RELEASE NOTES

Software 5.5.0 | May 2016 | 3725-42645-001A

# Polycom® UC Software 5.5.0

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

# Contents

# What’s New in Polycom UC Software 5.5.0

Polycom Unified Communications (UC) Software 5.5.0 is a release for all open SIP platforms. Note that UC Software 5.5.0 has not been qualified by Microsoft to use in Lync or Skype for Business deployments. Polycom will not support UC Software 5.5.0 use in Lync or Skype for Business deployments.

Polycom UC Software 5.5.0 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 5.5.0 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.

# New Features

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 specfifications 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 maxiumun level.

## Password Protection for Editing Contacts Directory

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

# 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* | *Sound Structure 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 | Yes | No |
| Enhanced Feature Keys | Yes | Yes | Yes | Yes | Yes | Yes | Yes | Yes | Yes | No |
| H.323 Video | No | No | No | No | No | No | Yes | Yes | Yes | No |
| 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.

# 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***<br>***Template*** | ***Permitted Values*** |
|---|---|
| call.shared.preferCallInfoCID<br>sip-interop.cfg | Specify whether Caller ID information is displayed.<br>0 (default) – Caller ID received from 200OK is ignored if NOTIFY message includes display information.<br>1 – Caller ID received from 200OK is displayed if NOTIFY message includes display information. |
| call.shared.reject<br>sip-interop.cfg | For shared line calls on the BroadWorks server.<br>0 – The phone displays a Reject soft key to reject an incoming call to a shared line.<br>1 – The Reject soft key does not display. |
| call.urlNumberModeToggling<br>site.cfg | Determines whether the phone uses Number mode or URL mode when a URL call is initiated.<br>0 (default) – URL mode is used for URL calls.<br>1 – Number mode is used for URL calls. |

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

| *Parameter*<br>*Template* | *Permitted Values* |
|---|---|
| `device.net.ipv6AddrDisc`<br>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.<br>1 (Default) -IPv6 global address and options are configured from DHCPv6.<br>2 - IPv6 global address is configured using prefixes received from Router Advertisements (RA) and options are configured from stateless DHCPv6.<br>0 - IPv6 global address and options must be configured manually. |
| `device.net.ipv6Address`<br>device.cfg, site.cfg | Specify a valid global IPv6 unicast address for the phone.<br>Null (default) |
| `device.net.ipv6Gateway`<br>device.cfg, site.cfg | Specify theIPv6 address of the default gateway for the phone.<br>Null (default) |
| `device.net.ipv6LinkAddress`<br>device.cfg, site.cfg | Specifies a valid Link Local IPv6 address for the phone.<br>Null (default) |
| `device.net.ipv6PrivacyExtension`<br>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.<br>0 (Default) - IPv6 global and link local addresses are in EUI-64 format.<br>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`<br>device.cfg, site.cfg | Specifies a valid Unique Local IPv6 address (ULA) for the phone.<br>Null (default) |
| `device.net.preferredNetwork`<br>device.cfg, site.cfg | Specify IPv4 or IPv6 as the preferred network in a Dual Stack mode.<br>1 (default) - Specifies IPv6 as a preferred network.<br>0 - Specifies IPv4 as a preferred network. |
| `device.theme`<br>device.cfg | Modern (default) - The phone uses the Modern theme.<br>Classic - The phone uses the Classic theme. |
| `device.theme.set`<br>device.cfg | 1 (Default) - The phone supports both the Classic and Modern theme. The `device.theme` parameter specifies which theme to use.<br>0 - The phone supports only Modern theme. |

| ***Parameter*** <br> ***Template*** | ***Permitted Values*** |
|---|---|
| `device.tr069.acs.password` <br> tr069.cfg | Sets the TR-069 ACS server password used to authenticate the phone. <br> Null (default) <br> String (256 maximum characters) |
| `device.tr069.acs.url` <br> tr069.cfg | Sets the URL for the TR-069 ACS server. <br> Null (default) <br> URL (256 maximum characters) |
| `device.tr069.acs.username` <br> tr069.cfg | Sets the TR-069 ACS server username used to authenticate the phone. <br> PlcmSpip (default) <br> String (256 maximum characters) |
| `device.tr069.cpe.password` <br> tr069.cfg | Specifies the TR-069 CPE password, which authenticates a connection request from the ACS server. <br> Null (default) <br> String (256 maximum characters) |
| `device.tr069.cpe.username` <br> tr069.cfg | Specifies the TR-069 CPE user name, which authenticates a connection request from the ACS server. <br> PlcmSpip (default) <br> String (256 maximum characters) |
| `device.tr069.periodicInform.enabled` <br> tr069.cfg | Indicates whether the CPE must periodically send CPE information to ACS using the **Inform** method call. <br> 0 (default) - Periodic **Inform** call is disabled. <br> 1 - Periodic **Inform** call is enabled. |
| `device.tr069.periodicInform.interval` <br> 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. <br> 18000 (default) <br> 0 to 36000 |
| `device.tr069.upgradesManaged.enabled` <br> tr069.cfg | Indicates whether the ACS manages image upgrades for the phone or not. <br> 0 (default) – The phone uses ACS or provisioning server for upgrade. <br> 1 - The phone upgrades only from the ACS server. |

| ***Parameter*** <br> ***Template*** | ***Permitted Values*** |
| --- | --- |
| `dir.local.passwordProtected` <br> features.cfg | Specifies whether you are prompted for an Admin or User password when adding, editing, or deleting contacts in the Contact Directory. <br> 0 (default) – No password prompt is displayed and pressing and holding the Line-key displays the Add or Edit menu. <br> 1 – You are prompted for your Admin or User password while adding, editing, or deleting contacts in the Contact Directory. |
| `feature.BSExecutiveAssistant.enabled` <br> features.cfg | 0 (default) - Disables the BroadSoft Executive-Assistant feature. <br> 1 - Enables the BroadSoft Executive-Assistant feature |
| `feature.BSExecutiveAssistant.regIndex` <br> features.cfg | The registered line assigned to the executive or assistant for the BroadSoft Executive-Assistant feature. <br> 1 (default) to 255 - The registered line for the Executive or Assistant. <br> 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` <br> features.cfg | ExecutiveRole (default) - Sets the registered line as an Executive line. <br> AssistantRole - Sets the registered line as an Assistant line. <br> Note: A phone can have a line set as an Executive or an Assistant; an Executive and an Assistant line cannot be set on the same phone. |
| `fs.unLockPhone.pin` | NULL (default) <br> 4 - 10 digits <br> Set a security pin for the Flexible Seating guest line on the host phone. |
| `hoteling.reg` <br> features.cfg | 1 (default) - Specifies the phone line on the host phone which hosts the guest line. |

| *Parameter*<br>*Template* | *Permitted Values* |
| --- | --- |
| hotelingMode.type | -1 (Default): The parameter does not exist on the Broadsoft server.<br>0 - Both Flexible Seating and Hoteling are disabled on the BroadSoft Device Management Server (DMS).<br>1 - Hoteling is enabled<br>2 - Flexible Seating is enabled but guest is not logged in.<br>3 - Flexible seating location is enabled and guest is logged in |
| httpd.cfg.lockWebUI.enable<br>site.cfg | Specifies whether web configuration login lock is enabled.<br>1 (default) – Enable the Web Configuration Login Lock feature.<br>0 - Disable the Web Configuration Login Lock feature. |
| httpd.cfg.lockWebUI.lockOutDuration<br>site.cfg | Specifies how long the user is locked out of the Web Configuration Utility.<br>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.<br>60 - 300 seconds |
| httpd.cfg.lockWebUI.noOfInvalidAttempts<br>site.cfg | Specifies the number of failed login attempts after which the user is locked out of the Web Configuration Utility.<br>5 (default)<br>3 - 20 |
| httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration<br>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.<br>60 seconds (default)<br>60 - 300 seconds |
| lcl.ml.lang.japanese.font.enabled[1]<br>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.<br>0 (default) – The phone does not use Japanese Kanji character font.<br>1 - The phone displays Japanese Kanji character font. |
| log.level.change.tr069<br>tr069.cfg | Sets the log levels for the TR-069 feature.<br>4 (default)<br>0 - 6 |

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

| *Parameter*<br>*Template* | *Permitted Values* |
|---|---|
| `reg.x.insertOBPAddressInRoute`<br>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.<br>0 – The outbound proxy address is not added to the route header.<br>1 (default) – The outbound proxy address is added as the top-most route header. |
| `reg.x.path`<br>debug.cfg | Specifies whether the path extension header field in the Register request message is supported for the specific line registration.<br>0 (default) – The path extension header field in the Register request message is not supported for the specific line registration.<br>1 – The path extension header field in the Register request message is supported for the specific line registration. |
| `reg.x.regevent`<br>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.<br>0 (default) – The phone is not subscribed to notifications for the specific phone line.<br>1 – The phone is subscribed to notifications for the specific phone line. |
| `reg.x.rejectNDUBInvite`<br>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.<br>0 (default) – Phone rejects the call with a 603 Decline response code.<br>1 – Phone accepts the call. |

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

| ***Parameter***<br>***Template*** | ***Permitted Values*** |
|---|---|
| `sec.TLS.profile.x.cipherSuite`<br>site.cfg, wireless.cfg | Specifies which cipher suite the phone uses for the TLS Application Profile.<br>Null (default)<br>1 – 8 – Choose the cipher suite for the TLS Application Profile. |
| `sec.TLS.profile.x.cipherSuiteDefault`<br>site.cfg, wireless.cfg | Specifies the default cipher suite for the TLS Application Profile.<br>1 (default) – Use the default cipher suite.<br>0 – Use the custom cipher suite for the TLS Application Profile. |
| `sec.TLS.webServer.cipherList`<br>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.<br>RSA:!EXP:!LOW:!NULL:!MD5:!RC4:@STRENGTH (default)<br>String |
| `up.deviceLock.createLockTimeout`<br>features.cfg | Specifies the timeout in minutes for the Create Lock Code prompt after Device Lock is enabled.<br>0 (default) – The Create Lock Code prompt does not time out.<br>1 – 3 minutes |
| `up.deviceLock.signOutOnIncorrectAttempts`<br>features.cfg | Configures phone behavior after six unsuccessful unlock attempts for Device Lock.<br>0 (default) – After six unsuccessful unlock attempts, phone prompts the user to wait 60 seconds before trying again.<br>1 – Signs the user out after six unsuccessful unlock attempts. |
| `up.LineViewCallStatus.enabled`<br>features.cfg | Specifies the Active Call Screen or Line Screen as default user interface for a call.<br>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.<br>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`<br>f | 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.<br>10 seconds (default)<br>2-9 seconds |

| ***Parameter***<br>***Template*** | ***Permitted Values*** |
|---|---|
| `up.OffHookIdleBrowserView.enabled`<br>features.cfg | Enables the microbrowser and associated soft keys to continue to display on the phone when the phone goes off-hook.<br>0 (Default) – The idle browser does not display on screen after the phone goes off-hook.<br>1 – The idle browser continues to display on screen after the phone goes off-hook. |
| `up.OffHookLineView.enabled`<br>features.cfg | Specifies the default user interface displayed when the phone goes off-hook.<br>0 (default) – Home Screen displays when the phone goes off-hook.<br>1 – Line Screen displays when the phone goes off-hook. |
| `up.ringer.minimumVolume`<br>site.cfg | Configure the minimum ringer volume. This parameter defines how many volume steps are accessible below the maximum level.<br>16 (default) – The full 16 steps of volume range are accessible.<br>1-15<br>0 – Ring volume is not adjustable by the user and the phone uses maximum ring volume.<br>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`<br>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.<br>30 dB (default)<br>3 – 90 dB |
| `voice.cn.hs.enable`<br>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.<br>0 (default) – No Comfort Noise is added.<br>1 – Comfort Noise is added to the handset. |

| ***Parameter*** <br> ***Template*** | ***Permitted Values*** |
|---|---|
| `voice.plcCnEnable` <br> 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. <br> 0 (default) – No Comfort Noise is added. <br> 1 – Comfort Noise is added. |
| `voice.plcCnGain` <br> 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. <br> 0 (default) <br> -20 – 20 dB |
| `voice.qualityMonitoring.processServiceRoute.enable` <br> features.cfg | Specifies whether the SIP route headers for the VQMon messages generated by the phone contain service route information. <br> 0 (default) – The VQMon messages generated by the phone do not contain service route information in SIP route headers. <br> 1 – The VQMon messages generated by phone, contain service route information, if available, in SIP route headers. |

| *Parameter*<br>*Template* | *Permitted Values* |
|---|---|
| `voIpProt.server.x.specialInterop` | Specifies the server-specific feature set supported for all line registrations.<br>VVX 101 = Standard<br>GENBAND<br>GENBAND-A2<br>ALU-CTS<br>DT<br><br>VVX 201 = Standard<br>GENBAND<br>GENBAND-A2<br>ALU-CTS<br>DT<br>ocs2007r2<br>lync2010<br><br>All other phones = Standard<br>GENBAND<br>GENBAND-A2<br>ALU-CTS<br>DT<br>ocs2007r2<br>lync2010<br>lcs2005 |
| `voipProt.SIP.anat.enabled`<br>sip-interop.cfg | Enables or disables Alternative Network Address Types (ANAT).<br>0 (default) - ANAT is disabled.<br>1 - ANAT is enabled. |
| `voIpProt.SIP.header.pEarlyMedia.support`<br>sip-interop.cfg | Specifies whether the caller phone supports the p-early-media header.<br>0 (Default) – The p-early-media header is not supported by the caller phone.<br>1 – The p-early-media header is supported by the caller phone. |

| *Parameter*<br>*Template* | *Permitted Values* |
|---|---|
| `voIpProt.SIP.IMS.enable`<br>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.<br>0 (Default) – Phone cannot support IMS features that are introduced in UC Software 5.5.0 or later.<br>1 – Phone supports IMS features that are introduced in UC Software 5.5.0 or later. |
| `voIpProt.SIP.looseContact`<br>sip-interop.cfg | Configures addition of the ephemeral port parameter to the contact header.<br>0 (default) - The ephemeral port is added to the contact header in TLS case.<br>1 – The port parameter is not added to the contact header or SIP messages. |
| `voIpProt.SIP.regevent`<br>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.<br>0 (default) – The phone is not subscribed to notifications for all phone lines.<br>1 – The phone is subscribed to notifications for all phone lines. |
| `voIpProt.SIP.rejectNDUBInvite`<br>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.<br>0 (default) – Phone rejects the call with a 603 Decline response code.<br>1 – Phone accepts the call. |
| `voIpProt.SIP.specialEvent.checkSync.downloadCallList`<br>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.<br>`0` (default) – Call list is not downloaded after receiving a check-sync event in the NOTIFY message.<br>`1` – Call list is not downloaded after receiving a check-sync event in the NOTIFY message. |

| ***Parameter***<br>***Template*** | ***Permitted Values*** |
|---|---|
| `voIpProt.SIP.supportFor199`<br>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.<br>0 (default) – The phone does not support 199 response code.<br>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 | |

# Release History

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

**Release History**

| *Release* | *Release Date* | *Description* |
|---|---|---|
| 5.5.0 | May 2016 | This release introduces support for Broadsoft Executive Assistant and Flexible Seating, TR-069, the 3GPP Technical Specification, the IPV6 protocol, Off-hook Call Status control, ability to lock the web configuration utility after failed login attempts, and user interface enhancements. |
| 5.4.3 | February 2016 | This release introduced the Polycom VVX D60 Wireless Handset and VVX D60 Base Station. |
| 5.4.1 | December 2015 | This release includes support for the following features:<br>• Introduced the Polycom VVX 301/311, 401/411, 501, and 601 business media phones.<br>• Flexible line key customization for Lync (EFLK)<br>• Master Key Identifiers (MKI)<br>• Shared Line appearance on Lync<br>• BToE for Windows 10<br>• Smart Search for Lync ABS<br>• Support for simplified Chinese font on VVX 101 |
| 5.4.0A | September 2015 | This release includes support for the following features:<br>Microsoft Office 365 and Skype for Business Online<br>Office365 and Skype for Business Provisioning and Manageability<br>Time and Date Initial Setup |

| *Release* | *Release Date* | *Description* |
|---|---|---|
| 5.4.0 | May 2015 | Added support for Alcatel-Lucent CTS features including<br>• Advanced Conference<br>• Shared Call Appearance with Bridge In<br>• Visitor Desk Phone<br>This release also included support for the following features:<br>• Barge In on Busy Lamp Field Lines<br>• DTMF Relay<br>• SIP Instance<br>• Comfort Noise<br>• Opus Codec<br>• DNS Server Address Override<br>• Global Directory Synchronization<br>• Basic Menu Lock<br>• 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 5.5.0

Consider the following information when installing or updating to Polycom UC Software 5.5.0.

**Caution: Updating VVX 1500 to UC Software 5.5.0**

Before updating your VVX 1500 phone to UC Software 5.5.0, 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 5.5.0, you can choose the combined UC Software package or the split UC Software package, both in ZIP file format. The combined version contains all files for all phone models. The split software package is smaller, downloads more quickly, and contains sip.ld files for each phone model, enabling you to choose provisioning software for your phone model and maintain software versions for each model in the same root directory.

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

# 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 | ✓ |
| 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) | ✓ | ✓ |

| *Distributed Files* | *File Purpose and Application* | *Combined* | *Split* |
|---|---|---|---|
| | from the file name | | |
| applications.cfg | Configuration parameters for microbrowser and browser applications | ✓ | ✓ |
| device.cfg | Configuration parameters for basic device configuration | ✓ | ✓ |
| features.cfg | Configuration parameters for telephony features | ✓ | ✓ |
| firewall-nat.cfg | Contains configuration parameters for telephony features | ✓ | ✓ |
| H323.cfg | Configuration parameters for the H.323 signaling protocol | ✓ | ✓ |
| lync.cfg | Contains Lync specific configuration parameters | ✓ | ✓ |
| pstn.cfg | Contains parameters for PSTN use | ✓ | ✓ |
| reg-advanced.cfg | Contains configuration parameters for the line and call registration and advanced phone feature settings | ✓ | ✓ |
| reg-basic.cfg | Configuration parameters for line and call registration and basic phone settings | ✓ | ✓ |
| region.cfg | Configuration parameters for regional and localization settings such as time and date and language | ✓ | ✓ |
| sip-basic.cfg | Configuration parameters for the VoIP server and softswitch registration | ✓ | ✓ |
| sip-interop.cfg | Configuration parameters for the VoIP server, softswitch registration, and interoperability configuration | ✓ | ✓ |
| site.cfg | Configuration parameters that are set for each site | ✓ | ✓ |
| video.cfg | Configuration parameters for video connectivity | ✓ | ✓ |
| video-integration.cfg | Configuration parameters for SoundStation IP 7000 and Polycom HDX system integration | ✓ | ✓ |
| VVX-dictionary.xml | Includes native support for the following languages:<br>• Arabic, UAE<br>• Chinese, Traditional<br>• Chinese, Simplified<br>• Danish, Denmark<br>• Dutch, Netherlands | ✓ | ✓ |

| *Distributed Files* | *File Purpose and Application* | *Combined* | *Split* |
|---|---|---|---|
| | • English, Canada<br>• English, United Kingdom<br>• English, United States<br>• French, Canada<br>• French, France<br>• German, Germany<br>• Italian, Italy<br>• Japanese, Japan<br>• Korean, Korea<br>• Norwegian, Norway<br>• Polish, Poland<br>• Portuguese, Brazil<br>• Russian, Russia<br>• Slovenian, Slovenia<br>• Spanish, Spain<br>• 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 | ✓ | ✓ |

# Known Issues

The following table lists the known issues and suggested workarounds for UC Software 5.5.0.

**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 phone 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. | |
| 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. | |
| 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. | |

| *Category* | *Issue No.* | *Release* | *Description* | *Workaround* |
|---|---|---|---|---|
| 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 Softkey is seen on the microbrowser home screen. | |
| User Interface | VOIP-116353 | 5.5.0 | On the D50 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. | |
| 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. | |
| Web Interface | VOIP-113192 | 5.5.0 | In the VVX system web interface Handset Settings, a mapped line is incorrectly listed twice. | |

| *Category* | *Issue No.* | *Release* | *Description* | *Workaround* |
|---|---|---|---|---|
| Web Interface | VOIP-113193 | 5.5.0 | The VVX D60 web interface line management page does not show the default line. | |

# Resolved Issues

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

**Resolved Issues in UC Software 5.5.0**

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

| *Category* | *Issue No.* | *Release* | *Description* |
|---|---|---|---|
| Calling | VOIP-113478 | 5.4.1 | When the SoundStructure VoIP Interface is in a call, sending a "set voip_send VoIP Out" command to the SoundStructure no longer causes the call to disconnect. Pressing a digit on a Polycom Touch Control paired with the SoundStructure during a call now works correctly. |
| Calling | VOIP-112886 | | Line seize behavior for accessing voicemail using Enhanced Feature Keys (EFK) has been improved. |
| Calling | VOIP-111991 | | Parameter `call.urlNumberModeToggling` now allows you to specify whether the phone uses Number mode or URL mode when a URL call is initiated. |
| Calling | VOIP-109593 | | The parameter `call.urlNumberModeToggling` was added to resolve a problem with URL dialing. |
| Calling | VOIP-109311 | | The phone now correctly sends “user=phone” in the invite message when a user enters a number that ends with “#” or “*”. |
| Calling | VOIP-107290 | | The phone now ignores any unrecognized parameters included in check-sync messages. |
| Calling | VOIP-115425 | | New parameter `call.shared.reject` was added to allow you to configure phones to display a Reject soft key for calls on a shared line. |
| Calling | VOIP-116273 | 4.0.9 | Phones now use the contact URI and TELURI in the request line of BYE message, so calls end correctly when the `reg.1.telUri` is enabled or disabled. |
| Calling | VOIP-116228 | 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. |

| *Category* | *Issue No.* | *Release* | *Description* |
|---|---|---|---|
| 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. |
| General | VOIP-110472 | 5.2.1 | VVX Keys are now optimized for the resposivess, 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 paramaters 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. |

| *Category* | *Issue No.* | *Release* | *Description* |
|---|---|---|---|
| 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 Lyncor 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. |
| 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 Audiocodes 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. |

| *Category* | *Issue No.* | *Release* | *Description* |
|---|---|---|---|
| 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. |
| 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. |

| *Category* | *Issue No.* | *Release* | *Description* |
|---|---|---|---|
| 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 no consistently displays the Encoding soft key on the Single Signin 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.

# The Polycom Community

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