Plantronics Voyager Legend UC no longer causes any abrupt call drops on VVX phones.
Audio	VOIP-113375		An issue was resolved that caused audio interruption on the Plantronics headset when a fourth caller tries to join a local three-way conference and then cancels.
Audio	VOIP-111806		Using the 3CX call park feature with the TCP trunk no longer causes one-way audio and no longer prevents unparking a call.

Category	*Issue No.*	*Release*	*Description*
Audio	VOIP-105505		A problem was resolved that caused audio drop when an attended transfer is triggered with the Competella Attendant Console.
Audio	VOIP-112844	5.3.1	When you enable the soft key using ESK, the user can access and launch the browser by pressing the soft key configured for the micro browser.
Busy Lamp Field	VOIP-115996		On phones with call waiting disabled, Busy Lamp Field activity no longer causes call waiting tones to be played.
Busy Lamp Field	VOIP-112438	5.3.1	A problem was resolved that caused Busy Lamp Field activity to trigger call waiting tones on phones where call waiting was disabled.
Calling	VOIP-115446		Blind transfer with SLA line and with `exposeAutoHold` enabled is now working as expected
Calling	VOIP-115285		The parameter `call.shared.preferCallInfoCID` was added to enable configuring whether Caller ID information is displayed.
Calling	VOIP-114466		A Polycom VVX phone configured to use Simultaneous Ring Personal no longer rings for an incoming call when Do Not Disturb is enabled.
Calling	VOIP-114287		An issue was resolved that caused an incoming click-to-dial call to play an incorrect tone.
Calling	VOIP-113925		Transferring an internal call between Polycom VVX phones when using the NUANCE dial-by-voice system now works as expected.
Calling	VOIP-113922		Joining a PSTN user to conference call now works on Skype for Business Online.
Calling	VOIP-113478	5.4.1	When the SoundStructure VoIP Interface is in a call, sending a "set voip_send VoIP Out" command to the SoundStructure no longer causes the call to disconnect. Pressing a digit on a Polycom Touch Control paired with the SoundStructure during a call now works correctly.
Calling	VOIP-112886		Line seize behavior for accessing voicemail using Enhanced Feature Keys (EFK) has been improved.
Calling	VOIP-111991		Parameter `call.urlNumberModeToggling` now allows you to specify whether the phone uses Number mode or URL mode when a URL call is initiated.
Calling	VOIP-109593		The parameter `call.urlNumberModeToggling` was added to resolve a problem with URL dialing.
Calling	VOIP-109311		The phone now correctly sends “user=phone” in the invite message when a user enters a number that ends with “#” or “*”.
Calling	VOIP-107290		The phone now ignores any unrecognized parameters included in check-sync messages.

Category	Issue No.	Release	Description
Calling	VOIP-115425		New parameter `call.shared.reject` was added to allow you to configure phones to display a Reject soft key for calls on a shared line.
Calling	VOIP-116273	4.0.9	Phones now use the contact URI and TELURI in the request line of BYE message, so calls end correctly when the `reg.1.telUri` is enabled or disabled.
Calling	VOIP-116228	5.4.2	Using blind transfer for calls to Exchange Auto attendant in a Skype for Business Online environment now works correctly.
Calling	VOIP-116207	5.4.2	An issue was resolved that caused a core dump after pressing Transfer and the extension.
Calling	VOIP-112294	5.4.0	An issue was resolved that was caused when an Inbound call was transferred, conferenced, and then transferred again.
Calling	VOIP-111987	4.0.8	The phone now uses the blind transfer behavior from the Enhanced Feature Key (EFK) soft keys and sends a HOLD message before the REFER message.
Contact Directory	VOIP-110651		Phones no longer reboot if a contact is selected and dialed within two seconds of receiving the first results in a Corporate Directory search.
Contact Directory	VOIP-110199		VVX 1500 integration with RPRM has been improved for IP, H323, E164, and Annex-O Phonebook storing and dialing.
Contacts	VOIP-115334		Parameter `voIpProt.SIP.looseContact` was added to control whether an ephemeral port is added to the contact header in a TLS environment.
Directory	VOIP-113115	5.4.2	The Contact Directory is now uploaded when sent check-sync;upload=directory is set.
General	VOIP-109359		EFK configured for dialing a number from shared Line 1 now works as expected, allowing users to dial out from Line 1.
General	VOIP-114622		The default User ID encoding mode for parameters `prov.login.userId.encodingMode` and `prov.login.password.encodingMode` was changed to abc/ASCII.
General	VOIP-113119	5.4.2	A problem was resolved that caused the Boss phone to reboot in a Boss-Admin situation where phones were running version 5.4.X software.
General	VOIP-111603	5.4.0	If the top of the route list's transport is UDP, phone now checks if it set by default or from the record route header and uses the same default transport mechanism for acknowledgement.
General	VOIP-111357	5.4.1	If the phone receives a 407 from the BYE message, it now responds adding the proxy-authorization header with credentials.

Category	*Issue No.*	*Release*	*Description*
General	VOIP-110472	5.2.1	VVX Keys are now optimized for responsiveness, speed, and stability even after a long period of uptime until phone is rebooted
General	VOIP-110017	5.4.1	DHCP stability issues on the VVX 310 phone have been resolved.
General	VOIP-113036		Several security configuration parameters were added to configure the phone to prompt users for SIP credentials at login. These credentials are then used for all SIP authorization. These parameters include: `prov.login.useProvAuth`, `voIpProt.SIP.specialEvent.checkSync.downloadCallList`, `prov.login.userId.encodingMode`, and `prov.login.password.encodingMode`.
Interoperability BroadSoft	VOIP-113154		Users can now search the BroadSoft directory using either the first name or the last name.
Interoperability BroadSoft	VOIP-109598	5.4.0	The star (*) and pound (#) symbols now display in the search field in the BroadSoft Directory.
Interoperability GENBAND	VOIP-115465	5.4.0	When saving a Genband Global Address Book (GAB) to the phone contact list, the contact's phone number is now saved correctly.
Interoperability GENBAND	VOIP-113314	5.4.1	A buddy's presence status is now updated on the Home screen when the parameter `voIpProt.SIP.presence.nortelShortMode` is set to True and the parameter `dir.local.serverFeatureControl.method` is set to GENBANDSOPI.
Interoperability GENBAND	VOIP-109599	5.4.1	On VVX phones, users can now watch buddies set in the GENBAND Personal Address Book when the parameter `feature.presence.enabled` is set to 1.
Interoperability Microsoft	VOIP-111382		Improvements have been made for Outlook calendar event synchronization.
Interoperability Microsoft	VOIP-113865	5.4.2	Stability issues in certain Lync or Skype for Business environments have been addressed.
Interoperability Microsoft	VOIP-111093	5.4.1	VVX Phones on Office 365 are now able to re-dial a number previously dialed using a Lync client pinned contact.
Interoperability Skype for Business	VOIP-115263	5.4.2	When you use the Boss-Admin for Skype for Business feature, only the Boss now gets the notification email regarding the admin's activity on Boss Number.
Microbrowser	VOIP-110527	5.3.1	The microbrowser now correctly displays the local time when the phone is set to the Lync profile.
Network	VOIP-108242		An issue was resolved that prevented the VVX phones from synchronizing after an interruption in Exchange connectivity.

Category	*Issue No.*	*Release*	*Description*
Network	VOIP-111998		Enabling SSLv3 on the LDAP server and disabling SSLv3 on the phone no longer causes issues on the phone.
Network	VOIP-115990	4.0.8	A problem was resolved that stopped NAT keep-alive messages when the provisioning server applies a firmware upgrade.
Network	VOIP-113928	5.4.1	VVX phones with edge registrations using an Audio codes gateway now negotiate ICE correctly.
Registration	VOIP-115741	5.4.2	An issue was resolved that caused the phone to unregister when ending an invalid URI call in a Lync environment.
Registration	VOIP-113016	5.4.1	When unregistered or powered off, the phone now correctly sends a notification event to unsubscribe from presence. When it registers, the phone now sends a notification event to subscribe for presence.
Reporting	VOIP-112424	5.4.0	The phone now displays the correct NLR and MOSQ when the phone generates DTMF in the generated VQMoN reports.
Reporting	VOIP-111764	5.4.1	Accurate overall Mean Opinion Scores (MOSs) are now created when there are several Synchronization Source range allocation (SSRC) changes that could occur as a result of codec changes. The phone will trigger a VQMon report as soon as an SSRC change is reported by DSP.
Reporting	VOIP-110308	5.4.0	The phone now displays the correct NLR and MOSQ when the phone generates DTMF in the generated VQMoN reports.
Security	VOIP-115481		The password used to authenticate to the GENBAND server (set with `parameter dir.corp.alt.password`) is now hidden in the configuration export.
Security	VOIP-110213		Multiple Denial of Service vulnerabilities in OpenSSL have been resolved.
Security	VOIP-109345		You can now use the parameter `dir.local.passwordProtected` to specify whether users are prompted for an Admin or User password when adding, editing, or deleting contacts from the Contact Directory.
Security	VOIP-113463		Parameters `sec.TLS.profile.webServer.cipherSuiteDefault` and `sec.TLS.webServer.cipherList` were added to allow configuration of the cipher suites for the web server profile.
Software Update	VOIP-113590		The user now remains signed in on the phone after upgrading the software.
Software Update	VOIP-113298	5.4.2	A problem was resolved that caused a problem when upgrading the phone from the Polycom hosted server on the phone's web interface.
User Interface	VOIP-115821		Parameter `lcl.ml.lang.japanese.font.enabled` was added to enable Japanese Kanji characters to display correctly.

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

Get Help

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

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

- Polycom VVX 101
- Polycom VVX 201
- Polycom VVX 300 and 310
- Polycom VXX 301 and 311
- Polycom VVX 400 and 410
- Polycom VVX 401 and 411
- Polycom VVX 500
- Polycom VVX 501
- Polycom VVX 600
- Polycom VVX 601
- Polycom VVX 1500
- Polycom VVX Camera
- Polycom VVX Expansion Modules
- Polycom VVX D60 Wireless Handset
- Polycom SoundStructure

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

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

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