ADMINISTRATOR GUIDE

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Polycom UC Software

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Before You Begin

Topics:

- Audience, Purpose, and Required Skills
- Get Help

This guide describes how to administer, configure, and provision Polycom phones and accessories.

The information in this guide applies to the following Polycom devices except where noted:

- Polycom® VVX® 101 business media phones
- Polycom® VVX® 201 business media phones
- Polycom® VVX® 300 series (300/301/310/311) business media phones
- Polycom® VVX® 400 series (400/401/410/411) business media phones
- Polycom® VVX® 500 series (500/501) business media phones
- Polycom® VVX® 600 series (600/601) business media phones
- Polycom® VVX® 1500 business media phones
- Polycom® D60 VVX Wireless Handset and Base Station
- Polycom® VVX Expansion Modules
- Polycom®VVX® 150 business IP phones
- Polycom®VVX® 250 business IP phones
- Polycom®VVX® 350 business IP phones
- Polycom®VVX® 450 business IP phones
- Polycom® SoundStructure™ VoIP Interface

Audience, Purpose, and Required Skills

This guide is written for a technical audience.

You must be familiar with the following concepts before beginning:

- Current telecommunications practices, protocols, and principles
- Telecommunication basics, video teleconferencing, and voice or data equipment
- Open SIP networks and VoIP endpoint environments

Get Help

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

Polycom and Partner Resources

In addition to this guide, the following documents and other resources provide details about Polycom UC Software:

- To access all Polycom UC Software releases and documentation, see Polycom Voice Support.
- To access Polycom Trio system documentation and support resources, see Polycom Trio on Polycom Support.
- You can find Request for Comments (RFC) documents by entering the RFC number at http://www.ietf.org/rfc.html.
- For information on IP PBX and softswitch vendors, see Polycom Desktop Phone Compatibility. If you're using the Polycom Trio solution, see Polycom Trio and SoundStation IP Platform Compatibility.

To find all Polycom partner solutions, see Strategic Global Partner Solutions.

Documentation Feedback

We welcome your feedback to improve the quality of Polycom documentation.

You can email Documentation Feedback for any important queries or suggestions related to this documentation.

Getting Started

Topics:

- Product Overview
- Phone Features and Licenses
- Working With Polycom UC Software

Polycom UC software is a binary file image and contains a digital signature that prevents tampering or the loading of rogue software images.

Each release of software is a new image file.

Product Overview

Polycom UC software manages the protocol stack, the digital signal processor (DSP), the user interface, and the network interaction on Polycom phones.

You can deploy Polycom UC software by configuring individual phones, but Polycom recommends setting up a provisioning server on your LAN or the internet for large-scale deployments.

UC software implements the following functions and features on the phones:

- VoIP signaling for a wide range of voice and video telephony functions using SIP signaling for call setup and control.
- SIP and H.323 signaling for video telephony. Support for H.323 varies by phone model.
- Industry-standard security techniques for ensuring that all provisioning, signaling, and media transactions are robustly authenticated and encrypted.
- Advanced audio signal processing for handset, headset, and speakerphone communications using a wide range of audio codecs.
- Flexible provisioning methods to support single phone, small business, and large multi-site enterprise deployments.

Supported Phones and Accessories

The following table lists the product names, model names, and part numbers for Polycom phones and devices that support Polycom UC Software.

Polycom VVX Product Name, Model Name, and Part Number

Product Name	Model Name	Part Number
SoundStructure VoIP Interface	SSTRVOIP	3111-33215-001
VVX D60 Wireless Handset	VVXD60	3111-17823-001
VVX 101	VVX101	3111-40250-001
VVX 150	VVX150	3111-48810-001

Product Name	Model Name	Part Number
VVX 201	VVX201	3111-40450-001
VVX 250	VVX250	3111-48820-001
VVX 300	VVX300	3111-46135-002
VVX 301	VVX301	3111-48300-001
VVX 310	VVX310	3111-46161-001
VVX 311	VVX311	3111-48350-001
VVX 350	VVX350	3111-48830-001
VVX 400	VVX400	3111-46157-002
VVX 401	VVX401	3111-48400-001
VVX 410	VVX410	3111-46162-001
VVX 411	VVX411	3111-48450-001
VVX 450	VVX450	3111-48840-001
VVX 500	VVX500	3111-44500-001
VVX 501	VVX501	3111-48500-001
VVX 600	VVX600	3111-44600-001
VVX 601	VVX601	3111-48600-001
VVX 1500	VVX1500	2345-17960-001

Phone Features and Licenses

You may need to purchase a feature license depending on the feature and phone model you are using.

The following tables list features available for each phone and indicate whether a feature license is required or not. In the following tables:

- No indicates that a phone does not support a feature.
- Yes indicates that a phone supports a feature and no license is required.
- Yes* indicates that the phone requires you to purchase a feature license from Polycom to support a feature.
- Yes** indicates that the phone requires you to purchase an honor-based license from Polycom to support a feature.

The following table lists VVX business media phone features and licenses.

Features and Licenses - Business Media Phones

Feature	VVX 101	VVX 201	VVX 300/301/ 310/311	VVX 400/410/ 401/411	VVX 500/501/ 600/601	VVX 1500	SoundStructure VoIP Interface
Asian Languages	No	Yes	Yes	Yes	Yes	Yes	No
Conference Management	Yes	Yes	Yes	Yes	Yes	Yes	No
Customize UI Background	No	No	Yes	Yes	Yes	Yes	No
Electronic Hookswitch	Yes	Yes	Yes	Yes	Yes	Yes	No
Enhanced BLF	No	No	Yes	Yes	Yes	Yes	No
Enhanced Feature Keys	Yes	Yes	Yes	Yes	Yes	Yes	No
H.323 Video	No	No	No	No	Yes	Yes	No
Skype for Business	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**	Yes**
Server-based Call Recording	Yes	Yes	Yes	Yes	Yes	Yes	No
USB Call Recording	No	No	No	400/410=No 401/411=Yes	Yes	Yes	No
Voice Quality Monitoring	Yes*	Yes*	Yes*	Yes*	Yes (Audio only)	Yes (Audio only)	No
XT9 Text Input (Pinyin)	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*	Yes*

The following table lists VVX business IP phone features and licenses.

Features and Licenses - VVX Business IP Phones

Feature	VVX 150	VVX 250	VVX 350	VVX 450
Asian Languages	Yes	Yes	Yes	Yes
Conference Management	Yes	Yes	Yes	Yes
Customize UI Background	No	Yes	Yes	Yes
Electronic Hookswitch	Yes	Yes	Yes	Yes
Enhanced BLF	No	Yes	Yes	Yes
Enhanced Feature Keys	Yes	Yes	Yes	Yes
H.323 Video	No	No	No	No
Skype for Business	No	Yes**	Yes**	Yes**
Server-based Call Recording	Yes	Yes	Yes	Yes
USB Call Recording	No	Yes	Yes	Yes
Voice Quality Monitoring	Yes*	Yes*	Yes*	Yes*
XT9 Text Input (Pinyin)	Yes*	Yes*	Yes*	Yes*

Working With Polycom UC Software

Polycom phones come installed with updater software that resides in the flash memory of the phone.

When you boot up or reboot the phone, the updater automatically updates, downloads, and installs new software versions or configuration files as needed, based on the server or phone settings.

Configuring Polycom Phones

Polycom provides several methods to configure or provision phones. The method you use depends on the number of phones, the phone model, and how you want to apply features and settings.

You can use multiple methods concurrently to provision and configure features, but some methods have a higher priority than others when you use multiple methods concurrently —settings you make using a higher priority configuration method override settings made using a lower priority method. When using multiple configuration methods, a setting you make using a lower-priority method does not apply to or override a duplicate setting made using a higher-priority method.

The provisioning and configuration methods in order of priority:

1. Quick Setup
2. Phone menu
3. Web Configuration Utility
4. Skype for Business in-band provisioning
5. USB
6. Polycom® Resource Manager
7. Centralized provisioning
8. Default phone values

Polycom phones can boot up without the use of configuration files, and you can specify a SIP server address and a registration address (the equivalent of a phone number) in a configuration file before or after the phone boots up. If a phone cannot locate a provisioning server upon boot up and has not been configured with settings from any other source, the phone operates with internally stored default values. If the phone has been previously configured with settings from a provisioning server and cannot locate the server when booting up, the phone operates with those previous settings.

Polycom phones support FTP, TFTP, HTTP, and HTTPS protocols and use FTP by default.

Record Version Information

In case you need to contact Polycom technical support, you should record the following information for future reference:

- Phone models
- Updater version
- UC Software version
- Partner Platform

Supported Network Configurations

Topics:

- Ethernet Line Rates
- Ethernet Network Connection Methods
- Link Layer Discovery Protocol and Supported Type Length Values
- DHCPv6 or DHCPv4 Parameters
- Parse Vendor ID Information

You need the following to operate Polycom phones as SIP endpoints in large-scale deployments:

- A working IP network
- Routers configured for VoIP
- VoIP gateways configured for SIP
- An active, configured call server to receive and send SIP messages and to register and authenticate voice endpoints.

  For information on IP PBX and softswitch vendors, see Polycom Desktop Phone Compatibility. If you are using the Polycom Trio Solution, see Polycom Trio and SoundStation IP Platform Compatibility.

  At minimum, your call server requires:

  - A call server address that registers voice endpoints with the SIP server
  - SIP authentication user name and password the phone uses to respond to any SIP authentication challenges from the SIP server.

In addition to these requirements, your deployment network should work within the Polycom-supported parameters of network settings, discovery methods such as DHCP, and supported Ethernet network settings.

Related Links

Network on page 499

Network Requirements for Provisioning on page 40

Ethernet Line Rates

The phones automatically negotiate the Ethernet rate and no special configuration is required.

Typical network equipment supports one of the three following Ethernet line rates:

- 10 Mbps
- 100 Mbps
- 1000 Mbps

While you can change the line rates and duplex configuration, Polycom recommends keeping the default settings.

Supported Denial of Service Filters

The phone supports two filters to prevent Denial of Service (DoS):

- Storm Filtering—This filter is enabled by default.
- VLAN Filtering—VLAN filtering cannot be disabled.

When these filters are enabled, Ethernet packets are filtered to prevent overflow caused by bad or excessive data. Support for Storm and VLAN filtering varies by device.

Supported 802.1x Configurations

Polycom phones support the following EAP authentication methods:

- EAP-TLS (requires Device and CA certificates)
- EAP-PEAPv0/MSCHAPv2 (requires CA certificates)
- EAP-PEAPv0/GTC (requires CA certificates)
- EAP-TTLS/MSCHAPv2 (requires CA certificates)
- EAP-TTLS/GTC (requires CA certificates)
- EAP-FAST (optional Protected Access Credential (PAC) file, if not using in-band provisioning)
- EAP-MD5

For more information about EAP methods, see RFC 3748: Extensible Authentication Protocol.

Ethernet Network Connection Methods

You can connect the phone to a network using Ethernet with the following methods:

- Virtual Local Area Networks (VLANs)
- lLink Layer Discovery Protocol and Supported Type Length Values
- lLink Layer Discovery Protocol and Supported Type Length Values

Virtual Local Area Networks (VLANs)

Settings from higher priority methods override settings from lower priority methods.

If the phone receives a Virtual Local Area Network (VLAN) setting from more than one of the following methods, the priority is as follows:

1. LLDP—Link Layer Discovery Protocol (LLDP) is a vendor-neutral Layer 2 protocol that allows a network device to advertise its identity and capabilities on the local network.
2. CDP—Cisco Discovery Protocol (CDP) is a proprietary Data Link Layer network protocol. CDP Compatible follows the same set of rules.
3. DVD (VLAN via DHCP)—Dynamic Host Configuration Protocol (DHCP) is an automatic configuration protocol used in IP networks. Note that use of DHCP for assigning VLANs is not standardized and is recommended only if the switch equipment does not support LLDP or CDP Compatible methods.
4. Static—The VLAN ID can be manually set by entering it through the phone's menu.

Virtual Local Area Network (VLAN) ID Assignment Using DHCP

In deployments where it is possible or desirable to assign a Virtual Local Area Network (VLAN) using LLDP, CDP, or Static methods, you can assign a VLAN ID to the phones by distributing the VLAN ID via DHCP.

When using this method to assign the phone's VLAN ID, the phone first boots on the Native VLAN/Data VLAN and then obtains its intended VLAN ID from the DHCP offer before it continues booting on the newly obtained VLAN.

Note: If a VLAN tag is assigned by CDP or LLDP, DHCP VLAN tags are ignored.

Valid DVD String DHCP Options

The DVD string in the DHCP option must meet the following conditions to be valid:

- Must start with “VLAN-A=” (case-sensitive)
- Must contain at least one valid ID
- VLAN IDs range from 0 to 4095
- Each VLAN ID must be separated by a “+” character
- The string must be terminated by a semi colon “;”
- All characters after the semi colon “;” are ignored
- There must be no white space before the semi colon “;”
- VLAN IDs may be decimal, hex, or octal

The following DVD strings result in the phone using VLAN 10:

- VLAN-A=10;
- VLAN-A=0x0a;
- VLAN-A=012;

Assign a VLAN ID Using DHCP

When the VLAN Discovery in the DHCP menu is set to **Fixed**, the phone examines DHCP options 128,144, 157, and 191 in that order for a valid Digital Versatile Disk DHCP VLAN Discovery string.

When set to **Custom**, a value set in the VLAN ID Option is examined for a valid DVD string.

If DHCP option 128 is configured for SIP outbound proxy, do not configure VLAN Discovery option 128 to **Fixed**.

Procedure

1. In the DHCP menu of the Main setup menu, set **VLAN Discovery** to **Fixed** or **Custom**.

Related Links

Device Parameters on page 601

Link Layer Discovery Protocol and Supported Type Length Values

A Link Layer Discovery Protocol (LLDP) frame must contain all mandatory Type Length Values (TLVs).

Polycom phones running UC Software support LLDP frames with both mandatory and optional TLVs.

The phones cannot determine their physical location automatically or provision to a statically configured location. Hence, they do not transmit location identification TLV in the LLDP frame. However, the location information from the switch is decoded and displayed on the phone's menu.

The LLDP feature supports VLAN discovery and LLDP power management, but not power negotiation. LLDP has a higher priority than Cisco Discovery Protocol (CDP) and DHCP VLAN discovery.

Supported TLVs

Polycom phones support the following mandatory and optional TLVs:

Mandatory:

- Chassis ID—Must be first TLV.
- Port ID—Must be second TLV.
- Time-to-live—Must be third TLV, set to 120 seconds.
- End-of-LLDPDU—Must be last TLV.
- LLDP-MED Capabilities.
- LLDP-MED Network Policy—VLAN, L2 QoS, L3 QoS.
- LLDP-MED Extended Power-Via-MDI TLV—Power Type, Power Source, Power Priority, Power Value.

Optional:

- Port Description
- System Name—Administrator assigned name.
- System Description—Includes device type, phone number, hardware version, and software version.
- System Capabilities—Set as ‘Telephone' capability.
- MAC / PHY configuration status—Detects duplex mismatch.
- Management Address—Used for network discovery.
- LLDP-MED Location Identification—Location data formats: Co-ordinate, Civic Address, ECS ELIN.
- LLDP-MED Inventory Management —Hardware Revision, Firmware Revision, Software Revision, Serial Number, Manufacturer's Name, Model Name, Asset ID.

The basic TLV format is as follows:

- TLV Type (7 bits) [0-6]
- TLV Length (9 bits) [7-15]
- TLV Information (0-511 bytes)

The following table lists the supported TLVs.

Name	Description	Typ e	Length	Type Length	Org. Unique Code (3 bytes)	Sub Type
Chassis-Id[1]	IP address of phone (4 bytes). Note that 0.0.0.0 is not sent until the phone has a valid IP address.	1	6	0x0206	-	5
Port-Id[1]	The MAC address of the phone (6 bytes).	2	7	0x0407	-	3
TTL	The TTL value is 120/0 sec.	3	2	0x0602	-	-
Port description	Port description 1.	4	1	0x0801	-	-
System name	Refer to System and Model Names.	5	min len > 0, max len <= 255	-	-	-
System description	Manufacturer's name - “Polycom”; Hardware version; Application version; BootROM version.	6	min len > 0, max len <= 255	-	-	-
Capabilities	System Capabilities: Telephone and Bridge if the phone has PC port support and it is not disabled. Enabled Capabilities: Telephone and Bridge if phone has PC port support, it is not disabled and PC port is connected to PC.	7	4	0x0e04	-	-
Management Address	Address String Len - 5, IPV4 subtype, IP address, Interface subtype - “Unknown”, Interface number - “0”, ODI string Len - “0”.	8	12	0x100c	-	-
IEEE 802.3 MAC/PHY config/ status[1]	Auto-Negotiation Supported - “1”, enabled/disabled, Refer to PMD Advertise and Operational MAU	127	9	0xfe09	0x00120f	1

Name	Description	Type	Length	Type Length	Org. Unique Code (3 bytes)	Sub Type
LLDP-MED capabilities	Capabilities - 0x33 (LLDP-Med capabilities, Network policy, Extended Power Via MDI-PD, Inventory) Class Type III. Note: After support for configuring location Identification information is locally available. Capabilities - 0x37 (LLDP-Med capabilities, Network policy, Location Identification, Extended Power Via MDI-PD, Inventory) Class Type III.	127	7	0xfe07	0x0012bb	1
LLDP-MED network policy[2]	ApplicationType: Voice (1), Policy: (Unknown(=1)/ Defined(=0) Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Network policy TLV is received from the switch.), Tagged/Untagged, VlanId, L2 priority, and DSCP.	127	8	0xfe08	0x0012bb	2
LLDP-MED network policy[2]	ApplicationType: Voice Signaling (2), Policy: (Unknown(=1)/Defined(=0) Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Network policy TLV is received from the switch.), Tagged/Untagged, VlanId, L2 priority and DSCP. Note: Voice signaling TLV is sent only if it contains configuration parameters that are different from voice parameters.	127	8	0xfe08	0x0012bb	2

Name	Description	Typ e	Length	Type Length	Org. Unique Code (3 bytes)	Sub Type
LLDP-MED network policy[2]	ApplicationType: Video Conferencing (6), Policy: (Unknown(=1)/Defined(=0). Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Network policy TLV is received from the switch.), Tagged/Untagged, VlanId, L2 priority and DSCP. Note: Video conferencing TLV is sent only from video-capable phones: VVX 500/501, 600/601, and 1500 business media phones.	127	8	0xfe08	0x0012bb	2
LLDP-MED location identification[3]	ELIN data format: 10 digit emergency number configured on the switch. Civic Address: physical address data such as city, street number, and building information.	127	min len > 0, max len <= 511	-	0x0012bb	3
Extended power via MDI	PowerType -PD device PowerSource-PSE&local Power Priority -Unknown PowerValue	127	7	0xfe07	0x0012bb	4
LLDP-MED inventory hardware revision	Hardware part number and revision.	127	min len > 0, max len <= 32	-	0x0012bb	5
LLDP-MED inventory firmware revision	BootROM revision.	127	min len > 0, max len <= 32	-	0x0012bb	6
LLDP-MED inventory software revision	Application (SIP) revision.	127	min len > 0, max len <= 32	-	0x0012bb	7
LLDP-MED inventory serial number	MAC Address (ASCII string).	127	min len > 0, max len <= 32	-	0x0012bb	8

Name	Description	Typ e	Length	Type Length	Org. Unique Code (3 bytes)	Sub Type
LLDP-MED inventory manufacturer name	Polycom	127	11	0xfe0b	0x0012bb	9
LLDP-MED inventory model name		127	min len > 0, max len <= 32	-	0x0012bb	10
LLDP-MED inventory asset ID	Empty (Zero length string).	127	4	0xfe08	0x0012bb	11
End of LLDP DU		0	0	0x0000	-	-

DHCPv6 or DHCPv4 Parameters

Polycom recommends using DHCP where possible to eliminate repetitive manual data entry.

After establishing network connectivity, the phone needs to acquire several IPv6 or IPv4 network settings. These settings are typically obtained automatically from a Dynamic Host Configuration Protocol (DHCPv6 or DHCPv4) server.

You have the option to configure IPv4 or IPV6 network settings manually from the phone screen or using `device.set` capability. When making the DHCP request, the phone includes information in Option 60 that can assist the DHCP server in delivering the appropriate settings to the device.

For more information on DHCP options, see RFC2131 and RFC 2132.

For more information on Using DHCP Vendor Identifying Options with Polycom Phones, see Technical Bulletin 54041 at Polycom Engineering Advisories and Technical Notifications.

IPv4 Network Parameters

The following table lists the ways a phone can obtain IPv4 and related parameters in an IPv4 network.

Parameter	DHCPv4 Option	DHCPv4	DHCPv4 INFORM	Configuration File (application only)	Device Settings
IPv4 address	No	Yes	No	No	Yes
Subnet mask	1	Yes	No	No	Yes

[1] [1] For other subtypes, refer to IEEE 802.1AB, March 2005.

[2] [2] For other application types, refer to TIA Standards 1057, April 2006.

[3] [3] At this time, this TLV is not sent by the phone.

Parameter	DHCPv4 Option	DHCPv4	DHCPv4 INFORM	Configuration File (application only)	Device Settings
IPv4 Gateway	3	Yes	No	No	Yes
Boot server address		Yes	Yes	No	Yes
SIP server address	151 You can change this value by changing the device setting.	Yes	No	Yes	Yes
SNTP server address	Look at option 42, then option 4.	Yes	No	Yes	Yes
SNTP GMT offset	2	Yes	No	Yes	Yes
Syslog		Yes	No	No	Yes
DNS server IP address	6	Yes	No	Yes	Yes
DNS INFORM server IP address	6	-	-	-	-
DNS domain	15	Yes	No	Yes	Yes
VLAN ID		Warning: Link Layer Discovery Protocol (LLDP) overrides Cisco Discovery Protocol (CDP). CDP overrides Local FLASH which overrides DHCP VLAN Discovery.			

IPv6 Network Parameters

The following table lists the ways a phone can obtain IPv6 and related parameters in an IPv6 network.

IPv6 Network Parameters

Parameter	SLAAC1	DHCPv6 Option	DHCPv6	DHCPv6 INFORM	Configuration File (application only)	Device Settings
IPv6 Global Address	Yes	No	Yes	No	No	Yes
IPv6 ULA Address	Yes	No	Yes	No	No	Yes
IPv6 Gateway	Yes	No	No	No	No	Yes
Boot server IPv6 Address	No	Custom2	Yes	No	No	Yes

Parameter	SLAAC1	DHCPv6 Option	DHCPv6	DHCPv6 INFORM	Configuration File (application only)	Device Settings
SIP server IPv6 Address	No	22/21	Yes	No	Yes	No
SNTP server IPv6 address	No	31	Yes	No	Yes	Yes
SNTP GMT offset	No	Custom2	Yes	No	Yes	Yes
Syslog IPv6 Address	No	Custom2	Yes	No	Yes	Yes
DNS server IPv6 address	No	23	Yes	No	Yes	Yes
IPv6 DNS domain	No	23	Yes	No	Yes	Yes
VLAN ID		Warning: Link Layer Discovery Protocol (LLDP) overrides Cisco Discovery Protocol (CDP). CDP overrides Local FLASH which overrides DHCP VLAN Discovery.				

Example Configuration: Polycom Vendor-Specific Information Options in DHCPv6

You can obtain the CDP Compatibility value from a connected Ethernet switch if the switch supports CDP.

In DHCPv6, there are no standard options defined for Boot Server IPv6 address, Syslog Server IPv6 Address, SNTP GMT Offset, and VLAN List. Polycom has defined subcodes for this specific information as part of the DHCPv6 Vendor-Specific Information Option.

You can use the `tcpIpApp.sntp.address.overrideDHCP` parameter values for the SNTP server address and SNTP GMT offset to override the DHCP value.

The following is an example configuration on a Linux DHCPv6 server for Polycom subcode definitions:

```
# Define PLCM options option space plcm code width 2 length width 2 hash size
4; option vsio.plcm code 13885 = encapsulate plcm; option plcm.boot-server
code 1 = string; option plcm.time-offset code 2 = signed integer 32; option
plcm.syslog-server code 3 = string; option plcm.vlan-list code 4 = string;
option plcm.boot-server "2620:0:1aa0:8071:d485:f47d:5de5:be04"; option
plcm.time-offset 19850; option plcm.syslog-server
"2620:0:1aa0:8071:d485:f47d:5de5:be04"; option plcm.vlan-list "VLAN-A=513;";
```

Example: DHCP Option 60 Packet Decode

The following example is a sample decode of a packet (DHCP Option 60) from the Polycom Trio 8800 system.

- Sub-option 2 (part), length, "Real PresencePolycom Trio-Polycom Trio_8800" `02 1a 52 65 61 6c 50 72 65 73 65 6e 63 65 54 72 69 6f 2d 54 72 69 6f 5f 38 38 30 30`

- Sub-option 3 (part number), length, "3111-65290-001,5" `03 10 33 31 31 31 2d 36 35 32 39 30 2d 30 30 31 2c 35`
- Sub-option 4 (Application version), length, "SIP/5.4.1.16972/04-Jan-16 16:05" `05 1d 53 49 50 2f 35 2e 34 2e 31 2e 31 36 39 37 32 2f 30 34 2d 4a 61 6e 2d 31 36 20 31 36 3a 30 35`

The following example is a sample decode of a packet (DHCP Option 60) from a VVX 500/501:

`3c 7a`

- Option 60, length of Option data (part of the DHCP specification) `00 00 36 3d`
- Polycom signature (always 4 octets) `75`
- Length of Polycom data `01 07 50 6f 6c 79 63 6f 6d`
- sub-option 1 (company), length, "Polycom" `02 0b 56 56 58 2d 56 56 58 5f 34 31 30`
- sub-option 2 (part), length, "VVX-VVX_500/501" `03 10 33 31 31 31 2d 34 36 31 36 32 2d 30 30 31 2c 37`
- sub-option 3 (part number), length, "3111-44500-001,7" `04 1e 53 49 50 2f 35 2e 32 2e 30 2e 58 58 58 58 2f 30 36 2d 41 75 67 2d 31 34 20 32 30 3a 35 35`
- sub-option 4 (Application version), length, "SIP/5.2.0.XXXX/06-Aug-14 20:55" `05 1d 55 50 2f 35 2e 34 2e 30 2e 58 58 58 58 2f 30 36 2d 41 75 67 2d 31 34 20 32 31 3a 30 34`
- sub-option 5 (Updater version), length, "UP/5.4.0.XXXX/06-Aug-14 21:04” `06 0c 64 73 6c 66 6f 72 75 6d 2e 6f 72 67`
- sub-option 6 “dslforum.org”

Vendor Specific DHCP Options

DHCP Option 60 controls how the phone identifies itself to a DHCP server for Polycom-specific options that must be returned.

If Option 60 format is set to RFC 3925, all returned values of Option 43 are ignored. If the format is set to an ASCII string, the Option 43 would have a hexadecimal string value encapsulating sub-options that override options received outside DHCP Option 43.

If you do not have control of your DHCP server or do not have the ability to set the DHCP options, enable the phone to automatically discover the provisioning server address. You can do this by connecting to a secondary DHCP server that responds to DHCP INFORM queries with a requested provisioning server value. For more information, see RFC 3361 and RFC 3925.

The following table lists supported DHCP Option 43 individual sub-options and combination sub-options:

Option	Results
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/ITP server address (RFC 868)	The phone parses the value.
Option 6 - Domain Name Server	The phone parses the value.

Option	Results
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/ SNTP server address (RFC 1769)	The phone parses the value.
Option 66 - Provisioning Server Address	The phone parses the value.
Option 128 - 255	Available option range for configuring a custom boot server address when option 66 is not used.
Sub-options configured in Option 43	
Options 1, 2, 3, 4, 5, 6, 7, 15, 42, and 66	The phone parses the value.
Option 128 - 255	Available option range for configuring a custom boot server address when option 66 is not used.

Parse Vendor ID Information

As a part of configuration, the Vendor ID information must be parsed with the Polycom phone.

Polycom follows RFC 3925 which specifies use of a unique Internet Assigned Numbers Authority (IANA) private enterprise number. The private enterprise number assigned to Polycom is 13885 (0x0000363D) and is represented as an array of binary data.

Procedure

1. Check for the Polycom signature at the start of the option: `4 octet: 00 00 36 3d`
2. Obtain the length of the entire list of sub-options: `1 octet`
3. Read the field code and length of the first sub-option, `1+1 octets`
4. If this is a field you want to parse, save the data.
5. Skip to the start of the next sub-option.
6. Repeat steps 3 to 5 until you have all the data or you encounter the End-of-Suboptions code (0xFF).

Required Ports

Topics:

- Ports Used on Polycom Phones

Polycom phones require certain network ports.

Ports Used on Polycom Phones

The following table lists the ports currently used by Polycom UC Software.

Telnet is disabled by default on VVX phones.

H.323 is available only on the VVX 500/501, 600/601, and 1500.

RTP and RTCP can use any even port between 2222 and 2269 (2317 on VVX 500/501, 600/601, or 1500), but you can configure ports by setting `tcpIpApp.port.rtp.mediaPortRangeStart` .

Ports Used by Polycom Phones

Port Number	Protocol	Outgoing	Incoming	UDP or TCP
21	FTP	Provisioning, Logs		TCP
22	SSH	Admin	Admin	TCP
23	Telnet	Admin		TCP
53	DNS			UDP
67	DHCP	Server		UDP
68	DHCP	Client		UDP
69	TFTP	Provisioning, Logs		UDP
80	HTTP	Provisioning, Logs, Pull Web interface, Poll		TCP
123	NTP	Time Server		UDP
389	LDAP			
443	HTTPS	Provisioning, Logs	HTTP Pull Web interface, HTTP Push	TCP
514	Syslog	Logs		
636	LDAP			
1719	H.323	RAS Signaling	RAS Signaling	

Port Number	Protocol	Outgoing	Incoming	UDP or TCP
1720	H.323	Signaling	Signaling	
2222	RTP	Media Packets	Media Packets	
2223	RTCP	Media Packet Statistics	Media Packet Statistics	
5060	SIP	SIP signaling	SIP signaling	
5061	SIP over TLS	Secure signaling	Secure signaling	
24800	PDC	PDC Client messages	PDC Server messages	TCP

Manually Configuring Phones

Topics:

- Configuring Phones Using the Phone Menu
- Configuring Phones Using the Web Configuration Utility

Polycom offers several methods to manually configure your phone.

You can use the phone menu to configure settings or access the phone through a web interface. When you use the web interface, you can copy settings from one phone to another.

If you need to set up more than 20 phones, Polycom recommends using a centralized provisioning server instead of manual configuration.

Configuring Phones Using the Phone Menu

You can use the menu system on your device as the sole configuration method or along with other methods.

Changes you make from the phone menu override the settings you configure using other methods.

You can access the administrator configuration settings on the **Advanced** menu, which requires an administrator password (the default is `456`). Some setting changes require a device restart or reboot.

Menu systems and interface settings vary by device and by UC Software release. For more information on using your device's phone menu, refer to your device's product documentation.

Configuring Phones Using the Web Configuration Utility

The Web Configuration Utility is a web-based interface that enables you to update the software and configure the phone's current settings.

Changes you make using the Web Configuration Utility override the settings you configure using a centralized provisioning server (if applicable).

You can also import and export configuration files using the Web Configuration Utility to configure multiple phones using the same settings.

For more information on using the Web Configuration Utility, see the *Polycom Web Configuration Utility User Guide* at the Polycom UC Software Support Center.

Configure a Phone Using Simple Setup

You can use the Web Configuration Utility to configure the minimum settings you need for your phone to work.

Procedure

1. Enter your phone's IP address into a web browser on your computer.
2. Select **Admin** as the login type, enter the admin password (the default is `456`), and click **Submit**.
3. Click **Simple Setup** and configure the following settings:
   - **Phone Language** Phone display language
   - **SNTP Server** Server that the phone uses to calculate the display time
   - **Time Zone** Time zone where the phone is located
   - **SIP Server** Server address and port that the phone uses for line registrations
   - **SIP Outbound Proxy** Outbound proxy server address and port that the phone uses to send all SIP requests
   - **SIP Line Identification** Information your phone needs to make calls, such as the phone display name, line address, authentication credentials, and line label
4. Click **Save**.

Configuring Phones by Importing Configuration Files

After you have configured a phone, its settings are saved in its configuration file.

To save time, you can export this configuration file and import it to other phones when you want the same configuration on multiple phones.

Export a Phone Configuration File

You can export the phone's configuration file using the Web Configuration Utility to make changes to the phone's current settings.

You can also export the file from one phone so you can import it into another one.

Procedure

1. Enter your phone's IP address into a web browser on your computer.
2. Select **Admin** as the login type, enter the admin password (the default is `456`), and click **Submit**.
3. Go to **Utilities** > **Import & Export Configuration**.
4. Choose the files to export from the **Export Configuration file** drop-down menu and click **Export**.

Import a Phone Configuration File

You can import a configuration file to your phone using the Web Configuration Utility.

Procedure

1. Enter your phone's IP address into a web browser on your computer.
2. Select **Admin** as the login type, enter the admin password (the default is `456`), and click **Submit**.
3. Go to **Utilities** > **Import & Export Configuration**.

4. Click **Choose File** to select the configuration file from your computer to import and click **Import**.

Reset to Default Settings

You can reset your phone settings to default using the Web Configuration Utility.

Procedure

1. Enter your phone's IP address into a web browser on your computer.
2. Select **Admin** as the login type, enter the admin password (the default is `456`), and click **Submit**.
3. Click **Simple Setup** and then click **Reset to Default**.

Provisioning Phones

Topics:

- Network Requirements for Provisioning
- Provisioning Server Discovery Methods
- Setting Up Your Provisioning Server
- Provisioning Phones

You can configure and provision multiple phones with the same settings for large-scale deployments.

If you need to set up more than 20 phones, Polycom recommends using a centralized provisioning server instead of manual configuration.

Network Requirements for Provisioning

Provisioning requires that your phones can securely reach your provisioning server and that your network time settings are in sync with your phones.

Related Links

Supported Network Configurations on page 23

User Accounts

Each phone user must have an account on your SIP call server.

Recommended Security Settings for Provisioning

Although optional, Polycom recommends using the following security settings when using a provisioning server.

- 802.1X
- VLAN
- File transfers using HTTPS
- SIP signaling over Transport Layer Security (TLS)
- Permissions for configuration and override files

Configure File Upload Permissions

When anyone modifies settings from the phone user interface or Web Configuration Utility, the phone attempts to upload override files with settings to the central server.

When your environment includes a provisioning server, you can permit the phone to upload the override file to the provisioning server by giving the phone write access to the provisioning server. Allowing the phone access to the provisioning server enables user settings to survive restarts, reboots, and software upgrades administrators apply to all phones from the provisioning server.

You can also use the override files to save user custom preferences and to apply specific configurations to a device or device group.

By default, provisioned phones attempt to upload phone-specific override and other configuration files to the server, but you must configure the server to allow these files to upload. Allowing these file uploads to the provisioning server gives you greater manageability for your phone deployment and help with troubleshooting issues.

Ensure that the file permissions you create provide the minimum required access and that the account has no other rights on the server. All other files that the phone needs to read, such as the application executable and standard configuration files, should be read-only.

If you reformat the phone's file system, the override file is deleted from the phone.

Procedure

1. Configure the server account with read, write, and delete permissions.
2. Create a separate directory on the server for each file type you want to upload and configure the permissions.

   Each directory can have different access permissions.

   Some example file directories include:

   - Log files
   - Override files
   - Contact directory
   - License directory
3. Edit the attributes of the master configuration file that correspond to the directories you created.
4. To allow a phone's override files to upload to the server, configure the override files with enable, read, and write access.

   The default override file names are the following:

   - Phone Menu `<MAC Address>-phone.cfg`
   - Web Configuration Utility `<MAC Address>-web.cfg`

Dynamic Host Configuration Protocol (DHCP)

Polycom recommends using DHCP where possible to eliminate repetitive manual data entry.

After establishing network connectivity, the phone needs to acquire several IPv6 or IPv4 network settings. These settings are typically obtained automatically from a Dynamic Host Configuration Protocol (DHCPv6 or DHCPv4) server.

Synchronized Time Settings

Its important to use a SNTP server in your network environment.

If SNTP settings are not available through DHCP, you may need to edit the SNTP GMT offset or SNTP server address, especially for the default daylight savings parameters outside of North America. Depending on your local security policy, you might need to disable the local web (HTTP) server or change its signaling port.

DNS

You need to set up Domain Name System (DNS).

Polycom supports the following DNS records types:

- DNS A record
- Service (SRV) record for redundancy
- Name Authority Pointer (NAPTR)

Provisioning Server Discovery Methods

After the phone has established network settings, it must discover a provisioning server to obtain software updates and configuration settings:

- **Static** You can manually configure the server address from the phone's user interface or the Web Configuration Utility, or you can provision a server address using `device.prov.serverName` and corresponding device parameters.
- **DHCP** A DHCP option is used to provide the address or URL between the provisioning server and the phone.
- **DHCP INFORM** The phone makes an explicit request for a DHCP option (which can be answered by a server that is not the primary DHCP server). For more information, see RFC 3361and RFC 3925.
- **Quick Setup** This feature takes users directly to a screen to enter the provisioning server address and information. This is simpler than navigating the menus to the relevant places to configure the provisioning parameters. For more information, see *Using Quick Setup with Polycom Phones: Technical Bulletin 45460 at* Polycom Engineering Advisories and Technical Notifications.
- **ZTP** If a provisioning server address is not discovered automatically using DHCP and a static address has not been entered, the phone contacts the Polycom ZTP server and requests initial configuration files, including the address of the service provider or enterprise provisioning server.

Supported Provisioning Protocols

By default, Polycom phones are shipped with FTP enabled as the provisioning protocol.

You can configure the phone using the following supported provisioning protocols:

- Trivial File Transfer Protocol (TFTP).
- File Transfer Protocol (FTP).
- Hyper Text Transfer Protocol - Secure (HTTPS).
- File Transfer Protocol - Secure (FTPS). When using FTPS as the provisioning protocol:
    - Set the value of `log.render.file.size` to 512.
    - Disable the Diffie-Hellman key exchange

Setting Up Your Provisioning Server

You can use a single provisioning server or configure multiple provisioning servers.

Your provisioning servers should be RFC compliant.

Install Provisioning Tools

Before you begin provisioning devices with UC Software, install tools on your computer and gather some information.

Procedure

1. If using Power over Ethernet (PoE) with the phone, obtain a PoE switch and network cable.
2. Install an XML editor, such as XML Notepad 2007, on your computer.
3. Install an FTP server application on your computer.

   FileZilla and **wftpd** are free FTP applications for windows and **vsftpd** is typically available with all standard Linux distributions.
4. Take note of the following:
   - **SIP server address** Host name or IP address of the call server that handles VoIP services on your network.
   - **SIP account information** SIP account credentials and the phone's registration address.
   - Although a user name and password are not required to get the phone working, Polycom strongly recommends using them for security reasons.
   - **Phone MAC addresses** Unique 12-digit serial number just above the phone's bar code on a label on the back of the phone. You need the MAC address for each phone in your deployment.
   - **Provisioning server IP address** IP address for the system used as the provisioning server. If you want to use your computer system as the provisioning server, then you need your computer's IP address.

Set Up a Single Provisioning Server

You can set up a single provisioning server for your phone deployment.

Procedure

1. Power on the phones and connect them to your VoIP network using a Power over Ethernet (PoE) switch or external adapter and a network cable.
2. Create a root FTP directory on the provisioning computer with full read and write access to all directories and files.

   This is where you need to place configuration files.
3. In your FTP server application, create a user account for the phone to use and take note of the user name and password.
4. Launch the FTP application.

   You must keep it running during provisioning so that the phones can communicate with the UC software.
5. Download Polycom UC Software from Polycom Support and uncompress the files into your root FTP directory.

   You can choose the combined UC software package or the split UC software package.
   - 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(s) and maintain software versions for each model in the same root directory.

Set Up Multiple Provisioning Servers

You can configure multiple (redundant) provisioning servers—one logical server with multiple addresses.

You can set up a maximum of eight provisioning servers.

You must be able to reach all of the provisioning servers with the same protocol, and the contents on each provisioning server must be identical.

Procedure

1. Power on the phones and connect them to your VoIP network using a Power over Ethernet (PoE) switch or external adapter and a network cable.
2. Create a root FTP directory on the provisioning computer with full read and write access to all directories and files.

   This is where you need to place configuration files.
3. In your FTP server application, create a user account for the phone to use and take note of the user name and password.
4. Launch the FTP application.

   You must keep it running during provisioning so that the phones can communicate with the UC software.
5. Download Polycom UC Software from Polycom Support and uncompress the files into your root FTP directory.

   You can choose the combined UC software package or the split UC software package.

   - 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(s) and maintain software versions for each model in the same root directory.
6. Map the provisioning server DNS name to a unique IP address for each server.
7. Configure the following settings:
   - Number of times a file transfer tries each server
   - How long to wait between each file transfer attempt
   - Maximum number of servers to which you want to try to transfer files

Test the Provisioning Settings

You can test your provisioning server setup by using the **Quick Setup** option on your device.

This option enables you to access the provisioning server and configure the phone for provisioning.

For more detail details on how to configure quick setup, see Technical Bulletin 45460: Using Quick Setup with Polycom Phones.

After the initial configuration is complete, you can continue to show or hide the **Quick Setup** option.

Related Links

Quick Setup Soft Key Parameters on page 615

Provisioning Phones

You provision phone features and settings with the UC software configuration files that you create and modify on your provisioning server.

You can also create and update specific phone configuration files, use variable substitution to update all phones in your deployment simultaneously, or configure phone groups.

When provisioning phones, you create configuration files as needed to support your deployment. When creating configuration files; however, do not use the following file names (the phones use these files to store override and logging information):

- `<MACaddress>-phone.cfg`
- `<MACaddress>-web.cfg`
- `<MACaddress>-app.log`
- `<MACaddress>-boot.log`
- `<MACaddress>-license.cfg`

Note: You can use the multiple key combination shortcut by simultaneously pressing **1-4-7** to display the following provisioning information on the phone:

- Phone IP address
- Phone MAC address
- VLAN ID
- Boot server type (FTP, TFTP, HTTP, HTTPS)

Provision Multiple Phones

You need to ensure that your phones are directed to the provision server.

You need to modify the default master configuration file with the provisioning server information.

Procedure

1. Create a `phone<MACaddress>.cfg` file for each phone you want to deploy.
2. Add the SIP server registration information and user account information to the appropriate parameters in the phone configuration file, such as `reg.1.address`, `reg.1.auth.userId`, `reg.1.auth.password`, `reg.1.label`, `reg.1.type` .
3. Create a `site<location>.cfg` file for each site location.

   Include SIP server or feature parameters such as `voIpProt.server.1.address` and `feature.corporateDirectory.enabled` .
4. Add the file name of each phone and site configuration file to the `CONFIG_FILES` attribute of the master configuration file, such as a reference to `phone<MACaddress>.cfg` and `sipVVX500.cfg` .
5. On each phone's **Home** screen or idle display, select **Settings** > **Advanced** > **Admin Settings** > **Network Configuration** > **Provisioning Server.**

When prompted for the administrative password, enter `456`.

6. Press **Select**.
7. Scroll down to **Server Type** and make sure that it is set to **FTP**.
8. Scroll down to **Server Address** and enter the IP address of your provisioning server.

   Press **Edit** to edit the value and then press **OK**.
9. Scroll down to **Server User** and **Server Password** and enter the user name and password of the account you created on your provisioning server.
10. Press **Back** twice.
11. Scroll down to **Save & Reboot**, and then press **Select**.

    The phone reboots and the UC software modifies the `APP_FILE_PATH` attribute of the master configuration file so that it references the appropriate `sip.ld` files.
12. Verify that the phones are provisioned:

    a. On the phone, press **Settings** (**Menu** if using a VVX 1500) and go to **Status** > **Platform** > **Application** > **Main** to see the UC software version and **Status** > **Platform** > **Configuration** to see the configuration files downloaded to the phone.

    b. Monitor the provisioning server event log and the uploaded event log files (if permitted).

    The phone uploads two logs files to the `LOG_DIRECTORY` directory: `<MACaddress>-app.log` and `<MACaddress>-boot.log` .

Provision Phones Using Variable Substitution

You can configure multiple phones in your deployment using variable substitution with a single master configuration file instead of a `<MACaddress>.cfg` file for each phone.

This method is useful if you need to maintain or modify settings common to all phones in your deployment or to specific phone groups based on variables such as phone model or part number. Additionally, if you want to add a new phone to your deployment, you need only create one new file.

Procedure

1. Create a configuration file for each phone containing the information you want to configure, such as registration information.

   You must identically name each of these phone-specific configuration files except for the information you plan to substitute with a variable string, such as phone's MAC address, part number, or phone model.

   For example, create phone-specific configuration files that contain registration information and name them `reg-basic_0004f2000001.cfg` , `reg-basic_0003a7100076.cfg` , `reg-basic_0004e5800094.cfg` , and so forth.
2. Copy one of the configuration file names and modify it by replacing the specific phone information with the corresponding variable as shown in the following table (make sure you include the square brackets).

   For example, change `reg-basic_0004f2000001.cfg` to `reg-basic_[PHONE_MAC_ADDRESS].cfg` or change `reg-basic_VVX500.cfg` to `reg-basic_[PHONE_MODEL].cfg` .

Variable	Description
[PHONE_MAC_ADDRESS]	Use to configure all phones in your deployment
[PHONE_PART_NUMBER]	Use to configure all phones with a specific part number
[PHONE_MODEL]	Use to configure a specific phone model

3. Add the file name with the variable substitution to the `CONFIG_FILES` attribute of the master configuration file.
4. Save the master configuration file.

Find a Phone's MAC Address

Each phone has a unique a-f hexadecimal digit called a MAC address, also known as the serial number (SN).

You can use the MAC address to create variables in the name of the master configuration file, or to specify phone-specific configuration files. There are three ways to find a phone's MAC address.

Procedure

1. Do one of the following:
   - Look on the label on the back of the phone.
   - On the phone, press **Settings** (**Menu** if using a VVX 1500) and go to **Status** > **Platform** > **Phone** > **S/N:**.
   - Use a multi-key shortcut by simultaneously pressing **1-4-7**.

Provision an Individual Phone

You can configure phones individually by creating an individual master configuration file for each phone.

This configuration method gives you a high degree of control over each phone, but for large deployments, the file naming scheme can require additional file management as you must create and edit at least two unique files for each phone in your deployment.

Procedure

1. Create a copy of the master configuration file template for the phone and name it `<MACaddress>.cfg` , replacing `000000000000` with the unique MAC address of the phone you want to configure.

   Note that you must use only numerals and lowercase letters in the file name.
2. Create a configuration file for the phone containing its unique information such as registration information.

   Name your files based on the file contents or purpose. You can use the template files in the UC software download, or you can create your own configuration file using parameters from the UC software template files.

   For example, you might use parameters from the `reg-basic.cfg` template file to create a registration file named `reg-basic_john_doe.cfg` .

3. Enter the name of the configuration files you created to the `CONFIG_FILES` attribute of the phone's `<MACaddress>.cfg` file.
4. Save the master configuration file.

Provision a Phone Group

You can apply features and settings to a phone group by phone model name or part number.

If you create configuration files for phone groups using the part number and model name for the same type of phone, the part number configuration file has priority over the phone model configuration file.

Procedure

1. Create a configuration file with the settings you want to apply.

   Name the file using the phone group's part number or phone model name, such as `3111-44500-001.cfg` or `VVX500.cfg`.
2. Add the file name to the `CONFIG_FILES` attribute of the master configuration file.
3. Save the master configuration file.

Working with Configuration Files

Topics:

- Master Configuration File

Polycom UC Software includes a number of resource files, template configuration files, and an XML schema file that provides examples of parameter types and permitted value types.

The resource and configuration files contains parameters you can use to configure features and apply settings to phones. You use configuration files when provisioning phones via a provisioning server, although you can also export and import configuration files between individual phones.

In order to work with configuration files, you'll need to install an XML editor.

Master Configuration File

The master configuration file maximizes the flexibility you have to customize features and settings for your devices in large deployments.

You can use the master configuration file to configure features and apply settings for any or all the phones in your deployment, including various groups of phones, specific phone models, or a single phone.

The default name for the master configuration file is `000000000000.cfg` . You can use the default name or rename the master configuration file to configure features and settings for your phone deployment. The file name must contain at least five characters and end with `.cfg` .

You can also specify the location of a master configuration file you want the phones to use, for example, `http://usr:pwd@server/dir/example1.cfg` . If the phone cannot find and download a file from that location, the phone uses an individual phone master configuration file or the default master configuration file.

The master configuration file applies the settings from the component configuration files listed in the `CONFIG_FILES` attribute in the following ways:

- The files you enter are read from left to right.
- Duplicate settings are applied from the configuration file in the order you list them.

The following table describes the XML field attributes in the master configuration file and the `APPLICATION` directories.

Master Configuration File XML Field Attributes

Attribute	Description
`APP_FILE_PATH`	The path name of the UC software application executable. The default value is `sip.ld` . Note that the phone automatically searches for the `sip.ld` and `<part number>.sip.ld` files. This field can have a maximum length of 255 characters. If you want the phone to search for a `sip.ld` file in a location other than the default or use a different file name, or both, modify the default. For example, you can specify a URL with its own protocol, user name, and password: `http://usr:pwd@server/dir/sip.ld` .
`DECT_FILE_PATH`	The path for the application executable for the Polycom VVX D60 Wireless Handset. The default value is 3111-17823-001.dect.ld. When the software for a VVX business media phone with a paired VVX D60 Base Station is updated, the phone also searches for the dect.ld for any updates to the base station software. If you want the phone to search for the 3111-17823-001.dect.ld in a location other than the default or use a different file name, or both, modify the default. For example, you can specify a URL with its own protocol, user name, and password: `http://usr:pwd@server/dir/3111-17823-001.dect.ld`.
`CONFIG_FILES`	Enter the names of your configuration files here as a comma-separated list. Each file name has a maximum length of 255 characters and the entire list of file names has a maximum length of 2047 characters, including commas and white space. If you want to use a configuration file in a different location or use a different file name, or both, you can specify a URL with its own protocol, user name and password, for example: `ftp://usr:pwd@server/dir/phone2034.cfg`. The files names you enter to the CONFIG_FILES field write are read from left to right. Duplicate settings are applied from the configuration file in the order you list them.
`MISC_FILES`	A comma-separated list of files. Use this to list volatile files that you want phones to download, for example, background images and ringtone.wav files. The phone downloads files you list here when booted, which can decrease access time.
`LOG_FILE_DIRECTORY`	An alternative directory for log files. You can also specify a URL. This field is blank by default.
`CONTACTS_DIRECTORY`	An alternative directory for user directory files. You can also specify a URL. This field is blank by default.
`OVERRIDES_DIRECTORY`	An alternative directory for configuration overrides files. You can also specify a URL. This field is blank by default.
`LICENSE_DIRECTORY`	An alternative directory for license files. You can also specify a URL. This field is blank by default
`USER_PROFILES_DIRECTORY`	An alternative directory for the `<user>.cfg` files.

Attribute	Description
CALL_LISTS_DIRECTORY	An alternative directory for user call lists. You can also specify a URL. This field is blank by default.
COREFILE_DIRECTORY	An alternative directory for Polycom device core files to use to debug problems. This field is blank by default.

Note: The directories labeled APPLICATION_SPIPXXX indicate phone models that are not compatible with the latest UC software version. If you are using any of the phone models listed in these directories, open the directory for the phone model you are deploying, and use the available fields to provision and configure your phones.

XML Resource Files

The UC software download contains optional resource configuration files you can apply to the phones.

In addition, you can allow phone-specific override files containing user settings to be uploaded to the central server. Resource and override files include:

- Language dictionaries for the phone menu and Web Configuration Utility
- Configuration override files that store settings made from the phone menu and Web Configuration Utility
- Ringtones
- Log files
- A template contact directory 000000000000-directory~.xml
- A licensing directory

Configuration Templates

Most configuration parameters are located in only one template file, but some are included in two or more files.

You can rearrange the parameters within the template, move parameters to new files, or create your own configuration files from parameters you want. This flexibility is especially useful when you want to apply specific settings to a group of phones. You can create and name as many configuration files as you want and your configuration files can contain any combination of parameters.

The following table lists the template directories and files included in the UC software download.

Note that techsupport.cfg is available from Polycom Customer Support for troubleshooting and debugging.

Configuration File Templates

Name	Description	Deployment Scenarios
Directories		

Name	Description	Deployment Scenarios
PartnerConfig	Contains configuration file specific to the following third-party servers: • Alcatel-Lucent • BroadSoft • GENBAND • Microsoft • Sylantro	For use with third-party servers.
Config		
applications.cfg	For applications, browser, microbrowser, XMP-API	Typical Hosted Service Provider Typical IP-PBX
device.cfg	Network Configuration parameters	Troubleshooting Administrative settings
features.cfg	Features including corporate directory, USB recording, presence, ACD	Typical Hosted Service Provider Typical IP-PBX
firewall-nat.cfg	Firewall parameters	
H323.cfg	H.323 video use	Typical Hosted Service Provider using VVX 500/501, 600/601, and 1500 for video calls
lync.cfg	Microsoft Skype for Business parameters	Typical Microsoft Skype for Business environment
pstn.cfg		
reg-advanced.cfg	Advanced call server, multi-line phones	Typical Hosted Service Provider Typical IP-PBX
reg-basic.cfg	Basic registration	Simple SIP device Typical Hosted Service Provider
region.cfg	Non-North American geographies	Typical Hosted Service Provider Typical IP-PBX
sip-basic.cfg	Basic call server	Simple SIP device Typical Hosted Service Provider
sip-interop.cfg	Advanced call server, multi-line phones	Typical Hosted Service Provider Typical IP-PBX

Name	Description	Deployment Scenarios
`site.cfg`	Multi-site operations	Typical Hosted Service Provider Typical IP-PBX
`techsupport.cfg`	Available by special request from Polycom Customer Support.	Use for troubleshooting and debugging only
`video.cfg`	VVX 500/501, 600/601, and 1500 video	Typical Hosted Service Provider if using VVX 500/501, 600/601, and 1500 for video calls
`video-integration.cfg`		

Using Correct Parameter XML Schema, Value Ranges, and Special Characters

The configuration parameters available in the UC software templates use a variety of value types.

UC software includes an XML schema file (`polycomConfig.xsd`) that provides information about parameter type, permitted values, default values, and valid enumerated type values. You can view this template file with an XML editor.

Polycom configuration parameters support the following value types:

- Boolean
- Enumerated
- Integer
- String

The following rules apply to UC software parameter values:

- Boolean values are not case sensitive.
- UC software interprets `Null` as empty.
- The values `0, false,` and `off` are supported and interchangeable.
- The values `1, true` , and `on` are supported and interchangeable. This administrator guide documents only `0` and `1` .

The following rules apply when you set a parameter with a numeric value outside of its valid range:

- If the value is greater than the allowable range, the maximum allowable value is used.
- If the value is less than the allowable range, the minimum allowable value is used.
- If you insert invalid parameter values into the configuration file, the value is ignored and the default value is used. Examples of invalid parameter values include enumerated values that do not match values defined in the UC software, numeric parameters set to non-numeric values, string parameters whose value is too long or short, and null strings in numeric fields. Invalid values are logged in the phone's log files.

You must use the appropriate XML code for special characters in a configuration file:

- & as `&amp;`
- ” as `&quot;`
- ' as `&apos;`

- < as `&lt;`
- > as `&gt;`
- random numbers as `&0x12;`

Microsoft Exchange Integration

Topics:

If you have a Skype for Business, Office 365, Lync Server 2010 or 2013 deployment, you can integrate with Microsoft Exchange Server.

You can set up visual voicemail, call log synchronization, Outlook contact search, and Skype for Business Address Book Service (ABS) adaptive search. Each of these features is enabled by default on Polycom phones registered with Skype for Business.

Note: If your Polycom phones are configured with G.722 and users find that they do not hear audio when retrieving voicemail from the Microsoft Skype for Business Server, you need to make the following changes to parameters in the site.cfg template file:

- Change `voice.codecPref.G7221.24kbps` from 0 to 5.
- Change `voice.codecPref.G7221.32kbps` from 5 to 0.
- Add `voice.audioProfile.G7221.24kbps.payloadType` and set it to 112.

After the phone is connected with the Exchange Server, you can:

- Verify the status of Exchange Server services on each phone.
- View the status of each service in the Web Configuration Utility.

Skype for Business

Skype for Business and Lync Server provides a unified communications (UC) solution that enables customers, colleagues, and business partners to communicate instantly by voice, video, or messaging through a single interface, regardless of their location or network.

Note that the concurrent failover/fallback feature is not compatible in a Microsoft environment.

For full administrator instructions on deploying and setting up features with Skype for Business and Lync Server, see the latest *Polycom UC Software with Skype for Business - Deployment Guide* on Polycom Support.

The features available when you are registered with Skype for Business Server vary with the Polycom phone model and Polycom UC Software version you are using. Polycom UC Software supports the following devices with Skype for Business and Lync Server:

- VVX 201, 300 series, 400 series, 500 series, and 600 series business media phones
- VVX 250, 350, and 450 business IP phones
- SoundStructure VoIP Interface

If you are using UC Software with Skype for Business and want to change default settings or customize your deployment, you must set up a provisioning server.

Polycom UC Software enables you to register only a single phone line with Skype for Business Server. When you register a line on a Polycom phone using Skype for Business Server you cannot register lines with another server.

Integrating with Microsoft Exchange

You can integrate with Microsoft Exchange using one of the following methods:

- Exchange Server auto-discover
- Provision the phone with the Microsoft Exchange address
- Web Configuration Utility

Note: If you enter sign-in credentials to the configuration file, phone users must enter credentials to the phone **Sign In** screen.

Provision the Microsoft Exchange Calendar

You can provision your phones with the Microsoft Exchange calendar.

Procedure

1. Add the following parameters to one of your configuration files:
    - `feature.exchangeCalendar.enabled=1`
    - `exchange.server.url=https://<example URL>`

Enable Microsoft Exchange Calendar Using the Web Configuration Utility

You can use the Web Configuration Utility to manually enable your phones with the Microsoft Exchange calendar.

This is useful for troubleshooting if auto-discovery is not working or misconfigured. This method applies only to a single phone at a time.

Procedure

1. Enable access to the Web Configuration Utility if the phone is registered with Skype for Business.
2. Log in to the Web Configuration Utility as Admin (default password `456`).
3. Go to **Settings** > **Applications** > **Exchange Applications,** and expand **Exchange Applications**.
4. In the **Exchange Calendar** field, select **Enable**.
5. Enter the exchange web services URL using a Microsoft Exchange Server URL, for example `https://<mail.com>/ews/exchange.asmx`.

   In this example, the URL part `<mail.com>` is specific to an organization
6. At the bottom of the browser page, click **Save**.
7. When the confirmation dialog displays, click **Yes**.

Your Exchange Calendar is successfully configured and the Calendar icon displays on your phone screen.

Verify the Microsoft Exchange Integration

You can verify if all of the Exchange services are working.

Procedure

1. Go to **Status** > **Diagnostics** > **Warnings** on the phone.
2. View the status of each service in the Web Configuration Utility.

Configuring the Microsoft Exchange Server

You should configure the following settings to take advantage of Microsoft Exchange services on your phones.

Note: Web Info: For help with Lync Server 2010, refer to Microsoft Configure Exchange Services for the Autodiscover Service.

For help with Lync Server 2013, refer to Microsoft Configuring Unified Messaging on Microsoft Exchange Server to work with Lync Server 2013.

Visual Voicemail

On the Exchange server, you can enable unified messaging and enable messages to play on the phone for each user.

If you disable `feature.exchangeVoiceMail.enabled` , the Message Center and Skype for Business Voice mail menus display the message: Skype for Business Server only plays voicemail and you cannot download voicemails or play locally on the phone.

Synchronizing Call Logs

On the Exchange server, you can enable the option to save calls logs to each user's conversation history in Outlook.

Call Log Synchronization Parameters

Use the following parameters to configure call logs.

Call Log Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.exchangeCallLog.enabled`	1 (default) - The Exchange call log feature is enabled and the user call log history of Missed, Received, and outgoing calls can be retrieved on the phone. You must also enable the parameter `feature.exchangeCalendar.enabled` to use the Exchange call log feature. If you disable `feature.exchangeCalendar.enabled` , also disable `feature.exchangeCallLog.enabled` to ensure call log functionality. 0 (default) - The Exchange call log feature is disabled and the user call logs history cannot be retrieved from the Exchange server.	No

Directory Search

You can enable the ABS service on the Exchange server.

There are three possible configurations.

- Outlook and ABS are both enabled by default. When both are enabled, the phone displays the Skype for Business Directory.
- If you disable Outlook and enable only ABS, the phone displays the Skype for Business Directory.
- If you enable Outlook and disable ABS, the Outlook Contact Search displays in Directories.

Microsoft Exchange Parameters

The following table lists parameters that configure the Microsoft Exchange integration.

Microsoft Exchange Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	exchange.meeting.alert.followOfficeHours	1 (default) - Audible alerts occur during business hours. 0 - Audible alerts occur at all times.	No
applications.cfg	exchange.meeting.alert.tonePattern	positiveConfirm (default) - Set the tone pattern of the reminder alerts using any tone specified by se.pat.*.	No
applications.cfg	exchange.meeting.alert.toneVolume	10 (default) - Set the volume level of reminder alert tones. 0 - 17	No
applications.cfg	exchange.meeting.allowScrollingToPast	0 (default) - Do not allow scrolling up in the Day calendar view to see recently past meetings. 1 - Allow scrolling up in the Day calendar view to see recently past meetings.	No
applications.cfg	exchange.meeting.parseOption	Indicates the field in the meeting invite from which the VMR or meeting number should be fetched. Location (default) All LocationAndSubject Description	No
applications.cfg	exchange.meeting.phonePattern	NULL (default) string The pattern used to identify phone numbers in meeting descriptions, where "x" denotes any digit and "|" separates alternative patterns (for example, xxx-xxx-xxxx|604.xxx.xxxx).	No
applications.cfg	exchange.meeting.reminderEnabled	1 (default) - Meeting reminders are enabled. 0 - Meeting reminders are disabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`exchange.meeting.reminderInterval`	300 seconds (default) 60 - 900 seconds Set the interval at which phones display reminder messages.	No
`applications.cfg`	`exchange.pollInterval`	The interval, in seconds, to poll the Exchange server for new meetings. 30000 (default) 4000 minimum 60000 maximum	No
`applications.cfg`	`exchange.meeting.reminderSound.enabled`	1 (default) - The phone makes an alert sound when users receive reminder notifications of calendar events. 0 - The phone does not make an alert sound when users receives reminder notifications of calendar events. Note that when enabled, alert sounds take effect only if `exchange.meeting.reminderEnabled` is also enabled.	No
`applications.cfg`	`exchange.meeting.reminderType`	Customize the calendar reminder and tone. 2 (default) - Reminder is always audible and visual. 1 - The first reminder is audible and visual reminders are silent. 0 - All reminders are silent.	No
`applications.cfg`	`exchange.server.url`	NULL (default) string The Microsoft Exchange server address.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`feature.EWSAutodiscover.enabled`	If you configure `exchange.server.url` and set this parameter to 1, preference is given to the value of `exchange.server.url` . 1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - Exchange autodiscovery is enabled and the phone automatically discovers the Exchange server using the email address or SIP URI information. 0 - Exchange autodiscovery is disabled on the phone and you must manually configure the Exchange server address.	No
`applications.cfg`	`feature.exchangeCalendar.enabled`	1 (default) - The calendaring feature is enabled. 0 - The calendaring feature is disabled. You must enable this parameter if you also enable `feature.exchangeCallLog.enabled` . If you disable `feature.exchangeCalendar.enabled` , also disable `feature.exchangeCallLog.enabled` to ensure call log functionality.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.exchangeCalendar.enabled`	Available for: ▪ Polycom Trio 8800 and 8500 systems ▪ VVX 300/301, 310/311, 400/401, 410/411, 500/501, 600/601 and 1500 business media phones ▪ VVX 250, 350, and 450 business IP phones ▪ CX5500 Unified Conference Station 1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 0 - The calendaring feature is disabled. 1 - The calendaring feature is enabled. You must enable this parameter if you also enable `feature.exchangeCallLog.enabled` . If you disable `feature.exchangeCalendar.enabled` , also disable `feature.exchangeCallLog.enabled` to ensure call log functionality.	No
`features.cfg`	`feature.exchangeContacts.enabled`	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - The Exchange call log feature is enabled and users can retrieve the call log histories for missed, received, and outgoing calls. 0 - The Exchange call log feature is disabled and users cannot retrieve call logs histories. You must also enable the parameter `feature.exchangeCallLog.enabled` to use the Exchange call log feature.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.exchangeContacts. enabled`	1 (default) - The Exchange call log feature is enabled and users can retrieve call logs history of Missed, Received, and outgoing calls on the phone. 0 - The Exchange call log feature is disabled and users cannot retrieve call logs history from the Exchange server. You must also enable the parameter `feature.exchangeCallLog.enabled` to use the Exchange call log feature.	No
`features.cfg`	`feature.exchangeVoiceMail.enabled`	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - The Exchange voicemail feature is enabled and users can retrieve voicemails stored on the Exchange server from the phone. 0 - The Exchange voicemail feature is disabled and users cannot retrieve voicemails from Exchange Server on the phone. You must also enable `feature.exchangeCalendar.enabled` to use the Exchange contact feature.	No
`features.cfg`	`feature.exchangeVoiceMail. enabled`	1 (default) - The Exchange voicemail feature is enabled and users can retrieve voicemails stored on the Exchange server from the phone. 0 - The Exchange voicemail feature is disabled and users cannot retrieve voicemails from Exchange Server on the phone. You must also enable `feature.exchangeCalendar.enabled` to use the Exchange contact feature.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.exchangeVoiceMail.skipPin.enabled`	0 (default) - Enable PIN authentication for Exchange Voicemail. Users are required to enter their PIN before accessing Exchange Voicemail. 1 - Disable PIN authentication for Exchange Voicemail. Users are not required to enter their PIN before accessing Exchange Voicemail.	No
`features.cfg`	`feature.lync.abs.enabled`	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - Enable comprehensive contact search in the Skype for Business address book service. 0 - Disable comprehensive contact search in the Skype for Business address book service.	No
`features.cfg`	`feature.lync.abs.maxResult`	Define the maximum number of contacts to display in a Skype for Business address book service contact search. 12 (default) 5 - 50	No
`features.cfg`	`up.oneTouchDirectory`	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 1 - The Skype for Business Search icon displays on the Home screen. 0 - The Skype for Business Search icon does not display on the Home screen.	No
`features.cfg`	`up.oneTouchVoiceMail`[1]	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 0 - The phone displays a summary page with message counts. The user must press the Connect soft key to dial the voicemail server. 1 - The phone dials voicemail services directly (if available on the call server) without displaying the voicemail summary.	No

Configuring Security Options

Topics:

Polycom UC Software enables you to optimize security settings.

These includes changing the passwords for the phone, enabling users to lock their phones, and blocking administrator functions from phone users.

Administrator and User Passwords

You can change the default administrator and user passwords.

When you set the Base Profile to Skype or update your phones to UC Software 5.x.x or later, the phones display a message prompting you to change the default administrator password (`456`). Polycom strongly recommends that you change the default password. This password is not the Skype for Business user Sign In password. The default administrator password enables administrators to access advanced settings menu on the phone menu and to log in to a phone's Web Configuration Utility as an administrator.

You can change the default password using any of the following methods:

- The popup prompt when the phone first registers
- Phone menu
- Web Configuration Utility
- Use the parameter `reg.1.auth.password` in the template configuration file

You must have a user or administrator password before you can access certain menu options on the phone and in the Web Configuration Utility. You can use the following default passwords to access menu options on the phone and to access the Web Configuration Utility:

- Administrative password: `456`
- User password: `123`

You can use an administrator password where a user password is required, and you will see all of the user options. If the phone requires the administrator password, you can use the user password, but you are presented with limited menu options. Note that the Web Configuration Utility displays different features and options depending on which password is used.

Change the Default Administrator Password on the Phone

If you do not change the default administrative password, the phone displays a warning and a reminder message each time the phone reboots.

If you are registering Polycom phones with Microsoft Skype for Business Server, a message displays on the phone screen prompting you to change the default password.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, and enter the default password.
2. Select **Administration Settings** > **Change Admin Password**.
3. Enter the default password, enter a new password, and confirm the new password.

Change the Default Passwords in the Web Configuration Utility

You can change the administrator and user passwords on a per-phone basis using the Web Configuration Utility.

If the default administrative password is in use, a warning displays in the Web Configuration Utility.

Procedure

1. In the Web Configuration Utility, select **Settings** > **Change Password**.
2. Update the passwords for the **Admin** and **User**.

Administrator and User Password Parameters

Use the parameters in the following table to set the administrator and user password and configure password settings.

Local Administrator and User Password Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.pwd.length.admin`	The minimum character length for administrator passwords changed using the phone. Use 0 to allow null passwords. 1 (default) 0 -32	Yes
`site.cfg`	`sec.pwd.length.user`	The minimum character length for user passwords changed using the phone. Use 0 to allow null passwords. 2 (default) 0 -32	Yes
`features.cfg`	`up.echoPasswordDigits`	1 (default) The phone briefly displays password characters before being masked by an asterisk. 0 - The phone displays only asterisks for the password characters.	No
`device.cfg, site.cfg`	`device.auth.localAdminPassword`	Specify a local administrator password. 0 - 32 characters You must use this parameter with `device.auth.localAdminPassword.set="1"`	No
`device.cfg, site.cfg`	`device.auth.localAdminPassword.set`	0 (default) - Disables overwriting the local admin password when provisioning using a configuration file. 1 - Enables overwriting the local admin password when provisioning using a configuration file.	No

Security Banner on the Web Configuration Utility

You can enable or disable the security banner on the Web Configuration Utility.

In addition, you can configure a custom text message to be displayed on the security banner of your phone's user interface.

Web Configuration Utility Security Banner Parameters

The following table includes the parameters of the web user interface for security banner parameters.

S

Web Configuration Utility Security Banner Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`feature.webSecurityBanner.enabled`	0 (default) - No security banner message displays on the phone's web user interface. 1 - A security banner with the configured message displays phone's web user interface. Use feature.webSecurityBanner.msg to configure the message.	
`site.cfg`	`feature.webSecurityBanner.msg`	Customize the text in security banner. "This is default text for the security log-on banner" (default) - This text displays because the security log-on banner has been enabled and the custom text to be displayed in the security log-on banner has not been configured. 2000 characters (maximum)	

Locking the Web Configuration Utility after Failed Login Attempts

For additional security, you can lock access to the Web Configuration Utility after a set amount of failed user login attempts and configure a period of time after which a user can attempt to log in again.

Web Configuration Utility Lock Parameters

Use the following parameters to configure how the Web Configuration Utility will behave after failed login attempts.

Lock Web Configuration Utility Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`httpd.cfg.lockWebUI.enable`	1 (default) - Enable the Web Configuration Login Lock feature. 0 - Disable the Web Configuration Login Lock feature.	No
`site.cfg`	`httpd.cfg.lockWebUI.lockOutDuration`	60 seconds (default) - The period of time the user is locked out of the Web Configuration Utility. The user can try logging in again after this time. 60 - 300 seconds The lock-out timer starts after the maximum number of unsuccessful attempts within the duration you configure. After the lock-out time has expired, the timers and the number of incorrect attempts resets to 60 seconds.	No
`site.cfg`	`httpd.cfg.lockWebUI.noOfInvalidAttempts`	5 (default) - After five failed login attempts, the user is locked out of the Web Configuration Utility. Specify the maximum number of failed login attempts after which the user is locked out of the Web Configuration Utility. 3 - 20 seconds	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration`	60 seconds (default) - After a user reaches the maximum failed login attempts within 60 seconds, the user is locked out of the Web Configuration Utility. After a user reaches the maximum failed login attempts within this time duration, the user is 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 - 300 seconds The timer starts again after the first incorrect password attempt.	No

Disabling External Ports and Features

You can disable unused external phone ports and features to increase the security of devices in your deployment.

You can disable the following ports and features:

- Web Configuration Utility
- PC port
- Aux port
- USB port
- Speakerphone
- Call forwarding
- Do Not Disturb
- Push-to-Talk (PTT)
- Auto Answer
- Applications icon
- Headset
- Handset
- Host and device ports

- Bluetooth
- NFC
- Wi-Fi

Note: At least one audio port must be enabled to send and receive calls.

Disable Unused Ports and Features Parameters

Use the parameters in the following table to disable external ports or specific features.

Disable Unused Ports and Features

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`	`device.net.etherModePC`	0 (default) - Disable the PC port mode that sets the network speed over Ethernet. 1 - Enable the PC port mode that sets the network speed over Ethernet.	No
`device.cfg`	`device.auxPort.enable`	0 (default) - Disable the phone auxiliary port. 1 - Enable the phone auxiliary port.	No
`site.cfg`	`httpd.enabled`	Base Profile = Generic 1 (default) - The web server is enabled. 0 - The web server is disabled. Base Profile = Skype 0 (default) - The web server is disabled. 1 - The web server is enabled.	Yes
`site.cfg`	`ptt.pttMode.enable`	0 (default) - Disable push-to-talk mode. 1 - Enable push-to-talk mode.	No
`features.cfg`	`feature.callRecording.enabled`	0 (default) - Disable the phone USB port for local call recording. 1 - Enable the phone USB port for local call recording.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.handsfreeMode	1(default) - Enable handsfree mode. 0 - disable handsfree mode.	No
features.cfg	feature.forward.enable	1(default) - Enable call forwarding. 0 - Disable call forwarding.	No
features.cfg	homeScreen.forward.enable	1(default) - Turn on display of the call forward icon on the phone Home screen. 0 - Turn on or off display of the call forward icon on the phone Home screen.	No
features.cfg	feature.doNotDisturb.enable	1(default) - Enable Do Not Disturb (DND). 0 - Disable Do Not Disturb (DND).	Yes
features.cfg	homeScreen.doNotDisturb.enable	1 (default) - Enables the display of the DND icon on the phone's Home screen. 0 - Disables the display of the DND icon on the phone's Home screen.	No
features.cfg	call.autoAnswerMenu.enable	1 (default) - Enables the phone's Autoanswer menu. 0 - Disables the phone's Autoanswer menu.	No
features.cfg	homeScreen.application.enable	1 (default) - Enables the Applications icon on the phone's Home screen. 0 - Disables the Applications icon on the phone's Home screen.	No
features.cfg	up.headsetModeEnabled	1 (default) - Enables the headset port. 0 - Enable or disable the headset port.	No
features.cfg	softkey.feature.doNotDisturb	1 (default) - Enables the DND soft key on the phone. 0 - Disables the DND soft key on the phone.	No

Visual Security Classification

The security classification of a call is determined by the lowest security classification among all participants connected to a call.

For example, a Top Secret classification displays when all participants in a call have a Top Secret classification level.

Note: Call classification is determined by the lowest classification among all participants in the call. You can safely exchange information classified no higher than the call's security classification. For example, if User A is classified as Top Secret and User B has a lower classification level of Restricted, both User A and B are connected to the call as Restricted.

Phone users can modify their assigned security classification level to a value lower than their assigned level during a call. When the call is over, the server resets the user's classification level to its original state.

Visual Security Classification Parameters

To enable the visual security classification feature, you must configure settings on the BroadSoft BroadWorks server v20 or higher and on the phones.

If a phone has multiple registered lines, administrators can assign a different security classification to each line.

An administrator can configure security classifications as names or strings and set the priority of each on the server in addition to the default security classification level Unclassified. The default security classification Unclassified displays until you set classifications on the server. When a user establishes a call to a phone not connected to this feature, the phone displays as Unclassified.

The following table lists the parameters you can use to configure visual security classification.

Configure Visual Security Classification

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.serverFeatureControl.securityClassification`	0 (default) - The visual security classification feature for all lines on a phone is disabled. 1 - The visual security classification feature for all lines on a phone is enabled.	Yes
`reg-advanced.cfg`	`reg.x.serverFeatureControl.securityClassification`	0 (default) - The visual security classification feature for a specific phone line is disabled. 1 - The visual security classification feature for a specific phone line is enabled.	No

Encryption

Polycom supports the use of encryption to protect configuration files, and phone calls.

Encrypting Configuration Files

Polycom phones can download encrypted files from the provisioning server and encrypt files before uploading them to the provisioning server.

You can encrypt all configuration files except the master configuration file, contact directory files, and configuration override files from the Web Configuration Utility and local device interface. You can also determine whether encrypted files are the same as unencrypted files and use the SDK to facilitate key generation. You cannot encrypt the master configuration file.

To encrypt files, you must provide the phone an encryption key. You can generate your own 32 hex-digit, 128 bit key or use the Polycom Software Development Kit (SDK) to generate a key and to encrypt and decrypt configuration files on a UNIX or Linux server.

Note: To request the SDK and quickly install the generated key, see *When Encrypting Polycom UC Software Configuration Files*: *Quick Tip 67442 at* Polycom Engineering Advisories and Technical Notifications.

You can use the following parameters to set the key on the phone:

- `device.set`
- `device.sec.configEncryption.key`
- `device.sec.configEncryption.key.set`

If the phone doesn't have a key, you must download the key to the phone in plain text, which is a potential security concern if you are not using HTTPS. If the phone already has a key, you can download a new key. Polycom recommends naming each key uniquely to identify which key was used to encrypt a file.

After encrypting a configuration file, it is useful to rename the file to avoid confusing it with the original version, for example, rename **site.cfg** to **site.enc**.

Note: If a phone downloads an encrypted file that it cannot decrypt, the action is logged, and an error message displays. The phone continues to do this until the provisioning server provides an encrypted file that can be read, an unencrypted file, or until the file is removed from the list in the master configuration file.

Change the Encryption Key on the Phone and Server

To maintain secure files, you can change the encryption key on the phones and the server.

Procedure

1. Place all encrypted configuration files that you want to use the new key on the provisioning server.

   The phone may reboot multiple times.

   The files on the server must be updated to the new key or they must be made available in unencrypted format. Updating to the new key requires decrypting the file with the old key, then encrypting it with the new key.

2. Put the new key into a configuration file that is in the list of files downloaded by the phone, specified in `000000000000.cfg` or `<MACaddress>.cfg` .
3. Use the `device.sec.configEncryption.key` parameter to specify the new key.
4. Provision the phone again so that it downloads the new key.

   The phone automatically reboots a second time to use the new key.

   Note that configuration files, contact directory files and configuration override files may all need to be updated if they were already encrypted. In the case of configuration override files, they can be deleted from the provisioning server so that the phone replaces them when it successfully boots.

Configuration File Encryption Parameters

The following table provides the parameters you can use to encrypt your configuration files.

Configuration File Encryption Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`	`device.sec.configEncryption.key`	Set the configuration encryption key used to encrypt configuration files. string	Yes
`site.cfg`	`sec.encryption.upload.callLists`	0 (default) - The call list is uploaded without encryption. 1 - The call list is uploaded in encrypted form.	Yes
`site.cfg`	`sec.encryption.upload.config`	0 (default) - The file is uploaded without encryption and replaces the phone specific configuration file on the provisioning server. 1 - The file is uploaded in encrypted form and replaces the existing phone specific configuration file on the provisioning server.	No
`site.cfg`	`sec.encryption.upload.dir`	0 (default) - The contact directory is uploaded without encryption and replaces the phone specific contact directory on the provisioning server. 1 - The contact directory is uploaded in encrypted form and replaces the existing phone specific contact directory on the provisioning server.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.encryption.upload.overrides`	0 (default) - The MAC address configuration file is uploaded without encryption and replaces the phone specific MAC address configuration file on the provisioning server. 1 - The MAC address configuration file is uploaded in encrypted form and replaces the existing phone specific MAC address configuration file on the provisioning server.	No

FIPS 140-2 Compliance Support

The Federal Information Processing Standard (FIPS 140-2) compliance is a cryptographic function.

You can configure phones to use the FIPS 140-2 compliant cryptography using any one of the following methods:

- Phones user interface
- Web Configuration Utility
- Phone's Updater user interface
- FIPS 140-2 parameters

FIPS 140-2 Parameters

The following table includes the new or modified parameter for the FIPS 140-2 feature.

FIPS 140-2 Parameter

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`device.sec.TLS.FIPS.enabled`	0 (default) - Does not allow the phone to use the FIPS-compliant cryptography feature. 1 - Allows the phone to use the FIPS-compliant cryptography feature.	No

Voice over Secure IP

You can configure phones to dynamically use either Secure Real Time Protocol (SRTP) or Real Time Protocol (RTP) depending on the media security mechanisms negotiated between phone and outbound proxy using Voice over Secure IP (VoSIP). When you enable this feature, the voice signals are transferred securely between endpoints without the need to introduce multiple lines in the Session Description Protocol (SDP).

The following are advantages for Voice over Secure IP (VoSIP):

- The voice signals are encrypted and secure allowing a safe transmission of signals between phones.
- Signaling and media to the cloud hosted product are encrypted.

VoSIP Parameters

The following table lists parameters to configure VoSIP.

Voice over Secure IP Parameter

Template	Parameter	Permitted Values	Change Causes Reboot or Restart
reg-advanced.cfg	`reg.X.rfc3329MediaSec.enable`	0 – Disables the media security mechanisms negotiated between Phone and Outbound proxy without the need of multiple m-lines in the Session Description Protocol. 1 – Enables the media security mechanisms negotiated between Phone and Outbound proxy without the need of multiple m-lines in the Session Description Protocol.	No

Securing Phone Calls with SRTP

Secure Real-Time Transport Protocol (SRTP) encrypts audio stream(s) to prevent interception and eavesdropping on phone calls.

When this feature is enabled, the phones negotiate the type of encryption and authentication to use for the session with the other endpoint.

SRTP authentication proves to the phone receiving the RTP/RTCP stream that the packets are from the expected source and have not been tampered with. Encryption modifies the data in the RTP/RTCP streams so that if the data is captured or intercepted it sounds like noise and cannot be understood. Only the receiver knows the key to restore the data.

If the call is completely secure (RTP authentication and encryption and RTCP authentication and RTCP encryption are enabled), a padlock symbol displays. Phone will send only one SRTP m-line for audio and video instead of multiple m-lines when VoSIP is enabled.

Related Links

TLS Parameters on page 99

SRTP Parameters

Use the session parameters in the following table to turn on or off authentication and encryption for RTP and RTCP streams.

You can also turn off the session parameters to reduce the phone's processor usage.

Secure Real Time Transport Protocol Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`mr.srtp.audio.require`	Enable or disable a requirement for SRTP encrypted audio media between MR hubs and devices. 1 (default) 0	Yes
`site.cfg`	`mr.srtp.video.require`	Enable or disable a requirement for SRTP encrypted video media between hubs and devices. 1 (default) 0	Yes
`sip-interop.cfg`	`sec.srtp.answerWithNewKey`	1 (default) - Provides a new key when answering a call. 0 - Does not provide a new key when answering the call.	No
`sip-interop.cfg`	`sec.srtp.enable`	1 (default) - The phone accepts the SRTP offers. 0 - The phone declines the SRTP offers. The defaults for SIP 3.2.0 is 0 when Null or not defined.	Yes
`sip-interop.cfg`	`sec.srtp.key.lifetime`	Specifies the lifetime of the key used for the cryptographic parameter in SDP. Null (default) - 0 - The master key lifetime is not set. Positive integer minimum 1024 or power of 2 notation - The master key lifetime is set. Setting this parameter to a non-zero value may affect the performance of the phone.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	sec.srtp.mki.enabled	0 (default) - The phone sends two encrypted attributes in the SDP, one with MKI and one without MKI when the base profile is set as Generic. 1 - The phone sends only one encrypted value without MKI when the base profile is set as Skype.	Yes
sip-interop.cfg	sec.srtp.mki.startSessionAtOne	0 (default) - The phone uses MKI value of 1. 1 - The MKI value increments for each new crypto key.	No
sip-interop.cfg	sec.srtp.offer	0 (default) - The secure media stream is not included in SDP of an SIP invite. 1 - The phone includes secure media stream along with the non-secure media description in SDP of an SIP invite.	Yes
sip-interop.cfg	sec.srtp.offer.HMAC_SHA1_32	0 (default) - The AES_CM_128_HMAC_SHA1_32 crypto suite in SDP is not included. 1 - The AES_CM_128_HMAC_SHA1_32 crypto suite in SDP is included.	Yes
sip-interop.cfg	sec.srtp.offer.HMAC_SHA1_80	1 (default) - The AES_CM_128_HMAC_SHA1_80 crypto suite in SDP is included. 0 - The AES_CM_128_HMAC_SHA1_80 crypto suite in SDP is not included.	Yes
sip-interop.cfg	sec.srtp.padRtpToFourByteAlignment	0 (default) - The RTP packet padding is not required when sending or receiving video. 1 - The RTP packet padding is required when sending or receiving video.	Yes
sip-interop.cfg	sec.srtp.require	0 (default) - The secure media streams are not required. 1 - The phone is only allowed to use secure media streams.	Yes
sip-interop.cfg	sec.srtp.requireMatchingTag	1 (default) - The tag values must match in the crypto parameter. 0 - The tag values are ignored in the crypto parameter.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	sec.srtp.sessionParams.noAuth.offer	0 (default) - The authentication for RTP offer is enabled. 1 - The authentication for RTP offer is disabled.	Yes
sip-interop.cfg	sec.srtp.sessionParams.noAuth.require	0 (default) - The RTP authentication is required. 1 - The RTP authentication is not required.	Yes
sip-interop.cfg	sec.srtp.sessionParams.noEncrypRTCP.offer	0 (default) - The encryption for RTCP offer is enabled. 1 - The encryption for RTCP offer is disabled.	Yes
sip-interop.cfg	sec.srtp.sessionParams.noEncrypRTCP.require	0 (default) - The RTCP encryption is required. 1 - The RTCP encryption is not required.	Yes
sip-interop.cfg	sec.srtp.sessionParams.noEncrypRTP.offer	0 (default) - The encryption for RTP offer is enabled. 1 - The encryption for RTP offer is disabled.	Yes
sip-interop.cfg	sec.srtp.sessionParams.noEncrypRTP.require	0 (default) - The RTP encryption is required. 1 - The RTP encryption is not required.	Yes
sip-interop.cfg	sec.srtp.simplifiedBestEffort	1 (default) - The SRTP is supported with Microsoft Description Protocol Version 2.0 Extensions. 0 - The SRTP is not supported with Microsoft Description Protocol Version 2.0 Extensions.	No

Enabling Users to Lock Phones

This feature enables users to lock their phones to prevent access to menus or directories.

If the enhanced feature key (EFK) feature is enabled, you can display a Lock button on the phone to enable users to quickly lock their phones.

After the phone is locked, users can only place calls to emergency and authorized numbers. You can specify which authorized numbers users can call.

If a user forgets their password, you can unlock the phone either by entering the administrator password or by disabling and re-enabling the phone lock feature. The latter method facilitates remote unlocking and avoids disclosing the administrator password to the user.

Note: If a locked phone has a registered shared line, calls to the shared line display on the locked phone and the phone's user can answer the call.

Phone Lock Parameters

Use the parameters in the following table to enable the phone lock feature, set authorized numbers for users to call when a phone is locked, and set scenarios when the phone should be locked.

Phone Lock is different from Device Lock for Skype for Business deployments. If you enable Phone Lock and Device Lock for Skype for Business at the same time on a phone with the Base Profile set to Skype, the Device Lock feature takes precedence over Phone Lock.

Phone Lock Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.enhancedFeatureKeys.enabled`	0 (default) - Disables the enhanced feature keys feature. 1 - Enables the enhanced feature keys feature.	No
`features.cfg`	`phoneLock.Allow.AnswerOnLock`	1(default) - The phone answers any incoming call without asking to UNLOCK. 0 - The phone asks to UNLOCK before answering.	No
`features.cfg`	`phoneLock.authorized.x.description`	The name or description of an authorized number. Null (default) String Up to five (x=1 to 5) authorized contacts that a user can call while their phone is locked. Each contact needs a description to display on the screen, and a phone number or address value for the phone to dial.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	phoneLock.authorized.x.value	The number or address for an authorized contact. Null (default) String Up to five (x=1 to 5) authorized contacts that a user can call while their phone is locked. Each contact needs a description to display on the screen, and a phone number or address value for the phone to dial.	No
features.cfg	phoneLock.browserEnabled	0 (default) - The microbrowser or browser is not displayed while the phone is locked. 1 - The microbrowser or browser is displayed while the phone is locked.	No
features.cfg	phoneLock.dndWhenLocked	0 (default) - The phone can receive calls while it is locked 1 - The phone enters Do-Not-Disturb mode while it is locked	No
features.cfg	phoneLock.enabled[1]	0 (default) - The phone lock feature is disabled 1 - The phone lock feature is enabled.	No
features.cfg	phoneLock.idleTimeout	The amount of time (in seconds) the phone can be idle before it automatically locks. If 0, automatic locking is disabled. 0 (default) 0 to 65535	No
features.cfg	phoneLock.lockState	0 (default) - The phone is unlocked. 1 - The phone is locked. The phone stores and uploads the value each time it changes via the MAC-phone.cfg. You can set this parameter remotely using the Web Configuration Utility.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	phoneLock.powerUpUnlocked	Overrides the phoneLock.lockState parameter. 0 (default) - The phone retains the value in phoneLock.lockState parameter. 1 - You can restart, reboot, or power cycle the phone to override the value for phoneLock.lockState in the MAC-phone.cfg and start the phone in an unlocked state. You can then lock or unlock the phone locally. Polycom recommends that you do not leave this parameter enabled	No

Locking the Basic Settings Menu

By default, all users can access the Basic settings menu available on the Polycom Trio 8800 system and VVX phones.

From this menu, users can customize non-administrative features on their phone. You can choose to lock the Basic settings menu to allow certain users access to the basic settings menu.

If enabled, you can use the default user password (123) or administrator password (456) to access the Basic settings menu, unless the default passwords are not in use.

Basic Settings Menu Lock Parameters

Use the parameter in the following table to lock the Basic settings menu.

Lock the Basic Settings Menu

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.basicSettingsPasswordEnabled	Specifies that a password is required or not required to access the **Basic Settings** menu. 0 (Default) - No password is required to access the **Basic Settings** menu. 1 - Password is required for access to the **Basic Settings** menu.	No

Secondary Port Link Status Report

Polycom devices can detect an externally connected host connection/disconnection, informing the authenticator switch to initiate the authentication process or drop an existing authentication.

This feature extends Cisco Discovery Protocol (CDP) to include a Second Port Status Type, Length, Value (TLV) that informs an authenticator switch of the status of devices connected to a device's secondary PC port.

This feature ensures the following:

- The port authenticated by the externally attached device switches to unauthenticated upon device disconnection so that other unauthorized devices cannot use it.
- The externally attached device can move to another port in the network and start a new authentication process.
- To reduce the frequency of CDP packets, the phone does not send link up status CDP packets before a certain time period. The phone immediately sends all link-down indication to ensure that the port security is not compromised.
- If the externally attached device (the host) supports 802.1X authentication, then the device can send an EAPOL-Logoff on behalf of the device after it is disconnected from the secondary PC port. This informs the authenticator switch to drop the authentication on the port corresponding with the previously attached device.

Secondary Port Link Status Report Parameters

You can use the parameters in the following table to configure options for the Secondary Port Link Status Report feature, including the required elapse or sleep time between two CDP UPs dispatching.

Secondary Port Link Status Report Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.dot1x.eapollogoff.enabled`	0 (default) - The phone does not send an EAPOL Logoff message. 1 - The phone sends an EAPOL Logoff message.	Yes
`site.cfg`	`sec.dot1x.eapollogoff.lanlinkreset`	0 (default) - The phone does not reset the LAN port link. 1 - The phone resets the LAN port link.	Yes
`site.cfg`	`sec.hostmovedetect.cdp.enabled`	0 (default) - The phone does not send a CDP packet. 1 - The phone sends a CDP packet.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.hostmovedetect.cdp.sleepTime`	Controls the frequency between two consecutive link-up state change reports. 1000 (default) 0 to 60000 If `sec.hostmovedetect.cdp.enabled` is set to 1, there is an x microsecond time interval between two consecutive link-up state change reports, which reduces the frequency of dispatching CDP packets.	Yes

802.1X Authentication

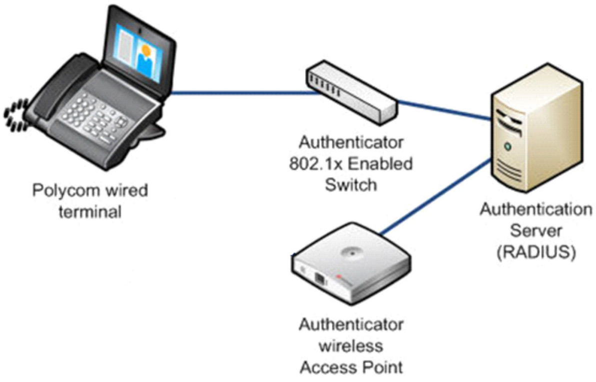
Polycom phones support standard IEEE 802.

1X authentication and the following EAP authentication methods:

- EAP-TLS (requires Device and CA certificates)
- EAP-PEAPv0/MSCHAPv2 (requires CA certificates)
- EAP-PEAPv0/GTC (requires CA certificates)
- EAP-TTLS/MSCHAPv2 (requires CA certificates)
- EAP-TTLS/GTC (requires CA certificates)
- EAP-FAST (optional Protected Access Credential (PAC) file, if not using in-band provisioning)
- EAP-MD5

The following figure shows a typical 802.1X network configuration with wired Polycom phones.

A typical 802.1X network configuration


Polycom wired terminal
Authenticator 802.1x Enabled Switch
Authentication Server (RADIUS)
Authenticator wireless Access Point

802.1X Authentication Parameters

To set up an EAP method that requires a device or CA certificate, you need to configure TLS Platform Profile 1 or TLS Platform Profile 2 to use with 802.

1X. You can use the parameters in the following table to configure 802.1X Authentication.

For more information on EAP authentication protocol, see RFC 3748: Extensible Authentication Protocol.

Set 802.1X Authentication Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg wireless.cfg	device.net.dot1x.enabled	Enable or disable 802.1X authentication. 0 1	Yes
device.cfg site.cfg wireless.cfg	device.net.dot1x.identity[1]	Set the identity (user name) for 802.1X authentication. String	Yes
device.cfg	device.net.dot1x.method	Specify the 802.1X EAP method. EAP-None - No authentication EAP-TLS, EAP-PEAPv0-MSCHAPv2, EAP-PEAPv0-GTC, EAP-TTLS-MSCHAPv2, EAP-TTLS-GTC, EAP-FAST, EAP-MD5	No
device.cfg site.cfg wireless.cfg	device.net.dot1x.password[1]	Set the password for 802.1X authentication. This parameter is required for all methods except EAP-TLS. String	Yes
device.cfg	device.net.dot1x.eapFastInBandProv	Enable EAP In-Band Provisioning for EAP-FAST. 0 (default) - Disabled 1 - Unauthenticated, active only when the EAP method is EAP-FAST.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`	`device.pacfile.data`	Specify a PAC file for EAP-FAST (optional). Null (default) 0-2048 - String length.	No
`device.cfg`	`device.pacfile.password`	The optional password for the EAP-FAST PAC file. Null (default). 0-255 - String length.	No

OpenSSL Versions List

This section lists OpenSSL versions used for each UC Software release.

OpenSSL Versions List

UC Software Version	OpenSSL Version
UC Software 5.5.3	OpenSSL 1.0.2j 26 Sep 2016
UC Software 4.0.13	OpenSSL 1.0.2j 26 Sep 2016 OpenSSL 0.9.8zg 11 Jun 2015 (for SoundStation IP 6000 and SoundStation IP 7000 phones)
UC Software 5.6.0	OpenSSL 1.0.2j 26 Sep 2016
UC Software 5.5.2	OpenSSL 1.0.2j 26 Sep 2016
UC Software 5.5.1	OpenSSL 1.0.1p 9 Jul 2015
UC Software 5.5.0	OpenSSL 1.0.1p 9 Jul 2015
UC Software 5.4.6	OpenSSL 1.0.1p 9 Jul 2015
UC Software 5.4.5	OpenSSL 1.0.1p 9 Jul 2015
UC Software 5.4.4	OpenSSL 1.0.1p 9 Jul 2015
UC Software 5.4.3	OpenSSL 1.0.1p 9 Jul 2015
UC Software 5.4.1	OpenSSL 1.0.1m 19 March 2015
UC Software 5.4.0	OpenSSL 1.0.1m 19 March 2015
UC Software 5.3.3	OpenSSL 1.0.1m 15 Oct 2014

UC Software Version	OpenSSL Version
UC Software 5.3.2	OpenSSL 1.0.1m 15 Oct 2014
UC Software 5.3.1	OpenSSL 1.0.1m 15 Oct 2014
UC Software 5.3.0	OpenSSL 1.0.1j 15 Oct 2014
UC Software 5.2.2	OpenSSL 1.0.1j 15 Oct 2014
UC Software 5.2.0	OpenSSL 1.0.1h 5 Jun 2014
UC Software 5.1.3	OpenSSL 1.0.1h 5 Jun 2014
UC Software 5.1.2	OpenSSL 1.0.1h 5 Jun 2014
UC Software 5.1.0	OpenSSL 1.0.1h 5 Jun 2014
UC Software 5.0.2	OpenSSL 1.0.1c 10 May 2012
UC Software 5.0.1	OpenSSL 1.0.1c 10 May 2012
UC Software 5.0.0	OpenSSL 1.0.1c 10 May 2012
UC Software 4.1.8	OpenSSL 1.0.1h 5 Jun 2014
UC Software 4.1.6	OpenSSL 1.0.1c 10 May 2012
UC Software 4.0.11	OpenSSL 0.9.8zg 11 Jun 2015
UC Software 4.0.10	OpenSSL 0.9.8zc 11 Jun 2015
UC Software 4.0.9	OpenSSL 0.9.8zc 15 Oct 2014
UC Software 4.0.8	OpenSSL 0.9.8zc 15 Oct 201
UC Software 4.0.0 - 4.0.7	OpenSSL 0.9.8k 25 Mar 2009

Certificates

Topics:

- Using the Factory-Installed Certificate
- Customizing Certificate Use
- Create a Certificate Signing Request
- Custom URL Location for LDAP Server CA Certificate

Security certificates are an important element in deploying a solution that ensures the integrity and privacy of communications involving Polycom® UC Software devices.

Polycom phones are installed with a Polycom-authenticated “built-in” device certificate that you can use or you can choose to customize your security by requesting additional certificates from a certificate authority of your choice.

You can customize security configuration options to determine type of device certificate is used for each of the secure communication options. By default, all operations will utilize the factory-installed device certificate unless you specify otherwise.

Note: You can install custom device certificates on your Polycom phones in the same way custom CA certificates are installed. See *Technical Bulletin 17877: Using Custom Certificates With Polycom Phones* for more information.

Certificates are used in the following situations:

- Mutual TLS Authentication: Allows a server to verify that a device is truly a Polycom device (and not a malicious endpoint or software masquerading as a Polycom device). This could be used for tasks like provisioning, or SIP signaling using TLS signaling. For example, certain partner provisioning systems use Mutual TLS as does Polycom® Zero Touch Provisioning (ZTP).
- Secure HTTP (https) access to the web server on the phone at https://<IP ADDRESS OF PHONE>. The web server is used for certain configuration and troubleshooting activities.
- Secure communications utilizing the Polycom Applications API.

There are different options for utilizing device certificates on the phone:

- Two platform device certificates. These certificates are loaded onto the device by the system administrator and can be configured to be used for any of the following purposes: 802.1X Authentication, provisioning, syslog, SIP signaling, browser communications, presence, and LDAP. Certificates for syslog, 802.1X, and provisioning must applied using TLS platform profiles.
- Six application device certificates. These certificates are loaded onto the device by the system administrator and can be used for all of the operations listed above for platform certificates. You cannot use TLS application profiles to applied certificates for 802.1X, syslog, and provisioning.

  Note: For details on installing digital credentials on VVX phones, see *Device Certificates on Polycom SoundPoint IP, SoundStation IP, and VVX Phones: Technical Bulletin 37148* at Polycom Engineering Advisories and Technical Notifications.

Related Links

TLS Platform Profile and Application Profile Parameters on page 91

Using the Factory-Installed Certificate

A factory-installed device certificate is installed at the time of manufacture and is unique to a device (based on the MAC address) and signed by the Polycom Certificate Authority (CA).

Since it is installed at the time of manufacture, it is the easiest option for out-of-box activities, especially phone provisioning.

You can use the factory-installed certificate for all your security needs. To configure your web servers and/or clients to trust the Polycom factory-installed certificates, you must download the Polycom Root CA certificate, which is available at http://pki.polycom.com/pki. You may also need to download the Intermediate CA certificates if determined by the authenticating server.

The location of the Certificate Revocation List (CRL)—a list of all expired certificates signed by the Polycom Root CA—is part of the Polycom Root CA digital certificate. If you enable Mutual TLS, you must have a root CA download (the Polycom Root CA certificate or your organization's CA) on the HTTPS server.

The certificate is set to expire on March 9, 2044.

Note: For more information on using Mutual TLS with Microsoft Internet Information Services (IIS) 6.0, see *Mutual Transport Layer Security Provisioning Using Microsoft Internet Information Services 6.0: Technical Bulletin 52609* at Polycom Engineering Advisories and Technical Notifications.

Check for a Device Certificate

The certificate and associated private key are stored on the phone in its non-volatile memory as part of the manufacturing process.

You can check if a phone has a certificate pre-installed.

Procedure

1. Navigate to **Settings** > **Advanced** > **Administration Settings** > **TLS Security** > **Custom Device Credentials**.
2. Select a credential and press Info to view the certificate.

   One of the following messages displays:

   - **Installed** or **Factory Installed** is displayed if the certificate is available in flash memory, all the certificate fields are valid (listed above), and the certificate has not expired.
   - **Not Installed** is displayed if the certificate is not available in flash memory (or the flash memory location where the device certificate is to be stored is blank).
   - **Invalid** is displayed if the certificate is not valid.

   Note: If your phone reports the device certificate as self-signed rather than **Factory Installed**, return the equipment to receive a replacement.

Customizing Certificate Use

You can add custom certificates to the phone and set up the phone to use the certificates for different features.

For example, the phone's factory-installed certificate can be used for authentication when phone provisioning is performed by an HTTPS server. You can use a different certificate when accessing content through a browser.

Determining TLS Platform Profiles or TLS Application Profiles

You use TLS Platform or TLS Application profiles to customize where your installed certificates are used for authentication.

After you install certificates on the phone, you can determine which TLS platform profiles or TLS application profiles use these certificates. By default, TLS Platform Profile 1 uses every CA certificate and the default device certificate. Also, each TLS application uses TLS Platform Profile 1 as the default profile. You can quickly apply a CA certificate to all TLS applications by installing it on the phone and keeping the default TLS profile and default TLS application values.

Alternatively, you can choose which TLS platform profile or application profile to use for each TLS application. You can use platform profiles for any of the following purposes: phone provisioning, for applications running on the microbrowser and browser, and for 802.1X, LDAP, and SIP authentication. You can use application profiles for all applications except 802.1X, syslog, and provisioning.

Note: For more information on using custom certificates, see *Technical Bulletin 17877: Using Custom Certificates With Polycom Phones*.

Related Links

TLS Platform Profile and Application Profile Parameters on page 91

TLS Platform Profile and Application Profile Parameters

By default, all Polycom-installed profiles are associated with the default cipher suite and use trusted and widely recognized CA certificates for authentication.

The following table shows parameters for TLS Platform Profile 1. To configure TLS Platform Profile 2, use a 2 at the end of the parameter instead of a 1. For example, set `device.sec.TLS.profile.caCertList2` instead of `.caCertList1` .

You can use the parameters in the following table to configure the following TLS Profile feature options:

- Change the cipher suite, CA certificates, and device certificates for the two platform profiles and the six application profiles.
- Map profiles directly to the features that use certificates.

TLS Platform Profile and Application Profile Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.sec.TLS.customCaCert1	Specify a custom certificate. Null (default) String (maximum of 12288 characters)	No
device.cfg, site.cfg	device.sec.TLS.profile.caCertList1	Specify which CA certificates to use. Null (default) String (maximum of 1024 characters)	No
device.cfg, site.cfg	device.sec.TLS.profile.cipherSuite1	Specify the cipher suite. Null (default) String (maximum of 1024 characters)	No
device.cfg, site.cfg	device.sec.TLS.profile.cipherSuiteDefault1	Null (default) 0 - Use the custom cipher suite. 1 - Use the default cipher suite.	No
device.cfg, site.cfg	device.sec.TLS.profile.deviceCert1	Specify which device certificates to use. Builtin (default) Builtin, Platform1, Platform2	No
site.cfg	sec.TLS.cipherList	Specifies the cipher list for all applications except web server. ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!MD5:!RC4:@STRENGTH (default) String (maximum of 1024 characters)	No
site.cfg	sec.TLS.customCaCert.x	The custom certificate for TLS Application Profile x (x= 1 to 6). Null (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.customDeviceKey.x`	The custom device certificate private key for TLS Application Profile x (x= 1 to 6). Null (default) String	No
`site.cfg`	`sec.TLS.exchangeServices.cipherList`	Specifies the cipher list for Exchange services profile. (default) ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!MD5:!RC4:@STRENGTH String (maximum of 1024 characters) The format for the cipher list uses OpenSSL syntax found at https://www.openssl.org/docs/man1.0.2/apps/ciphers.html	No
`site.cfg`	`sec.TLS.profile.exchangeServices.cipherSuiteDefault`	1 (default) - Use the default cipher suite of Exchange services for the TLS Application Profile. 0 - Use the custom cipher suite of Exchange services for the TLS Application Profile.	No
`site.cfg`	`sec.TLS.profile.x.caCert.application1`	1 (default) - Enable a CA Certificate for TLS Application Profile 1. 0 - Disable a CA Certificate for TLS Application Profile 1.	No
`site.cfg`	`sec.TLS.profile.x.caCert.application2`	1 (default) - Enable a CA Certificate for TLS Application Profile 2. 0 - Disable a CA Certificate for TLS Application Profile 2.	No
`site.cfg`	`sec.TLS.profile.x.caCert.application3`	1 (default) - Enable a CA Certificate for TLS Application Profile 3. 0 - Disable a CA Certificate for TLS Application Profile 3.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.profile.x.caCert.application4`	1 (default) - Enable a CA Certificate for TLS Application Profile 4. 0 - Disable a CA Certificate for TLS Application Profile 4.	No
`site.cfg`	`sec.TLS.profile.x.caCert.application5`	1 (default) - Enable a CA Certificate for TLS Application Profile 5. 0 - Disable a CA Certificate for TLS Application Profile 5.	No
`site.cfg`	`sec.TLS.profile.x.caCert.application6`	1 (default) - Enable a CA Certificate for TLS Application Profile 6. 0 - Disable a CA Certificate for TLS Application Profile 6.	No
`site.cfg`	`sec.TLS.profile.x.caCert.application7`	1 (default) - Enable a CA Certificate for TLS Application Profile 7. 0 - Disable a CA Certificate for TLS Application Profile 7.	No
`site.cfg`	`sec.TLS.profile.x.caCert.defaultList`	Specifies the list of default CA Certificate for TLS Application Profile x (x=1 to 7). Null (default) String	No
`site.cfg`	`sec.TLS.profile.x.caCert.platform1`	1 (default) - Enable a CA Certificate for TLS Platform Profile 1. 0 - Disable a CA Certificate for TLS Platform Profile 1.	No
`site.cfg`	`sec.TLS.profile.x.caCert.platform2`	1 (default) - Enable a CA Certificate for TLS Platform Profile 2. 0 - Disable a CA Certificate for TLS Platform Profile 2.	No
`site.cfg`	`sec.TLS.profile.x.cipherSuite`	Specifies the cipher suite for TLS Application Profile x (x=1 to 8). Null (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.profile.x.cipherSuiteDefault`	1 (default) - Use the default cipher suite for TLS Application Profile x (x= 1 to 8). 0 - Use the custom cipher suite for TLS Application Profile x (x= 1 to 8).	No
`site.cfg`	`sec.TLS.profile.x.deviceCert`	Specifies the device certificate to use for TLS Application Profile x (x = 1 to 7). Polycom (default) Platform1, Platform2, Application1, Application2, Application3, Application4, Application5, Application6,Application7	No
`site.cfg`	`sec.TLS.webServer.cipherList`	Specify the cipher list for web server. ALL:!aNULL:!eNULL:!DSS:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!AES256-SHA:!AES128-SHA:!MD5:!RC4:@STRENGTH (default) String (maximum of 1024 characters)	No

Related Links
Certificates on page 89
Determining TLS Platform Profiles or TLS Application Profiles on page 91

TLS Protocol Configuration for Supported Applications

You can configure the TLS Protocol for the following supported applications:

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

- 802.1x

Related Links

TLS Protocol Parameters on page 96

TLS Protocol Parameters

The following table includes the parameters for the TLS protocol supported applications.

TLS Protocol Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.sec.TLS.protocol.dot1x	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and 802.1x authentication. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No
device.cfg, site.cfg	device.sec.TLS.protocol.prov	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and provisioning. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No
device.cfg, site.cfg	device.sec.TLS.protocol.syslog	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and Syslog. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	sec.TLS.protocol.browser	Configure the lowest TLS/SSL version to use for handshake negotiation between the phone and phone browser. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2 The microbrowser restarts when there is a change in the browser TLS protocol or TLS cipher settings, and the last web page displayed is not restored.	No
device.cfg, site.cfg	sec.TLS.protocol.exchangeServices	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and Exchange services. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.ldap	Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Lightweight Directory Access Protocol (LDAP). The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	sec.TLS.protocol.sip	Configures the lowest TLS/SSL version to use for handshake negotiation between the phone and SIP signaling. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.sopi	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and SOPI. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.webServer	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and web server. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No
device.cfg, site.cfg	sec.TLS.protocol.xmpp	Configures the lowest TLS/SSL version to use for handshake negotiation between phone and XMPP. The phone handshake starts with the highest TLS version irrespective of the value you configure. TLSv1_0 (default) SSLv2v3 TLSv1_1 TLSv1_2	No

Related Links

TLS Protocol Configuration for Supported Applications on page 95

TLS Parameters

The next table lists configurable TLS parameters.

For the list of configurable ciphers, refer to the Secure Real-Time Transport Protocol table.

TLS Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	sec.TLS.browser.cipherList	The cipher list is for browser. The format for the cipher list uses OpenSSL syntax found at: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html. NoCipher (default) String	No
site.cfg	sec.TLS.customDeviceCert.x	The custom device certificate for TLS Application Profile x (x= 1 to 6). Null (default) String	No
site.cfg	sec.TLS.LDAP.cipherList	The cipher list for the corporate directory. The format for the cipher list uses OpenSSL syntax found here: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html. NoCipher (default) String	No
site.cfg	sec.TLS.LDAP.strictCertCommonNameValidation	1 (default) - Requires to validate the server certificate during an LDAP or LDAPS connection over TLS. 0 - Does not require to validate the server certificate during an LDAP or LDAPS connection over TLS.	No
site.cfg	sec.TLS.profileSelection.SOPI	Select the platform profile required for the phone. PlatformProfile1 (default) 1 - 7	No
site.cfg	sec.TLS.profile.webServer.cipherSuiteDefault	1 (default) - The phone uses the default cipher suite for web server profile. 0 - The custom cipher suite is used for web server profile.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	sec.TLS.prov.cipherList	The cipher list for provisioning. The format for the cipher list uses OpenSSL syntax found here: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html. NoCipher (default) String	No
site.cfg	sec.TLS.SIP.cipherList	The cipher list for SIP. The format for the cipher list uses OpenSSL syntax found here: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html. NoCipher (default) String	No
site.cfg	sec.TLS.SIP.strictCertCommonNameValidation	1 (default) - The common name validation is enabled for SIP. 0 - The common name validation is not enabled for SIP.	No
site.cfg	sec.TLS.SOPI.cipherList	Selects a cipher key from the list of available ciphers. NoCipher (default) 1 - 1024 character string	No
site.cfg	sec.TLS.SOPI.strictCertCommonNameValidation	Controls the strict common name validation for the URL provided by the server. 1 (default) - The SOPI verifies the server certificate to match commonName/SubjectAltName against the server hostname. 0 - The SOPI will not verify the server certificate for commonName/SubjectAltName against the server hostname.	No
site.cfg	sec.TLS.syslog.cipherList	The cipher list for syslog. The format for the cipher list uses OpenSSL syntax found here: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html NoCipher (default) String	No

Related Links

Securing Phone Calls with SRTP on page 77

TLS Profile Selection Parameters

You can configure the parameters listed in the next table to choose the platform profile or application profile to use for each TLS application.

The permitted values are:

- PlatformProfile1
- PlatformProfile2
- ApplicationProfile1
- ApplicationProfile2
- ApplicationProfile3
- ApplicationProfile4
- ApplicationProfile5
- ApplicationProfile6
- ApplicationProfile7

TLS Profile Selection Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.profileSelection.browser`	Specifies to select a TLS platform profile or TLS application profile for the browser or a microbrowser. PlatformProfile1 (default) TLS profile	No
`site.cfg`	`sec.TLS.profileSelection.LDAP`	Specifies to select a TLS platform profile or TLS application profile for the corporate directory. PlatformProfile1 (default) TLS profile	No
`site.cfg`	`sec.TLS.profileSelection.SIP`	Specifies to select a TLS platform profile or TLS application profile for SIP operations. PlatformProfile1 (default) TLS profile	No
`site.cfg`	`sec.TLS.profileSelection.syslog`	Specifies to select a TLS platform profile for the syslog operations. PlatformProfile1 (default) PlatformProfile1 or PlatformProfile2	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.profileSelection.SOPI`	Specifies to select a TLS platform profile or TLS application profile for the GENBAND “Subscriber Open Provisioning Interface” (*SOPI*). PlatformProfile1 (default) TLS profile	No

Configurable TLS Cipher Suites

You can configure which cipher suites to offer and accept during TLS session negotiation. The following table lists supported cipher suites. NULL cipher is a special case that does not encrypt the signaling traffic.

TLS Cipher Suites

Cipher	Cipher Suite
ADH	ADH-RC4-MD5, ADH-DES-CBC-SHA, ADH-DES-CBC3-SHA, ADH-AES128-SHA, ADH-AES256-SHA
AES128	AES128-SHA
AES256	AES256-SHA
DES	DES-CBC-SHA, DES-CBC3-SHA
DHE	DHE-DSS-AES128-SHA, DHE-DSS-AES256-SHA, DHE-RSA-AES128-SHA, DHE-RSA-AES256-SHA
EXP	EXP-RC4-MD5, EXP-DES-CBC-SH, EXP-EDH-DSS-DES-CBC-SHA, EXP-DES-CBC-SHA, EXP-ADH-RC4-MD5, EXP-ADH-DES-CBC-SHA, EXP-EDH-RSA-DES-CBC-SHA
EDH	EDH-RSA-DES-CBC-SHA, EDH-DSS-DES-CBC3-SHA, EDH-DSS-CBC-SHA
NULL	NULL-MD5, NULL-SHA
RC4	RC4-MD5, RC4-SHA

TLS Cipher Suite Parameters

You can use the parameters listed in the following table to configure TLS Cipher Suites.

TLS Cipher Suite Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.TLS.cipherList`	String (1 - 1024 characters) RC4:@STRENGTH (default) ALL:!aNULL:!eNULL:!DSS:!SEED :!ECDSA:!IDEA:!MEDIUM:!LOW:! EXP:!ADH:!ECDH:!PSK:!MD5:! RC4:@STRENGTH The global cipher list parameter. The format for the cipher list uses OpenSSL syntax found at: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html.	No
`site.cfg`	`sec.TLS.<application>.cipherList`	Specify the cipher list for a specific TLS Platform Profile or TLS Application Profile.	No

Create a Certificate Signing Request

You generate a certificate signing request directly from the Polycom device.

By default, the phone requests a 2048-bit certificate with 'sha256WithRSAEncryption' as the signature algorithm. You can use OpenSSL or another certificate signing request utility if you require a stronger certificate.

Polycom supports the use of Subject Alternative Names (SAN) with TLS security certificates. Polycom does not support the use of the asterisk (*) or wildcard characters in the Common Name field of a Certificate Authority's public certificate. If you want to enter multiple hostnames or IP addresses on the same certificate, use the SAN field.

You must have a provisioning server in place before generating the certificate signing request.

Procedure

1. Navigate to **Settings** > **Advanced** > **Admin Settings** > **Generate CSR**.
2. When prompted, enter the administrative password and press Enter.

   The default administrative password is 456.
3. From the **Generate CSR Screen**, fill in the Common Name field - the Organization, Email Address, Country, and State fields are optional.
4. Press **Generate**.

A message “CSR generation completed” displays on the phone's screen. The MAC.csr (certificate request) and MAC-private.pem (private key) are uploaded to the phone's provisioning server.

5. Forward the CSR to a Certificate Authority (CA) to create a certificate.

   If your organization doesn't have its own CA, you need to forward the CSR to a company like Symantec.

Download Certificates to a Polycom Phone

You can download and install up to eight CA certificates and eight device certificates on a Polycom phone.

After installing the certificates, you can refresh the certificates when they expire or are revoked, and you can delete any CA certificate or device certificate that you install.

You can download certificate(s) to a phone in the following ways:

- Using a configuration file
- Through the phone's user interface
- Through the Web Configurable Utility

Note: For VVX 1500 phones, the maximum certificate size on Platform CA1 is 1536KB and 4KB for Platform CA2.

Procedure

1. Navigate to **Settings** > **Advanced** > **Administrative Settings** > **TLS Security and select Custom CA Certificates or Custom Device Certificates**.
2. Select **Install**.
3. Enter the URL where the certificate is stored.

   For example, `http://bootserver1.polycom.com/ca.crt`

   The certificate is downloaded, and the certificate's MD5 fingerprint displays to verify that the correct certificate is to be installed.
4. Select **Accept**.

   The certificate is installed successfully.

Custom URL Location for LDAP Server CA Certificate

You can set the URL from where Polycom phones can download a CA certificate or a chain of CA certificates required to authenticate the LDAP server.

By default, all Polycom-installed profiles are associated with the default cipher suite and use trusted and widely recognized CA certificates for authentication. You can download and install up to seven custom CA certificates onto a Polycom phone. The certificates are installed in descending order starting with the Application CA 7 slot and continues to Application CA 1 slot depending on how many certificates are in the chain.

Note: If the custom application CA certificate slots already have CA certificates installed on your Polycom phones, downloading LDAP server CA certificates will overwrite any existing certificates on the phone in descending order starting with the seventh certificate.

Custom URL Location for LDAP Server Certificates Parameters

Use the parameter in the following table to configure this feature.

In addition to the parameter in the following table, you must also configure the following Corporate Directory parameters:

- `sec.TLS.proflieSelection.LDAP = ApplicationProfile1`

Custom URL Location for LDAP Server Certificates Parameters

Template	Parameter	Permitted Values	Change Causes Reboot or Restart
`site.cfg`	`sec.TLS.LDAP.customCaCertUrl`	Enter the URL location from where the phone can download LDAP server certificates. String (default) 0 - Minimum 255 - Maximum You must configure parameters `dir.corp.address` and `feature.corporateDirectory.enabled` as well to enable this parameter.	No

Confirm the Installed LDAP Server Certificates on the Phone

After you set the URL for the location where the phone can download the chain of CA certificates using the parameter `sec.TLS.LDAP.customCaCertUrl` and enabled the parameters `dir.corp.address` and `feature.corporateDirectory.enabled` as well, the certificates are automatically updated on the phones. You can confirm that the correct certificates were downloaded and installed on the phone.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, and enter the administrator password.
2. Select **Administrative Settings** > **TLS Security** > **Custom CA Certificates** > **Application CA placeholders**.
3. Check that correct certificates were installed on the phone.

Upgrading the Software

Topics:

You can upgrade the software that is running on the Polycom phones in your organization.

The upgrade process varies with the version of Polycom UC Software that is currently running on your phones and with the version that you want to upgrade to.

- You can upgrade software with the user-controlled software upgrade feature.
- If you are upgrading software from UC Software 4.0.x, update the phones from your 4.0.x version.

Upgrading UC Software on a Single Phone

You can use the software upgrade tool in the Web Configuration Utility to update the software version running on a single phone.

For instructions, see *Use the Software Upgrade Tool in the Web Configuration Utility: Feature Profile 67993* at Polycom Engineering Advisories and Technical Notifications.

Configuration changes made to individual phones using the Web Configuration Utility override configuration settings made using central provisioning.

User-Controlled Software Update

This feature enables phone users to choose when to accept software updates you send to the phones.

You can send an earlier or a later software version than the current version on the phone.

User-controlled updates apply to configuration changes and software updates you make on the server and Web Configuration Utility. If a user postpones a software update, configuration changes and software version updates from both the server and Web Utility are postponed. When the user chooses to update, configuration and software version changes from both the server and Web Utility are sent to the phone.

This feature does not work if you have enabled ZTP or Skype for Business Device Update, and it is not available with Skype for Business.

User-Controlled Software Update Parameters

You can set a polling policy and polling time period at which the phone polls the server for software updates and displays a notification on the phone to update software.

For example, if you set the polling policy to poll every four hours, the phone polls the server for new software every four hours and displays a notification letting the user know that a software update is available. Users can choose to update the software or they postpone it to a maximum of three times for up to six hours. The phone automatically updates the software after three postponements or after six hours, whichever comes first.

The polling policy is disabled after the phone displays the software update notification.

After the software postponement ends, the phone displays the software update notification again.

User-Controlled Software Update Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.usercontrol.enabled	0 (default) - The phone does not display the software update notification and options and the phone reboots automatically to update the software. 1 - The phone displays the software update notification and options and the user can control the software download.	No
site.cfg	prov.usercontrol.postponeTime	Sets the time interval for software update notification using the HH:MM format. 02:00 (default) 00:15 01:00 02:00 04:00 06:00	No

Reverting to a Previous UC Software Release

If you want to revert to a previous software release, follow the instructions in *Upgrading Polycom Phones to and Downgrading Phones from Polycom UC Software 4.0.0*:

T*echnical Bulletin 64731 at* Polycom Engineering Advisories and Technical Notifications.

Upgrade Phones from UC Software 4.0.x

If your Polycom phones are running UC Software 4.

0.x or later, you can upgrade to a later UC Software version. If your phones are running a software release earlier than UC Software 4.0.x, you should first upgrade to UC Software 4.0.x following the instructions in *Technical Bulletin 64731: Upgrading Polycom Phones to and Downgrading Phones From Polycom UC Software 4.0.0* at Polycom Engineering Advisories and Technical Notifications.

Note: To ensure predictable phone behavior, the configuration files listed in CONFIG_FILES attribute of the master configuration file must be updated when the software is updated.

Procedure

1. Back up your existing application and configuration files.
2. Create your new configuration using UC Software 4.1.0.

   Configuration file changes and enhancements are explained in the Release Notes that accompany the software.
3. Save the new configuration files and images (such as `sip.ld`) on your provisioning server.
4. Reboot the phones using an automatic method such as polling or check-sync.
   - Reboot your phone manually as a backup option only if another reboot method fails.
   - You can boot the phones remotely through the SIP signaling protocol.

You can configure the phones to periodically poll the provisioning server for changed configuration files or application executables. If a change is detected, the phone may reboot to download the change.

Diagnostics and Status

Topics:

- View the Phone's Status
- Test Phone Hardware
- Upload a Phone's Configuration
- Perform Network Diagnostics
- Reboot the Phone
- Restart the Phone
- Reset the Phone and Configuration
- Monitoring the Phone's Memory Usage
- Remote Packet Capture
- Uploading Logs to a USB Flash Drive

Polycom phones running Polycom UC Software provide a variety of screens and logs that allow you to review information about the phone and its performance, help you diagnose and troubleshoot problems, view error messages, and test the phone's hardware.

Review the latest UC Software Release Notes for your voice product on Voice on Polycom UC Software Support Center for known problems and possible workarounds. If you don't find your problem in this section or in the latest Release Notes, contact your Certified Polycom Reseller for support.

The phone includes a variety of information screens and tools that can help you monitor the phone and resolve problems.

View the Phone's Status

You can troubleshoot phone issues by viewing the phone's Status menu.

Procedure

1. Select **Settings** > **Status** > **Select**.
2. Scroll to a **Status** menu item and press **Select**.

   The following table lists available options:

Menu Item	Menu Information
Platform	• Phone's serial number or MAC address • Current IP address • Updater version • Application version • Name of the configuration files in use • Address of the provisioning server
Network	• TCP/IP Setting • Ethernet port speed • Connectivity status of the PC port (if it exists) • Statistics on packets sent and received since last boot • Last time the phone rebooted • Call Statistics showing packets sent and received on the last call
Lines	• Detailed status of each of the phone's configured lines
Diagnostics	• Hardware tests to verify correct operation of the microphone, speaker, handset, and third party headset, if present • Hardware tests to verify correct operation of the microphones and speaker. • Tests to verify proper functioning of the phone keys • List of the functions assigned to each of the phone keys • Real-time graphs for CPU, network, and memory use

Test Phone Hardware

You can test the phone's hardware directly from the user interface.

Procedure

1. Go to **Settings** > **Status** > **Diagnostics**.
2. Choose from these tests:
    - **Audio Diagnostics** Test the speaker, microphone, handset, and a third party headset.
    - **Keypad Diagnostics** Verify the function assigned to each keypad key.
    - **Display Diagnostics** Test the LCD for faulty pixels.
    - **LED Diagnostics** Test the LED lights on your phone.
    - **Touch Screen Diagnostics** Test the touch screen response.
    - **Display Diagnostics** Test the LCD for faulty pixels.
    - **Touch Screen Diagnostics** Test the touch screen response.

Upload a Phone's Configuration

You can upload the phone's current configuration files from the phone menu to help you debug configuration problems.

A number of files can be uploaded to the provisioning server, one for every active source as well as the current non-default configuration set.

You can use the Web Configuration Utility to upload the files.

Procedure

1. Navigate to **Settings** > **Advanced** > **Admin Settings** > **Upload Configuration**.
2. Choose which files to upload: All Sources, Configuration Files, Local, MR, Web, or SIP.

   If you use the Web Configuration Utility, you can also upload Device Settings.
3. Press **Upload**.
4. The phone uploads the configuration file to the location you specified in the parameter `prov.configUploadPath` .

   For example, if you select All Sources, a file <MACaddress>-update-all.cfg is uploaded.

Perform Network Diagnostics

You can use ping and traceroute to troubleshoot network connectivity problems.

Procedure

1. Go to **Settings** > **Status** > **Diagnostics** > **Network**.
2. Enter a URL or IP address.
3. Press **Enter**.

Reboot the Phone

You can reboot the phone from the phone menu when you want to send configuration changes requiring a reboot or restart to the phone.

Parameters that require a reboot or restart are marked in the parameter tables in this guide. If a configuration change does not require a reboot or restart, you can update configuration.

Procedure

1. On the phone, go to **Settings** > **Advanced** > **Reboot Phone**.

Restart the Phone

You can restart the phone from the phone menu when you want to send configuration changes requiring a reboot or restart to the phone.

Parameters that require a reboot or restart are marked in the parameter tables in this guide. For configuration changes that do not require a reboot or restart, you can update configuration.

Procedure

1. On the phone, go to **Settings** > **Basic** > **Update Configuration**.

If new Updater or Polycom UC Software is available on the provisioning server, the phone downloads the software. If new software is available on the provisioning server, the phone downloads the software and restarts.

Update Configuration from the Phone Menu

You can update the phone configuration from the phone menu when you want to send configuration changes to the phone.

Some configuration changes require a reboot or restart and parameters that require a reboot or restart are marked in the parameter tables in this guide. If there are configuration file changes or new software available on the provisioning server, your phone restarts or reboots if required.

Procedure

1. On the phone, go to **Settings** > **Basic** > **Update Configuration**.

Reset the Phone and Configuration

You can reset part or all of the phone and phone configuration.

Procedure

1. On the phone, go to **Settings** > **Advanced** > **Administration Settings** > **Reset to Defaults**.

   The following table describes the phone reset options.

Setting	Description
Reset Local Configuration	Clears the override file generated by changes using the phone user interface.
Reset Web Configuration	Clears the override file generated by changes using the Web Configuration Utility.
Reset Device Settings	Resets the phone's flash file system settings that are not stored in an override file. These are your network and provisioning server settings and include custom certificates and encryption keys. Local, web, and other configuration files remain intact.

Setting	Description
Format File System	Formats the phone's flash file system and deletes the UC Software application, log files, configuration, and override files. Note that if the override file is stored on the provisioning server, the phone re-downloads the override file when you provision the phone again. Formatting the phone's file system does not delete those device settings affecting network and provisioning, and any certificates and encryption keys remain on the phone.
Reset to Factory	Removes the Web and local override files, any stored configuration files in the flash file system, as well as any custom certificates and encryption keys. All network and provisioning settings are reset but the UC Software application and updater remain intact.

Reset to Factory Parameter

By default, only administrators can initiate a factory reset. However, you can configure the phone using configuration parameter to make **Reset to Factory** setting available to users. The following table lists the parameter you need to configure this setting.

Reset to Factory Parameter

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.basicSettings.factoryResetEnabled`	0 (default) - Reset to Factory option is not displayed under **Basic** settings menu. 1 – Reset to Factory option is displayed under **Basic** settings menu	No

Monitoring the Phone's Memory Usage

To ensure that your phones and their configured features operate smoothly, verify that the phones have adequate available memory resources.

If you are using a range of phone features, customized configurations, or advanced features, you might need to manage phone memory resources.

If your deployment includes a combination of phone models, consider configuring each phone model separately with its own features instead of applying all phone features to all phone models.

For best performance, the phone should use no more 95% of its available memory. When the phone memory resources are low, you may notice one or more of the following symptoms:

- The phones reboot or freeze up.
- The phones do not download all ringtones, directory entries, backgrounds, or XML dictionary files.
- Applications running in the microbrowser or browser stop running or do not start.

Check Memory Usage from the Phone

You can view a graphical representation of the phone's memory usage directly on the phone.

Procedure

1. Load and configure the features and files you want to make available on the phone's interface.
2. Navigate to **Settings** > **Status** > **Diagnostics** > **Graphs** > **Memory Usage**.

View Memory Usage Errors in the Application Log

Each time the phone's minimum free memory goes below about 5%, the phone displays a message in the application log that the minimum free memory has been reached.

The application log file is enabled by default. The file is uploaded to the provisioning server directory on a schedule you can configure.

You can also upload a log file manually.

Phone Memory Resources

If you need to free memory on your phone, review the following table for the amount of memory each customizable feature uses and consider strategies for reducing the amount of memory you need the feature to use.

Feature	Typical Memory Size	Description
Idle Browser	Varies, depending on number and complexity of application elements.	To reduce memory resources used by the idle browser: • Display no more than three or four application elements. • Simplify pages that include large tables or images.
Custom Idle Display Image	15 KB	The average size of the Polycom display image is 15 KB. Custom idle display image files should also be no more than 15 KB.
Main Browser	Varies, depending on number and complexity of applications.	To reduce memory resources used by the main browser: • Display no more than three or four application elements. Simplify pages.
Local Contact Directory	42.5 KB	Polycom phones are optimized to display a maximum of 250 contacts. Each contact has four attributes and requires 170 bytes. A local contact directory of this size requires 42.5 KB. To reduce memory resources used by the local contact directory: • Reduce the number of contacts in the directory Reduce the number of attributes per contact

Feature	Typical Memory Size	Description
Corporate Directory	Varies by server	Polycom phones are optimized to corporate directory entries with 5 - 8 contact attributes each. The size of each entry and the number of entries in the corporate directory vary by server. If the phone is unable to display directory search results with more than five attributes, make additional memory resources available by reducing memory requirements of another feature.
Ringtones	16 KB	The Polycom ringtone files range in size from 30KB to 125KB. If you use custom ringtones, Polycom recommends limiting the file size to 16KB. To reduce memory resources required for ringtones: Reduce the number of available ringtones.
Background Images	8 - 32 KB	Polycom phones are optimized to display background images of 50KB. To reduce memory resources required for background images: Reduce the number and size of available background images.
Phone Interface Language	90 - 115 KB, depending on language	The language dictionary file used for the phone's user interface ranges from 90KB to 115KB for languages that use an expanded character set. To conserve memory resources, Polycom recommends using XML language files for only the languages you need.
Web Configuration Utility Interface	250 KB - 370 KB	

Phone Memory Alert

You can set a threshold for the phone’s free memory below which the phone displays a warning message.

You can configure a threshold as a percentage of the phone’s free memory. If the phone’s free memory falls below this threshold, for example, 20%, the phone displays a warning message. You can also configure the interval, in minutes, that the phone's free memory is checked.

Phone Memory Alert Parameters

The following table lists parameters that configure the phone memory alert feature.

Phone Memory Alert Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.sysFreeMemThresholdPercent	Set the threshold of free memory, in percentage, below which the phone displays a warning message. 20 percent (default) 20 - 30 percent	No
	up.lowSysMemWarn.timeInMins	Set the interval, in minutes, that the phone's free memory is checked. 0 (default) 0 - 1440 minutes	No

Remote Packet Capture

You can configure phones to capture packets. Using parameters you can enable the remote packet capture feature. The VVX 101 business media phone does not support this feature.

Related Links

Device Diagnostics Details

Remote Packet Capture Parameters

Use these parameters to enable and set up the remote packet capture feature.

Remote Packet Capture Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	diags.dumpcore.enabled	Determine whether the phone generates a core file if it crashes. 1 (default) 0	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`techsupport.cfg`	`diags.pcap.enabled`	Enable or disable all on-board packet capture features. 0 (default) 1	No
`techsupport.cfg`	`diags.pcap.remote.enabled`	Enable or disable the remote packet capture server. 0 (default) 1	No
`techsupport.cfg`	`diags.pcap.remote.password`	Enter the remote packet capture password. <MAC Address> (default) alphanumeric value	No
`techsupport.cfg`	`diags.pcap.remote.port`	Specify the TLS profile to use for each application. 2002 (default) Valid TCP Port	No

Related Links
Device Diagnostics Details

Uploading Logs to a USB Flash Drive

You can configure your VVX phones to copy application and boot logs to a USB flash drive connected to the phone.

You can configure the phone to copy the application logs to the USB flash drive when the log file size reaches the limit defined in the `log.render.file.size` parameter. Similarly, you can configure the phone to application copy logs to the USB flash drive periodically using `log.render.file.upload.period` parameter.

The following VVX phones support this feature:

- VVX 401 business media phones
- VVX 411 business media phones
- VVX 500 series business media phones
- VVX 600 series business media phones
- VVX 250 business IP phones
- VVX 350 business IP phones
- VVX 450 business IP phones

USB Logging Parameter

The following table lists the parameter to configure the USB logging feature.

USB Logging Parameter

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.usbLogging.enabled`	0 (default) – Disables USB logging feature on VVX phones. 1 – Enables USB logging feature on VVX phones.	No

System Logs

Topics:

- Configuring Log Files
- Logging Levels
- Upload Logs to the Provisioning Server

System log files can assist when troubleshooting issues.

System log files contain information about system activities and the system configuration profile. After setting up system logging, you can retrieve a system log file.

The detailed technical data in the system log files can help Polycom Global Services resolve problems and provide technical support for your system. In such a situation, your support representative may ask you to download log archives and send them to Polycom Global Services.

You must contact Polycom Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure log levels.

Configuring Log Files

You can configure log files using the logging parameters.

Log file names use the following format: `[MAC address]_[Type of log].log` . For example, if the MAC address of your phone is `0004f2203b0` , the app log file name is `0004f2203b0-app.log` .

The phone writes information into several different log files. The following table describes the type of information in each.

Log File	Description
Boot Log	Boot logs are sent to the provisioning server in a boot.log file collected from the Updater/ BootROM application each time the phone boots up. The BootROM/Updater application boots the application firmware and updates is new firmware is available.
Application Log	The application log file contains complete phone functionality including SIP signaling, call controls and features, digital signal processor (DSP), and network components.
Syslog	For more information about Syslog, see Syslog on Polycom Phones - Technical Bulletin 17124.

Severity of Logging Event Parameters

You can configure the severity of the events that are logged independently for each module of the Polycom UC Software.

This enables you to capture lower severity events in one part of the application, and high severity events for other components. Severity levels range from 0 to 6, where 0 is the most detailed logging and 6 captures only critical errors. Note that user passwords display in level 1 log files.

You must contact Polycom Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure log levels.

Severity of Events Logged

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`techsupport.cfg`	`log.level.change.`*`module_name`*	Set the severity level to log for the module name you specify. Not all modules are available for all phone models. For a list of available module names, module descriptions, and log level severity, see the Web Configuration Utility at Settings > Logging > Module Log Level Limits.	

Log File Collection and Storage Parameters

You can configure log file collection and storage using the parameters in the following table.

You must contact Polycom Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure log file collection and storage.

There is no way to prevent the system log file [MAC address]-plcmsyslog.tar.gz from uploading to the server and you cannot control it using the parameters `log.render.file.upload.append.sizeLimit` and `log.render.file.upload.append.limitMode` . However, you can control the frequency of uploads using `log.render.file.upload.system.period` .

Log File Collection and Storage Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`techsupport.cfg`	`log.render.level`	Specify the events to render to the log files. Severity levels are indicated in brackets. 0 SeverityDebug (7) 1 SeverityDebug (7) - default 2 SeverityInformational (6) 3 SeverityInformational (6) 4 SeverityError (3) 5 SeverityCritical (2) 6 SeverityEmergency (0)	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	log.render.file.size	Set the maximum file size of the log file. When the maximum size is about to be exceeded, the phone uploads all logs that have not yet been uploaded and erases half of the logs on the phone. You can use a web browser to read logs on the phone. 512 kb (default) 1 - 10240 kB	
techsupport.cfg	log.render.file.upload.period	Specify the frequency in seconds between log file uploads to the provisioning server. Note: The log file is not uploaded if no new events have been logged since the last upload. 172800 seconds (default) - 48 hours	
techsupport.cfg	log.render.file.upload.append	1 (default) - Log files uploaded from the phone to the server are appended to existing files. You must set up the server to append using HTTP or TFTP. 0 - Log files uploaded from the phone to the server overwrite existing files. Note that this parameter is not supported by all servers.	
techsupport.cfg	log.render.file.upload.append.sizeLimit	Specify the maximum size of log files that can be stored on the provisioning server. 512kb (default)	
techsupport.cfg	log.render.file.upload.append.limitMode	Specify whether to stop or delete logging when the server log reaches its maximum size. delete (default) - Delete logs and start logging again after the file reaches the maximum allowable size specified by log.render.file.upload.append.sizeLimit . stop - Stop logging and keep the older logs after the log file reaches the maximum allowable size.	

Scheduled Logging Parameters

Scheduled logging can help you monitor and troubleshoot phone issues.

Use the parameters in this table to configure scheduled logging.

You must contact Polycom Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure scheduled logging.

Scheduled Logging Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	log.sched.x.name	Configure the number of debug commands you want to schedule an output for. You can configure x = 1-10 debug commands per phone. If x = 1, the default command name is ‘showCpuLoad'. 9 (default) If x = 2, the default command name is ‘showBatteryStat'. 22 (default) 3 - 10 = No default value Values: NULL memShow checkStack ls ifShow ifShowVerbose showProcesses showCpuUsage showCpuLoad ethBufPoolShow sysPoolShow netPoolShow netRxShow endErrShow routeShow netCCB arpShow fsShow ipStatShow udpStatShow sipPrt showBatteryStat	No

Logging Levels

The event logging system supports the classes of events listed in the table Logging Levels.

Two types of logging are supported:

- Level, change, and render
- Schedule

Note: Logging parameter changes can impair system operation. Do not change any logging parameters without prior consultation with Polycom Technical Support.

Logging Levels

Logging Level	Interpretation
0	Debug only
1	High detail class event
2	Moderate detail event class
3	Low detail event class
4	Minor error—graceful recovery
5	Major error—will eventually incapacitate the system
6	Fatal error

Each event in the log contains the following fields separated by the | character:

- Time or time/date stamp, in one of the following formats:
  - 0 - milliseconds 011511.006 - 1 hour, 15 minutes, 11.006 seconds since booting
  - 1 - absolute time with minute resolution 0210281716 - 2002 October 28, 17:16
  - 2 - absolute time with seconds resolution 1028171642 - October 28, 17:16:42
- 1-5 character component identifier (such as “so”)
- Event class
- Cumulative log events missed due to excessive CPU load
- The event description

Logging Level, Change, and Render Parameters

This configuration parameter is defined in the following table.

Logging Level, Change, and Render Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	log.level.change.xxx	Controls the logging detail level for individual components. These are the input filters into the internal memory-based log system. 4 (default) 0 - 6 Possible values for xxx are acom, ares, app1, bluet, bdiag, brow, bsdir, cap, cdp, cert, cfg, cipher, clink, clist, cmp, cmr, copy, curl, daa, dapi, dasvc, dbs, dbuf, dhcpc, dis, dock, dot1x, dns, drvbt, ec, efk, ethf, flk, h323, hset, httpa, httpd, hw, ht, ib, key, ldap, lic, lldp, loc, log, mb, mobil, net, niche, ocsp, osd, pcap, pcd, pdc, peer, pgui, pmt, poll, pps, pres, pstn, ptt, push, pwrsv, rdisk, res, restapi, rtos, rtls, sec, sig, sip, slog, so, srtp, sshc, ssps, style, sync, sys, ta, task, tls, trace, ttrs, usb, usbio, util, utilm, vsr, wdog, wmgr, and xmpp.	No
techsupport.cfg	log.level.change.flk	Sets the log level for the FLK logs. 4 (default) 0 - 6	No
techsupport.cfg	log.level.change.mr	Initial logging level for the Networked Devices log module. 4 (default) 0 - 6	No
techsupport.cfg	log.level.change.mraud	Initial logging level for the Networked Devices Audio log module. 4 (default) 0 - 6	No
techsupport.cfg	log.level.change.mrmgr	Initial logging level for the Networked Devices Manager log module. 4 (default) 0 - 6	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	log.level.change.prox	Initial logging level for the Proximity log module. 4 (default) 0 - 6	No
techsupport.cfg	log.level.change.ptp	Initial logging level for the Precision Time Protocol log module. 4 (default) 0 - 6	No
techsupport.cfg	log.level.change.sopi	Specify the SOPI service log level for the GENBAND Global Address Book and Personnel Address Book. 4 (default) 0 - 6	No
techsupport.cfg	log.render.file	Polycom recommends that you do not change this value. 1 (default) 0	No
techsupport.cfg	log.render.realtime	Polycom recommends that you do not change this value. 1 (default) 0	No
techsupport.cfg	log.render.stdout	Polycom recommends that you do not change this value. 0 (default) 1	No
techsupport.cfg	log.render.type	Refer to the Event Timestamp Formats table for timestamp type. 2 (default) 0 - 2	No

Logging Schedule Parameters

The phone can be configured to schedule certain advanced logging tasks on a periodic basis.

Polycom recommends that you set the parameters listed in the next table in consultation with Polycom Technical Support. Each scheduled log task is controlled by a unique parameter set starting with `log.sched.x` where `x` identifies the task. A maximum of 10 schedule logs is allowed.

Logging Schedule Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`techsupport.cfg`	`log.sched.x.level`	The event class to assign to the log events generated by this command. 3 (default) 0 - 5 This needs to be the same or higher than `log.level.change.slog` for these events to display in the log.	No
`techsupport.cfg`	`log.sched.x.period`	Specifies the time in seconds between each command execution. 15 (default) positive integer	No
`techsupport.cfg`	`log.sched.x.startDay`	When startMode is abs, specifies the day of the week to start command execution. 1=Sun, 2=Mon, ..., 7=Sat 7 (default) 0 - 7	No
`techsupport.cfg`	`log.sched.x.startMode`	Starts at an absolute or relative time to boot. Null (default) 0 - 64	No
`techsupport.cfg`	`log.sched.x.startTime`	Displays the start time in seconds since boot when startMode is rel or displays the start time in 24-hour clock format when startMode is abs. Null (default) positive integer, hh:mm	No

Upload Logs to the Provisioning Server

You can manually upload logs to the provisioning server.

When you manually upload log files, the word `now` is inserted into the name of the file, for example, `0004f200360b-now-boot.log` .

Procedure

1. Press the multiple key combination `1-5-9` on the phone.

Troubleshooting

Topics:

The following sections cover some of the errors you might see, along with suggested actions.

Updater Error Messages and Possible Solutions

If a fatal error occurs, the phone does not boot up.

If the error is not fatal, the phone boots up but its configuration might be changed. Most updater errors are logged to the phone's boot log. However, if the phone is having trouble connecting to the provisioning server, the phone is not likely to upload the boot log.

The following table describes possible solutions to updater error messages.

Error Message	Possible Solution
Failed to get boot parameters via DHCP	The phone does not have an IP address and therefore cannot boot. • Check that all cables are connected, the DHCP server is running, and that the phone has not been set to a VLAN that is different from the DHCP server. • Check the DHCP configuration.

Error Message	Possible Solution
Application <file name> is not compatible with this phone!	An application file was downloaded from the provisioning server, but it cannot be installed on this phone. Install a compatible software image on the provisioning server. Be aware that there are various hardware and software dependencies.
Could not contact boot server using existing configuration	The phone cannot contact the provisioning server. Possible causes include: • Cabling issues • DHCP configuration • Provisioning server problems The phone can recover from this error so long as it previously downloaded a valid application BootROM image and all of the necessary configuration files.
Error, application is not present!	The phone does not have an application stored in device settings and, because the application could not be downloaded, the phone cannot boot. • Download compatible Polycom UC Software to the phone using one of the supported provisioning protocols. If no provisioning server is configured on the phone, enter the provisioning server details after logging in to the Updater menu and navigating to the Provisioning Server menu.

Polycom UC Software Error Messages

If an error occurs in the UC Software, an error message and a warning icon displays on the phone.

The location of the Warnings menu varies by model:

- VVX 1500—**Menu** > **Status** > **Diagnostics** > **Warnings**
- VVX phones—**Settings** > **Status** > **Diagnostics** > **Warnings**
- Polycom Trio —**Settings** > **Status** > **Diagnostics** > **Warnings**.

The following table describes Polycom UC Software error messages.

Polycom UC Software Error Messages

Error Message	Cause
Config file error: Files contain invalid params: <filename1>, <filename2>,... Config file error: <filename> contains invalid params The following contain pre-3.3.0 params: <filename>	These messages display if the configuration files contain these deprecated parameters: • tone.chord.ringer.x.freq.x • se.pat.callProg.x.name • ind.anim.IP_500.x.frame.x.duration • ind.pattern.x.step.x.state • feature.2.name • feature.9.name This message also displays if any configuration file contains more than 100 of the following errors: • Unknown parameters • Out-of-range values • Invalid values. To check that your configuration files use correct parameter values, refer to Using Correct Parameter XML Schema, Value Ranges, and Special Characters.
Line: Unregistered	This message displays if a line fails to register with the call server.
Login credentials have failed. Please update them if information is incorrect.	This message displays when the user enters incorrect login credentials on the phone: Status > Basic > Login Credentials.
Missing files, config. reverted	This message displays when errors in the configuration and a failure to download the configuration files force the phone to revert to its previous (known) condition with a complete set of configuration files. This also displays if the files listed in the <MAC Address>.cfg file are not present on the provisioning server.
Network link is down	Indicates that the phone cannot establish a link to the network and persists until the link problem is resolved. Call-related functions, and phone keys are disabled when the network is down but the phone menu works.

Network Authentication Failure Error Codes

This message displays if 802.

1X authentication with the Polycom phone fails. The error codes display on the phone when you press the **Details** key. Error codes are also included in the log files.

Event Code	Description	Comments
1	Unknown events	An unknown event by ‘1' can include any issues listed in this table.
2	Mismatch in EAP Method type Authenticating server's list of EAP methods does not match with clients'.	
30xxx	TLS Certificate failure 000 - Represents a generic certificate error. The phone displays the following codes: 042 - bad cert 043 - unsupported cert 044 - cert revoked 045 - cert expired 046 - unknown cert 047 - illegal parameter 048 - unknown CA	See section 7.2 of RFC 2246 for further TLS alert codes and error codes.
31xxx	Server Certificate failure ‘xxx' can use the following values: •009 - Certificate not yet Valid •010 - Certificate Expired •011 - Certificate Revocation List (CRL) not yet Valid •012 - CRL Expired	
4xxx	Other TLS failures ‘xxx' is the TLS alert message code). For example, if the protocol version presented by the server is not supported by the phone, then ‘xxx' is 70, and the EAP error code is 4070.	See section 7.2 of RFC 2246 for further TLS alert codes and error codes.
5xxx	Credential failures 5xxx - wrong user name or password	
6xxx	PAC failures 080 - No PAC file found 081 - PAC file password not provisioned 082 - PAC file wrong password 083 - PAC file invalid attributes	

Event Code	Description	Comments
7xxx	Generic failures 001 - dot1x can not support (user) configured EAP method 002 - dot1x can not support (user) configured security type 003 - root certificate could not be loaded 174 - EAP authentication timeout 176 - EAP Failure 185 - Disconnected	

Power and Startup Issues

The following table describes possible solutions to power and startup issues.

Power or Startup Issue	Possible Solutions:
The phone has power issues or the phone has no power.	Determine whether the problem is caused by the phone, the AC outlet, or the PoE switch. Do one of the following: • Verify that no lights appear on the unit when it is powered up. • Check to see if the phone is properly plugged into a functional AC outlet. • Make sure that the phone is not plugged into an outlet controlled by a light switch that is turned off. • If the phone is plugged into a power strip, try plugging directly into a wall outlet instead.
The phone does not boot.	If the phone does not boot, there may be a corrupt or invalid firmware image or configuration on the phone: • Ensure that the provisioning server is accessible on the network and a valid software load and valid configuration files are available. • Ensure that the phone is configured with the correct address for the provisioning server on the network.

Dial Pad Issues

The following table describes possible solutions to issues with the dial pad.

Issues	Possible Solutions
The dial pad does not work.	If the dial pad on your phone does not respond, do one of the following: • Check for a response from other feature keys. • Place a call to the phone from a known working telephone. Check for display updates. • On the phone, go to Menu > System Status > Server Status to check if the telephone is correctly registered to the server. • On the phone, go to Menu > System Status > Network Statistics. Scroll down to see whether LAN port shows Active or Inactive. Check the termination at the switch or hub end of the network LAN cable. Ensure that the switch/hub port that is connected to the telephone is operational.

Screen and System Access Issues

The following table describes possible solutions to screen and system access issues.

Issue	Possible solution
There is no response from feature key presses.	If your phone keys do not respond to presses: • Press the keys more slowly. • Check to see whether or not the key has been mapped to a different function or disabled. • Make a call to the phone to check for inbound call display and ringing. If successful, try to press feature keys while a call is active to access a directory or buddy status. • On the phone, go to Navigate to Menu > Status > Lines to confirm the line is actively registered to the call server. Reboot the phone to attempt re-registration to the call server.

Issue	Possible solution
The display shows the message Network Link is Down.	This message displays when the LAN cable is not properly connected. Do one of the following: • Check the termination at the switch or hub end of the network LAN cable. • Check that the switch or hub is operational (flashing link/ status lights). • On the phone, go to **Menu** > **Status** > **Network**. Scroll down to verify that the LAN is active. • Ping the phone from a computer. Reboot the phone to attempt re-registration to the call server. Navigate to Menu > Settings > Advanced > Reboot Phone).

Calling Issues

The following table provides possible solutions to generic calling issues.

Issue	Possible Solution
There is no dial tone.	If there is no dial tone, power may not be correctly supplied to the phone. Try one of the following: • Check that the display is illuminated. • Make sure the LAN cable is inserted properly at the rear of the phone; try unplugging and re-inserting the cable. If you are using in-line powering, check that the switch is supplying power to the phone.
The dial tone is not present on one of the audio paths.	if dial tone is not present on one of the audio paths, do one of the following: • Switch between handset, headset (if present), or handsfree speakerphone to see whether or not dial tone is present on another path. • If the dial tone exists on another path, connect a different handset or headset to isolate the problem. Check configuration for gain levels.
The phone does not ring.	If there is no ringtone but the phone displays a visual indication when it receives an incoming call, do the following: • Adjust the ring level from the front panel using the volume up/down keys. Check the status of handset, headset (if connected), and handsfree speakerphone.

Issue	Possible Solution
The line icon shows an unregistered line icon.	If the phone displays an icon indicating that a line is unregistered, do the following: Try to re-register the line and place a call.

Display Issues

The following table provides tips for resolving display screen issues.

Issue	Possible Solution
There is no display or the display is incorrect.	If there is no display, power may not be correctly supplied to the phone. Do one of the following: • Check that the display is illuminated. • Make sure the power cable is inserted properly at the rear of the phone. • If your are using PoE powering, check that the PoE switch is supplying power to the phone. Use the screen capture feature to verify whether the screen displays properly in the capture. Refer to Capture Your Device's Current Screen.
The display is too dark or too light.	The phone contrast may be set incorrectly. To adjust the contrast, do one of the following: • Adjust the contrast. • Reboot the phone to obtain the default level of contrast.
The display is flickering.	Certain types of older fluorescent lighting cause the display to flicker. If your phone is in an environment lit with fluorescent lighting, do one of the following: Angle or move the Polycom phone away from the lights.
The time and date are flashing.	If the time and date are flashing, the phone is disconnected from the LAN or there is no SNTP time server configured. Do one of the following: • Reconnect the phone to the LAN. • Configure an SNTP server. Disable the time and date if you do not want to connect your phone to a LAN or SNTP server.

Audio Issues

The following table describes possible solutions to audio issues.

Issue	Possible Workaround
There is no audio on the headset	If there is no audio on your headset, the connections may not be correct. Do one of the following: • Ensure the headset is plugged into the jack marked Headset at the rear of the phone. Ensure the headset amplifier (if present) is turned on and adjust the volume.

Licensed Feature Issues

The following table describes issues for features that require a license.

Issue	Possible Solutions
Voice Quality Monitoring or H.323 is not available on the phone.	If you cannot access features, check your licenses on the phone by navigating to Menu > Status > Licenses. • You require a license key to activate the VQMon feature on the following VVX business media phones: 101, 201, 300, 301, 310, 311, 400, 401, 410, 411. The following phones do not require a license key to activate the VQMon feature: 500, 600, 1500, Polycom Trio. • You require a license key to activate the VQMon feature on the following VVX business IP phones: 150, 250, 350, and 450. • You need a license to use H.323 on VVX 1500. You do not need a license to use H.323 on the VVX 500/501, 600/601. Note that H.323 is not supported on VVX 300/301, 310/311, 400/401, 410/411, and SoundStructure VOIP Interface. If your phone is not installed with UC Software version 4.0.0 or later, you also require a license for conference management, corporate directory, and call recording.

Software Upgrade Issues

The following table describes possible solutions to issues that may occur during or after a software upgrade.

Issue	Possible Solutions
Some settings or features are not working as expected on the phone.	The phone's configuration may be incorrect or incompatible. Check for errors on the phone by navigating to Menu > Status > Platform > Configuration. If there are messages stating Errors Found, Unknown Params, or Invalid values, correct your configuration files and restart the phone.
The phone displays a Config file error message for five seconds after it boots up.	You are using configuration files from a UC Software version earlier than the UC Software image running on the phones. Configuration parameters and values can change each release and specific parameters may or may not be included. • Correct the configuration files, remove the invalid parameters, and restart the phone. See the UC Software Administrator's Guide and Release Notes for the UC Software version you have installed on the phones.

Issue	Possible Solutions
When using the Web Configuration Utility to upgrade phone software, the phone is unable to connect to the Polycom Hosted Server.	Occasionally, the phone is unable to connect to the Polycom hosted server because of the following: • The Polycom hosted server is temporarily unavailable. • There is no software upgrade information for the phone to receive. • The network configuration is preventing the phone from connecting to the Polycom hosted server. Note: UC Software 4.0.0 does not support internet access for software upgrades through a web proxy. To troubleshoot the issue: • Try upgrading your phone later. • Verify that new software is available for your phone using the *Polycom UC Software Release Matrix for VVX Phones*. • Verify that your network's configuration allows the phone to connect to http://downloads.polycom.com. If the issue persists, try manually upgrading your phone's software.Occasionally, the phone is unable to connect to the Polycom hosted server because of the following: • The Polycom hosted server is temporarily unavailable. • There is no software upgrade information for the phone to receive. • The network configuration is preventing the phone from connecting to the Polycom hosted server. Note: UC Software 4.0.0 does not support internet access for software upgrades through a web proxy. To troubleshoot the issue: • Try upgrading your phone later. • Verify that new software is available for your phone using the *Polycom UC Software Release Matrix for VVX Phones*. • Verify that your network's configuration allows the phone to connect to http://downloads.polycom.com. If the issue persists, try manually upgrading your phone's software.

Wireless Handset Software Upgrade Issues

If any wireless handset fails to update its software, the base station retries pushing the update to the wireless handset three times before moving to the next registered handset.

If the base station or wireless handset fails to update or restarts during the update process, the base station or wireless handset restarts with the previous software version.

Try the following solutions if the base station or any of the wireless handsets fail to update:

- Manually update the wireless handset software.
- Restart the base station, then pair the base station with the VVX business media phone again.
- To restart the base station, using a paper clip, press and hold the Reset button on the back of the base station for five seconds.

- After the software process is complete for all registered wireless handsets, unregister and register any wireless handsets that failed to update.

If the base station fails to pair with the VVX business media phone after successfully updating, you need to repair the base station with the phone manually.

Related Links

Pairing a VVX Phone with a VVX D60 Base Station on page 155

Provisioning Issues

If settings you make from the central server are not working, check first for priority settings applied from the phone menu system or Web Configuration Utility, and second for duplicate settings in your configuration files.

Hardware and Accessories

Topics:

This section provides information on configuring power management options, pairing hardware, adding expansion modules and configuring the Polycom Desktop Connector for your users.

Powering VVX Phones with an Ethernet Switch Connection

VVX business media phones and business IP phones have two Ethernet ports—labeled LAN and PC—and an embedded Ethernet switch that runs at full line rate.

The SoundStructure VoIP Interface has one Ethernet port, labeled LAN. The Ethernet switch enables you to connect a personal computer and other Ethernet devices to the office LAN by daisy-chaining through the phone, eliminating the need for a stand-alone hub.

You can power each phone through an AC adapter or through a Power over Ethernet (PoE) cable connected to the phone's LAN port. If you are using a VLAN, ensure that the 802.1p priorities for both default and real-time transport protocol (RTP) packet types are set to 2 or greater so that audio packets from the phone have priority over packets from the PC port.

Power-Saving

The power-saving feature automatically turns off the phone's LCD display when not in use.

Power-saving is not available on the VVX 101 business media phone or SoundStructure VoIP Interface.

Power-saving is enabled by default for the VVX 500/501, 600/601, and 1500 phones.

You can configure the following power-saving options:

- Turn on the phone's power-saving feature during non-working hours and working hours.

  If you want to turn on power-saving during non-working hours, you can configure the power-saving feature around your work schedule.
- On the VVX 1500, use the `powerSaving.userDetectionSensitivity.*` parameters to configure the sensitivity of the built-in motion detection system and an idle time after which the phone enters the power-saving mode.

When you enable power-saving mode and the phone is in low power state, the red LED indicator flashes at three second intervals to show that the phone still has power.

Note: When you enable power-saving mode on VVX 500 and 600, the phone display screen does not automatically turn back on after going idle.

Power-Saving Parameters

Use the parameters in the following table to configure the power-saving features and feature options.

Power-Saving Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	powerSaving.enable	Enable or disable the power-saving feature. The default value varies by phone model. VVX 201=0 (default) VVX 300/301/310/311=0 (default) VVX 400/401/410/411=0 (default) VVX 500/501, 600/601, 1500=1 (default) 1 - Enable the LCD power-saving feature. 0 - Disable The LCD power-saving feature. Note that when the phone is in power-saving mode, the LED Message Waiting Indicator (MWI) flashes. To disable the MWI LED when the phone is in power saving mode, set the parameter `ind.pattern.powerSaving.step.1.state.x` to 0 where x=your phone's model. For example, enter the parameter as `ind.pattern.powerSaving.step.1.state.VVX500` to disable the MWI for your VVX 500 phone.	No
site.cfg	powerSaving.idleTimeout.offHours	The number of idle minutes during off hours after which the phone enters power saving. 1 (default) 1 - 10	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	powerSaving.idleTimeout.officeHours	The number of idle minutes during office hours after which the phone enters power saving. 30 (default) 1 - 600	No
site.cfg	powerSaving.idleTimeout.userInputExtension	The number of minutes after the phone is last used that the phone enters power saving. 10 (default) 1 - 20	No
site.cfg	powerSaving.officeHours.duration.Monday powerSaving.officeHours.duration.Tuesday powerSaving.officeHours.duration.Wednesday powerSaving.officeHours.duration.Thursday powerSaving.officeHours.duration.Friday powerSaving.officeHours.duration.Saturday powerSaving.officeHours.duration.Sunday	Set the duration of the office working hours by week day. Monday - Friday = 12 (default) Saturday - Sunday = 0 0 - 24	No
site.cfg	powerSaving.officeHours.startHour.x	Specify the starting hour for the day's office working hours. 7 (default) 0 - 23 Set x to Monday, Tuesday, Wednesday, Thursday, Friday, Saturday, and Sunday (refer to powerSaving.officeHours.duration for an example).	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	powerSaving.userDetectionSensitivity.offHours	Available on the VVX 1500 only. The sensitivity used to detect the presence of the phone's user during off hours. 2 (default) - The default value was chosen for good performance in a typical office environment and is biased for difficult detection during office hours. 0 - The feature is disabled. 1 - 10 - Set the sensitivity.	No
site.cfg	powerSaving.userDetectionSensitivity.officeHours	Available on the VVX 1500 only. The sensitivity used to detect the presence of the phone's user during office hours. 7 (default) - The default value was chosen for good performance in a typical office environment and is biased for easy detection during office hours. 0 - The feature is disabled. 1 - 10 - Set the sensitivity.	No

Headset and Speakerphone

All VVX phones are equipped with a handset and a dedicated RJ9 headset port.

While handsets are shipped with all VVX phones, headsets are not provided. The following VVX phones are also equipped with a USB port you can use for a USB headset or other USB device:

- VVX 401/411 business media phones
- VVX 500/501 business media phones
- VVX 600/601 business media phones
- VVX 250/350/450 business IP phones

By default, VVX phones have dedicated keys to switch to speakerphone or headset. You can enable or disable the handsfree speakerphone mode and headset mode.

Headset and Speakerphone Parameters

You can use the parameters in the following table to enable and disable the headset or speakerphone and control other options for the headset and speakerphone.

Configure the Headset and Speakerphone

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.analogHeadsetOption	Electronic Hookswitch mode for the phone's analog headset jack. 0 (Default) - No EHS-compatible headset is attached. 1 - Jabra EHS-compatible headset is attached. 2 - Plantronics EHS-compatible headset is attached. 3 - Sennheiser EHS-compatible headset is attached.	No
features.cfg	up.audioMode	Specify whether you want to use the handset or headset for audio. 0 (Default) 1	No
features.cfg	up.handsfreeMode	1(default) - Enable handsfree mode. 0 - disable handsfree mode.	No
features.cfg	up.headset.phoneVolumeControl	Controls the phone's behavior when you adjust volume at the headset. Auto (Default) - Phone automatically selects which of the above two behaviors to apply based on the type and model of headset that you attach. Disable - Phone ignores volume up/down events from the headset; pressing the headset's volume controls has no effect on the phone. Enable - Phone responds to volume up/down events from the headset, displays the volume widget in the phone's user interface and adjusts the phone's internal volume.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`voice.volume.persist.handsfree`	Specify if the speakerphone volume persists between calls. 1 (default) - The speakerphone volume at the end of a call persists between calls. 0 - The speakerphone volume does not persist between calls and is reset each new call.	No
`site.cfg`	`voice.volume.persist.usbHeadset`	0 (default) – The USB headset volume does not persist between calls and is reset each new call. 1 - The USB headset volume at the end of a call persists between calls.	No

Polycom VVX Expansion Modules

The Polycom VVX Expansion Modules are consoles you can connect to VVX business media phones to add additional lines.

VVX Expansion Modules enable users to efficiently perform the following tasks:

- Handle large call volumes on a daily basis
- Expand the functions of their phone
- Accept, screen, dispatch, and monitor calls
- Reduce the number of lost customer calls
- Shorten transaction times
- Increase the accuracy of call routing

Polycom VVX Expansion Modules are available for the following Polycom VVX business media phones running UC Software 4.1.6 or later:

- VVX 300 series and 400 series
- VVX 500 series and 600 series

Note: For all documents that help you set up and use the Polycom VVX expansion modules with your VVX phones see Polycom VVX Expansion Modules Support page.

VVX Expansion Module Features

The following features are available on the VVX LCD Color Expansion Modules and VVX Expansion Modules with a paper display:

- VVX Expansion Modules - LCD Color Display—VVX Color Expansion Modules feature an easy-to-navigate 272x480 LCD display. Each color expansion module provides users with 28 line keys and 3 display pages, supporting a total of 84 lines that you can set up as registrations, favorites, busy lamp field contacts, or Microsoft Skype for Business presence contacts. You can connect up to three color expansion modules to a phone to support an additional 252 line keys per phone.

If you are registering Polycom phones with Skype for Business Server, you can use only the LCD color display expansion modules; you cannot use the paper display expansion modules for phones registered with Skype for Business Server.

- VVX Expansion Modules - Paper Display—VVX Expansion Modules with a paper display provide users with 40 line keys that you can set up as registrations, favorites, or busy lamp field contacts. You can connect up to three expansion modules to your phone to support an additional 120 line keys per phone.

The following figure illustrates the LCD color and paper expansion modules.

Expansion Module LCD color display and paper display

Expansion Module Line Keys

The line keys on VVX phones and expansion modules are numbered sequentially, and the line key numbering on an expansion module depends on how many lines the phone supports.

For example, a VVX 600/601 phone supports 16 lines, numbered 1-16. The first line on an expansion module connected to a VVX 600/601 phone is line 17.

Flexible Call Appearances

VVX Expansion Module Power Values

Polycom VVX phones use more power when you connect an expansion module.

The following table outlines the power each phone uses when you connect an expansion module, as well as the power value sent in LLDP-MED. For a list of power values for all Polycom phones without an expansion module attached, see Power Values.

Model	Power Usage (Watts)	Power Value Sent in LLDP-MED Extended Power Via MDI TLV
VVX 300/301	5.0	5000mW
VVX 310/311	5.0	5000mW
VVX 400/401	5.0	5000mW
VVX 410/411	5.0	5000mW

Model	Power Usage (Watts)	Power Value Sent in LLDP-MED Extended Power Via MDI TLV
VVX 500/501	8.0	8000mW
VVX 600/601	8.0	8000mW

Generate a Line Key PDF for Paper VVX Expansion Modules

Using the Web Configuration Utility, you can generate and download a PDF file with the line key configuration for each paper display expansion module connected to a VVX phone.

The generated PDF enables you to print line key information for line keys on your expansion modules and insert the PDF as a directory card on expansion modules.

Procedure

1. In a web browser, enter your phone's IP address into the address bar.
2. Log in as an Admin, enter the password, and select Submit.
3. Select Utilities > EM Directory.
4. Select the expansion module you want to generate a PDF for.

   For example, select EM1.
5. In the confirmation dialog, select Yes to download the PDF for the configured lines for your expansion module.
6. Select Save > Open.

   The PDF with the configured line key information for your expansion module displays.

After you download the PDF with configured line key information for your expansion module, you can print the PDF and insert the PDF as the directory card for the expansion module.

Smart Paging on VVX Expansion Modules

The smart paging feature arranges line key assignments and distributes pages on the VVX Color Expansion Modules based on the number of expansion modules connected to a VVX phone.

Smart paging is automatically enabled for color expansion modules connected to VVX phones with UC Software 5.1.0 or later, and is not available on the VVX Expansion Modules with a paper display.

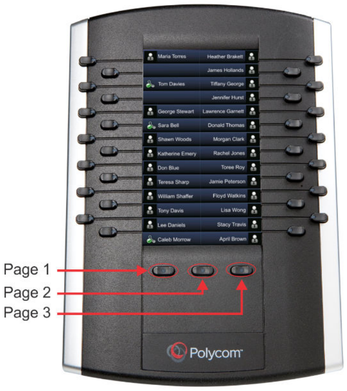
Note that when the flexible line key feature is enabled, the expansion module ignores the smart paging configuration and line key assignments display on the designated line key.

Note: Smart paging applies only when you connect more than one expansion module to a VVX phone. If you connect one expansion module, the order of pages is sequential even if smart paging is disabled.

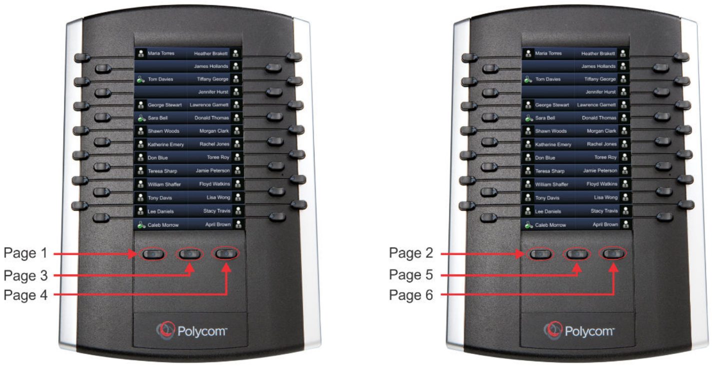
Smart Paging Distribution Scenarios

When you enable smart paging, the pages on the color expansion module are distributed across the connected expansion modules, as described in the following scenarios.

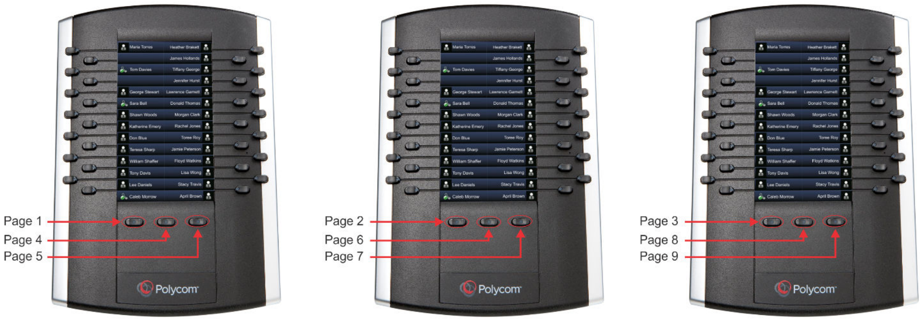
- If only one expansion module is connected to the VVX phone, the pages are ordered sequentially on the module from left to right: pages 1, 2, and 3.



- If two expansion modules are connected, the pages are ordered non-sequentially from left to right across both expansion modules where pages 1, 3, and 4 are on the first expansion module, and pages 2, 5, and 6 are on the second expansion module.



- If you are using three connected expansion modules, the pages are distributed across all modules from left to right where pages 1, 4, and 5 are on the first expansion module, pages 2, 6, and 7 are on the second expansion module, and pages 3, 8, and 9 are on the third expansion module.



Smart Paging Parameters

The following table lists the configuration parameter you need to enable and disable the smart paging feature.

Configuring Smart Paging

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
em.cfg	up.em.smartpaging.enabled	Enable or disable line key assignments and page distribution on VVX Expansion Modules. 1 (Default) - Smart Paging is enabled. 0 - Smart Paging is disabled. The flexible line key configuration overrides Smart Paging for the expansion module, and Smart Paging is disabled for VVX Expansion Modules with a paper display.	No

Polycom Desktop Connector

With the Polycom®Desktop Connector™ application installed on a computer, users can use their mouse and keyboard to enter information and navigate screens on VVX business media phones and VVX business IP phones without having to use the phone's keypad or touchscreen.

The Desktop Connector application is compatible with computers running Microsoft® Windows XP®, Windows Vista®, and Windows® 7.

After users install the Polycom Desktop Connector, they need to use one of two methods to pair the VVX phone and the computer:

- Direct—If the phone is connected directly to the computer over Ethernet, users can select Reconnect to connect the phone with the desktop application.
- Indirect—If the phone is connected to the computer through a switch or hub, users can enter the computer's IP address into the phone's user interface and select Reconnect.

Note: For details on how to install Polycom Desktop Connector application and enable it for use on VVX phones, see the latest *Polycom VVX Business Media Phones User Guide* at Latest Polycom UC Software Release.

While pairing, the Polycom Desktop Connector application shows a pop-up to the user with phone's Secure Shell (SSH) server RSA key. When authentication for the requested connection is successful, the key gets permanently stored in the application. When the user accepts the connection by selecting **Yes**, the application stores the key for future connections and does not prompt again. However, if user selects **No**, the connection establishes but the application will prompt upon new connection.

Polycom Desktop Connector Parameters

To use this feature, the phone and computer must be on the same network or directly connected through the phone's PC port.

You can configure this feature using configuration parameters shown in the following table or by using the Web Configuration Utility.

Enable Polycom Desktop Connector Integration

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	apps.ucdesktop.adminEnabled	1 (default) - Enable he Polycom Desktop Connector for administrator configuration. 0 - Disable the Polycom Desktop Connector for administrator configuration.	Yes
applications.cfg	apps.ucdesktop.desktopUserName	The user's name, supplied from the user's computer, for example, bsmith . NULL (default) string	No
applications.cfg	apps.ucdesktop.enabled	0 (default) - Disable the Polycom Desktop Connector for users. 1 - Enable the Polycom Desktop Connector for users.	No
applications.cfg	apps.ucdesktop.orientation	The location of the VVX 500/501 and 1500 with respect to the user's computer. For example, to the Left of the computer. Unspecified (default) Left Right	No
applications.cfg	apps.ucdesktop.ServerAddress	The user's computer as a fully qualified domain name (FQDN), for example, computer@yourcompany.com . NULL (default) string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`apps.ucdesktop.ServerPort`	The port number. Note: This value should be the same as the one that is used on the user's computer, otherwise the connection is not established. 24800 (default) 1 to 65535	No

USB Port Lock

The USB port lock down feature enables you to choose which of the phone's USB ports to power on or off.

The port lock down feature is available on phones that have USB ports:

- VVX 401/411, 500/501, 600/601, and 1500 phones
- VVX 250, 350, and 450 business IP phones

Also note the following:

- VVX 250, 401/411, and 1500 have a single USB port.
- VVX 250, 350, 450, 500/501 and 600/601 phones support two USB ports.
- The top USB port on the VVX 500/501 and 600/601 supports the VVX Camera. Top and rear USB ports are enabled by default.

The phone ports support various USB devices such as USB mass storage devices and a USB headset. The following features are not available when you disable a USB port:

- Call recording
- Picture frame
- USB headset
- USB camera for video calls on the VVX 500/501 and 600/601 - no video calls
- USB charging device on the rear port of the VVX 500/501 and 600/601

Note: When you connect a power adapter to a VVX 500/501, the USB ports are powered on even if the parameters f `eature.usbTop.power.enabled` and `feature.usbRear.power.enabled` are disabled. This can cause issues during phone reboots when USB devices are connected to the phone.

USB Port Lock Down Parameters

You can use the parameters in the following table to control the USB ports on the supported phones.

Note the following when setting parameters:

- The parameter `feature.usbTop.power.enabled` applies only to the VVX 1500 right-side port.
- The parameter `feature.usbRear.power.enabled` applies only to the VVX 401/411 rear port.

- You can control the VVX 500/501 and 600/601 top and rear USB ports independently using `feature.usbTop.power.enabled` to control the top USB port and `feature.usbRear.power.enabled` to control the rear USB port.
- If you set the parameter `feature.usbTop.power.enabled` to 0 to disable the top USB port on VVX 500/501 and 600/601 phones, you must set the parameter `video.enable` to 0 as well.

Note: Two parameters `feature.usbTop.power.enabled` and `feature.usbRear.power.enabled` replace `feature.usb.power.enabled` . You must replace `feature.usb.power.enabled` with these two new parameters in your configuration file and set both parameters to 0 to disable USB ports.

USB Port Lock Down Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.usbTop.power.enabled`	This parameter applies to the top port of the VVX 1500 business media phones and to the side port of the VVX 250, 350, and 450 business IP phones. 1 (default) - Enable power to the USB port (port 1). 0 - Disable power to the USB port and the phone does not detect USB devices connected to the USB port.	No
`features.cfg`	`feature.usbRear.power.enabled`	This parameter applies to all VVX business media phones except the VVX 1500. This parameter applies to the rear port of VVX 350 and 450 business IP phones. 1 (default) - Enable power to the rear USB port (port 2). 0 - Disable power to the rear USB port and the phone does not detect USB devices connected to the rear USB port.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.enable`	To ensure the USB port is disabled on when you set `feature.usbTop.power.enabled` to 0, you must also disable this parameter. 1 (default) - Enables video in outgoing and incoming calls. 0 - Disables video. The G.722.1C and Siren 14 codecs are disabled when you enable video on the VVX 500 and 600 business media phones.	No

Polycom VVX D60 Wireless Handset and Base Station

Topics:

You can pair the Polycom® D60 Wireless Handset to VVX business media phones and VVX business IP phones to allow users mobile access to calls and call controls.

You can pair one base station and register up to five wireless handsets to the VVX 300 series, 400 series, 500 series, and 600 series business media phones, and to VVX 250, 350, and 450 business IP phones.

Features Supported on VVX D60 Wireless Handsets

The following table includes common features that are and are not supported on VVX D60 wireless handsets.

Feature	Supported
Busy Lamp Field (BLF)	Yes (on paired VVX)
Hunt Groups	Yes (on paired VVX)
Local Conference Calling	Yes
Push-to-Talk	No
Shared Line Appearance/Shared Call Appearances	Yes (BroadSoft only)
Simultaneous Calls (G.729 Encode/Decode)	Yes (maximum 4 active calls)
Skype for Business Line Registration	No
USB Call Media Recording (CMR)	No
VVX Camera	Yes

Feature	Supported
VVX Expansion Module	Yes
Flexible Line Keys (FLK)	Yes (on paired VVX)
Automatic Call Distribution (ACD) / Hoteling	Yes (on paired VVX)
CDP support on VVX D60 base station	Yes
Call HandOff Between VVX D60 Handsets and VVX Business Media Phones on twinned lines	Yes
Configure maximum number of handsets	Yes
Pairing using Mac address of VVX D60	Yes

The VVX D60 base station can access Voice VLAN through Link-Layer Discovery Protocol (LLDP) and Cisco Discovery Protocol (CDP). The VVX D60 base station supports CDP and is enabled by default on the VVX D60 base station.

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

Pairing a VVX Phone with a VVX D60 Base Station

You can pair the VVX D60 base station and register the wireless handset to a VVX business media phone using the local phone interface, the Web Configuration Utility, or through the provisioning server.

You can use the following methods to pair the base station with a VVX business media phone:

- PC Port pairing
- Automatic pairing
- Manual pairing
- MAC address pairing

Related Links
Update the Wireless Handset Software Manually on page 163
Wireless Handset Software Upgrade Issues on page 138

Limitations to MAC Address Pairing

The limitations for pairing of VVX business media phones and the VVX D60 base station through MAC address are as follows:

- User actions are given higher precedence. Consider the user unpairs the VVX D60 base station that is paired to the VVX business media phone using the configuration file and then the user pairs manually through automatic pairing or PC port pairing. In this case, if the VVX business media phone restarts due to a power outage or software update, then the VVX business media phone re-pairs with the VVX base station which is paired using the non-MAC based pairing mode.

- If the user unpairs the VVX D60 base station, then the base station does not pair automatically with the VVX business media phone.
- If the device is currently paired and the current pairing mode is other than through the MAC address, the VVX business media phone logs a warning provided the configuration parameter VVXD60.base.mac is set.
- The configuration parameter VVXD60.base.mac is applied only if `feature.dect.enabled` is enabled.

Obtain the Base Station IP Address

If you use Manual Pairing to pair the base station with the VVX business media phone, you need to use your computer to get the IP address of the base station.

You can use either the Static or DHCP IP address to pair the base station with the phone.

Procedure

1. Connect the Ethernet cable from the PC port on the base station to an Ethernet port on a computer.
2. On the computer, navigate to **Network and Sharing Center**, then select **Local Area Connection**.
3. Click Properties, select **Internet Protocol Version 4 (TCP/IPv4)**, then click **Properties**.
4. Select **Use the following IP address**, then enter the following values into the associated fields:
   - IP Address: 192.168.0.10
   - Subnet mask: 255.255.255.0
   - Default Gateway: 192.168.0.1
5. Click **OK**.
6. In a web browser, enter https://192.168.0.2.
7. In the Web Configuration Utility, enter the following default credentials:
   - User name: `Polycom`
   - Password: `456`
8. Navigate to **Settings** > **Network Settings**.

   The IP address of the base station displays in the IP Settings tab.

Pairing the Base Station using the Web Configuration Utility

You can pair the VVX D60 base station using the Web Configuration Utility in the following methods:

- PC Port pairing
- Automatic pairing
- Manual pairing
- MAC address pairing

By default, users are not allowed to pair or unpair a VVX D60 base station with a VVX business media phone. Administrators can control whether users are allowed to pair a base station only, unpair a base station only, or pair and unpair a base station with a VVX business media phone. You can use the parameter `VVXD60.allowPairing` to configure this feature.

Pair using PC Port Pairing

When the Ethernet cable is connected to the LAN port on the VVX D60 base station and the PC port on a VVX phone, the phone initiates SSH tunneling and pairs with the base station automatically.

Procedure

1. Sign into the Web Configuration Utility, and navigate to **Settings** > **VVX D60 Settings**.
2. Click the **VVX D60 Profile** check box and select **PC Port** in the **Pairing Mode** drop-down menu.

Pair using Automatic Pairing

When you connect the Ethernet cable from the base station LAN port into a LAN outlet, the phone pairs with the base station automatically.

All base stations on the network are displayed automatically in the VVX Web Configuration Utility as long as the devices are on the same subnetwork and VLAN.

Procedure

1. Sign into the Web Configuration Utility, and navigate to **Settings** > **VVX D60 Settings**.
2. Click the **VVX D60 Profile** check box and select **Automatic** in the **Pairing Mode** drop-down menu.

   A list of discovered base stations and wireless handsets display.
3. Select a base station and click **Pair**.

Pair using Manual Pairing

When the Ethernet cable is connected from the base station LAN port to the VVX phone's PC port or when the Ethernet cable is connected from the base station LAN port into a LAN outlet, you can manually enter the base station IP address to pair with a VVX business media phone.

Manual pairing enables you to pair the base station with the phone without the base station being on the same subnetwork or VLAN as the VVX phone.

Note: When the VVX D60 base station is connected to a subnet that is different than that of the VVX business media phone, Polycom recommends that you either configure the IP address of the VVX D60 base station statically or via static DHCP. This will help to minimize pairing issues if the IP address of the VVX D60 base station changes.

Procedure

1. Sign into the Web Configuration Utility, and navigate to **Settings** > **VVX D60 Settings**.
2. Click the **VVX D60 Profile** check box and select **Manual** in the **Pairing Mode** drop-down menu.
3. Enter the IP address of the base station.

   The base station's information displays.

Pair using MAC Address Pairing

When the Ethernet cable is connected from the base station LAN port into a LAN outlet or an outlet with external power supply of the VVX D60 base station, you can manually select to pair using the MAC address.

Procedure

1. Sign into the Web Configuration Utility, and navigate to **Settings** > **VVX D60 Settings**.
2. Click the **VVX D60 Profile** check box and select **MAC Address** in the **Pairing Mode** drop-down menu.
3. Enter the base station MAC Address.
4. Click **Pair**.

Pairing the Base Station using the Local Phone Interface

You can pair the VVX D60 base station using the local phone interface in the following methods:

- PC Port pairing
- Automatic pairing
- Manual pairing
- MAC address pairing

Pair using PC Port Pairing

When the Ethernet cable is connected from the base station LAN port to the PC port on the VVX phone, the phone pairs with the base station automatically.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, and enter the password.
2. Select Administration **Settings** > **VVX D60 Configuration**.
3. Select **VVX D60 Profile**, then select **Enable**.
4. On the **VVX D60 Configuration** screen, select **Base Station**, then select **PC Port Pairing**.

Pair using Automatic Pairing

When the Ethernet cable is connected from the base station LAN port into a LAN outlet, the phone pairs with the base station automatically.

All base stations on the network are displayed automatically on the VVX phone as long as the devices are on the same subnetwork or VLAN.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, and enter the password.
2. Select **Administration Settings** > **VVX D60 Configuration**.
3. Select **VVX D60 Profile**, then select **Enable**.

4. On the **VVX D60 Configuration** screen, select **Base Station**, then select **Auto Pairing**.

Pair using Manual Pairing

When the Ethernet cable is connected from the base station LAN port to the VVX PC port or when the Ethernet cable is connected from the base station LAN port into a LAN outlet, you can manually enter the base station IP address to pair with a VVX phone.

Manual pairing enables you to pair the base station with the phone without the base station being on the same subnetwork or VLAN as the VVX phone.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, and enter the password.
2. Select **Administration Settings** > **VVX D60 Configuration**.
3. Select **VVX D60 Profile**, then select **Enable** and go to the previous menu
4. On the **VVX D60 Configuration** screen, select **Base Station**, then select **Manual Pairing**.
5. Enter the IP address of the base station, then select **Pair**.

   The base station's information displays.

Pair using MAC Address Pairing

When the Ethernet cable is connected from the base station LAN port into a LAN outlet, you can manually select to pair using the MAC address.

If the phone is already configured with a MAC address using configuration parameter, you can choose **Skip**.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, and enter the password.
2. Select **Administration Settings** > **VVX D60 Configuration**.
3. On the **VVX D60 Configuration** screen, do one of the following:
   - Select **Skip** to manually pair with a different base station.
   - Select **Continue** to pair with the configured MAC address.If you select neither Skip nor Continue, a timer is displayed and the VVX business media phone pairs with the configured MAC address.
4. On the **Manual Pairing** screen, select **Base Station MAC ID**.
5. Edit the configured base station MAC ID with the new address.

   The configuration parameter for the VVX business media phone gets updated with the new MAC address.

Unpairing the Base Station for MAC Address-Based Pairing

You can unpair the VVX D60 base station by removing the corresponding MAC address in the configuration parameter VVXD60.

base.mac. If the MAC address configured in the parameter VVXD60.base.mac is modified, the VVX business media phone unpairs the existing VVX D60 base station and tries to pair with VVX D60 base station with the modified MAC address.

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

If the VVX phone unpairs from a previously paired VVX D60 base station for any reason, such as a power outage, the phone will continuously attempt to pair with the base station again until the phone and base station are successfully paired.

This is achieved with the following mechanisms:

- A unicast re-pairing beacon packet is sent to the last known IP address of the VVX D60 base station.
- Three seconds later, a broadcast re-pairing beacon packet is sent to the broadcast address. This is used in case the IP address of the VVX D60 base station has changed.
- The VVX phone waits for a random time interval, between 30 and 60 seconds before resending the unicast and broadcast re-pairing beacon packets.

If the VVX D60 base station and the VVX phone are in the same subnet, the VVX phone tries to send the unicast re-pairing beacon packet three times; after the third attempt, only the broadcast re-pairing beacon packet is tried indefinitely. If the VVX D60 base station and VVX phone are in different subnets, the VVX phone tries to send resend the unicast and broadcast re-pairing beacon packets.

If a user no longer wants the base station to pair with the phone, the user must contact a system administrator to cancel the pairing attempt.

After powering on, the VVX D60 base station may take up to 60 seconds to re-pair with the VVX phone.

Registering Handsets for VVX D60 Base Station

You can control the number of handsets that can be registered to the VVX D60 base station.

A minimum of one and a maximum of five handsets can be configured for a VVX D60 base station. This is configurable from the VVX phone and Web Configuration Utility. Upon pairing, the VVX phone makes sure that maximum handsets registered to the paired VVX D60 base station is lesser or equal to the configured value. After reaching the maximum limit of handsets, the VVX D60 base station and the VVX phone do not provide any mechanism to register a new handset. The administrator can use the parameter `VVXD60.handset.maxCount` to configure this feature.

If the VVX D60 base station is registered with more handsets than the configured number of handsets, then the handsets will be deleted in the following order:

- Blocked
- Unavailable
- Available (the last handset that was registered among the available handsets)

Maximum Number of Handsets

You can use the Web Configuration Utility or local phone interface to configure the maximum number of handsets that can be registered to the VVX D60 base station.

Set the Maximum Number of Registered Handsets using the Web Configuration Utility

You can configure the number of VVX D60 handsets that can be configured for a VVX base station using the Web Configuration Utility.

Procedure

1. On the Web Configuration Utility, login as the administrator and navigate to **Settings** > **VVX D60 Settings**.
2. In the **Max Configurable Handsets** page displayed, enter the number of headsets to be registered to the VVX D60 base station.

Set the Maximum Number of Registered Handsets using the Local Phone Interface

You can configure the number of VVX D60 handsets that can be configured for a VVX base station using the VVX phone interface.

Procedure

1. On the VVX phone, navigate to **Settings** > **Advanced Settings** > **Administration Settings** > **VVX D60 Settings** > **Handset Configuration**.
2. In the **Max Configurable Handsets** page displayed, enter the number of headsets to be registered to the VVX D60 base station.

Register a VVX D60 Wireless Handset

After the base station is paired with the VVX phone, you can register up to five wireless handsets to the base station.

Procedure

1. On the wireless handset, navigate to Settings > Features > Registration.
2. Select Register.
3. Press and hold the **Find** button on the base station for a few seconds.
4. On the wireless handset, confirm the registration with the base station.

Unregister a VVX D60 Wireless Handset

You can unregister a wireless handset from the base station when you need to replace a wireless handset with another one.

Procedure

1. On the wireless handset, navigate to Settings > Features > Registration.
2. Select Deregister.
3. Confirm you want to unregister the wireless handset.

Set a Unique Name for the Base Station and Wireless Handset

In the Web Configuration Utility, you can set a unique name for each base station and wireless handset to distinguish between multiple sets of base stations and wireless handsets.

You can also set a unique name for the base station and wireless handsets from the local phone interface.

Note that the Intercom feature must be enabled to change the name of a wireless handset. You cannot set a unique name for a wireless handset if the Intercom feature is disabled.

Procedure

1. In the Web Configuration Utility, navigate to **Settings** > **VVX D60 Settings**.
2. Under base station Settings, enter a unique name in the Name field.
3. Under Handset Settings, enter a unique name in the Display Name field for each registered handset.

Assigning Lines to the VVX D60 Wireless Handset

After you have paired the base station to a VVX phone and registered wireless handsets to the base station, you can assign lines to each wireless handset.

You can assign up to five lines to each wireless handset.

When assigning lines, keep the following in mind:

- The first line is assigned to the VVX phone.
- For Private Lines, you can assign each line to the VVX phone or the Wireless Handset or both.
- For Shared lines (SCA/SLA), you can assign each line only to one device: VVX phone or Wireless Handset.

Assign Lines using the Web Configuration Utility

You can assign lines to the wireless handset using the Web Configuration Utility.

Procedure

1. In the Web Configuration Utility, navigate to **Settings** > **VVX D60** Settings.
2. In the **Handset Settings** section, click **Map Lines**.
3. Choose the lines you want to map to a registered wireless handset.
4. Click **Update**.

Assign Lines using the Phone Interface

You can assign lines to the wireless handset from the Advanced settings menu on the VVX phone.

Procedure

1. On the phone, navigate to **Settings** > **Advanced**, then enter your password.
2. Select **Administration Settings** > **VVX D60 Configuration** > **Map Lines**.
3. Choose a line, then choose a registered wireless handset for the line.

Update the VVX D60 Wireless Handset Software

When you update the VVX host phone with the latest supported software version using the master configuration file that includes the file path to the dect.

ld, the software on the base station and wireless handsets update automatically within two minutes after they are paired and registered with the VVX phone. The base station updates first, then each wireless handset is updated sequentially with the first registered handset updating first, followed by each remaining handset.

Procedure

1. Place the handset in the base station or charging cradle, and ensure the handset battery is charged to at least 50%.
2. When prompted, accept the update notification.

   If you do not accept the update notification, the wireless handset will begin the update 20 seconds after the notification displays.

Update the Wireless Handset Software Manually

If the software update notification does not display on the wireless handset within five minutes of registering the wireless handset, you can check for configuration updates and manually update the software from the VVX host phone.

Procedure

1. Place the handset in the base station or charging cradle, and ensure the handset battery is charged to at least 50%.
2. On the VVX host phone, navigate to **Settings** > **Basic** > **Update Configuration**.

   If there is a software update available, the wireless handsets update sequentially with the first registered handset updating first.

Related Links

Pairing a VVX Phone with a VVX D60 Base Station on page 155

Configure VVX D60 Network Settings

By default, you can edit network settings for the VVX D60 base station.

You can use the Web Configuration Utility to make changes to the base station's network settings.

Procedure

1. In a web browser, enter `https://<IP address of D60 base station>`
2. In the Web Configuration Utility, enter the following default credentials:
   - User name: `Polycom`
   - Password: `456`
3. Navigate to **Settings** > **Network Settings**.
4. Update the desired network settings - IP settings, LLDP, CDP, VLAN, QOS, SNTP address and DNS.
5. Click **Submit.**

Parameters for VVX D60 Wireless Handsets

The following table lists the configuration parameters you need to configure the VVX D60 feature.

Configuring the VVX D60 Accessories

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
`features.cfg, dect.cfg`	`feature.dect.enabled`	0 (default) - Disables communication and pairing with the VVX D60 Wireless Handset and Base Station accessories. The VVX D60 menu options do not display. 1 - Enables communication and pairing with the VVX D60 Wireless Handset and Base Station accessories. The VVX D60 menu options display on the phone and in the Web Configuration Utility.	No
`dect.cfg`	`VVXD60.base.mac`	Specifies the VVX D60 Base Station MAC address from the provisioning server. NULL (default) string (maximum 12 alphanumeric characters)	No

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
`dect.cfg`	`VVXD60.Handset.X.outGoingLineIndex`	Controls the registration index that is used as the default line for outgoing calls placed on the wireless handset without selecting a line first. X refers to the wireless handset where X can be 1-5. 1 (default) 1 - 34	No
`dect.cfg`	`VVXD60.Handset.X.line.Y`	Sets the lines that will be accessible from the wireless handset where X is the wireless handset (1-5) and Y is the registered line on the VVX phone that will be mapped to the wireless handset. You can map up to five lines to a wireless handset. 0 (default) 0 to 34	No
`reg-advanced.cfg`	`reg.x.terminationType`	Determines the type of termination that is used for the line where the line can be managed automatically on the VVX, the wireless handset, or on both. X = each registration index. NULL (default) VVX, DECT, or VVX-DECT	No
`techsupport.cfg, dect.cfg`	`log.level.change.dect`	Sets the logging detail level for the VVX D60 accessory. 4 (default) 0 - 6	No

Template	Parameter Template	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.VVXD60.allowLineMappings	0 (default) - The Map Lines menu is available only as a password-protected option in the Administrator menu and administrators can map lines on VVX phones to the Polycom D60 handset. 1 - The Map Lines menu is available to administrators and to users on VVX phones at Menu > Settings > Features > VVX D60 Configuration to map lines on VVX phones to the Polycom D60 handset.	No
dect.cfg, features.cfg	feature.VVXD60.allowPairing	None (default) - Users are not allowed to pair or unpair a base station from the VVX phone. Pairing - Users are allowed to pair the base station with the VVX phone, but unpairing is not allowed. Unpairing - Users are allowed to unpair the base station from the phone, but pairing is not allowed. Both - Users are allowed to pair and unpair the base station with the VVX phone.	No
site.cfg	VVXD60.handset.maxCount	Const_NumHandSets (default) 1	No

Audio Features

Topics:

After you set up your Polycom phones on the network, users can send and receive calls using the default configuration.

However, you might consider configuring modifications that optimize the audio quality of your network.

This section describes the audio sound quality features and options you can configure for your Polycom phones. Use these features and options to optimize the conditions of your organization's phone network system.

Automatic Gain Control

Automatic Gain Control (AGC) is available for conference phone models and is used to boost the gain of the near-end conference participant.

Gain control helps conference participants hear your voice. This feature is enabled by default.

Background Noise Suppression

Background noise suppression is designed primarily for handsfree operation and reduces background noise, such as from fans, projectors, or air conditioners, to enhance communication.

This feature is enabled by default.

Comfort Noise

Comfort Noise ensures a consistent background noise level to provide a natural call experience and is enabled by default.

Comfort noise fill is unrelated to Comfort Noise packets generated if Voice Activity Detection is enabled.

Voice Activity Detection

Voice activity detection (VAD) conserves network bandwidth by detecting periods of silence in the transmit data path so the phone doesn't have to transmit unnecessary data packets for outgoing audio.

For compression algorithms without an inherent VAD function, such as G.711, the phone uses the codec-independent comfort noise transmission processing specified in RFC 3389. The RFC 3389 algorithm is derived from G.711 Appendix II, which defines a comfort noise (CN) payload format (or bit stream) for G. 711 use in packet-based, multimedia communication systems.

Voice Activity Detection Parameters

The following table lists the parameters you can use to configure Voice Activity Detection.

Voice Activity Detection Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.vad.signalAnnexB	0—There is no change to SDP. If voice.vadEnable is set to 0, add parameter line a=fmtp:18 annexb="no" below the a=rtpmap … parameter line (where “18” could be replaced by another payload). 1 (default)—Annex B is used and a new line is added to SDP depending on the setting of voice.vadEnable . If voice.vadEnable is set to 1, add parameter line a=fmtp:18 annexb="yes" below a=rtpmap … parameter line (where '18' could be replaced by another payload).	No
site.cfg	voice.vadEnable	0 - Disable Voice activity detection (VAD). 1 - Enable VAD.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.vadThresh	The threshold for determining what is active voice and what is background noise in dB. 25 (default) Integer from 0 - 30 Sounds louder than this value are considered active voice, and sounds quieter than this threshold are considered background noise. This does not apply to G.729AB codec operation which has its own built-in VAD function.	No

Comfort Noise Payload Packets

When enabled, the Comfort Noise payload type is negotiated in Session Description Protocol (SDP) with the default of 13 for 8 KHz codecs, and a configurable value between 96 and 127 for 16 KHz codecs.

Comfort Noise Payload Packets Parameters

The following table includes the parameters you can use to configure Comfort Noise payload packets.

Comfort Noise Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.CNControl	Publishes support for Comfort Noise in the SDP body of the INVITE message and includes the supported comfort noise payloads in the media line for audio. 1 (default)—Either the payload type 13 for 8 KHz sample rate audio codec is sent for Comfort Noise, or the dynamic payload type for 16 KHz audio codecs are sent in the SDP body. 0—Does not publish support or payloads for Comfort Noise.	No
site.cfg	voice.CN16KPayload	Alters the dynamic payload type used for Comfort Noise RTP packets for 16 KHz codecs. 96 to 127 122 (default)	No

Synthesized Call Progress Tones

Polycom phones play call signals and alerts, called call progress tones, that include busy signals, ringback sounds, and call waiting tones.

The built-in call progress tones match standard North American tones. If you want to customize the phone's call progress tones to match the standard tones in your region, contact Polycom Support.

Jitter Buffer and Packet Error Concealment

The phone employs a high-performance jitter buffer and packet error concealment system designed to mitigate packet inter-arrival jitter and out-of-order, or lost or delayed (by the network) packets.

The jitter buffer is adaptive and configurable for different network environments. When packets are lost, a concealment algorithm minimizes the resulting negative audio consequences. This feature is enabled by default.

Dual-Tone Multi-Frequency Tones

The phone generates dual-tone multi-frequency (DTMF) tones, also called touch tones, in response to user dialing on the dialpad.

These tones are transmitted in the real-time transport protocol (RTP) streams of connected calls.

The phone can encode the DTMF tones using the active voice codec or using RFC 2833-compatible encoding. The coding format decision is based on the capabilities of the remote endpoint. The phone generates RFC 2833 (DTMF only) events but does not regenerate—or otherwise use—DTMF events received from the remote end of the call.

DTMF Tone Parameters

The following table includes the parameters you can use to set up DTMF tones.

DTMF Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`reg.1.telephony`	1 (default) - Allows the phone to publish its capability in an SDP offer or answer to send and receive the DTMF tones over RFC-2833. 0 - Disables the phone's capability to send and receive the DTMF tones through RFC-2833 in an SDP offer or answer.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	tone.dtmf.chassis.masking	0 (default) - DTMF tones play through the speakerphone in handsfree mode. 1 - Set to 1 only if tone.dtmf.viaRtp is set to 0. DTMF tones are substituted with non-DTMF pacifier tones when dialing in handsfree mode to prevent tones from broadcasting to surrounding telephony devices or inadvertently transmitted in-band due to local acoustic echo.	Yes
sip-interop.cfg	tone.dtmf.level	The level of the high frequency component of the DTMF digit measured in dBm0; the low frequency tone is two dB lower. -15 -33 to 3	Yes
sip-interop.cfg	tone.dtmf.offTime	When a sequence of DTMF tones is played out automatically, specify the length of time in milliseconds the phone pauses between digits. This is also the minimum inter-digit time when dialing manually. 50 ms Positive integer	Yes
sip-interop.cfg	tone.dtmf.onTime	Set the time in milliseconds DTMF tones play on the network when DTMF tones play automatically. The time you set is also the minimum time the tone plays when manually dialing. 50 ms (default) 1 - 65535 ms	Yes
sip-interop.cfg	tone.dtmf.rfc2833Control	Specify if the phone uses RFC 2833 to encode DTMF tones. 1 (default) - The phone indicates a preference for encoding DTMF through RFC 2833 format in its Session Description Protocol (SDP) offers by showing support for the phone-event payload type. This does not affect SDP answers and always honor the DTMF format present in the offer.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	tone.dtmf.rfc2833Payload	Specify the phone-event payload encoding in the dynamic range to be used in SDP offers. Skype (default) - 101 Generic (default) -127 96 to 127	Yes
sip-interop.cfg	tone.dtmf.rfc2833Payload_OPUS	Sets the DTMF payload required to use Opus codec. 126 (default) 96 - 127	Yes
sip-interop.cfg	tone.dtmf.viaRtp	1 (default) - Encode DTMF in the active RTP stream. Otherwise, DTMF may be encoded within the signaling protocol only when the protocol offers the option. If you set this parameter to 0, you must set tone.dtmf.chassis.masking to 1.	Yes
sip-interop.cfg	tone.localDtmf.onTime	Set the time in milliseconds DTMF tones play for when the phone plays out a DTML tone sequence automatically. 50 ms (default) 1 - 65535 ms	No

Acoustic Echo Cancellation

Polycom phones use advanced acoustic echo cancellation (AEC) for handsfree operation using the speakerphone.

The phones significantly reduce echo while permitting natural communication.

You can configure the Acoustic Echo Cancellation (AEC) feature to remove the echo of the local loudspeaker from the local microphone without removing the near-end speech.

The AEC feature includes the following:

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

The phones also support headset echo cancellation.

Acoustic Echo Cancellation Parameters

The following table includes the parameters you can use to set up Acoustic Echo Cancellation.

Acoustic Echo Cancel (AEC) Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`voice.aec.hf.enable`	1 (default)—Enables the AEC function for handsfree options. 0—Disables the AEC function for handsfree options. Polycom does not recommend disabling this parameter.	No
`site.cfg`	`voice.aec.hs.enable`	0—Disables the AEC function for the handset. 1 (default)—Enables the AEC function for the handset.	No
`debug.cfg`	`voice.aes.hf.duplexBalance`	0 - Max Echo Control (default) - Balances the Acoustic Echo Suppression to maximize the echo control, allowing the near-end and far-end users to speak simultaneously with minimal full duplex in handsfree mode. 1 - Max Full Duplex: Balances the Acoustic Echo Suppression to maximize full duplex. This makes the phone handsfree more susceptible to echo during continuous double-talk or when moving the phone or objects near the phone.	No

Context-Sensitive Volume Control

In some countries, regulations state that a phone's receiver volume must be reset to a nominal level for each new call.

Transmit levels are fixed according to the TIA/EIA-810-A standard. The next table lists parameters that configure the receiver volume to reset and persist across calls each time a user makes changes to the default volume level. You can adjust the volume of phone sound effects—such as the ringer and the volume of receiving call audio—separately for the speakerphone, handset, and headset.

Context Sensitive Volume Control Parameters

The following table includes the parameters you can use to configure Context Sensitive Volume Control.

Context Sensitive Volume Control Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`voice.volume.persist.bluetooth.headset`	0 (default) - The Bluetooth headset volume does not persist between calls and resets to a nominal level each new call. 1 - The volume for each call is the same as the previous call.	No
`site.cfg`	`voice.volume.persist.handset`	Specifies whether the handset's volume level persists and stays at the same level between calls. 0 (default) - The handset volume automatically resets to a nominal level after each call. 1 - The volume for each call is the same as the previous call.	No
`site.cfg`	`voice.volume.persist.handsfree`	Specify if the speakerphone volume persists between calls. 1 (default) - The speakerphone volume at the end of a call persists between calls. 0 - The speakerphone volume does not persist between calls and resets to a nominal level each new call.	No
`site.cfg`	`voice.volume.persist.usb.handsfree`	Specifies if a USB headset should be used for every call 0 (default)- Does not use USB headset automatically for calls. 1- Uses the USB headset automatically for all calls.	No
`site.cfg`	`voice.volume.persist.usbHeadset`	0 (default) – The USB headset volume does not persist between calls and resets to a nominal level each new call. 1 - The USB headset volume at the end of a call persists between calls.	No

Polycom Acoustic Fence

Available on all VVX phones, the Polycom Acoustic Fence feature improves background noise suppression when you are using the phone handset or a headset connected to the headset port.

This feature is particularly useful in call center environments where background noise can impact far-end audio quality.

Only headsets connected using an RJ-9 port at the rear of the phone (the headset port) can make use of the Acoustic Fence.

The Acoustic Fence is not available for USB or Bluetooth headset use.

Acoustic Fence Parameters

The following table includes the noise suppression parameters you can use to configure Polycom Acoustic Fence.

Acoustic Fence Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.acousticFenceUI.enabled`	0 (default) - Hide display of the Acoustic Fence Configuration setting on the phone. 1 - Displays the Acoustic Fence Configuration setting on the phone.	No
`features.cfg`	`voice.ns.hd.enable`	0 (default) — Disables noise suppression for headsets. 1 — Enables noise suppression for headsets.	No
`features.cfg`	`voice.ns.hd.enhanced`	The parameter `voice.ns.hd.enable` must be set to 1 to use this parameter. 0 (default) - Disables Acoustic Fence noise suppression for headsets. 1 - Enables Acoustic Fence noise suppression for headsets.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`voice.ns.hd.nonStationaryThresh`	Sets the Acoustic Fence noise suppression threshold for headsets. A lower value allows more background sound to enter, and a higher value suppresses background noise. 1 to 10 8 (default) High values can suppress the speaker's voice and impact far-end audio quality.	No
`features.cfg`	`voice.ns.hs.enable`	0 (default) - Disables noise suppression for handsets. 1 - Enables noise suppression for handsets.	No
`features.cfg`	`voice.ns.hs.enhanced`	The parameter `voice.ns.hs.enable` must be set to 1 to use this parameter. 1 (default) - Enables Acoustic Fence noise suppression for handsets. 0 - Disables Acoustic Fence noise suppression for handsets.	No
`features.cfg`	`voice.ns.hs.nonStationaryThresh`	Sets the Acoustic Fence noise suppression threshold for handsets. A lower value allows more background sound to enter, and a higher value suppresses background noise. 1 to 10 8 (default) High values can suppress the speaker's voice and impact far-end audio quality.	No

Bluetooth Device Support

You can enable VVX 600 and 601 business media phones to pair and connect with Bluetooth devices such as smartphones and headsets to handle audio calls.

By default, this feature is disabled. After you enable this feature, the user can either pair and connect a smartphone or Bluetooth headset to your VVX business media phone. Users can also manage calls and enter DTMF digits from the VVX phone by setting the phone as the audio device for their Bluetooth device.

Note that using a Bluetooth headset can affect voice quality on the phone due to inherent limitations with Bluetooth technology. Users may not experience the highest voice quality when using a Bluetooth headset while the 2.4 GHz band is enabled or while they are in an environment with many other Bluetooth devices.

Bluetooth Device Support Parameters

The following table lists the parameters you can use to configure Bluetooth devices like headset and smartphone features.

Bluetooth Device Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.bluetooth.enabled`	For high security environments. 1 (default)- The Bluetooth feature is enabled. 0 - The Bluetooth feature is disabled.	No
`features.cfg`	`bluetooth.radioOn`	0 (default)—Turns off the Bluetooth radio. 1—Turns on the Bluetooth radio to enable other devices to detect and connect to the device over Bluetooth.	No
`features.cfg`	`bluetooth.device.discoverable`	Specify the discovery mode to make VVX 600/601 phones visible to other Bluetooth devices. 1 (Default) - Enable the discoverable mode 0 - Disable the discoverable mode.	No
`features.cfg`	`bluetooth.device.name`	Sets the Bluetooth device name for VVX phones to identify with the same name on the smartphone. String (default) 1 – Minimum 20 – Maximum	No

Location of Audio Alerts

You can choose where all audio alerts, including incoming call alerts, are played on Polycom phones.

You can specify the audio to play from the handsfree speakerphone (default), the handset, the headset, or the active location. If you choose the active location, audio alerts play out through the handset or headset if they are in use. Otherwise, alerts play through the speakerphone.

Audio Alert Parameters

Use the parameters in the following table to specify where audio alerts and sound effects play.

Audio Alert and Sound Effect Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`techsupport.cfg`	`se.appLocalEnabled`	1 (default)—Enable audio alerts and sound effects. 0—Disable audio alerts and sound effects	Yes
`reg-advanced.cfg`	`se.destination`	Set where alerts and sound effects play out. chassis (default) —Alerts and sound effects play out through the phone's speakerphone. headset (if connected) handset active —Alerts play from the destination that is currently in use. For example, if a user is in a call on the handset, a new incoming call rings on the handset.	No
`site.cfg`	`se.stutterOnVoiceMail`	1 (default)—A stuttered dial tone is used instead of a normal dial tone to indicate that one or more voicemail messages are waiting at the message center. 0—A normal tone is used to indicate that one or more voicemail messages are waiting at the message center.	No

Ringtones

Ringtones are used to define a simple ring class that is applied based on credentials carried within the network protocol.

The ring class includes parameters such as call-waiting and ringer index, if appropriate.

The ring class can use one of the following types of rings:

- Ring   Plays a specified ring pattern or call waiting indication.
- Visual   Provides a visual indication (no audio) of an incoming call, no ringer needs to be specified.
- Answer   Provides auto-answer on an incoming call.
- Ring-answer   Provides auto-answer on an incoming call after a certain number of rings.

Note: that auto-answer for an incoming call works only when there is no other call in progress on the phone, including no other calls in progress on shared or monitored lines. However, if a phone initiates a call on a shared or monitored line, auto-answer works.

Supported Ring Classes

The phone supports the following ring classes:

- default
- visual
- answerMute
- autoAnswer
- ringAnswerMute
- ringAutoAnswer
- internal
- external
- emergency
- precedence
- splash
- custom<y> where y is 1 to 17.

Ringtone Parameters

The following parameters configure ringtones.

Ringtone Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`se.rt.enabled`	0—The ringtone feature is not enabled. 1 (default)—The ringtone feature is enabled.	No
`reg-advanced.cfg`	`se.rt.modification.enabled`	Determines whether or not users are allowed to modify the pre-defined ringtone from the phone's user interface. 0 1 (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	se.rt.<ringClass>.callWait	The call waiting tone used for the specified ring class. The call waiting pattern should match the pattern defined in Call Progress Tones. callWaiting (default) callWaitingLong precedenceCallWaiting	No
sip-interop.cfg	se.rt.<ringClass>.name	The answer mode for a ringtone, which is used for to identify the ringtone in the user interface. UTF-8 encoded string	No
sip-interop.cfg	se.rt.<ringClass>.ringer	The ringtone used for this ring class. The ringer must match one listed in Ringtones. default ringer1 to ringer24 ringer2 (default)	No
sip-interop.cfg	se.rt.<ringClass>.timeout	The duration of the ring in milliseconds before the call is auto answered, which only applies if the type is set to ring-answer. 1 to 60000 2000 (default	No
sip-interop.cfg	se.rt.<ringClass>.type	The answer mode for a ringtone. ring visual answer ring-answer	No

Related Links
Distinctive Ringtones on page 180

Distinctive Ringtones

This feature enables you to apply a distinctive ringtone to a registered line, a specific contact, or type of call, including internal or external calls.

You can set up distinctive ringing using more than one of the following methods, however, the phone uses the highest priority method:

- Assign ringtones to specific contacts in the contact directory. This option is the first and highest in priority.
- Use the voIpProt.SIP.alertInfo.x.value and voIpProt.SIP.alertInfo.x.class parameters in the sip-interop.cfg template to map calls to specific ringtones. The value you enter depends on the call server. This option requires server support and is second in priority.
- Users can select a ringtone for each registered line on the phone from the phone menu. This option has the lowest priority.

Note: You can use the SIP alert-info header to delay the auto-answer feature. If you set **delay=0** in the SIP.alert-Info header, the phone immediately auto-answers incoming calls without ringing. If you set **delay=x** where x=time in seconds, the phone rings for that duration of time before auto-answering incoming calls.

Related Links

Call Progress Tones on page 187

Ringtone Parameters on page 179

Distinctive Ringtone Parameters

The following table includes the parameters you can use to configure distinctive ringtones for a line, contact, or type of call.

Distinctive Ringtone Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.alertInfo.x.class	Alert-Info fields from INVITE requests are compared to parameters as specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class is applied. default (default) See the list of ring classes in Ringtone Parameters.	No
sip-interop.cfg	voIpProt.SIP.alertInfo.x.value	Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE. NULL (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.ringType`	The ringer to be used for calls received by this registration. The default is the first non-silent ringer. If you use the configuration parameters ringer13 and ringer14 on a single registered line, the phone plays SystemRing.wav. default (default) ringer1 to ringer24	No

Ringtone Patterns

The following table lists the ring pattern names and their default descriptions.

Note that sampled audio files 1 to 10 listed in the table all use the same built-in file unless that file has been replaced with a downloaded file.

Sampled audio files 1 to 10 all use the same built-in file unless that file has been replaced with a downloaded file.

Ringtone Pattern Names

Parameter Name	Ringtone Name	Description
ringer1	Silent Ring	Silent ring Note: Silent ring provides a visual indication of an incoming call, but no audio indication.
ringer2	Low Trill	Long single A3 Db3 major warble
ringer3	Low Double Trill	Short double A3 Db3 major warble
ringer4	Medium Trill	Long single C3 E3 major warble
ringer5	Medium Double Trill	Short double C3 E3 major warble
ringer6	High Trill	Long single warble 1
ringer7	High Double Trill	Short double warble 1
ringer8	Highest Trill	Long single Gb3 A4 major warble
ringer9	Highest Double Trill	Short double Gb3 A4 major warble
ringer10	Beeble	Short double E3 major
ringer11	Triplet	Short triple C3 E3 G3 major ramp

Parameter Name	Ringtone Name	Description
ringer12	Ringback-style	Short double ringback
ringer13	Low Trill Precedence	Long single A3 Db3 major warble Precedence
ringer14	Ring Splash	Splash
ringer15	-	Sampled audio file 1
ringer16	-	Sampled audio file 2
ringer17	-	Sampled audio file 3
ringer18	-	Sampled audio file 4
ringer19	-	Sampled audio file 5
ringer20	-	Sampled audio file 6
ringer21	-	Sampled audio file 7
ringer22	-	Sampled audio file 8
ringer23	-	Sampled audio file 9
ringer24	-	Sampled audio file 10

Sound Effects

The phone uses built-in sampled audio files (SAF) in wave file format for some sound effects.

You can customize the audio sound effects that play for incoming calls and other alerts using synthesized tones or sampled audio files with .wav files you download from the provisioning server or Internet.

Ringtone files are stored in volatile memory which allows a maximum size of 600 kilobytes (614400 bytes) for all ringtones.

Sampled Audio Files

The phone uses built-in sampled audio files (SAF) in wave file format for some sound effects.

You can add files downloaded from the provisioning server or from the Internet. Ringtone files are stored in volatile memory, which allows a maximum size of 600 kilobytes (614400 bytes) for all ringtones.

The phones support the following sampled audio WAVE (.wav) file formats:

- mono 8 kHz G.711 u-Law—Supported on all phones
- mono G.711 (13-bit dynamic range, 8-khz sample rate)
- G.711 A-Law—Supported on all phones
- mono L16/8000 (16-bit dynamic range, 8-kHz sample rate)—Supported on all phones
- mono 8 kHz A-law/mu-law—Supported on all phones

- L8/16000 (16-bit, 8 kHz sampling rate, mono)—Supported on all phones
- mono L16/16000 (16-bit dynamic range, 16-kHz sample rate)
- L16/16000 (16-bit, 16 kHz sampling rate, mono)—Supported on all phones
- L16/32000 (16-bit, 32 kHz sampling rate, mono)—Supported on VVX 500/501, 600/601, and 1500
- L16/44100 (16-bit, 44.1 kHz sampling rate, mono)—Supported on VVX 500/501, 600/601, and 1500
- L16/48000 (16-bit, 48 kHz sampling rate, mono)—Supported on VVX 500/501, 600/601, and 1500

Default Sample Audio Files

The following table defines the phone's default use of the sampled audio files.

Default Sample Audio File Usage

Sampled Audio File Number	Default Use (Pattern Reference)
1	Ringer 12 (`se.pat.misc.welcome`) Ringer 15 (`se.pat.ringer.ringer15`)
2	Ringer 16 (`se.pat.ringer.ringer16`)
3	Ringer 17 (`se.pat.ringer.ringer17`)
4	Ringer 18 (`se.pat.ringer.ringer18`)
5	Ringer 19 (`se.pat.ringer.ringer19`)
6	Ringer 20 (`se.pat.ringer.ringer20`)
7	Ringer 21 (`se.pat.ringer.ringer21`)
8	Ringer 22 (`se.pat.ringer.ringer22`)
9	Ringer 23 (`se.pat.ringer.ringer23`)
10	Ringer 24 (`se.pat.ringer.ringer24`)
11 to 24	Not Used

Sampled Audio File Parameters

Your custom sampled audio files must be available at the path or URL specified in the parameter `saf.x` so the phone can download the files. Make sure to include the name of the file and the .wav extension in the path.

Use the parameters in the following tables to customize this feature.

In the following table, x is the sampled audio file number.

Sample Audio File Parameter

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	saf.x	Specify a path or URL for the phone to download a custom audio file. Null (default)—The phone uses a built-in file. Path Name —During startup, the phone attempts to download the file at the specified path in the provisioning server. URL— During startup, the phone attempts to download the file from the specified URL on the Internet. Must be a RFC 1738-compliant URL to an HTTP, FTP, or TFTP wave file resource. If using TFTP, the URL must be in the following format: `tftp://<host>/[pathname]<filename>` . For example: `tftp://somehost.example.com/sounds/example.wav` . To use a welcome sound, enable the parameter `up.welcomeSoundEnabled` and specify a file in `saf.x` . The default UC Software welcome sound file is `Welcome.wav` .	No

Sound Effect Patterns

You can specify the sound effects that play for different phone functions and specify the sound effect patterns and the category.

Sound effects are defined by patterns: sequences of chord-sets, silence periods, and wave files. You can also configure sound effect patterns and ringtones. The phones use both synthesized and sampled audio sound effects.

Patterns use a simple script language that allows different chord sets or wave files to be strung together with periods of silence. The script language uses the instructions shown in the next table.

Sound Effects Pattern Types

Instruction	Meaning	Example
sampled (n)	Play sampled audio file n	`se.pat.misc.SAMPLED_1.inst.1.type =”sampled”` (sampled audio file instruction type) `se.pat.misc.SAMPLED_1.inst.1.value =”2”` (specifies sampled audio file 2)

Instruction	Meaning	Example
chord (n, d)	Play chord set n (d is optional and allows the chord set ON duration to be overridden to d milliseconds)	`se.pat.callProg.busyTone.inst.2.type = “chord”` (chord set instruction type) `se.pat.callProg.busyTone.inst.2.value = “busyTone”` (specifies sampled audio file busyTone) `se.pat.callProg.busyTone.inst.2.param = “2000”` (override ON duration of chord set to 2000 milliseconds)
silence (d)	Play silence for d milliseconds (Rx audio is not muted)	`se.pat.callProg.bargeIn.inst.3.type = “silence”` (silence instruction type) `se.pat.callProg.bargeIn.inst.3.value = “300”` (specifies silence is to last 300 milliseconds)
branch (n)	Advance n instructions and execute that instruction (n must be negative and must not branch beyond the first instruction)	`se.pat.callProg.alerting.inst.4.type = “branch”` (branch instruction type) `se.pat.callProg.alerting.inst.4.value = “-2”` (step back 2 instructions and execute that instruction)

Sound Effect Pattern Parameters

There are three categories of sound effect patterns that you can use to replace `cat` in the parameter names: `callProg` (Call Progress Patterns), `ringer` (Ringer Patterns) and `misc` (Miscellaneous Patterns).

Keep the following in mind when using the parameters in the following table:

- Xis the pattern name.
- Y is the instruction number.
- Both x and y need to be sequential.
- `Cat` is the sound effect pattern category.

Sound Effects Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`region.cfg`	`se.pat.callProg.secondaryDialTone.name`	1-255	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`region.cfg`	`se.pat.callProg.secondaryDialTone.inst.1.type`	0-255	No
`region.cfg`	`se.pat.callProg.secondaryDialTone.inst.1.value`	0-50	No
`region.cfg`	`se.pat.callProg.secondaryDialTone.inst.1.atte`	Sound effects name, where cat is `callProg` , `ringer` , or `misc` . UTF-8 encoded string	No
`region.cfg`	`se.pat.cat.x.inst.y.type`	Sound effects name, where cat is `callProg` , `ringer` , or `misc` . sample chord silence branch	No
`region.cfg`	`se.pat.cat.x.inst.y.value`	The instruction: `sampled` - sampled audio file number, `chord` - type of sound effect, `silence` - silence duration in ms, `branch` - number of instructions to advance. String	No

Call Progress Tones

The following table lists the call progress pattern names and their descriptions.

Call Progress Tone Pattern Names

Call Progress Pattern	Description
alerting	Alerting
bargeIn	Barge-in tone
busyTone	Busy tone
callWaiting	Call waiting tone

Call Progress Pattern	Description
callWaitingLong	Call waiting tone long (distinctive)
callWaitingRingback	Call Waiting RingBack Tone
confirmation	Confirmation tone
dialTone	Dial tone
howler	Howler tone (off-hook warning)
intercom	Intercom announcement tone
msgWaiting	Message waiting tone
precedenceCallWaiting	Precedence call waiting tone
precedenceRingback	Precedence ringback tone
preemption	Preemption tone
precedence	Precedence tone
recWarning	Record warning
reorder	Reorder tone
ringback	Ringback tone
secondaryDialTone	Secondary dial tone
stutter	Stuttered dial tone

Related Links

Distinctive Ringtones on page 180

Miscellaneous Patterns

The following table lists the miscellaneous patterns and their descriptions.

Miscellaneous Pattern Names

Parameter Name	Miscellaneous Pattern Name	Description
instantmessage	instant message	New instant message
localHoldNotification	local hold notification	Local hold notification
messageWaiting	message waiting	New message waiting indication
negativeConfirm	negative confirmation	Negative confirmation

Parameter Name	Miscellaneous Pattern Name	Description
positiveConfirm	positive confirmation	Positive confirmation
remoteHoldNotification	remote hold notification	Remote hold notification
welcome	welcome	Welcome (boot up)

Supported Audio Codecs

The following table details the supported audio codecs and priorities for Polycom phone models.

Note the following limitations when using the Opus codec:

- VVX 301, 311, 401, 411, 500, 501, 600, and 601 business media phones support a single Opus stream. Users can establish only one call at a time when using the Opus codec on these phones.
- VVX 150 business IP phone does not support Opus codec.
- VVX 250, 350, and 450 business IP phones support a single Opus stream. Users can establish only one call at a time when using the Opus codec on these phones.
- VVX 500 and 600 do not support video when using Opus.
- VVX 500 and 600 do not support local conferences when using Opus.
- Opus is not compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC are not published; if you set G.729 and iLBC to the highest priority, Opus is not published.

Note: On the VVX 500/501 and 600/601, when you enable video, the G.722.1C codec is disabled.

Audio Codec Priority

Phone	Supported Audio Codecs	Priority
VVX 101 VVX 201 VVX 150	G.711µ-law	6
	G.711a-law	7
	G.722	4
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0

Phone	Supported Audio Codecs	Priority
VVX 300/301, 310/311, 400/401, 410/411 VVX 250, 350, 450 * Note: VVX 301, 311, 401, 411 support a single Opus stream. VVX 300, 310, 400, 410 do not support Opus.	G.711µ-law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Opus*	0
	Siren 7	0
VVX 500/501, 600/601 ◦ VVX 500 and 600 support a single Opus stream. ◦ VVX 500 and 600 do not support both Opus and video.	G.711 µ -law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	Opus*	0
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Siren 7	0
VVX 1500	G.711 µ -law	6
	G.711a-law	7
	G.719 (64kbps)	0
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	Siren14 (48kbps)	3
	iLBC (13.33kbps, 15.2kbps)	0, 0

Phone	Supported Audio Codecs	Priority
SoundStructure VoIP Interface ◦ SoundStructure VoIP Interface supports a single Opus stream. ◦ SoundStructure VoIP Interface does not support both Opus and video.	G.711 µ -law	6
	G.711a-law	7
	G.722	4
	G.722.1 (24kbps, 32kbps)	5
	G.722.1C (48kbps)	2
	G.729AB	8
	iLBC (13.33kbps, 15.2kbps)	0, 0
	Siren 7	0

Supported Audio Codec Specifications

The following table summarizes the specifications for audio codecs supported on Polycom phones.

Audio Codec Specifications

Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
G.711 µ -law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
G.711 a-law	RFC 1890	64 Kbps	80 Kbps	8 Ksps	20 ms	3.5 KHz
G.719	RFC 5404	32 Kbps48 Kbps64 Kbps	48 Kbps64 Kbps80 Kbps	48 Ksps	20 ms	20 KHz
G.711	RFC 1890	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz
G.722[1]	RFC 3551	64 Kbps	80 Kbps	16 Ksps	20 ms	7 KHz
G.722.1	RFC 3047	24 Kbps32 Kbps	40 Kbps48 Kbps	16 Ksps	20 ms	7 KHz
G.722.1C	G7221C	224 Kbps32 Kbps48 Kbps	40 Kbps48 Kbps64 Kbps	32 Ksps	20 ms	14 KHz
G.729AB	RFC 1890	8 Kbps	24 Kbps	8 Ksps	20 ms	3.5 KHz
Opus	RFC 6716	8 - 24 Kbps	24 - 40 Kbps	8 Ksps 16 Ksps	20 ms	3.5 KHz 7 KHz

Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
Lin16	RFC 1890	128 Kbps256 Kbps512 Kbps705.6 Kbps768 Kbps	132 Kbps260 Kbps516 Kbps709.6 Kbps772 Kbps	8 Ksps16 Ksps32 Ksps44.1 Ksps48 Ksps	10 ms	3.5 KHz7 KHz14 KHz20 KHz22 KHz
Siren 7	SIREN7	16 Kbps24 Kbps32 Kbps	32 Kbps40 Kbps48 Kbps	16 Ksps	20 ms	7 KHz
Siren14	SIREN14	24 Kbps32 Kbps48 Kbps	40 Kbps48 Kbps64 Kbps	32 Ksps	20 ms	14 KHz
Siren22	SIREN22	32 Kbps48 Kbps64 Kbps	48 Kbps64 Kbps80 Kbps	48 Ksps	20 ms	22 KHz
iLBC	RFC 3951	13.33 Kbps15.2 Kbps	31.2 Kbps24 Kbps	8 Ksps	30 ms20 ms	3.5 KHz
SILK	SILK	Skype SILK	6 - 20 Kbps	36 Kbps	8 Ksps	3.5 KHz
			7 - 25 Kbps	41 Kbps	12 Ksps	5.2 KHz
			8 - 30 Kbps	46 Kbps	16 Ksps	7 KHz
			12 - 40 Kbps	56 Kbps	24 Ksps	11 KHz

[1] Per RFC 3551. Even though the actual sampling rate for G.722 audio is 16,000 Hz (16ksps), the RTP clock rate advertised for the G.722 payload format is 8,000 Hz because that value was erroneously assigned in RFC 1890 and must remain unchanged for backward compatibility.

Note: The network bandwidth necessary to send the encoded voice is typically 5-10% higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48kbps for both the receive and transmit signals consumes about 100kbps of network bandwidth (two-way audio).

Audio Codec Parameters

You can configure a set of codec properties to improve consistency and reduce workload on the phones.

Use the parameters in the following table to specify the priority for audio codecs on your Polycom phones. If 0 or Null, the codec is disabled. A value of 1 is the highest priority.

If a phone does not support a codec, it treats the setting as if it were 0 and not offer or accept calls with that codec. The phone ignores the unsupported codec and continues to the codec next in priority. For example, using the default values, the VVX 310 doesn't support G.722.1C or G.719 and uses G.722.1 as the highest-priority codec.

Audio Codec Parameters

Template	Parameter	Permitted Value	Default	Change Causes Restart or Reboot
site.cfg	voice.codecPref.G711_A	0 to 27	7	No
site.cfg	voice.codecPref.G711_Mu	0 to 27	6	No
site.cfg	voice.codecPref.G719.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.G719.48kbps	0 to 27	0	No
site.cfg	voice.codecPref.G719.64kbps	0 to 27	0	No
site.cfg	voice.codecPref.G722	0 to 27	4	No
site.cfg	voice.codecPref.G7221.24kbps	0 to 27	0	No
site.cfg	voice.codecPref.G7221.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.G7221_C.24kbps	0 to 27	5	No
site.cfg	voice.codecPref.G7221_C.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.G7221_C.48kbps	0 to 27	2	No
site.cfg	voice.codecPref.G729_AB	0 to 27	8	No
site.cfg	voice.codecPref.iLBC.13_33kbps	0 to 27	0	No
site.cfg	voice.codecPref.iLBC.15_2kbps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.8ksps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.16ksps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.32ksps	0 to 27	0	No

Template	Parameter	Permitted Value	Default	Change Causes Restart or Reboot
site.cfg	voice.codecPref.Lin16.44_1ksps	0 to 27	0	No
site.cfg	voice.codecPref.Lin16.48ksps	0 to 27	0	No
site.cfg	voice.codecPref.Siren7.16kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren7.24kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren7.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren14.24kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren14.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren14.48kbps	0 to 27	3	No
site.cfg	voice.codecPref.Siren22.32kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren22.48kbps	0 to 27	0	No
site.cfg	voice.codecPref.Siren22.64kbps	0 to 27	1	No
site.cfg	voice.codecPref.SILK.8ksps	0 to 27	0	No
site.cfg	voice.codecPref.SILK.12ksps	0 to 27	0	No
site.cfg	voice.codecPref.SILK.16ksps	0 to 27	0	No
site.cfg	voice.codecPref.SILK.24ksps	0 to 27	0	No

SILK Audio Codec

Polycom VVX 501 and 601 business media phones support the SILK audio codec.

SILK Audio Codec Parameters

Use the following parameters to configure the SILK audio codec.

SILK Audio Codec Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.codecPref.SILK.8ksps	Set the SILK audio codec preference for the supported codec sample rates. 0 (default)	No
site.cfg	voice.codecPref.SILK.12ksps	Set the SILK audio codec preference for the supported codec sample rates.	No
site.cfg	voice.codecPref.SILK.16ksps	Set the SILK audio codec preference for the supported codec sample rates. 0 (default)	No
site.cfg	voice.codecPref.SILK.24ksps	Set the SILK audio codec preference for the supported codec sample rates. 0 (default)	No
site.cfg	voice.audioProfile.SILK.8ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 20 kbps (default) 6 – 20 kbps	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.audioProfile.SILK.12ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 25 kbps (default) 7 – 25 kbps	No
site.cfg	voice.audioProfile.SILK.16ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 30 kbps (default) 8 – 30 kbps	No
site.cfg	voice.audioProfile.SILK.24ksps.encMaxAvgBitrateKbps	Set the maximum average encoder output bitrate in kilobits per second (kpbs/s) for the supported SILK sample rate. 40 kbps (default) 12 – 40 kbps	No
site.cfg	voice.audioProfile.SILK.encComplexity	Specify the SILK encoder complexity. The higher the number the more complex the encoding allowed. 2 (default) 0-2	No
site.cfg	voice.audioProfile.SILK.encDTXEnable	0 (default) – Disable Enable Discontinuous transmission (DTX). 1 – Enable DTX in the SILK encoder. Note that DTX reduces the encoder bitrate to 0bps during silence.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.audioProfile.SILK.encExpectedPktLossPercent	Set the SILK encoder expected network packet loss percentage. A non-zero setting allows less inter-frame dependency to be encoded into the bitstream, resulting in increasingly larger bitrates but with an average bitrate less than that configured with voice.audioProfile.SILK.*. 0 (default) 0-100	No
site.cfg	voice.audioProfile.SILK.encInbandFECEnable	0 (default) - Disable inband Forward Error Correction (FEC) in the SILK encoder. 1 - Enable inband FEC in the SILK encoder. A non-zero value here causes perceptually important speech information to be sent twice: once in the normal bitstream and again at a lower bitrate in later packets, resulting in an increased bitrate.	No
site.cfg	voice.audioProfile.SILK.MaxPTime	Specify the maximum SILK packet duration in milliseconds (ms). 20 ms	No
site.cfg	voice.audioProfile.SILK.MinPTime	Specify the minimum SILK packet duration in milliseconds (ms). 20 ms	No
site.cfg	voice.audioProfile.SILK.pTime	The recommended received SILK packet duration in milliseconds (ms). 20 ms	No

IEEE 802.1p/Q

The phone tags all Ethernet packets it transmits with an 802.

1Q VLAN header when the following occurs:

- A valid VLAN ID is specified in the phone's network configuration.
- The phone is instructed to tag packets through Cisco Discovery Protocol (CDP) running on a connected Ethernet switch.
- A VLAN ID is obtained from DHCP or LLDP

EEE 802.1p/Q Parameters

Use the following table to set values for IEEE 802.

1p/Q parameters. You can configure the user_priority specifically for RTP and call control packets, such as SIP signaling packets, with default settings configurable for all other packets.

The phone tags all Ethernet packets it transmits with an 802.1Q VLAN header when the following occurs:

- A valid VLAN ID specified in the phone's network configuration.
- The phone is instructed to tag packets through Cisco Discovery Protocol (CDP) running on a connected Ethernet switch.
- A VLAN ID is obtained from DHCP or CDP.

IEEE 802.1p/Q Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`qos.ethernet.other.user_priority`	Set user priority for packets without a per-protocol setting. 2 (Default) 0 - 7	No
`site.cfg`	`qos.ethernet.rtp.video.user_priority`	Set user-priority used for Video RTP packets. 5 (Default) 0 - 7	No
`site.cfg`	`qos.ethernet.rtp.user_priority`	Choose the priority of voice Real-Time Protocol (RTP) packets. 5 (Default) 0 - 7	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	qos.ethernet.callControl.user_priority	Set the user-priority used for call control packets. 5 (Default) 0 - 7	No

Voice Quality Monitoring (VQMon)

You can configure the phones to generate various quality metrics that you can use to monitor sound and listening quality.

These metrics can be sent between the phones in RTCP XR packets, which are compliant with RFC 3611—RTP Control Extended Reports (RTCP XR). The packets are sent to a report collector as specified in draft RFC Session initiation Protocol Package for Voice Quality Reporting Event. The metrics can also be sent as `SIP PUBLISH` messages to a central voice quality report collector.

You can use Real Time Control Protocol Extended Report (RTCP XR) to report voice quality metrics to remote endpoints. This feature supports RFC6035 compliance as well as draft implementation for voice quality reporting.

You need a license key to activate the VQMon feature on the VVX 300/301, 310/311, 400/401, and 410/411 business media phones and VVX business IP phones: 150, 250, 350, 450.

This feature is available for open SIP environments, but is not available with Skype for Business Server. For more information on VQMon, contact your Certified Polycom Reseller.

VQMon Reports

You can enable three types of voice quality reports:

- Alert—Generated when the call quality degrades below a configurable threshold.
- Periodic—Generated during a call at a configurable period.
- Session—Generated at the end of a call.

You can generate a wide range of performance metrics using the parameters shown in the following table. Some are based on current values, such as jitter buffer nominal delay and round trip delay, while others cover the time period from the beginning of the call until the report is sent, such as network packet loss. Some metrics are generated using other metrics as input, such as listening Mean Opinion Score (MOS), conversational MOS, listening R-factor, and conversational R-factor.

VQMon Parameters

All of the parameters that configure Voice Quality Monitoring in the following table are located in the `features.cfg` template.

Voice Quality Monitoring Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`voice.qualityMonitoring.collector.alert.moslq.threshold.critical`	Specify the threshold value of listening MOS score (MOS-LQ) that causes the phone to send a critical alert quality report. Configure the desired MOS value multiplied by 10. For example, a value of 28 corresponds to the MOS score 2.8. 0 (default) - Critical alerts are not generated due to MOS-LQ. 0 - 40	Yes
`features.cfg`	`voice.qualityMonitoring.collector.alert.moslq.threshold.warning`	Specify the threshold value of listening MOS score (MOS-LQ) that causes phone to send a warning alert quality report. Configure the desired MOS value multiplied by 10. For example, a configured value of 35 corresponds to the MOS score 3.5. 0 (default) - Warning alerts are not generated due to MOS-LQ. 0 - 40	Yes
`features.cfg`	`voice.qualityMonitoring.collector.alert.delay.threshold.critical`	Specify the threshold value of one way-delay (in milliseconds) that causes the phone to send a critical alert quality report. One-way delay includes both network delay and end system delay. 0 (default) - Critical alerts are not generated due to one-way delay. 0 - 2000 ms	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	voice.qualityMonitoring.collector.alert.delay.threshold.warning	Specify the threshold value of one-way delay (in milliseconds) that causes the phone to send a critical alert quality report. One-way delay includes both network delay and end system delay. 0 (default) - Warning alerts are not generated due to one-way delay. 0 - 2000 ms	Yes
features.cfg	voice.qualityMonitoring.collector.enable.periodic	0 (default) - Periodic quality reports are not generated. 1 - Periodic quality reports are generated throughout a call.	Yes
features.cfg	voice.qualityMonitoring.collector.enable.session	0 (default) - Quality reports are not generated at the end of each call. 1 - Reports are generated at the end of each call.	Yes
features.cfg	voice.qualityMonitoring.collector.enable.triggeredPeriodic	0 (default) - Alert states do not cause periodic reports to be generated. 1 - Periodic reports are generated if an alert state is critical. 2 - Period reports are generated when an alert state is either warning or critical. Note: This parameter is ignored when voice.qualityMonitoring.collector.enable.periodic is 1, since reports are sent throughout the duration of a call.	Yes
features.cfg	voice.qualityMonitoring.collector.period	The time interval (in milliseconds) between successive periodic quality reports. 5 (default) 5 - 900 ms	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	voice.qualityMonitoring.collector.server.x.address	The server address of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages. Set x to 1 as only one report collector is supported at this time. NULL (default) IP address or hostname	Yes
features.cfg	voice.qualityMonitoring.collector.server.x.outboundProxy.address	This parameter directs SIP messages related to voice quality monitoring to a separate proxy. No failover is supported for this proxy, and voice quality monitoring is not available for error scenarios. NULL (default) IP address or FQDN	No
features.cfg	voice.qualityMonitoring.collector.server.x.outboundProxy.port	Specify the port to use for the voice quality monitoring outbound proxy server. 0 (default) 0 to 65535	No
features.cfg	voice.qualityMonitoring.collector.server.x.outboundProxy.transport	Specify the transport protocol the phone uses to send the voice quality monitoring SIP messages. DNSnaptr (default) TCPpreferred UDPOnly TLS TCPOnly	No
features.cfg	voice.qualityMonitoring.collector.server.x.port	Set the port of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages. Set x to 1 as only one report collector is supported at this time. 5060 (default) 1 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`voice.qualityMonitoring.failover.enable`	1 (default) - The phone performs a failover when voice quality SIP PUBLISH messages are unanswered by the collector server. 0 - No failover is performed; note, however, that a failover is still triggered for all other SIP messages. This parameter is ignored if `voice.qualityMonitoring.collector.server.x.outboundProxy` is enabled.	No
`features.cfg`	`voice.qualityMonitoring.location`	Specify the device location with a valid location string. If you do not configure a location value, you must use the default string ‘Unknown'. Unknown (default)	No
`features.cfg`	`voice.qualityMonitoring.rfc6035.enable`	0 (default) - The existing draft implementation is supported. 1 - Complies with RFC6035.	No
`features.cfg`	`voice.qualityMonitoring.rtcpxr.enable`	0 (default) - RTCP-XR packets are not generated. 1 - The packets are generated.	Yes

Video Features

Topics:

After you set up Polycom phones on your network with the default configuration, you can make custom configurations to optimize video calling for Polycom phones, if supported.

Polycom Open SIP video is compatible with the following RFCs:

- RFC 3984 - RTP Payload Format for H.264 video
- RFC 4629 - RTP Payload Format for ITU-T Rec. H.263 Video,
- RFC 5168 - XML Schema for Media Control

The following Polycom phones support transmission and reception of high quality video images:

- VVX® Camera with VVX 500/501 and 600/601 business media phones.
- Built-in camera on VVX 1500 business media phones.

Video and Camera Options

By default, at the start of a video call, the VVX Camera transmits an RTP encapsulated video stream with images captured from the local camera.

Users can stop and start video transmission by pressing the Video key, and then selecting the Stop or Start soft key.

You can use the parameters in the following sections to configure video transmission, the video and local camera view, and video camera options.

Video Quality Parameters

Use the parameters in the following table to configure video transmission.

Video Transmission Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.quality`	The optimal quality for video that is sent in a call or a conference. Motion (default) — for VVX 500 and 600 series business media phones. Sharpness (default) — for VVX 1500 business media phone. Motion - for outgoing video that has motion or movement. Sharpness - for outgoing video that has little or no movement. Note: If `motion` is not selected, moderate to heavy motion can cause some frames to be dropped.	No
`video.cfg`	`video.autoFullScreen`	0 (default) — Video calls only use the full screen layout if it is explicitly selected by the user. 1 — Video calls use the full screen layout by default, such as when a video call is first created or when an audio call transitions to a video call	No
`video.cfg`	`video.autoStartVideoTx`	This parameter controls video sent to the far side. Video from the far side always displays if available, and far side users can control when to send video. 1 (default) — Video transmission to the far side begins when a user starts a call. 0 — Video transmission does not start until a user manually starts video using the Start Video soft key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.callRate`	The default call rate (in kbps) to use when initially negotiating bandwidth for a video call. 512 (default) - The overlay does not time out. 128 - 2048	No
`video.cfg`	`video.forceRtcpVideoCodecControl`	0 (default) — RTCP feedback messages depend on a successful SDP negotiation of a=rtcp-fb and are not used if that negotiation is missing. 1 — The phone is forced to send RTCP feedback messages to request fast I-frame updates along with SIP INFO messages for all video calls irrespective of a successful SDP negotiation of a=rtcp-fb. For an account of all parameter dependencies when setting I-frame requests, refer to the section I-Frames.	No
`video.cfg`	`video.maxCallRate`	Sets the maximum call rate that the users can select. The value set on the phone cannot exceed this value. If `video.callRate` exceeds this value, this parameter overrides `video.callRate` and this value is used as the maximum. 768 (default) 128 - 2048	No

Video and Camera Parameters

You can use configuration parameters to configure the video and camera options for supported cameras.

Video and Camera Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.camera.brightness`	Sets the brightness level of the video stream. The value range is from 0 (dimmest) to 1000 (brightest). NULL (default) 0 - 1000	No
`video.cfg`	`video.camera.contrast`	Sets the contrast level of the video stream for all supported USB cameras. The value range is from 0 (no contrast increase) to 3 (most contrast increase), and 4 (noise reduction contrast). NULL (default) 0 - 1000	No
`video.cfg`	`video.camera.flickerAvoidance`	Sets the flicker avoidance for all supported USB cameras. NULL (default) 0 - Flicker avoidance is automatic. 1 - 50hz AC power frequency flicker avoidance (Europe/Asia). 2 - 60hz AC power frequency flicker avoidance (North America).	No
`video.cfg`	`video.camera.frameRate`	Sets the target frame rate (frames per second) for all supported USB cameras. Values indicate a fixed frame rate from 5 (least smooth) to 30 (most smooth). 25 (default) 5 - 30 If `video.camera.frameRate` is set to a decimal number, the value 25 is used instead.	No
`video.cfg`	`video.camera.saturation`	Sets the saturation level of video captured by any supported USB camera. NULL (default) 0 - 1000	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.camera.sharpness	Sets the sharpness level of video captured. NULL (default) 0 - 1000	No
video.cfg	video.screenMode	Specify the view of the video window in normal viewing mode. normal (default) full crop	No
video.cfg	video.screenModeFS	Specify the view of the video window in full screen viewing mode. normal (default)	No

Video Codec Parameters

Use the parameters in the following table to prioritize and adjust the video codecs.

Video Codec Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.codecPref.H261	0 - 8 6(default)	No
video.cfg	video.codecPref.H264	0 - 8 4 (default)	No
video.cfg	video.codecPref.H2631998	0 - 8 4 (default)	No
video.cfg	video.codecPref.H263	0 - 8 5 (default)	No

Video Profile Parameters

These settings include a group of low-level video codec parameters.

For most use cases, the default values are appropriate. Polycom does not recommend changing the default values unless specifically advised to do so.

Video Profile Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.profile.H261.annexD`	1 (default) - Enables Annex D when negotiating video calls. 0 - Disables Annex D when negotiating video calls.	Yes
`video.cfg`	`video.profile.H261.CifMpi`	Specifies the frame rate divider used by the phone when negotiating CIF resolution for a video call. 1 (default) 0 - 4 To disable, enter 0.	Yes
`video.cfg`	`video.profile.H261.jitterBufferMax`	The largest jitter buffer depth to be supported (in milliseconds). 2000ms (default) (`video.profile.H261.jitterBufferMin` + 500ms) to 2500ms. Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
`video.cfg`	`video.profile.H261.jitterBufferMin`	The smallest jitter buffer depth (in milliseconds) that must be achieved before the first play out. 150ms (default) 33ms to 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H261.jitterBufferShrink	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms to 1000ms Smaller values (33 ms) minimize the delay on trusted networks. Larger values (1000ms) minimize packet loss on networks with large jitter (3000 ms).	Yes
video.cfg	video.profile.H261.payloadType	Specifies the RTP payload format type for H261 MIME type. 31 (default) 0 -127	Yes
video.cfg	video.profile.H261.QcifMpi	Specifies the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 4 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H263.CifMpi	Specifies the frame rate divider that the phone uses when negotiating CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H263.jitterBufferMax	The largest supported jitter buffer depth (in milliseconds). 2000ms (default) (video.profile.H263.jitterBufferMin + 500ms) to 2500ms Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H263.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) to be achieved for the first time, before play out begins. 150ms (default) 33ms to 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes
video.cfg	video.profile.H263.jitterBufferShrink	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms to 1000ms Smaller values (33 ms) minimize the delay on trusted networks. Larger values (1000ms) minimize packet loss on networks with large jitter (3000 ms).	Yes
video.cfg	video.profile.H263.payloadType	Specifies the RTP payload format type for H263 MIME type. 34 (default) 0 - 127	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H263.QcifMpi	Specifies the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H263.SqcifMpi	Specifies the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H2631998.annexF	0 (default) - Enables Annex F when negotiating video calls. 1 - Disables Annex F when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.annexI	0 (default) - Enables Annex I when negotiating video calls. 1 - Disables Annex I when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.annexJ	0 (default) - Enables Annex J when negotiating video calls. 1 - Disables Annex J when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.annexK	Specifies the value of Annex K to use when negotiating video calls. 0 (default) - Enables Annex K when negotiating video calls. 1 - Disables Annex K when negotiating video calls. 2,3,4	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H2631998.annexN	Specifies the value of Annex N to use when negotiating video calls. 0 (default) - Enables Annex N when negotiating video calls. 1 - Disables Annex N when negotiating video calls. 2,3,4	Yes
video.cfg	video.profile.H2631998.annexT	0 (default) - Enables Annex T when negotiating video calls. 1 - Disables Annex T when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.CifMpi	Specifies the frame rate divider that the phone uses when negotiating CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 to 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H2631998.jitterBufferMax	The largest supported jitter buffer depth (in milliseconds). 2000ms (default) (video.profile.H2631998.jitterBufferMin + 500ms) to 2500ms Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H2631998.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) to be achieved for the first time before play out begins. 150ms (default) 33ms - 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H2631998.jitterBufferShrink	The absolute minimum time duration (in milliseconds) of RTP packet Rx, with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms - 1000ms Use smaller values (33 ms) to minimize the delay on trusted networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).	Yes
video.cfg	video.profile.H2631998.payloadType	Specifies the RTP payload format type for H263-1998/90000 MIME type. 96 (default) 96 to 127	Yes
video.cfg	video.profile.H2631998.QcifMpi	Specifies the frame rate divider used by the phone when negotiating Quarter CIF resolution of a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H2631998.SqcifMpi	Specifies the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H264.jitterBufferMax	The largest jitter buffer depth to be supported (in milliseconds). 2000ms (default) (video.profile.H264.jitterBufferMin + 500ms) to 2500ms Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H264.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. 150ms (default) 33ms to 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes
video.cfg	video.profile.H264.jitterBufferShrink	The absolute minimum duration time (in milliseconds) of RTP packet Rx, with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms to 1000ms Use smaller values (33 ms) to minimize the delay on trusted networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.profile.H264.packetizationMode`	Set to control the H.264 decoding capabilities of supported VVX phones for incoming SDP offers from remote endpoints. 0 (default) - Supports only Single NAL unit mode: • If the remote endpoint supports only Non-interleaved mode, the VVX phone rejects the call. • If the remote endpoint supports Single NAL unit mode, the VVX phone answers the incoming call. 1 - Supports Single NAL unit and Non-interleaved modes. The VVX phone answers the incoming call when the remote endpoint supports either Non-interleaved or Single NAL unit mode. **Note:** When packetization mode = 1, the VVX phone supports FU NAL type only, and the phone does not send offer SDP with packetization mode = 1..	No
`video.cfg`	`video.profile.H264.payloadType`	Specifies the RTP payload format type for H264/90000 MIME type. 109 (default) 96 to 127	Yes
`video.cfg`	`video.profile.H264.profileLevel`	Specifies the highest profile level within the baseline profile supported in video calls. 1.3 (default) 1, 1b, 1.1, 1.2, 1.3, and 2 VVX 500/501 and VVX 600/601 phones support H.264 with a profile level of 2, and VVX 1500 phones support H.264 with a profile level of 1.3.	Yes

Supported Video Codecs

See the following table for a summary of video codecs supported on VVX business media phones.

Video Codec Specifications

Algorithm	MIME Type	Frame Size	Bit Rate (kbps)	Frame Rate (fps)
H.261	H261/90000	Tx Frame size: CIF, QCIF, SQCIF RX Frame size: CIF, QCIF	64 to 768	5 to 30
H.263	H263/90000,H263-1998/90000	Tx Frame size:CIF, QCIF Rx Frame size:CIF, QCIF, SQCIF, QVGA, SVGA, SIF	64 to 768 kbps	5 to 30
H.264	H264/90000	Tx Frame size:CIF, QCIF VVX 5xx and 6xx with a VVX Camera support sending 720p resolution for Tx Frame size Rx Frame size:CIF, QCIF, SQCIF, QVGA, SVGA, SIF	64 to 768	5 to 30

H.323 Protocol

VVX 1500 phones and VVX camera-enabled VVX 500/501 and 600/601 phones support telephony signaling via the H.323 protocols.

H.323 protocol enables direct communication with H.323 endpoints, gatekeepers, call servers, media servers, and signaling gateways.

Note: You need a license key to activate H.323 video on your VVX 1500 phone; the license is installed on the VVX 1500D. For more information, contact your Certified Polycom Channel Partner.

SIP and H.323 Protocol

The VVX 500/501, 600/601, and 1500 phones can support both SIP and H.

323 signaling simultaneously, and the phones support bridging both types of calls during multi-party conference calls. The phone can automatically detect the correct or optimal signaling protocol when dialing a call from the contact directory or the corporate directory.

While SIP supports server redundancy and several transport options, only a single configured H.323 gatekeeper address per phone is supported. The phone does not require H.323 gatekeepers, but you can use them if available. If an H.323 gatekeeper is not configured or is unavailable, you can still enable the phones to make H.323 calls.

Support of the SIP protocol for telephony signaling can be disabled on the VVX 500/501, 600/601, and 1500 such that all calls route via the H.323 protocol.

H.323 and SIP Protocol Limitations and Restrictions

The following information should be noted for H.

323 Protocol:

- If the phone has only the H.323 protocol enabled, the phone cannot be used to answer SIP calls.
- If the phone has only the SIP protocol enabled, the phone cannot be used to answer H.323 calls.
- If both SIP and H.323 protocols are disabled by mistake, the phone continues to work as a SIP-only phone; however, the phone is not registered (you are able to send and receive SIP URL calls).
- The protocol to be used when placing a call from the user's local contact directory is unspecified by default. The user can select SIP or H.323 from the directory.
- The protocol that is used when placing a call from the user's corporate directory depends on the order of the attributes in the corporate directory. If only `SIP_address` is defined, then the SIP protocol is used. If only `H323_address` is defined, then the H.323 protocol is used. If both are defined, then the one that is defined first is used.

  For example, if `dir.corp.attribute.4.type` is `SIP_address` and `dir.corp.attribute.5.type` is `H323_address` , then the SIP protocol is used.
- By default, when more than one protocol is available, each protocol displays as a soft key and the user can choose which protocol to use.
- Calls made using H.323 cannot be forwarded or transferred, and the following conditions apply:
  - The Transfer and Forward soft keys do not display during an H.323 call.
  - The Forward soft key does not display on the idle screen if the primary line is an H.323 line.
  - If a user presses the Transfer soft key during an H.323 call, no action is taken.
  - The auto-divert field in the local contact directory entry is ignored when a call is placed to that contact using H.323.
  - If a conference host ends a three-way conference call and one of the parties is connected by H.323, that party is not transferred to the other party that was part of the conference call.

Supported H.323 Video Standards

The following table lists the standards the H.

323 feature supports.

Supported Video Standards

Standard	Description
ITU-T Recommendation H.323 (2003)	Packet-based multimedia communications systems
ITU-T Recommendation Q.931 (1998)	ISDN user-network interface layer 3 specification for basic call control
ITU-T Recommendation H.225.0 (2003)	Call signaling protocols and media stream packetization for packet-based multimedia communications systems
ITU-T Recommendation H.245 (5/2003)	Control protocol for multimedia communication
ITU-T Recommendation H.235.0 - H.235.9 (2005)	Security and encryption for H Series (H.323 and other H.245 based) multimedia terminals

H.323 Protocol Parameters

Use the parameters in the following table to:

- Configure SIP and H.323 protocols
- Set up a SIP and H.323 dial plan

  Numbers with the format 0xxx are placed on a SIP line and numbers with the format 33xx are placed on an H.323 line.
- Set up manual protocol routing using soft keys

  If the protocol to use to place a call cannot be determined, the Use SIP and Use H.323 soft keys display, and users must select one to place the call.
- Configure auto-answering on H.323 calls only.
- Set the preferred protocol to SIP.
- Set to configure one SIP line, one H.323 line, and a dual protocol line—both SIP and H.323 can be used.
- Set the preferred protocol for off-hook calls on the third (dual protocol) line to SIP.

H.323 Protocol Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.manualProtocolRouting`	Specifies whether to choose a protocol routing or use a default protocol. 1 (Default) - User is presented with a protocol routing choice in situations where a call can be placed using either protocol (for example, with SIP and H.323 protocols). 0 - Default protocol is used and the user does not choose. Supports VVX 500/501, 600/601, and 1500 phones.	No
`features.cfg`	`up.manualProtocolRouting.softKeys`	Display soft keys that control **Manual Protocol Routing** options. 1 (Default) - Soft keys are enabled. Use soft keys to choose between the SIP or H.323 protocol. 0 - Soft keys for protocol routing do not display. You can use this parameter with the VVX 500/501, 600/601, and 1500 phones.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg, h323.cfg	call.autoAnswer.H323	This parameter is available for the VVX 500/501, 600/601, and 1500. 0 (default) - Disable auto-answer for H.323 calls. 1 - Enable auto-answer for H.323 calls.	No
sip-interop.cfg	call.enableOnNotRegistered	Lync Base Profile – 0 (default) Generic Base Profile – 1 (default) 1 - Users can make calls when the phone is not registered. When set to 1, Polycom VVX 500/501, 600/601, and 1500 business media phones can make calls using the H.323 protocol even though an H.323 gatekeeper is not configured. 0 - Calls are not permitted without registration.	Yes
reg-advanced.cfg, video.cfg	call.autoAnswer.videoMute	You can use this parameter for the VVX 500/501, 600/601, and 1500 business media phones. 0 (default) - Video begins transmitting (video Tx) immediately after a call is auto-answered. 1 - Video transmission (video Tx) is initially disabled after a call is auto-answered.	No
video.cfg	call.autoRouting.preferredProtocol	You can use this parameter for the VVX 500/501, 600/601, and 1500 business media phones. SIP (default) - Calls are placed via SIP if available or via H.323 if SIP is not available. H323 - Calls are placed via H.323 if available, or via SIP if H.323 is not available.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	call.autoRouting.preference	You can use this parameter for the VVX 500/501, 600/601, and 1500 business media phones. line - Calls are placed via the first available line, regardless of its protocol capabilities. If the first available line has both SIP and H.323 capabilities, the preferred protocol is used (call.autoRouting.preferredProtocol). protocol - The first available line with the preferred protocol activated is used, if available. If not available, the first available line is used. Note that auto-routing is used when manual routing selection features (up.manualProtocolRouting) are disabled.	No
sip-interop.cfg	reg.x.protocol.H323	You can use this parameter for the VVX 500/501, 600/601, and 1500. 0 (default) - H.323 signaling is not enabled for registration x. 1 - H.323 signaling is enabled for registration x.	No
site.cfg	reg.x.server.H323.y.address	Address of the H.323 gatekeeper. Null (default) IP address or hostname	No
site.cfg	reg.x.server.H323.y.port	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. 0 (default) 0 to 65535	No
site.cfg	reg.x.server.H323.y.expires	Desired registration period. 3600 positive integer	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
h323.cfg	voIpProt.H323.autoGateKeeperDiscovery	1 (default) - The phone will attempt to discover an H.323 gatekeeper address via the standard multi cast technique, provided that a statically configured gatekeeper address is not available. 0 - The phone will not send out any gatekeeper discovery messages.	Yes
h323.cfg	voIpProt.H323.blockFacilityOnStartH245	0 (default) - facility messages when using H.245 are not removed. 1 - facility messages when using H. 245 are removed.	Yes
h323.cfg	voIpProt.H323.dtmfViaSignaling.enabled	1 (default) - The phone will use the H. 323 signaling channel for DTMF key press transmission. 0 - The phone will not use H.323 signaling channel for DTMF key press transmission.	Yes
h323.cfg	voIpProt.H323.dtmfViaSignaling.H245alphanumericMode	1 (default) - The phone will support H. 245 signaling channel alphanumeric mode DTMF transmission. 0 - The phone will not support H.245 signaling channel alphanumeric mode DTMF transmission Note: If both alphanumeric and signal modes can be used, the phone gives priority to DTMF.	Yes
h323.cfg	voIpProt.H323.dtmfViaSignaling.H245signalMode	1 (default) - The phone will support H. 245 signaling channel signal mode DTMF transmission. 0 - The phone will not support H.245 signaling channel signal mode DTMF transmission.	Yes
h323.cfg	voIpProt.H323.enable	0 (default) - The H.323 protocol is not used for call routing, dial plan, DTMF, and URL dialing. 1 - The H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
h323.cfg	voIpProt.H323.local.port	Local port for sending and receiving H.323 signaling packets. 0 - 1720 is used for the local port but is not advertised in the H.323 signaling. 0 to 65535 - The value is used for the local port and it is advertised in the H.323 signaling.	Yes
sip-interop.cfg	voIpProt.H323.local.RAS.port	Specifies the local port value for RAS signaling. 1719 (default) 1 to 65535	Yes
h323.cfg	voIpProt.server.H323.x.address	Address of the H.323 gatekeeper. Only one H.323 gatekeeper per phone is supported. If more than one is configured, only the first is used. Null (default) IP address or hostname	No
h323.cfg	voIpProt.server.H323.x.port	Port to be used for H.323 signaling. The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP. 1719 (default) 0 to 65535	No
h323.cfg	voIpProt.server.H323.x.expires	Desired registration period. 3600 (default) positive integer.	No
site.cfg	sec.H235.mediaEncryption.enabled	1 (default) - The H.235 media encryption is enabled and negotiated. 0 - The H.235 media encryption is disabled.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`sec.H235.mediaEncryption.offer`	0 (default) - The media encryption offer is not initiated with the far-end. 1 - If the `sec.H235.mediaEncryption.enabled` is also 1, media encryption negotiations is initiated with the far-end; however, successful negotiations are not a requirement for the call to complete.	Yes
`site.cfg`	`sec.H235.mediaEncryption.require`	0 (default) - The media encryption requirement is not required. 1 - If the `sec.H235.mediaEncryption.enabled` is also 1, media encryption negotiations are initiated or completed with the far end, and if negotiations fail, the call is dropped.	Yes

FQDN Support for H.323 Gatekeeper Failover

This enhancement, available only for registration failover scenarios, enables fully qualified domain name (FQDN) configuration for H.

323 Gatekeeper. Gatekeeper IP addresses are resolved from a DNS server when the Gatekeeper sends a DNS A query or through the local static cache. This enhancement supports a maximum of two IP addresses based on the DNS response irrespective of the number of records received.

Note that this enhancement does not apply if you are using the parameter `voIpProt.H323.autoGateKeeperDiscovery` for auto-discovery.

Toggling Between Audio-only or Audio-Video Calls

You can toggle between audio-only and audio-video calls.

When this feature is enabled on the VVX 1500, and VVX camera-enabled VVX 500/501 and 600/601 business media phones, a soft key displays to enable users to toggle calls between audio-only or audio-video.

When the phone is registered, you can:

- Use `video.callMode.default` to begin calls as audio-video or audio only. By default, calls begin as audio-video. After a video call has ended, the phone returns to audio-only.
- Use `up.homeScreen.audioCall.enabled` to enable a Home screen icon that allows you to make audio-only calls. Far-end users can add video during a call if the far-end device is video capable.

Audio-only or Audio-Video Call Parameters

The following parameters configure whether the phone starts a call with audio and video.

Audio-only or Audio-Video Call Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.homeScreen.audioCall.enabled	0 (default) - Disable a Home screen icon that allows users to make audio-only calls. 1 - Enable a Home screen icon that allows users to make audio-only calls. Devices that support video calling show an 'Audio Call' button on the Home screen to initiate audio-only calls.	No
video.cfg	video.autoStartVideoTx	1 (default) - Automatically begin video to the far side when you start a call. 0 - Video to the far side does not begin. Note that when the phone Base Profile is set to Skype or Lync, the default is 1.	No
video.cfg	video.callMode.default	VVX phones Allow the user to begin calls as audio-only or with video. When you set this parameter to 'video', the VVX 500/501 and 600/601 display a Video Mode soft key and the VVX 1500 displays a video icon. audio (default) Calls begin with audio only. video - Calls begin with video. Allow the user to begin calls as audio-only or with video. video (default) audio - Set the initial call to audio only and video may be added during a call. On Polycom Trio solution, you can combine this parameter with `video.autoStartVideoTx` .	No

I-Frames

When video streams initialize, devices transmit video packets called I-frames (reference frames) that contain information to display a complete picture.

Subsequent video packets, known as P-frames, are smaller and not as complete to consume less bandwidth. Due to packet loss, jitter, or corruption, devices occasionally need to make multiple requests for a complete I-frame in order to reset the full frame, after which devices can revert to P-frame updates.

You can set parameters to control an I-frame request. The following table indicates parameter dependencies and messaging behavior when setting an I-frame request method.

I-Frame Parameter Dependencies

video.forceRtcpVideoCodecControl	video.dynamicControlMethod	voIpProt.SDP.offer.rtcpVideoCodecControl	Behavior when requesting video I-frame updates
0	0 (n/a)	0	Only SIP INFO messages are sent. No RTCP-FB is offered in SDP.
0	1 (n/a)	0	Only SIP INFO messages are sent. No RTCP-FB is offered in SDP.
0	0 (n/a)	1	RTCP-FB is offered in SDP. If SDP responses do not contain the required RTCP-FB attribute, then only SIP INFO requests are used.
0	1 (n/a)	1	RTCP-FB is offered in SDP. If SDP responses do not contain the required RTCP-FB attribute, then only SIP INFO requests are used.
1	0	0	The SDP attribute a=rtcp-fb is not included in SDP offers. Both RTCP-FB and SIP INFO messages are attempted.
1	1	0	The SDP attribute a=rtcp-fb is not included in SDP offers. Both RTCP-FB and SIP INFO messages are attempted. If no RTCP-FB messages are received, only SIP INFO messages are sent. If no response is received for SIP INFO messages then, again, both RTCP-FB and SIP INFO messages are attempted.
1	0	1	RTCP-FB is offered in SDP. Even if the SDP response does not include an accepted a=rtcp-fb attribute both RTCP-FB and SIP INFO messages are sent.
1	1	1	RTCP-FB is offered in SDP. Even if the SDP response does not include an accepted a=rtcp-fb attribute both RTCP-FB and SIP INFO messages are sent initially. If no RTCP-FB response is received, only SIP INFO messages are sent afterwards.

Video Parameters

The parameters in the table are supported on the VVX 500/501, VVX 600/601, and VVX 1500, and Polycom Trio solution.

Video Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`video.cfg`	`video.allowWithSource`	Restricts sending video codec negotiation in Session Description Protocol (SDP). 0 (default) 0 or 1 This parameter applies only for VVX 500/501 and VVX 600/601.	No
`video.cfg`	`video.autoFullScreen`	0 (default) - Video calls use the full screen layout, only if explicitly selected by the user. 1 - Video calls use the full screen layout by default.	No
`video.cfg`	`video.dynamicControlMethod`	0 (default) 1 - The first I-Frame request uses the method defined by `video.forceRtcpVideoCodecControl` and subsequent requests alternate between RTCP-FB and SIP INFO. To set other methods for I-frame requests, refer the parameter `video.forceRtcpVideoCodecControl.`	No
`video.cfg`	`video.iFrame.delay`	0 (default) 1 -10 seconds - Transmits an extra I-frame after the video starts. The amount of delay from the start of video until the I-frame is sent is configurable up to 10 seconds.	Yes
`video.cfg`	`video.iFrame.minPeriod`	Time taken before sending a second I-frame in response to requests from the far end. 2 (default) 1 - 60	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.iFrame.onPacketLoss	0 (default) 1 - Transmits an I-frame to the far end when video RTP packet loss occurs.	No
video.cfg	video.iFrame.period.onBoard	Set the I-Frame interval used for the VC4 encoder. 180 (default) 300 maximum	No

Video Codec Preference Parameters

The following table lists video codec parameters and specifies the video codec preferences for the Polycom Trio solution.

The following table lists video codec and specifies the video codec preferences for the VVX 500/501, 600/601, and 1500 phones. To disable codecs, set the value to 0. A value of 1 indicates the codec is the most preferred and has highest priority. The VVX 500/501 and 600/601 support H.263 and H.264 and do not support H.261 or H.263 1998.

Video Codec Preference Parameters

Template	Parameter	Permitted Value	Change Causes Restart or Reboot
video.cfg	video.codecPref.H261	Sets the H.261 payload type. 6 (default) 0 - 8	No
video.cfg	video.codecPref.H264	Sets the H.264 payload type. 4 (default) 0 - 8	No
video.cfg	video.codecPref.H263 1998	Sets the H.263 payload type. 5 (default) 0 - 8	No
video.cfg	video.codecPref.H263	5 (default) 0 - 8	No
video.cfg	video.codecPref.H264	4 (default) 0 - 8	No

Video Profile Parameters

These settings include a group of low-level video codec parameters.

For most use cases, the default values are appropriate. Polycom does not recommend changing the default values unless specifically advised to do so.

Video Profile Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H261.annexD	1 (default) - Enables Annex D when negotiating video calls. 0 - Disables Annex D when negotiating video calls.	Yes
video.cfg	video.profile.H261.CifMpi	Specifies the frame rate divider used by the phone when negotiating CIF resolution for a video call. 1 (default) 0 - 4 To disable, enter 0.	Yes
video.cfg	video.profile.H261.jitterBufferMax	The largest jitter buffer depth to be supported (in milliseconds). 2000ms (default) (video.profile.H261.jitterBufferMin + 500ms) to 2500ms. Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H261.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) that must be achieved before the first play out. 150ms (default) 33ms to 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H261.jitterBufferShrink	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms to 1000ms Smaller values (33 ms) minimize the delay on trusted networks. Larger values (1000ms) minimize packet loss on networks with large jitter (3000 ms).	Yes
video.cfg	video.profile.H261.payloadType	Specifies the RTP payload format type for H261 MIME type. 31 (default) 0 -127	Yes
video.cfg	video.profile.H261.QcifMpi	Specifies the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 4 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H263.CifMpi	Specifies the frame rate divider that the phone uses when negotiating CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H263.jitterBufferMax	The largest supported jitter buffer depth (in milliseconds). 2000ms (default) (video.profile.H263.jitterBufferMin + 500ms) to 2500ms Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H263.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) to be achieved for the first time, before play out begins. 150ms (default) 33ms to 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes
video.cfg	video.profile.H263.jitterBufferShrink	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms to 1000ms Smaller values (33 ms) minimize the delay on trusted networks. Larger values (1000ms) minimize packet loss on networks with large jitter (3000 ms).	Yes
video.cfg	video.profile.H263.payloadType	Specifies the RTP payload format type for H263 MIME type. 34 (default) 0 - 127	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H263.QcifMpi	Specifies the frame rate divider that the phone uses when negotiating Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H263.SqcifMpi	Specifies the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H2631998.annexF	0 (default) - Enables Annex F when negotiating video calls. 1 - Disables Annex F when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.annexI	0 (default) - Enables Annex I when negotiating video calls. 1 - Disables Annex I when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.annexJ	0 (default) - Enables Annex J when negotiating video calls. 1 - Disables Annex J when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.annexK	Specifies the value of Annex K to use when negotiating video calls. 0 (default) - Enables Annex K when negotiating video calls. 1 - Disables Annex K when negotiating video calls. 2,3,4	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H2631998.annexN	Specifies the value of Annex N to use when negotiating video calls. 0 (default) - Enables Annex N when negotiating video calls. 1 - Disables Annex N when negotiating video calls. 2,3,4	Yes
video.cfg	video.profile.H2631998.annexT	0 (default) - Enables Annex T when negotiating video calls. 1 - Disables Annex T when negotiating video calls.	Yes
video.cfg	video.profile.H2631998.CifMpi	Specifies the frame rate divider that the phone uses when negotiating CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 to 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H2631998.jitterBufferMax	The largest supported jitter buffer depth (in milliseconds). 2000ms (default) (video.profile.H2631998.jitterBufferMin + 500ms) to 2500ms Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H2631998.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) to be achieved for the first time before play out begins. 150ms (default) 33ms - 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H2631998.jitterBufferShrink	The absolute minimum time duration (in milliseconds) of RTP packet Rx, with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms - 1000ms Use smaller values (33 ms) to minimize the delay on trusted networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).	Yes
video.cfg	video.profile.H2631998.payloadType	Specifies the RTP payload format type for H263-1998/90000 MIME type. 96 (default) 96 to 127	Yes
video.cfg	video.profile.H2631998.QcifMpi	Specifies the frame rate divider used by the phone when negotiating Quarter CIF resolution of a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes
video.cfg	video.profile.H2631998.SqcifMpi	Specifies the frame rate divider that the phone uses when negotiating Sub Quarter CIF resolution for a video call. 1 (default) - Enables the frame rate divider used by the phone when negotiating the resolution of a video call. 0 - 32 0 - Disables the frame rate divider used by the phone when negotiating the resolution of a video call.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H264.jitterBufferMax	The largest jitter buffer depth to be supported (in milliseconds). 2000ms (default) (video.profile.H264.jitterBufferMin + 500ms) to 2500ms Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.	Yes
video.cfg	video.profile.H264.jitterBufferMin	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. 150ms (default) 33ms to 1000ms Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.	Yes
video.cfg	video.profile.H264.jitterBufferShrink	The absolute minimum duration time (in milliseconds) of RTP packet Rx, with no packet loss between jitter buffer size shrinks. 70ms (default) 33ms to 1000ms Use smaller values (33 ms) to minimize the delay on trusted networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
video.cfg	video.profile.H264.packetizationMode	Set to control the H.264 decoding capabilities of supported VVX phones for incoming SDP offers from remote endpoints. 0 (default) - Supports only Single NAL unit mode: • If the remote endpoint supports only Non-interleaved mode, the VVX phone rejects the call. • If the remote endpoint supports Single NAL unit mode, the VVX phone answers the incoming call. 1 - Supports Single NAL unit and Non-interleaved modes. The VVX phone answers the incoming call when the remote endpoint supports either Non-interleaved or Single NAL unit mode. **Note:** When packetization mode = 1, the VVX phone supports FU NAL type only, and the phone does not send offer SDP with packetization mode = 1..	No
video.cfg	video.profile.H264.payloadType	Specifies the RTP payload format type for H264/90000 MIME type. 109 (default) 96 to 127	Yes
video.cfg	video.profile.H264.profileLevel	Specifies the highest profile level within the baseline profile supported in video calls. 1.3 (default) 1, 1b, 1.1, 1.2, 1.3, and 2 VVX 500/501 and VVX 600/601 phones support H.264 with a profile level of 2, and VVX 1500 phones support H.264 with a profile level of 1.3.	Yes

Phone Display and Appearances

Topics:

- Time Zone Location Description
- Time and Date
- Phone Theme
- Default Phone Screen
- Graphic Display Background
- Digital Picture Frame
- Background Image Lock
- Phone Languages
- Pinyin Text Input
- Hide the MAC Address
- Digital Phone Label
- Unique Line Labels for Registration Lines
- LED Indicators
- Capture Your Device's Current Screen

This section provides information on setting up features involving the phone's user interface.

Time Zone Location Description

The following two parameters configure a time zone location description for their associated GMT offset:

- `device.sntp.gmtOffsetcityID` If you are not provisioning phones manually from the phone menu or Web Configuration Utility and you are setting the `device.sntp.gmtOffset` parameter, then you must configure `device.sntp.gmtOffsetcityID` to ensure that the correct time zone location description displays on the phone menu and Web Configuration Utility. The time zone location description is set automatically if you set the `device.sntp.gmtOffset` parameter manually using the phone menu or Web Configuration Utility.
- `tcpIpApp.sntp.gmtOffsetcityID` If you are not provisioning phones manually from the Web Configuration Utility and you are setting the `tcpIpApp.sntp.gmtOffset` parameter, then you must configure `tcpIpApp.sntp.gmtOffsetcityID` to ensure that the correct time zone location description displays on the Web Configuration Utility. The time zone location description is set automatically if you set the `tcpIpApp.sntp.gmtOffset` parameter manually using the Web Configuration Utility.

Related Links

Time and Date Display Parameters on page 240

Time Zone Location Parameters

The following parameters configure time zone location.

Time Zone Location Parameters

Permitted Values		Permitted Values	
0	(GMT -12:00) Eniwetok,Kwajalein	61	(GMT +2:00) Helsinki,Kyiv
1	(GMT -11:00) Midway Island	62	(GMT +2:00) Riga,Sofia
2	(GMT -10:00) Hawaii	63	(GMT +2:00) Tallinn,Vilnius
3	(GMT -9:00) Alaska	64	(GMT +2:00) Athens,Istanbul
4	(GMT -8:00) Pacific Time (US & Canada)	65	(GMT +2:00) Damascus
5	(GMT -8:00) Baja California	66	(GMT +2:00) E.Europe
6	(GMT -7:00) Mountain Time (US & Canada)	67	(GMT +2:00) Harare,Pretoria
7	(GMT -7:00) Chihuahua,La Paz	68	(GMT +2:00) Jerusalem
8	(GMT -7:00) Mazatlan	69	(GMT +2:00) Kaliningrad (RTZ 1)
9	(GMT -7:00) Arizona	70	(GMT +2:00) Tripoli
10	(GMT -6:00) Central Time (US & Canada)		
11	(GMT -6:00) Mexico City	71	(GMT +3:00) Moscow
12	(GMT -6:00) Saskatchewan	72	(GMT +3:00) St.Petersburg
13	(GMT -6:00) Guadalajara	73	(GMT +3:00) Volgograd (RTZ 2)
14	(GMT -6:00) Monterrey	74	(GMT +3:00) Kuwait,Riyadh
15	(GMT -6:00) Central America	75	(GMT +3:00) Nairobi
16	(GMT -5:00) Eastern Time (US & Canada)	78	(GMT +3:00) Baghdad
17	(GMT -5:00) Indiana (East)	76	(GMT +3:00) Minsk
18	(GMT -5:00) Bogota,Lima	77	(GMT +3:30) Tehran
19	(GMT -5:00) Quito	79	(GMT +4:00) Abu Dhabi,Muscat
20	(GMT -4:30) Caracas	80	(GMT +4:00) Baku,Tbilisi
21	(GMT -4:00) Atlantic Time (Canada)	81	(GMT +4:00) Izhevsk,Samara (RTZ 3)
22	(GMT -4:00) San Juan	82	(GMT +4:00) Port Louis
23	(GMT -4:00) Manaus,La Paz	83	(GMT +4:00) Yerevan
24	(GMT -4:00) Asuncion,Cuiaba	84	(GMT +4:30) Kabul
25	(GMT -4:00) Georgetown	85	(GMT +5:00) Ekaterinburg (RTZ 4)
26	(GMT -3:30) Newfoundland	86	(GMT +5:00) Islamabad
27	(GMT -3:00) Brasilia	87	(GMT +5:00) Karachi
28	(GMT -3:00) Buenos Aires	88	(GMT +5:00) Tashkent
29	(GMT -3:00) Greenland	89	(GMT +5:30) Mumbai,Chennai
30	(GMT -3:00) Cayenne,Fortaleza	90	(GMT +5:30) Kolkata,New Delhi

Permitted Values		Permitted Values	
31	(GMT -3:00) Montevideo	91	(GMT +5:30) Sri Jayawardenepura
32	(GMT -3:00) Salvador	92	(GMT +5:45) Kathmandu
33	(GMT -3:00) Santiago	93	(GMT +6:00) Astana,Dhaka
34	(GMT -2:00) Mid-Atlantic	94	(GMT +6:00) Almaty
35	(GMT -1:00) Azores	95	(GMT +6:00) Novosibirsk (RTZ 5)
36	(GMT -1:00) Cape Verde Islands	96	(GMT +6:30) Yangon (Rangoon)
37	(GMT 0:00) Western Europe Time	97	(GMT +7:00) Bangkok,Hanoi
38	(GMT 0:00) London,Lisbon	98	(GMT +7:00) Jakarta
39	(GMT 0:00) Casablanca	99	(GMT +7:00) Krasnoyarsk (RTZ 6)
40	(GMT 0:00) Dublin	100	(GMT +8:00) Beijing,Chongqing
41	(GMT 0:00) Edinburgh	101	(GMT +8:00) Hong Kong,Urumqi
42	(GMT 0:00) Monrovia	102	(GMT +8:00) Kuala Lumpur
43	(GMT 0:00) Reykjavik	103	(GMT +8:00) Singapore
44	(GMT +1:00) Belgrade	104	(GMT +8:00) Taipei,Perth
45	(GMT +1:00) Bratislava	105	(GMT +8:00) Irkutsk (RTZ 7)
46	(GMT +1:00) Budapest	106	(GMT +8:00) Ulaanbaatar
47	(GMT +1:00) Ljubljana	107	(GMT +9:00) Tokyo,Seoul,Osaka
48	(GMT +1:00) Prague	108	(GMT +9:00) Sapporo,Yakutsk (RTZ 8)
49	(GMT +1:00) Sarajevo,Skopje	109	(GMT +9:30) Adelaide,Darwin
50	(GMT +1:00) Warsaw,Zagreb	110	(GMT +10:00) Canberra
51	(GMT +1:00) Brussels	111	(GMT +10:00) Magadan (RTZ 9)
52	(GMT +1:00) Copenhagen	112	(GMT +10:00) Melbourne
53	(GMT +1:00) Madrid,Paris	113	(GMT +10:00) Sydney,Brisbane
54	(GMT +1:00) Amsterdam,Berlin	114	(GMT +10:00) Hobart
55	(GMT +1:00) Bern,Rome	115	(GMT +10:00) Vladivostok
56	(GMT +1:00) Stockholm,Vienna	116	(GMT +10:00) Guam,Port Moresby
57	(GMT +1:00) West Central Africa	117	(GMT +11:00) Solomon Islands
58	(GMT +1:00) Windhoek	118	(GMT +11:00) New Caledonia
59	(GMT +2:00) Bucharest,Cairo	119	(GMT +11:00) Chokurdakh (RTZ 10)
60	(GMT +2:00) Amman,Beirut	120	(GMT +12:00) Fiji Islands

Permitted Values	Permitted Values
	121 (GMT +12:00) Auckland,Anadyr 122 (GMT +12:00) Petropavlovsk-Kamchatsky (RTZ 11) 123 (GMT +12:00) Wellington 124 (GMT +12:00) Marshall Islands 125 (GMT +13:00) Nuku'alofa 126 (GMT +13:00) Samoa

Time and Date

A clock and calendar display on the phones by default.

You can choose how to display the time and date for your time zone in several formats, or you can disable the display of the time and date. You can also set the time and date format to display differently when the phone is in certain modes. For example, the display format can change when the phone goes from idle mode to an active call.

To have the most accurate time, you have to synchronize the phone to the Simple Network Time Protocol (SNTP) time server. Until a successful SNTP response is received, the phone continuously flashes the time and date to indicate that they are not accurate.

The time and date display on the phones in PSTN mode and are set by an incoming call with a supported caller ID standard, or when the phone is connected to Ethernet and you enable the date and time display.

Time and Date Display Parameters

Use the parameters in the following table to configure time and display options.

Time and Date Display Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.localClockEnabled`	Specifies whether or not the date and time are shown on the idle display. 1 (Default) - Date and time and shown on the idle display. 0 - Date and time are not shown on the idle display.	No
`site.cfg`	`lcl.datetime.date.dateTop`	1 (default) - Displays the date above time. 0 - Displays the time above date.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	lcl.datetime.date.format	The phone displays day and date. "D,dM" (default) String The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time. For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday.	No
site.cfg	lcl.datetime.date.longFormat	1 (default) - Displays the day and month in long format (Friday/November). 0 - Displays the day and month in abbreviated format (Fri/Nov).	No
site.cfg	lcl.datetime.time.24HourClock	1 (default) - Displays the time in 24-hour clock mode. 0 - Does not display the time in 24-hour clock mode.	No
site.cfg	tcpIpApp.sntp.address	Specifies the SNTP server address. NULL (default) Valid hostname or IP address.	No
site.cfg	tcpIpApp.sntp.AQuery	Specifies a query to return hostnames. 0 (default) - Queries to resolve the SNTP hostname are performed using DNS SRV. 1 - Query the hostname for a DNS A record.	No
site.cfg	tcpIpApp.sntp.address.overrideDHCP	0 (Default) - DHCP values for the SNTP server address are used. 1 - SNTP parameters override the DHCP values.	No
site.cfg	tcpIpApp.sntp.daylightSavings.enable	1 (Default) - Daylight savings rules apply to the displayed time. 0 - Daylight savings time rules are not applied to the displayed time.	No
site.cfg	tcpIpApp.sntp.daylightSavings.fixedDayEnable	0 (Default) - Month , date , and dayOfWeek are used in the DST calculation. 1 - Only month and date are used in the DST calculation.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`tcpIpApp.sntp.daylightSavings.start.date`	Start date for daylight savings time. Range is 1 to 31. 8 (Default) - Second occurrence in the month after DST starts. 0 - If `fixedDayEnable` is set to 0, this value specifies the occurrence of `dayOfWeek` when DST should start. 1 - If `fixedDayEnable` is set to 1, this value is the day of the month to start DST. 15 - Third occurrence. 22 - Fourth occurrence. Example: If value is set to 15, DST starts on the third `dayOfWeek` of the month.	No
`site.cfg`	`tcpIpApp.sntp.daylightSavings.start.dayOfWeek`	Specifies the day of the week to start DST. Range is 1 to 7. 1 (Default) - Sunday 2 - Monday... 7 - Saturday This parameter is not used if `fixedDayEnable` is set to 1.	No
`site.cfg`	`tcpIpApp.sntp.daylightSavings.start.dayOfWeek.lastInMonth`	0 (Default) 1 - DST starts on the last `dayOfWeek` of the month and the `start.date` is ignored. This parameter is not used if `fixedDayEnable` is set to 1.	No
`site.cfg`	`tcpIpApp.sntp.daylightSavings.start.month`	Specifies the month to start DST. Range is 1 to 12. 3 (Default) - March 1 - January 2 - February... 12 - December	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.sntp.daylightSavings.start.time	Specifies the time of day to start DST in 24-hour clock format. Range is 0 to 23. 2 (Default) - 2 a.m. 0 - 12 a.m. 1 - 1 a.m.... 12 - 12 p.m. 13 - 1 p.m... 23 - 11 p.m.	No
site.cfg	tcpIpApp.sntp.daylightSavings.stop.date	Specifies the stop date for daylight savings time. Range is 1 to 31. 1 (Default) - If fixedDayEnable is set to 1, the value of this parameter is the day of the month to stop DST. Set 1 for the first occurrence in the month. 0 - If fixedDayEnable is set to 0, this value specifies the dayOfWeek when DST should stop. 8 - Second occurrence. 15 - Third occurrence. 22 - Fourth occurrence. Example: If set to 22, DST stops on the fourth dayOfWeek in the month.	No
site.cfg	tcpIpApp.sntp.daylightSavings.stop.dayOfWeek	Day of the week to stop DST. Range is 1 to 7. 1 (default) - Sunday 2 - Monday 3 - Tuesday 7 - Saturday Parameter is not used if fixedDayEnable is set to 1.	No
site.cfg	tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth	1 - DST stops on the last dayOfWeek of the month and the stop.date is ignored). Parameter is not used if fixedDayEnable is set to 1.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.sntp.daylightSavings.stop.month	Specifies the month to stop DST. Range is 1 to 12. 11 - November 1 - January 2 - February… 12 - December	No
site.cfg	tcpIpApp.sntp.daylightSavings.stop.time	Specifies the time of day to stop DST in 24-hour clock format. Range is 0 to 23. 2 (Default) - 2 a.m. 0 - 12 a.m. 1 - 1 a.m.... 12 - 12 p.m. 13 - 1 p.m... 23 - 11 p.m.	No
site.cfg	tcpIpApp.sntp.gmtOffset	Specifies the offset in seconds of the local time zone from GMT. 0 (Default) - GMT 3600 seconds = 1 hour -3600 seconds = -1 hour Positive or negative integer	No
site.cfg	tcpIpApp.sntp.gmtOffsetcityID	Range is 0 to127. NULL (Default) For descriptions of all values, refer to Time Zone Location Description.	No
site.cfg	tcpIpApp.sntp.gmtOffset.overrideDHCP	0 (Default) - The DHCP values for the GMT offset are used. 1 - The SNTP values for the GMT offset are used.	No
site.cfg	tcpIpApp.sntp.resyncPeriod	Specifies the period of time (in seconds) that passes before the phone resynchronizes with the SNTP server. 86400 (Default). 86400 seconds is 24 hours. Positive integer	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.sntp.retryDnsPeriod	Sets a retry period for DNS queries. 86400 (Default). 86400 seconds is 24 hours. 60 - 2147483647 seconds The DNS retry period is affected by other DNS queries made on the phone. If the phone makes a query for another service during the retry period, such as SIP registration, and receives no response, the Network Time Protocol (NTP) DNS query is omitted to limit the retry attempts to the unresponsive server. If no other DNS attempts are made by other services, the retry period is not affected. If the DNS server becomes responsive to another service, NTP immediately retries the DNS query.	No

Related Links

Time Zone Location Description on page 237

Date Formats

Use the following table to choose values for the `lcl.`

`datetime.date.format`and `lcl.datetime.date.longformat` parameters. The table shows values for Friday, August 19, 2011 as an example.

Date Formats

lcl.datetime.date.format	lcl.datetime.date.longformat	Date Displayed on Phone
dM,D	0	19 Aug, Fri
dM,D	1	19 August, Friday
Md,D	0	Aug 19, Fri
Md,D	1	August 19, Friday
D,dM	0	Fri, 19 Aug
D,dM	1	Friday, August 19
DD/MM/YY	n/a	19/08/11
DD/MM/YYYY	n/a	19/08/2011
MM/DD/YY	n/a	08/19/11

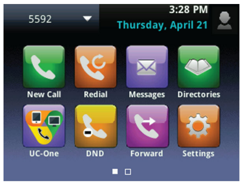
lcl.datetime.date.format	lcl.datetime.date.longformat	Date Displayed on Phone
MM/DD/YYYY	n/a	08/19/2011
YY/MM/DD	n/a	11/08/19
YYYY/MM/DD	n/a	2011/08/11

Phone Theme

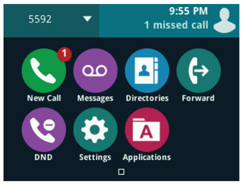
You can configure a phone's theme depending on your phone model.

The VVX 500/501 and 600/601 business media phones include three display themes (Classic (default), Modern, and BroadSoft). The VVX 300 and 400 series business media phones and 250, 350, and 450 business IP phones include two themes (Classic and BroadSoft) that determine how the user interface and icons display on the phone.

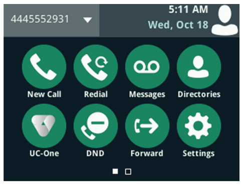
The following figures show the differences between the themes.



Classic



Modern



BroadSoft

Phone Theme Parameters

Use the parameters in the following table to configure a theme for the VVX 300, 400, 500, and 600 series business media phones and VVX 250, 350, and 450 business IP phones.

Note: If the parameters `reg.x.server.y.specialInterop` and `voIpProt.server.x.specialInterop` are configured for any value other than Standard, the phone displays the Classic theme in place of the BroadSoft theme.

Phone Theme Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg	device.theme	Choose the user interface color scheme and icons that displays on the phone. Classic (default) Modern (only on 500/501 and 600/601) BroadSoft	Yes
device.cfg	device.theme.set	0 (default) - The phone does not apply the user interface theme specified in the device.theme parameter, and the default theme displays after a reboot. 1 - The phone applies the theme specified in the device.theme parameter, and the selected them displays after a reboot.	No
reg-advanced.cfg	reg.x.server.y.specialInterop	Specify the server-specific feature set for the line registration. Standard (Default) VVX 101: Standard, GENBAND, ALU-CTS, DT VVX 150, 201: Standard, GENBAND, ALU-CTS, ocs2007r2, lync2010, DT All other phones: Standard, GENBAND, ALU-CTS, ocs2007r2, lync2010, lcs2005, DT	No
sip-interop.cfg	voIpProt.server.x.specialInterop	Enables server-specific features for all registrations. Standard (default) VVX 101 = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT VVX 150, 201 = Standard, GENBAND, GENBAND-A2, ALU-CTS, ocs2007r2, lync2010, DT All other phones = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT, ocs2007r2, lync2010, lcs2005, DT	No

Default Phone Screen

On all VVX phones, you can configure the default phone screen that displays when the phone goes off-hook or is in an active call.

Off-Hook Phone Screen

When the phone goes off-hook, you can configure the phone to display either the dialer view or the Lines screen by default.

If the Lines screen is set to be the default; when the user dials a number, a dialer screen would display. When the Lines Screen is set as default; if the user dials a number, a dialer screen is displayed. If the user selects the New Call soft key, a line screen would display.

Displaying the Lines screen when the phone goes off-hook enables users to quickly select a favorite or BLF line to dial. In this scenario, when users start to enter a number from the keypad, the phone switches to the dialer screen.

Off-Hook Phone Screen Parameters

Use the parameters in the following table to set the default screen that displays when the phone is off-hook.

Off-Hook Phone Screen Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.OffHookLineView.enabled`	0 (Default) - After the phone goes off-hook, the phone displays the dialing screen. 1 - After the phone goes off hook, the phone displays the line screen.	No

Active Call Phone Screen

In an active call, you can configure the phone to display the active call screen or the Lines screen.

You can configure the phones to display the screens as follows:

- The normal active call screen or call overlay showing active call information.
- The Lines screen, showing active call information in the ribbon at the top of the screen.

Displaying the Lines screen during an active call enables users to see the status of any lines, buddies, and BLF contacts they are monitoring without active call information getting in the way.

It is still possible to switch between the normal active call display and the lines view regardless of the default screen you set.

Active Call Screen Parameters

Use the parameters in the following table to set the default screen that displays when the phone is in call.

Start writing here.

Active Call screen parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.LineViewCallStatus.enabled`	0 (Default) - In an active call, the active call screen displays. Any incoming or outgoing call triggers the display of the active call screen. 1 - During an incoming call and in an active call, the line view displays and call details display on the status ribbon.	No
`features.cfg`	`up.LineViewCallStatus.timeout`	Specify the timeout period after which the phones go back to the Line Screen when the user goes to the Active Call Screen from the Line View. 10 (default) - The phone returns to the line screen after 10 seconds. 2 - 10 - Specify the seconds after which the phone returns to the line screen.	No

Graphic Display Background

You can display a graphic image on the background of all VVX business media phones, VVX 350 and 450 business IP phones, and connected VVX Color Expansion Modules.

The phones display a default background picture. You can configure a specified background picture or design, such as a company logo, or you can import a set of custom images that users can choose from.

Polycom phones support JPEG, BMP, and PNG image file formats; progressive/multi-scan JPEG images are not supported.

Maximum Image Size

Refer to the following table for the maximum image size supported for each VVX phone.

For detailed instructions on adding a graphic display to a VVX phone, see the *Polycom VVX Business Media Phones User Guide*.

Maximum Phone Screen Image Size

Phone	Screen Size
VVX 250 business IP phone	320x240 pixels
VVX 300 series business media phones	208x104 pixels (Grayscale)
VVX 550 business IP phone	320x240 pixels
VVX 400 series business media	320x240 pixels
VVX 450 business IP phones	480x272 pixels
VVX 500 series business media phones	320x240 pixels
VVX 600 series business media phones	480x272 pixels
VVX 1500 business media phones	800x480 pixels
VVX Color Expansion Module	272x480 pixels

Graphic Display Background Parameters

The configured background image displays across the entire phone screen, and the time, date, line and key labels display over the background.

If you want the background image to display more visibly from behind line key labels, use `up.transparentLines` to render line key labels transparent—this option is available only on the VVX 500/501 and 600/601 business media phones.

Use the parameters in the following table to configure graphic display background on VVX business IP phones, VVX business media phones and connected expansion modules.

Graphic Display Background Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	bg.background.enabled	0 (default) - The user cannot set the background image of the phone screen and the background image option is not available on the phone menu or in the Web Configuration Utility when logged in as a user. In addition, the icon to set the displayed image as a background in the picture frame menu does not display. 1 - The user can set the background image on the phone screen from the phone menu or when logged into the Web Configuration Utility.	No
features.cfg	bg.color.bm.x.em.name	Specify the name of the Expansion Module (EM) background image file including extension with a URL or file path of a BMP or JPEG image. Note: If the file is missing or unavailable, the built-in default solid pattern is displayed.	No
features.cfg	bg.color.bm.x.name	Specify the name of the phone screen background image file including extension with a URL or file path of a BMP or JPEG image. Note: If the file is missing or unavailable, the built-in default solid pattern is displayed.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	bg.color.selection	Specify which type of background (w) and index (x) for that type is selected on reboot. The default selection is 1,1 the first solid background. Use w=1 and x=1 (1,1) to select the built-in image. Use w=2 and x= 1 to 4 to select one of the four `solid` backgrounds. Use w=3 and x= 1 to 6 to select one of the six background `bm` images You can set backgrounds for specific phone models by adding the model name, for example: `bg.color.VVX500.selection`, `bg.color.VVX1500.selection` Note that although the VVX 300 series phones use a grayscale background, you can use this parameter to set the background. 1,1 (default) w,x	No
features.cfg	up.transparentLines	Enable or disable transparent line key labels on the VVX 500/501 and 600/601.	
features.cfg	up.transparentLines	0 (Default) - Line keys block display of the background image. 1 - Line keys are transparent and allow the background image to display behind the line labels. Applies only to the VVX 500/501 and 600/601 business media phones.	No

Digital Picture Frame

On VVX 401/411, 500/501, 600/601, and 1500 business media phones, and VVX 250, 350, and 450 business IP phones, you can display a slide show of images stored on a USB drive on the phone's idle screen.

For images to display, the images must be saved in JPEG, BMP, or PNG format on the top directory of a USB device that is attached to the phone. The phone can display a maximum image size of 9999x9999 pixels and a maximum of 1000 images.

Although 9999x9999 images and progressive/multi-scan JPEG images are supported, the maximum image size that can be downloaded is restricted by the available memory in the phone.

You can access the digital picture frame on the web using PicFrame:// URL.

Digital Picture Frame Parameters

The parameters you can configure are listed in the following table.

Digital Picture Frame Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.pictureFrame.enabled`	For VVX 401/411, 500/501, 600/601, and 1500 business media phones, and VVX 250, 350, and 450 business IP phones. 1 (default) - Enable the digital picture frame. 0 - Disable the digital picture frame.	Yes
`features.cfg`	`up.pictureFrame.folder`	Path name for images. NULL (Default) - Images stored in the root folder on the USB flash drive are displayed. string - 0 to 40 characters Example: If images are stored in the /images/phone folder on the USB flash drive, set this parameter to `images/phone` . For the VVX 500/501, 600/601, and 1500 only.	No
`features.cfg`	`up.pictureFrame.timePerImage`	For the VVX 500/501, 600/601, and 1500 only. The number of seconds to display each picture frame image. Range is 3 to 300 seconds. 5 (Default)	No

Background Image Lock

By default, users can set a background image for their phones using the phone, a USB drive attached to the phone, or the Web Configuration Utility.

You can disable the user's ability to set images as a background when viewing images on a USB attached to the phone.

Disabling this feature removes the following options for users:

- Access to the Background menu on the phone

- The Set Background icon to set a background from an image on a USB drive attached to the phone
- The Background menu option in the Preferences menu in the Web Configuration Utility

Phone Languages

All phones support the following languages: Arabic, Simplified Chinese, Traditional Chinese, Danish, Dutch,English, French, German, Italian, Japanese, Korean, Norwegian, Polish, Brazilian Portuguese, Russian, Slovenian, International Spanish, and Swedish.

Each language is stored as a language file in the **VVXLocalization** folder, which is included with the Polycom UC Software package. If you want to edit the language files, you must use a Unicode-compatible XML editor such as XML Notepad 2007 and familiarize yourself with the guidelines on basic and extended character support.

At this time, the updater is available in English only.

Phone Language Parameters

You can select the language that displays on the phone using the parameters in the following table.

Phone Language Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`	`device.spProfile`	Set the default language that displays on the phone. NULL (default) - The default language is an empty string (lcl.ml.lang=""), which is English. DT - The default language is German (lcl.ml.lang="DTGerman_Germany").	No
`site.cfg`	`lcl.ml.lang`	Null (default) - Sets the phone language to US English. String - Sets the phone language specified in the `lcl.ml.lang.menu.x.label` parameter.	No
`site.cfg`	`lcl.ml.lang.menu.x`	Specifies the dictionary files for the supported languages on the phone. Null (default) String Dictionary files must be sequential. The dictionary file cannot have caps, and the strings must exactly match a folder name of a dictionary file.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	lcl.ml.lang.menu.x.label	Specifies the phone language menu label. The labels must be sequential. Null (default) String	No

Multilingual Parameters

The multilingual parameters listed in the following table are based on string dictionary files downloaded from the provisioning server.

These files are encoded in XML format and include space for user-defined languages.

Multilingual Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	lcl.ml.lang.charset	Provides the language character set. Null (default) String	Yes
site.cfg	lcl.ml.lang.clock.x.24HourClock	Overrides the lcl.datetime.time.24HourClock parameter. 1 (default) - Displays the time in 24-hour clock mode. 0 - Does not display the time in 24-hour clock mode.	No
site.cfg	lcl.ml.lang.clock.x.dateTop	Overrides the lcl.datetime.date.dateTop parameter. 1 (default) - Displays date above time. 0 - Displays date below time.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	lcl.ml.lang.clock.x.format	Overrides the lcl.datetime.date.format parameterto display the day and date . "D,dM" (default) String The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time. For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday.	No
site.cfg	lcl.ml.lang.clock.x.longFormat	Overrides the lcl.datetime.date.longFormat parameter. 1 (default) - Displays the day and month in long format (Friday/November). 0 - Displays the day and month in abbreviated format (Fri/Nov).	No
site.cfg	lcl.ml.lang.japanese.font.enabled	0 (default) - The phone does not display the Japanese Kanji character font. 1 - The phone displays the Japanese Kanji character font. This parameter applies to Polycom Trio, VVX 400, 401, 410, 411, 500, 501, 600, 601, and 1500.	Yes
region.cfg	lcl.ml.lang.list	Displays the list of languages supported on the phone. All (default) String	Yes

The basic character support includes the Unicode character ranges listed in the next table.

Unicode Ranges for Basic Character Support

Name	Range
C0 Controls and Basic Latin	U+0000 - U+007F
C1 Controls and Latin-1 Supplement	U+0080 - U+00FF
Cyrillic (partial)	U+0400 - U+045F

Add a Language for the Phone Display and Menu

Use the multilingual parameters to add a new language to your provisioning server directory to display on the phone screen and menu.

Procedure

1. Create a new dictionary file based on an existing one.
2. Change the strings making sure to encode the XML file in UTF-8 but also ensuring the UTF-8 characters chosen are within the Unicode character ranges indicated in the tables below.
3. Place the file in an appropriately named folder according to the format `language_region` parallel to the other dictionary files under the VVXLocalization folder on the provisioning server.
4. Add an `lcl.ml.lang.clock.menu.x` parameter to the configuration file.
5. Add `lcl.ml.lang.clock.x.24HourClock, lcl.ml.lang.clock.x.format, lcl.ml.lang.clock.x.longFormat` , and `lcl.ml.lang.clock.x.dateTop` parameters and set them according to the regional preferences.
6. (Optional) Set `lcl.ml.lang` to be the new `language_region` string.

Pinyin Text Input

Pinyin is the phonetic system used to transcribe Mandarin pronunciation of Chinese into Latin characters.

The pinyin text input feature on VVX phones uses Nuance XT9 Smart Input to enable users to enter Chinese characters into text input fields using the phone's dial pad keys or on-screen keyboard. The pinyin text input feature is not supported on VVX 101, 150, and 201 phones.

To enable users to use the pinyin text input feature on Polycom phones, download a license key to the user's phone.

Note: For complete information on the pinyin text input feature, see the *Polycom VVX Business Media Phones User Guide*.

Hide the MAC Address

You can configure the phone to hide MAC address displayed on the phone. When you enable this feature, users cannot view or retrieve the MAC address from the phone. The MAC address is available to administrators only.

Hide MAC Address Parameters

The following table lists parameters that configure the display of MAC address.

Hide MAC Address Parameters

Template	Parameter	Permitted Values	Change Causes Reboot or Restart
`device.cfg`	`device.mac.hide.set`	Allows you to use the `device.mac.hide` parameter to control the display of MAC address information of VVX phones to users. Null (default) 0 – Disables the ability to control the display of MAC address information. 1 – Enables the ability to control the display of Mac address information.	No
`device.cfg`	`device.mac.hide`	0 (default) – MAC information displays. 1 – MAC address information is hidden.	No

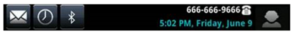
Digital Phone Label

You can configure the Digital Phone Label feature to display the complete registration line address in the status bar of the phone's screen.

The following phones support the Digital Phone Label feature:

- VVX 3xx/4xx/5xx/6xx business media phones.
- VVX 250, 350, and 450 business IP phones.

The following illustrates a successfully configured registration line in the address bar.



Digital Phone Label Parameters

When enabled, the `lcl.status.LineInfoAtTopText` parameter provides the text to be displayed in the status bar of the phone. You can enable the feature by setting the value of the `lcl.status.LineInfoAtTop` parameter to 1.

Registration Line Address Bar Parameter

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`lcl.status.LineInfoAtTop`	0 (default) - Does not display the registration line address in the status bar of the phone's screen. 1 - Display the complete registration line address in the status bar of the phones screen. Set the text to be displayed using `lcl.status.LineInfoAtTopText`.	No
`site.cfg`	`lcl.status.LineInfoAtTopText`	Provides the text be displayed on the phones screen. Null (default) 0 - 14 digits. Use of characters is permitted but might lead to truncation. You must enable `lcl.status.LineInfoAtTop` to configure this parameter.	No

Unique Line Labels for Registration Lines

You can configure unique labels on line keys for registration lines.

You must configure multiple line keys on the phone for a registration in order to configure unique line labels. For example, you can set different names to display for the registration 4144 that displays on four line keys.

If you configure the line to display on multiple line keys without a unique label assigned to each line, the lines are labeled automatically in numeric order. For example, if you have four line keys for line 4144 labeled Polycom, the line keys are labeled as 1_Polycom, 2_ Polycom, 3_ Polycom, and 4_ Polycom. This also applies to lines without labels.

Unique Line Labels for Registration Lines Parameters

When using this feature with the parameter `reg.x.label.y` where x=2 or higher, multiple line keys display for the registered line address.

Configure Unique Line Labels

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`reg.x.line.y.label`	Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when `up.cfgUniqueLineLabel=1` . If `reg.x.linekeys=1` , this parameter does not have any effect. x = the registration index number starting from 1. y = the line index from 1 to the value set by `reg.x.linekeys` . Specifying a string sets the label used for the line key registration on phones with multiple line keys. If no parameter value is set for `reg.x.line.y.label` , the phone automatically numbers multiple lines by prepending “<y>_” where <y> is the line index from 1 to the value set by `reg.x.linekeys` . • The following examples show labels for line 1 on a phone with user registration 1234, where `reg.x.linekeys=2` : ◦ If no label is configured for registration, the labels are “1_1234” and “2_1234”. ◦ If `reg.1.line.1.label=Polycom` and `reg.1.line.2.label=VVX` , the labels display as ‘Polycom' and ‘VVX'.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.cfgLabelElide	Controls the alignment of the line label. When the line label is an alphanumeric or alphabetic string, the label aligns right. When the line label is a numeric string, the label aligns left. None (Default) Right Left	No
features.cfg	up.cfgUniqueLineLabel	Allow unique labels for the same registration that is split across multiple line keys using reg.X.linekeys. 0 (Default) - Use the same label on all line keys. 1 - Display a unique label as defined by reg.X.line.Y.label. If reg.X.line.Y.label is not configured, then a label of the form <integer>_ will be applied in front of the applied label automatically.	No

LED Indicators

The LED indicators on VVX business IP phones, VVX business media phones and expansion modules alert users to the different states of the phone and remote contacts.

You can turn LED indicators on or off, and set the pattern, color, and duration of a pattern for all physical keys on the phones.

You can set the pattern, color, and duration for the following LED indicators on VVX phones:

- Line keys
- Message Waiting Indicator (MWI)
- Headset key (excluding VVX 101, 150, and 201)

LED Behavior Parameters

The LED pattern parameters listed in the following table configure the pattern state, color, and duration of the LED indicators and the pattern types on Polycom devices and expansion modules.

For each parameter, specify x, y, and a permitted value:

- Specify an LED pattern using the LED pattern parameters.
- For x, specify an LED pattern type.
- For y, specify the step in the LED pattern with a number between 1-20.

Use the parameters in the following table lists to set the pattern state, color, and duration of the LED indicators on VVX phones and expansion modules.

LED Behavior Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	ind.pattern.x.step.y.state	0 (default) - Turn off the LED indicator. 1 - Turn on the LED indicator.	No
features.cfg	ind.pattern.x.step.y.color	Specify the color of the LED indicator. The Yellow value is available only for VVX 300 and 400 series phones. The Yellow value is not available for line key indicators on VVX 101 and 201 phones or VVX Expansion Modules. Red (default) Green Yellow	No
features.cfg	ind.pattern.x.step.y.duration	Specify the duration of the pattern in milliseconds. 0 (default) 0 - 32767	No

LED Indicator Pattern Types

Enter one of the values in the following table to indicate the LED indicator pattern type.

LED Indicator Pattern Type

Pattern Type	Function
powerSaving	Sets the behavior for Message Waiting Indicator when the phone is in Power Saving mode.
active	Sets the pattern for line keys during active calls.
on	Turns on the LED indicator pattern.
off	Turns off the LED indicator pattern.
offering	Sets the pattern for line keys during incoming calls.
flash	Sets the pattern for line keys during held calls and the Message Waiting Indicator when there are unread voicemail messages.
lockedOut	Sets the pattern for line keys when a remote party is busy on a shared line.

Pattern Type	Function
FlashSlow	Sets the pattern for the Headset key when Headset Memory Mode is enabled.
held	Sets the pattern for line keys during a held call.
remoteBusyOffering	Sets the pattern for line keys for monitored BLF contacts when the BLF is in an active call and receives a new incoming call.

LED Pattern Examples

This section includes example configurations you can use to set the patterns of LED indicators for VVX phones and expansion modules.

Example: Disable the Headset Key LED in Headset Memory Mode

By default, the Headset key on all VVX phones, excluding VVX 101 and 201, glows green for analog headsets and blue for USB headsets.

The Headset key also flashes by default if Headset Memory Mode is enabled.

The default configuration is listed in the following table.

Headset Key Indicator Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	ind.pattern.flashSlow.step.1.state	1 (default) - Turns on the LED indicator for Headset key. 0 - Turns off the LED indicator for Headset key.	No
debug.cfg	ind.pattern.flashSlow.step.1.duration	Specify the duration of the pattern in milliseconds for Headset key LED. 100 (default) 0 - 32767	No
debug.cfg	ind.pattern.flashSlow.step.2.state	0 (default) - Turns off the LED indicator for the specified duration of the pattern. for Headset key. 1 - Turns on the LED indicator for the specified duration of the pattern for Headset key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	`ind.pattern.flashSlow.step.2.duration`	Set the duration of the pattern in milliseconds to which the LED indicator is turned off for Headset key. 2900 (default) 0 - 32767 After the specified duration, the pattern repeats.	No
debug.cfg	pres.idleTimeoutoffHours.period	The number of minutes to wait while the phone is idle during off hours before showing the Away presence status. 15 (default) 1 - 600	
debug.cfg	pres.idleTimeout.officeHours.periods	The number of minutes to wait while the phone is idle during office hours before showing the Away presence status 15 (default) 1 - 600	

You can disable and turn off the flash pattern for the Headset key when Headset Memory Mode is enabled.

Procedure

1. Set the parameter `ind.pattern.flashSlow.step.1.state` to 0.

Example: Set an LED Pattern for Active Calls

In the following example, during an active call, the line key alternates green and red.

Procedure

1. Configure the pattern as follows:
    - `ind.pattern.active.step.1.color=` "Green"
    - `ind.pattern.active.step.1.state=` "1"
    - `ind.pattern.active.step.1.duration=` ”1000”
    - `ind.pattern.active.step.2.color=` "Red"
    - `ind.pattern.active.step.2.state=` "1"
    - `ind.pattern.active.step.2.duration=` ”1000”

Example: Turn Off the Message Waiting Indicator in Power Saving Mode

When Power Saving mode is enabled, the screen darkens, and the MWI flashes red.

By default, the powerSaving pattern has two steps before the pattern is repeated: a quick on period and then a long off period.

By default, the following parameters set the behavior of the MWI during Power Saving mode.

Power Saving Mode Indicator Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	ind.pattern.powerSaving.step.1.state	1 (default) - Turns on the LED indicator for power saving mode. 0 - Turns off the LED indicator for power saving mode.	No
debug.cfg	ind.pattern.powerSaving.step.1.duration	Specify the duration of the pattern in milliseconds for power saving mode. 100 (default) 0 - 32767	No
debug.cfg	ind.pattern.powerSaving.step.2.state	0 (default) - Turns off the LED indicator for the specified duration of the pattern for power saving mode. 1 - Turns on the LED indicator for the specified duration of the pattern for power saving mode.	No
debug.cfg	ind.pattern.powerSaving.step.2.duration ="2900"	Set the duration of the pattern in milliseconds for power saving mode to which the LED indicator is turned off. 2900 (default) 0 - 32767 After the specified duration, the pattern repeats.	No
debug.cfg	pres.idleTimeoutoffHours.period	The number of minutes to wait while the phone is idle during off hours before showing the Away presence status. 15 (default) 1 - 600	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	pres.idleTimeout.officeHours.periods	The number of minutes to wait while the phone is idle during office hours before showing the Away presence status 15 (default) 1 - 600	

You can turn off the MWI or change the duration of the pattern steps.

Procedure

1. Set the parameter `ind.pattern.powerSaving.step.1.state` to 0.

Example: Change the Color of Line Key Indicators for Incoming Calls

When a phone receives an incoming call, the line key LED indicator flashes green.

You can change the color of the indicator to Yellow or Red for incoming calls.

By default, the following parameters set the behavior of the line key LED indicators for incoming calls.

Incoming Call Indicator Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`debug.cfg`	`ind.pattern.offering.step.1.state`	1 (default) - Turns on the LED indicator for incoming call. 0 - Turns off the LED indicator for incoming call.	No
debug.cfg	`ind.pattern.offering.step.1.duration`	Specify the duration of the pattern in milliseconds for incoming call. 5000 (default) 0 - 32767	No
debug.cfg	`ind.pattern.offering.step.1.color`	Sets the color of the LED indicator for the pattern for incoming call. Green (default) Yellow Red	No
debug.cfg	`ind.pattern.offering.step.2.state`	0 (default) - Turns off the LED indicator for incoming call in step 2. 1 - Turns on the LED indicator for incoming call in step 2.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	`ind.pattern.offering.step.2.duration`	Specify the duration of the pattern in milliseconds for incoming call in step 2. 5000 (default) 0 - 32767	No
debug.cfg	`ind.pattern.offering.step.2.color`	Sets the color of the LED indicator for the pattern for incoming call in step 2. Yellow (default) Green Red If `ind.pattern.offering.step.2.state=0` , this parameter value is ignored.	No
debug.cfg	`ind.pattern.offering.step.3.state`	1 (default) - Turns on the LED indicator for incoming call in step 3. 0 - Turns off the LED indicator for incoming call in step 3.	No
debug.cfg	`ind.pattern.offering.step.3.duration`	Specify the duration of the pattern in milliseconds for incoming call in step 3. 5000 (default) 0 - 32767	No
debug.cfg	`ind.pattern.offering.step.3.color`	Sets the color of the LED indicator for the pattern for incoming call in step 3. Red (default) Green Yellow	No
debug.cfg	`pres.idleTimeoutoffHours.period`	The number of minutes to wait while the phone is idle during off hours before showing the Away presence status. 15 (default) 1 - 600	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	`pres.idleTimeout.officeHours.periods`	The number of minutes to wait while the phone is idle during office hours before showing the Away presence status 15 (default) 1 - 600	

Procedure

1. Set the parameter `ind.pattern.offering.step.1.color` to Yellow.

Capture Your Device's Current Screen

You can capture your phone or expansion module's current screen.

VVX business IP phones and the Polycom Trio solution do not support expansion modules.

Before you can take a screen capture, you must provide power and connect the expansion module to a phone, and enable the phone's web server using the parameter `httpd.enabled`.

Procedure

1. In the `sip-interop.cfg` template, locate the parameter `up.screenCapture.enabled` .

   You can add the sip-interop.cfg template to the CONFIG-FILES field of the master configuration file, or copy the parameter to an existing configuration file.
2. Set the value to `1` and save the configuration file.
3. On the device, go to **Settings** > **Basic** > **Preferences** > **Screen Capture**.

   Note you must repeat step 3 each time the device restarts or reboots.
4. Locate and record the phone's IP address at **Status** > **Platform** > **Phone** > **IP Address**.
5. Set the phone to the screen you want to capture.
6. In a web browser address field, enter `https://<phoneIPaddress>/captureScreen` where `<phoneIPaddress>` is the IP address you obtained in step 5.

   The web browser displays an image showing the phone's current screen. You can save the image as a BMP or JPEG file.

Capture Your Device's Current Screen Parameters

Use the following parameters to get a screen capture of the current screen on your device.

Device's Current Screen Parameters

Template	Parameter	Permitted Values	Change Causes Reboot or Restart
`sip-interop.cfg`	`up.screenCapture.enabled`	0 (Default) - The Screen Capture menu is hidden on the phone. 1 - The Screen Capture menu displays on the phone. When the phone reboots, screen captures are disabled from the Screen Capture menu on the phone.	Yes
`sip-interop.cfg`	`up.screenCapture.value`	0 (Default) - The Screen Capture feature is disabled. 1 - The Screen Capture feature is enabled.	No

User Profiles

Topics:

- User Profile Parameters
- Remotely Logging Out Users
- Authentication of User Profiles

When you set up user profiles, you enable users to access their personal phone settings, including their contact directory, speed dials, and other phone settings from any phone on the network.

This feature is particularly useful for remote and mobile workers who do not have a dedicated work space and conduct their business in more than one location. This feature is also useful if an office has a common conference phone from which multiple users need to access their personal settings.

Note: You can configure all company phones so that anyone can call authorized and emergency numbers when not logged in to a phone. For more information, see `dialplan.routing.emergency.outboundIdentity`.

If you set up the user profile feature, a user can log in to a phone by entering their user ID and password. The default password is 123. If the user profile feature is set up on your company's phones, users can:

- Log in to a phone to access their personal phone settings.
- Place a call to an authorized number from a phone that is in the logged out state.
- Change their user password.
- Log out of a phone after they finish using it.

If a user changes any settings while logged in to a phone, the settings save and display the next time the user logs in to another phone. When a user logs out, the user's personal phone settings are no longer displayed.

Related Links

Local Contact Directory File Size Parameters on page 626

User Profile Parameters

Before you configure user profiles, you must complete the following:

- Create a phone configuration file, or update an existing file, to enable the feature's settings.
- Create a user configuration file in the format <user>.cfg to specify the user's password, registration, and other user-specific settings that you want to define.

  Important: You can reset a user's password by removing the password parameter from the override file. This causes the phone to use the default password in the <user>.cfg file.

When you set up the user profile feature, you can set the following conditions:

- If users are required to always log in to use a phone and access their personal settings.

- If users are required to log in and have the option to use the phone as is without access to their personal settings.
- If users are automatically logged out of the phone when the phone restarts or reboots.
- If users remain logged in to the phone when the phone restarts or reboots.

Use the parameters in the following table to enable users to access their personal phone settings from any phone in the organization.

User Profile Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.login.automaticLogout	Specify the amount of time before a non-default user is logged out. 0 minutes (default) 0 to 46000 minutes	No
site.cfg	prov.login.defaultOnly	0 (default) - The phone cannot have users other than the default user. 1 - The phone can have users other than the default user.	No
site.cfg	prov.login.defaultPassword	Specify the default password for the default user. NULL (default)	No
site.cfg	prov.login.defaultUser	Specify the name of the default user. If a value is present, the user is automatically logged in when the phone boots up and after another user logs out. NULL (default)	No
site.cfg	prov.login.enabled	0 (default) - The user profile is disabled. 1 - The user profile feature is enabled.	No
site.cfg	prov.login.localPassword.hashed	0 (default) - The user's local password is formatted and validated as clear text. 1 - The user's local password is created and validated as a hashed value.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.login.localPassword	Specify the password used to validate the user login. The password is stored either as plain text or as an encrypted SHA1 hash. 123 (default)	No
site.cfg	prov.login.persistent	0 (default) - Users are logged out if the handset reboots. 1 - Users remain logged in when the phone reboots.	No
site.cfg	prov.login.required	0 (default) - The user does not have to log in. 1 - The user must log in when the login feature is enabled.	No
site.cfg	prov.login.useProvAuth	0 (default) - The phone do not user server authentication. 1 - The phones use server authentication and user login credentials are used as provisioning server credentials.	No
site.cfg	voIpProt.SIP.specialEvent.checkSync.downloadCallList	0 (default) - The phone does not download the call list for the user after receiving a checksync event in the NOTIFY. 1 - The phone downloads the call list for the user after receiving a checksync event in the NOTIFY.	No

Remotely Logging Out Users

Note that if an unexpected reboot occurs while a user is logged in, the user is not logged out and the phone returns to the user profile after reboot.

If a user is not logged out from a phone and other users are not prevented from logging in, the user can ask the administrator to log out remotely. Administrators can log out a user remotely with a checksync event in the NOTIFY by setting the parameter profileLogout=remote .

Authentication of User Profiles

When using the User Profiles feature, you can authenticate users with phone-based or server-based authentication methods. Phone-based authentication authenticates credentials entered by the user against the crednetials in the `<user>.cfg` file. Server-based authentication passes user credentials to the provisioning server for authentication.

Server Authentication of User Profiles

Instead of phone-based authentication of user profiles, you can configure server authentication.

When you enable server authentication, you set up user accounts on the provisioning server and each user can authenticate their phone by entering correct server credentials.

The phone downloads log files app.log and boot.log from the generic profile on the provisioning server regardless of user logins.

Create a Generic Profile Using Server Authentication

Create a generic profile and generic credentials on the provisioning server when a user is not logged into the phone.

If you enable server authentication of user profiles, the following parameters do not apply and you do not need to configure them:

- `prov.login.defaultUser`
- `prov.login.defaultPassword`
- `prov.login.defaultOnly`
- `prov.login.localPassword`
- `prov.login.localPassword.hashed`

Procedure

1. On the server, create an account and directory for the generic profile, for example, '*Generic_Profile*'.
2. In the *Generic_Profile* directory, create a configuration file for a generic profile the phone uses by default, for example, *genericprofile*.cfg.
3. In *genericprofile*.cfg, include registration and server details and set all phone feature parameters.

   You must set the following parameters to use server authentication:

   - `prov.login.enabled="1"`
   - `prov.login.useProvAuth="1"`
   - `prov.login.persistent="1"` Note that if you enable `prov.login.enabled=1` and do not enable `prov.login.useProvAuth=0` , users are authenticated by a match with credentials you store in the user configuration file <*user*>.cfg.
4. Create a master configuration file 000000000000.cfg for all the phones, or a <*MACAddress*>.cfg for each phone, and add *genericprofile*.cfg to the CONFIG_FILES field.
5. Set the provisioning server address and provisioning server user name and password credentials for the generic user account on the phone at **Settings** > **Advanced** > **Provisioning Server** details and inform users of their user profile credentials.

The following override files are uploaded to the generic profile directory:

- Log files
- Phone menu settings
- Web Configuration Utility settings

- Call logs
- Contact directory file

Create a User Profile Using Server Authentication

Create a user profile in the Home directory of each user with a user-specific configuration file that you store on the provisioning server with a unique name as well as user-specific files such as settings, directory, and call lists.

When a user logs in with credentials, the phone downloads the user profile from the provisioning server. When the user logs out, the phone downloads the default user profile using the generic credentials.

Procedure

1. On the server, create an account and a directory for each user, for example, '*User1*', '*User2*”.
2. In each user directory, create a configuration file for each user, for example, *User1*.cfg, *User2*.cfg, that contains the user's registration details and feature settings.

The following override files are uploaded to the generic profile account on the server:

- Log files
- Web Configuration Utility settings

The following override files are uploaded to the user profile account on the server:

- Phone menu settings
- Contact directory file

Phone Authentication of User Profiles

You can create default credentials and user profiles without use of server authentication.

Create Default Credentials and a Profile for a Phone

You can choose to define default credentials for a phone, which the phone uses to automatically log itself in each time an actual user logs out or the phone restarts or reboots.

When the phone logs itself in using the default login credentials, a default phone profile displays, and users retain the option to log in and view their personal settings.

You can create a new phone configuration file for the default profile, then add and set the attributes for the feature. You can also update an existing phone configuration file to include the user login parameters you want to change.

Important: Polycom recommends that you create a single default user password for all users.

Procedure

1. Create a site.cfg file for the phone and place it on the provisioning server.

   You can base your file on the sample configuration template in your software package. To find the file, navigate to <provisioning server location>/Config/site.cfg.
2. In site.cfg, open the <prov.login/> attribute, then add and set values for the user login attributes.

Create a User Configuration File

Create a configuration file for each user that you want to enable to log in to the phone.

The name of the file should specify the user's login ID. In the file, specify any user-specific settings that you want to define for the user.

If a user updates their password or other user-specific settings on the phone, the updates are stored in <user>-phone.cfg, not <MACaddress>-phone.cfg.

If a user updates their contact directory while logged in to a phone, the updates are stored in <user>-directory.xml. Directory updates display each time the user logs in to a phone. For certain phones (for example, the VVX 1500 phone), an up-to-date call lists history is defined in <user>-calls.xml. This list is retained each time the user logs in to their phone. The following is a list of configuration parameter precedence (from first to last) for a phone that has the user profile feature enabled:

- <user>-phone.cfg
- Web Configuration Utility
- Configuration files listed in the master configuration file (including <user>.cfg)
- Default values

Note: To convert a phone-based deployment to a user-based deployment, copy the <MACaddress>-phone.cfg file to <user>-phone.cfg and copy phoneConfig<MACaddress>.cfg to <user>.cfg.

Procedure

1. On the provisioning server, create a user configuration file for each user.
2. Name each file the ID the user will use to log in to the phone.

   For example, if the user's login ID is user100, the name of the user's configuration file is user100.cfg.
3. In each <user>.cfg file, you are required to add and set values for the user's login password.
4. Add and set values for any user-specific parameters, such as:
   - Registration details such as the number of lines the profile displays and line labels.
   - Feature settings such as microbrowser settings).

   Caution: If you add optional user-specific parameters to <user>.cfg, add only those parameters that will not cause the phone to restart or reboot when the parameter is updated.

Directories and Contacts

Topics:

You can configure phones with a local contact directory and link contacts to speed dial buttons.

Additionally, call logs stored in the Missed Calls, Received Calls, and Placed Calls call lists let you view user phone events like remote party identification, time and date of call, and call duration. This section provides information on contact directory, speed dial, and call log parameters you can configure on your Polycom phone.

Local Contact Directory

Polycom phones feature a contact directory file you can use to store frequently used contacts.

The UC Software package includes a template contact directory file named `000000000000-directory~.xml` that is loaded to the provisioning server the first time you boot up a phone with UC Software or when you reset the phone to factory default settings.

When you first boot the phone out of the box or when you reset the phone to factory default settings, the phone looks for contact directories in the following order:

- An internally stored local directory
- A personal `<MACaddress>-directory.xml` file
- A global `000000000000-directory.xml` file when the phone substitutes `<000000000000>` for its own MAC address.

The Contact Directory is the central database for several phone features including speed dial, distinctive incoming call treatment, presence, and instant messaging.

You can configure the phones to hide the Contact Directory and Favorites options from all screens in the user interface on all VVX phones except the VVX 1500 phone. You can also set the local directory as read-only and restrict users from modifying the speed dials only.

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

Related Links

Speed Dials on page 280
Speed Dial Contacts Parameters on page 281
Contact Directory Macros on page 478
Expanded Macros on page 481
Macro Actions on page 479
Prompt Macro Substitution on page 480

Local Contact Directory Parameters

The following parameters configure the local contact directory.

Local Contact Directory Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.local.contacts.maxNum	Set the maximum number of contacts that can be stored in the Local Contact Directory. The maximum number varies by phone model, refer to section ‘Maximum Capacity of the Local Contact Directory'. VVX 101, 150, 201: Default 99 contacts, Maximum 99 contacts VVX 3xx, 4xx, 5xx, 6xx, and business media phones and business IP phones: Default 500 contacts, Maximum 500 contacts Polycom Trio 8800 and 8500: • 2000 (default) • Maximum 3000 contacts	Yes
features.cfg	dir.local.passwordProtected	0 (default) - Disable password protection of the local Contact Directory. 1 - Enables password protection of the local Contact Directory.	No
features.cfg	dir.local.readonly	0 (default) - Disable read only protection of the local Contact Directory. 1 - Enable read-only protection of the local Contact Directory.	No
features.cfg	feature.directory.enabled	0 (default) - The local contact directory is disabled when the Polycom Trio solution Base Profile is set to Lync. 1 - The local directory is enabled when the Polycom Trio solution Base Profile is set to Lync.	No
features.cfg	dir.search.field	Specify whether to search the directory by first name or last name. 0 (default) - Contact directory searches are sorted by contact's last name. 1 - Contact directory searches are sorted by first name.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voIpProt.SIP.specialEvent.checkSync.downloadDirectory	0 (default) - The phone downloads updated directory files after receiving a checksync NOTIFY message. 1 - The phone downloads the updated directory files along with any software and configuration updates after receiving a checksync NOTIFY message. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates. Note: The parameter hotelingMode.type set to 2 or 3 overrides this parameter.	No
features.cfg	dir.local.passwordProtected	Specify whether you are prompted for an Admin/User password when adding, editing, or deleting contacts in the Contact Directory.	No
features.cfg	dir.local.UIenabled	1 (default) – The Directory menus provide access to Favorites/Speed Dial and Contact Directory entries and display the Favorites quick access menu on the Home screen of the VVX 500/501 and 600/601 business media phones. 0 – The local Contact Directory and Favorites/Speed Dial menu entries are not available. The Favorites quick access menu on the Home screen are not available on the VVX 500/501 and 600/601 business media phones. Set to 0 when dir.local.readOnly is set to 1 to add speed dials and macros on the phone and prevent user modification. If your call control platform provides direct contact integration and you want to prevent any access to the local directory, set feature.directory.enabled=0.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.pauseAndWaitDigitEntryControl.enabled`	1 (default) - Enable processing of control characters in the contact phone number field. When enabled, ',' or 'p' control characters cause a one second pause. For example, ',' or 'p' control characters cause a one second pause. ';' or 'w' control character cause a user prompt that allows a user-controlled wait. Subsequent digits entered to the contact field are dialed automatically. 0 - Disable processing of control characters.	No
`features.cfg`	`up.regOnPhone`	0 (default) – Contacts you assign to a line key display on the phone in the position assigned. 1 – Contacts you assign to a line key are pushed to the attached expansion module.	Yes

Related Links

Distinctive Incoming Call Treatment on page 305
Instant Messaging on page 399
Presence Status on page 306

Maximum Capacity of the Local Contact Directory

The following table lists the maximum number of contacts and maximum file size of the local Contact Directory for each phone.

To conserve phone memory, use the parameter `dir.local.contacts.maxNum` to set a lower maximum number of contacts for the phones.

Maximum File Size and Number of Contacts

Phone	Maximum File Size	Maximum Number of Contacts in File
VVX 101, 150, 201	Not available	99
VVX 3xx series	4MB	500
VVX 4xx series	4MB	500
VVX 500/501 and 600/601	4MB	500

Phone	Maximum File Size	Maximum Number of Contacts in File
VVX 1500	102400 bytes Non-volatile: 100KB	9999
SoundStructure VoIP Interface	Not applicable	Not applicable

Creating Per-Phone Directory Files

To create a per-phone, personal directory file, replace <**000000000000**> in the global file name with the phone's MAC address: **<MACaddress** > **-directory.xml**.

Any changes users make to the contact directory from the phone are stored on the phone drive and uploaded to the provisioning server in the personal directory (**<MACaddress** > **-directory.xml**) file, which enables you to preserve a contact directory during reboots.

To create a global directory file that you can use to maintain the directory for all phones from the provisioning server, remove the tilde (~) from the template file name **000000000000-directory.xml**. When you update the global directory file on the provisioning server, the updates are downloaded onto the phone and combined with the phone specific directory.

Maintaining Per-Phone Directory Files

Using the parameter `voIpProt.SIP.specialEvent.checkSync.downloadDirectory` , you can configure the phones to download updated directory files. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates.

Any changes to either the global or personal directory files are reflected in the directory on the phone after a restarts. When merging the two files, the personal directory always takes precedence over the changes in the global directory. Thus, if a user modifies a contact from the global directory, the contact is saved in the personal directory file, and the contact from the global directory is ignored when the files are next uploaded.

The phone requests both the per-phone <MACaddress>-directory.xml and global contact directory 000000000000-directory.xml files and merges them for presentation to the user. If you created a per-phone <MACaddress>-directory.xml for a phone, and you want to use the 000000000000-directory.xml file, add the 000000000000-directory.xml file to the provisioning server and update the phone's configuration.

Note: You can duplicate contacts in the Contact Directory on phones registered with the GENBAND server.

Note: To avoid users accidentally deleting the definitions in the contact directory, make the contact directory file read only.

Speed Dials

You can link entries in the local contact directory to speed dial contacts to line keys on the Home or Lines screen to enable users to place calls quickly using dedicated speed dial buttons.

The number of supported speed dial entries varies by phone model

Speed Dial Index Ranges

Phone Model	Range
VVX 101, 150, 201	1 - 99
VVX 250, 300 series, 400 series, 500 series, and 600 series	1 - 500
VVX 1500	1 - 9999 The maximum number may be limited by the phone's available memory.
SoundStructure VoIP Interface	Not applicable.

Related Links
Local Contact Directory on page 276

Speed Dial Contacts Parameters

After setting up your per-phone directory file (**<MACaddress** > **-directory.xml**), enter a number in the speed dial `<sd>` field to display a contact directory entry as a speed dial contact on the phone. Speed dial entries automatically display on unused line keys on the phone and are assigned in numerical order.

On some call servers, enabling presence for an active speed dial contact displays that contact's status on the speed dial's line key label.

Use the parameters in the following table, which identifies the directory XML file and the parameters you need to set up your speed dial contacts.

Speed Dial Parameters

Template	Parameter	Permitted Values

Related Links
Local Contact Directory on page 276

Corporate Directory

You can connect phones to a corporate directory server that supports the Lightweight Directory Access Protocol (LDAP), version 3.

After you set up the corporate directory on the phones, users can search for contacts in the directory, place calls to directory contacts, and save entries to the local contact directory on the phone.

Polycom phones support corporate directories that support server-side sorting and those that do not. For servers that do not support server-side sorting, sorting is performed on the phone.

Note: Polycom recommends using corporate directories that have server-side sorting for better performance. Consult your LDAP administrator when making any configuration changes for the corporate directory. For more information on LDAP attributes, see RFC 4510 - Lightweight Directory Access Protocol (LDAP): Technical Specification Road Map

Securely Store LDAP Credentials on VVX Phones

You can enable multiple users to enter their LDAP user credentials directly onto the phone to access the Corporate (LDAP) Directory, and you can enable VVX phones to store those credentials on the phone.

Any LDAP credentials entered on the phone are encrypted and stored on the phone only, and the credentials persist after the phone restarts or reboots.

When this feature is configured for phones with BroadSoft Flexible Seating, the phones can store up to 50 user credentials. If the number of user credentials reaches 50, the user who has the longest inactivity period is removed from the phone when any additional users are added.

Procedure

1. Set the parameter `dir.corp.persistentCredentials` to **1**.

Corporate Directory Parameters

Use the parameters in the following table to configure the corporate directory.

Note that the exact configuration of a corporate directory depends on the LDAP server you use.

Note: For detailed explanations and examples of all currently supported LDAP directories, see *Technical Bulletin 41137*: *Best Practices When Using Corporate Directory on Polycom Phones* at Polycom Engineering Advisories and Technical Notifications.

Use the Corporate Directory

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`dir.corp.address`	Set the IP address or hostname of the LDAP server interface to the corporate directory. Null (default) IP address Hostname FQDN	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.allowCredentialsFromUI.enabled	Enable users to enter LDAP credentials on the phone. 0 (default) – Users are not prompted to enter credentials on the phone when they access the Corporate Directory. 1 – Users are prompted to enter credentials on the phone when accessing the Corporate Directory for the first time. **Note:** Users are only prompted to enter their credentials when credentials are not added through configuration or after a login failure.	No
features.cfg	dir.corp.alt.address	Enter the URL address of the GAB service provided by the server. Null (default) Hostname FQDN	No
features.cfg	dir.corp.alt.attribute.x.filter	Enter a filter to use to set a predefined search string through configuration files. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.attribute.x.label	Enter a label to identify a user. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.attribute.x.name	Enter the name of the parameter to match on the server. Each name must be unique; however, a global address book entry can have multiple parameters with the same name. You can configure up to eight parameters (x = 1 to 8). Null (default) UTF-8 encoding string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.alt.attribute.x.sticky	0 (default) —the filter string criteria for attribute x is reset after a reboot. 1—the filter string criteria is retained through a reboot. If you set an attribute to be sticky (set this parameter to 1), a '*' displays before the label of the attribute on the phone.	No
features.cfg	dir.corp.alt.attribute.x.type	Define how x is interpreted by the phone. Entries can have multiple parameters of the same type. first_name last_name (default) phone_number SIP_address Other—for display purposes only. If the user saves the entry to the local contact directory on the phone, first_name, last_name, and phone_number are copied. The user can place a call to the phone_number and SIP_address from the global address book directory.	No
features.cfg	dir.corp.alt.auth.useLoginCredentials	0 (default) 1	No
features.cfg	dir.corp.alt.autoQuerySubmitTimeout	0 (default) 0 - 60	No
features.cfg	dir.corp.alt.password	Enter the password used to authenticate to the GENBAND server. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.port	Set the port that connects to the server if a full URL is not provided. 0 (default) Null 1 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.alt.protocol	Set a directory protocol used to communicate to the corporate directory. sopi (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.transport	Choose a transport protocol used to communicate to the corporate directory. TCP (default) TLS	No
features.cfg	dir.corp.alt.user	Enter the user name used to authenticate to the GENBAND server. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.viewPersistence	Determine if the results from the last address directory search displays on the phone. 0 (default) 1	No
features.cfg	dir.corp.attribute.x.addstar	Determine if the wild-card character, asterisk(*), is appended to the LDAP query field. 0 1 (default)	Yes
features.cfg	dir.corp.attribute.x.filter	Set the filter string for this parameter, which is edited when searching. Null (default) UTF-8 encoding string	Yes
features.cfg	dir.corp.attribute.x.label	Enter the label that shows when data is displayed. Null (default) UTF-8 encoding string	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.attribute.x.name	Enter the name of the parameter to match on the server. Each name must be unique; however, a global address book entry can have multiple parameters with the same name. You can configure up to eight parameters (x = 1 to 8). Null (default) UTF-8 encoding string	Yes
features.cfg	dir.corp.attribute.x.searchable	Determine whether quick search on parameter x (if x is 2 or more) is enabled or disabled. 0 (default) 1	Yes
features.cfg	dir.corp.attribute.x.sticky	0 (default) —the filter string criteria for attribute x is reset after a reboot. 1—the filter string criteria is retained through a reboot. If you set an attribute to be sticky (set this parameter to 1), a ‘*' displays before the label of the attribute on the phone.	Yes
features.cfg	dir.corp.attribute.x.type	Define how x is interpreted by the phone. Entries can have multiple parameters of the same type. first_name last_name (default) phone_number SIP_address H323_address URL other If the user saves the entry to the local contact directory on the phone, first_name, last_name, and phone_number are copied. The user can place a call to the phone_number and SIP_address from the global address book directory.	Yes
features.cfg	dir.corp.auth.useLoginCredentials	0 (default) 1	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.autoQuerySubmitTimeout	Set the timeout (in seconds) between when the user stops entering characters in the quick search and when the search query is automatically submitted. 0 (default)—there is no timeout and automatic submit is disabled. 0 - 60 seconds	Yes
features.cfg	dir.corp.backGroundSync	Determine if background downloading from the LDAP server is allowed. 0 (default) 1	Yes
features.cfg	dir.corp.backGroundSync.period	Set the time (in seconds) the corporate directory cache is refreshed after the corporate directory feature has not been used for the specified period of time. 86400 (default) 3600 to 604800	Yes
features.cfg	dir.corp.baseDN	Enter the base domain name, which is the starting point for making queries on the LDAP server. Null (default) UTF-8 encoding string	Yes
features.cfg	dir.corp.bindOnInit	Determine if bind authentication is used on initialization. 1 (default) 0	Yes
features.cfg	dir.corp.cacheSize	Set the maximum number of entries that can be cached locally on the phone. 128 (default) 32 to 256 For VVX 101, the permitted values are 32 to 64 where 64 is the default.	Yes
features.cfg	dir.corp.customError	Enter the error message to display on the phone when the LDAP server finds an error. Null (default) UTF-8 encoding string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.domain	0 to 255	No
features.cfg	dir.corp.filterPrefix	Enter the predefined filter string for search queries. (objectclass=person) (default) UTF-8 encoding string	Yes
features.cfg	dir.corp.pageSize	Set the maximum number of entries requested from the corporate directory server with each query. 32 (default) 8 to 64 For VVX 101, the permitted values are 8 to 32 where 16 is the default.	Yes
features.cfg	dir.corp.password	Enter the password used to authenticate to the LDAP server. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.persistentCredentials	Set to securely store and encrypt LDAP directory user credentials on the phone. Enable dir.corp.allowCredentialsFromUI.enabled to allow users to enter credentials on the phone. 0 (default) 1 **Note:** If you disable the feature after enabling it, then all the saved user credentials are deleted for all users.	
features.cfg	dir.corp.port	Enter the port that connects to the server if a full URL is not provided. 389 (default for TCP) 636 (default for TLS) 0 Null 1 to 65535	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.querySupportedControlOnInit	Determine if the phone makes an initial query to check the status of the server when booting up. 0 1 (default)	No
features.cfg	dir.corp.scope	sub (default)—a recursive search of all levels below the base domain name is performed. one —a search of one level below the base domain name is performed. base—a search at the base domain name level is performed.	Yes
features.cfg	dir.corp.serverSortNotSupported	0 (default) – The server supports server-side sorting. 1 – The server does not support server-side sorting, so the phone handles the sorting.	No
features.cfg	dir.corp.sortControl	Determine how a client can make queries and sort entries. 0 (default)—leave sorting as negotiated between the client and server. 1—force sorting of queries, which causes excessive LDAP queries and should only be used to diagnose LDAP servers with sorting problems.	Yes
features.cfg	dir.corp.transport	Specify whether a TCP or TLS connection is made with the server if a full URL is not provided. TCP (default) TLS Null	Yes
features.cfg	dir.corp.user	Enter the user name used to authenticate to the LDAP server. Null (default) UTF-8 encoding string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.viewPersistence	0 (default) — the corporate directory search filters and browsing position are reset each time the user accesses the corporate directory. 1— the search filters and browsing position from the previous session are displayed each time the user accesses the corporate directory.	Yes
features.cfg	dir.corp.vlv.allow	Determine whether virtual view list (VLV) queries are enabled and can be made if the LDAP server supports VLV. 0 (default) 1	Yes
features.cfg	dir.corp.vlv.sortOrder	Enter the list of parameters, in exact order, for the LDAP server to use when indexing. For example: sn, givenName, telephoneNumber . Null (default) list of parameters	Yes
features.cfg	feature.contacts.enabled	1 (default) - The Contacts icon displays on the Home screen, the global menu, and in the dialer. 0 - Disable display of the Contacts icon.	No
features.cfg	feature.corporateDirectory.enabled	0 (default) - The corporate directory feature is disabled and the icon is hidden. 1 (default) - The corporate directory is enabled and the icon shows.	No

Call Logs

The phone records and maintains user phone events to a call log, which contains call information such as remote party identification, time and date of the call, and call duration.

The log is stored on the provisioning server as an XML file named <MACaddress>-calls.xml. If you want to route the call logs to another server, use the CALL_LISTS_DIRECTORY field in the master configuration file. All call logs are enabled by default.

The phones automatically maintain the call log in three separate call lists that users can access: Missed Calls, Received Calls, and Placed Calls. Users can clear lists manually on their phones, or delete individual records or all records in a group (for example, all missed calls).

Call Log Parameters

Use the parameters in the following table to configure call logs

Call Log Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg, features.cfg	callLists.collapseDuplicates	Lync Base Profile – 0 (default) Generic Base Profile – 1 (default) 1 – Consecutive incomplete calls to/ from the same party and in the same direction are collapsed into one record in the calls list. The collapsed entry displays the number of consecutive calls. 0 – Each call is logged individually in the calls list.	No
site.cfg, features.cfg	callLists.logConsultationCalls	ync Base Profile – 1 (default) Generic Base Profile – 0 (default) 0 – Consultation calls not joined into a conference call are not logged as separate calls in the calls list. 1 – Each consultation calls is logged individually in the calls list.	No
features.cfg	feature.callList.enabled	1 (default) - Allows you to enable the missed, placed, and received call lists on all phone menus including the Home screen and dial pad. 0 - Disables all call lists. Hiding call lists from the Home screen and dial pad requires UCS 5.4.2 RevAA or higher.	No
features.cfg	feature.callListMissed.enabled	0 (Default) - The missed call list is disabled 1 - The missed call list is enabled. To enable the missed, placed, or received call lists, feature.callList.enabled must be enabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.callListPlaced.enabled	0 (Default) - The placed call list is disabled 1 - The placed call list is enabled. To enable the missed, placed, or received call lists, feature.callList.enabled must be enabled.	No
features.cfg	feature.callListReceived.enabled	0 (Default) - The received call list is disabled 1 - The received call list is enabled. To enable the missed, placed, or received call lists, feature.callList.enabled must be enabled.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.exchangeCallLog.enabled	If Base Profile is: Generic - 0 (default) Skype for Business - 1 (default) 1 - The Exchange call log feature is enabled, user call logs are synchronized with the server, and the user call log history of Missed, Received, and outgoing calls can be retrieved on the phone. You must also enable the parameter feature.callList.enabled to use the Exchange call log feature. • The value of the configuration parameter callLists.collapseDuplicates that collapses call lists has no effect in a Skype for Business environment. • The local call logs are not generated when the following parameters are disabled: ◦ feature.callListMissed.enabled ◦ feature.callListPlaced.enabled ◦ feature.callListReceived.enabled 0 - The Exchange call log feature is disabled, the user call logs history cannot be retrieved from the Exchange server, and the phone generates call logs locally.	

Call Log Elements and Attributes

The following table describes each element and attribute that displays in the call log.

You can place the elements and attributes in any order in your configuration file.

Call Log Elements and Attributes

Element	Permitted Values
direction	In, Out
Call direction with respect to the user.	

Element	Permitted Values
disposition Indicates what happened to the call. When a call entry is first created, the disposition is set to Partial.	Busy, Forwarded, Normal, Partial, Preempted, Rejected, RemotelyHandled, Transferred
line The line (or registration) index.	Positive integer
protocol The line protocol.	SIP or H323
startTime The start time of the call. For example: 2010-01-05T12:38:05 in local time.	String
duration The duration of the call, beginning when it is connected and ending when the call is terminated.For example: `PT1H10M59S` .	String
count The number of consecutive missed and abandoned calls from a call destination.	Positive Integer
destination The original destination of the call. For outgoing calls, this parameter designates the outgoing call destination; the name is initially supplied by the local phone (from the name field of a local contact entry) but may later be updated via call signaling. This field should be used for basic redial scenarios. For incoming calls, the called destination identifies the requested party, which may be different than any of the parties that are eventually connected (the destination may indicate a SIP URI which is different from any SIP URI assigned to any lines on the phone).	Address
source The source of the call (caller ID from the call recipient's perspective).	Address
Connection	Address

Element	Permitted Values
An array of connected parties in chronological order. As a call progresses, the connected party at the far end may change, for example, if the far end transfers the call to someone else. The connected element allows the progression of connected parties, when known, to be saved for later use. All calls that contain a connected state must have at least one connection element created.	
finalDestination The final connected party of a call that has been forwarded or transferred to a third party.	Address

Call Controls

Topics:

- Local Call Recording
- Centralized Call Recording
- Busy Lamp Field (BLF)
- Instant Messaging
- Local and Centralized Conference Calls
- Conference Management
- Local Digit Map
- Enhanced 911 (E.911)
- Assured Services - Session Initiation Protocol (AS-SIP)
- Bluetooth Support for VVX Business Media Phones
- International Dialing Prefix

This section shows you how to configure call control features.

Microphone Mute

All phones have a microphone mute button.

By default, when you activate microphone mute, a red LED glows or a mute icon displays on the phone screen, depending on the phone model you are using.

You cannot configure the microphone mute feature.

Persistent Microphone Mute

With this feature, you can enable the microphone mute to persist across all calls managed on a phone.

By default, users can mute the microphone during an active call, and the microphone is unmuted when the active call ends. With persistent microphone mute enabled, when a user mutes the microphone during an active call, the microphone remains muted for all following calls until the user unmutes the microphone or the phone restarts.

\When a user mutes the microphone when the phone is idle, the mute LED glows but no icon displays on the screen. When a user initiates a new active call with the microphone muted, the mute LED glows and a Mute icon displays on the phone screen.

Persistent Microphone Mute Parameters

Use the following parameter to enable persistent microphone mute.

Persistent Microphone Mute Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.persistentMute.enabled	0 (default) - Mute ends when the active call ends or when the phone restarts. 1 - Enable the persistent mute feature.	Yes

Call Timer

By default, a call timer displays on the phone's screen during calls, and a separate call duration timer displays the hours, minutes, and seconds for each call in progress.

You cannot configure the display of the call timer.

Called Party Identification

By default, the phone displays and logs the identity of all parties called from the phone.

The phone obtains called party identities from network signaling. Because called party identification is a default feature, the phone displays caller IDs matched to the call server and does not match IDs to entries in the contact directory or corporate directory.

Calling Party Identification Parameters

Use the parameters in the following table to configure Calling Party Identification.

Calling Party Identification Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-basic.cfg`	`call.callsPerLineKey`	Set the maximum number of concurrent calls per line key. This parameter applies to all registered lines. Note that this parameter can be overridden by the per-registration parameter `reg.x.callsPerLineKey` . The maximum number of concurrent calls per line key varies by phone model and is listed for each phone in the column Calls Per Line Key in the table Flexible Call Appearances. 24 1 - 24 VVX 101, 201 8 (default) 1- 8	No
`features.cfg`	`up.useDirectoryNames`	1 (default) - The name field in the local contact directory is used as the caller ID for incoming calls from contacts in the local directory. Note: Outgoing calls and corporate directory entries are not matched. 0 - Names provided through network signaling are used for caller ID.	No

Related Links

Flexible Call Appearances on page 381

Connected Party Identification

By default, the phone displays and logs the identities of remote parties you connect to if the call server can derive the name and ID from network signaling.

In cases where remote parties have set up certain call features, the remote party you connect to—and the caller ID that displays on the phone—may be different than the intended party's. For example, Bob places a call to Alice, but Alice has call diversion configured to divert Bob's incoming calls to Fred. In this case, the phone logs and displays the connection between Bob and Fred. The phone does not match party IDs to entries in the contact directory or the corporate directory.

Calling Party Identification

By default, the phone displays the identity of incoming callers if available to the phone through the network signal.

If the incoming call address has been assigned to the contact directory, you can enable the phones to display the name assigned to contacts in the contact directory. However, the phone cannot match the identity of calling parties to entries in the corporate directory.

Calling Party Identification Parameters

Use the parameters in the following table to configure Calling Party Identification.

Calling Party Identification Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.useDirectoryNames`	1 (default) - The name field in the local contact directory is used as the caller ID for incoming calls from contacts in the local directory. Note: Outgoing calls and corporate directory entries are not matched. 0 - Names provided through network signaling are used for caller ID.	No

Related Links

Flexible Call Appearances on page 381

Remote Party Caller ID from SIP Messages

You can specify which SIP request and response messages to use to retrieve caller ID information.

Remote Party Caller ID from SIP Messages Parameters

Use the following parameters to specify which SIP request and response messages to use to retrieve caller ID information.

Remote Party Caller ID from SIP Messages Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.CID.request.sourceSipMessage	Specify which header in the SIP request to retrieve remote party caller ID from. You can use: • voIpProt.SIP.callee.sourcePreference • voIpProt.SIP.caller.sourcePreference • voIpProt.SIP.CID.sourcePreference UPDATE takes precedence over the value of this parameter. NULL (default) - Remote party caller ID information from INVITE is used. INVITE PRACK ACK This parameter does not apply to shared lines.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.CID.response.sourceSipMessage	Specify which header in the SIP request to retrieve remote party caller ID from. You can use: • voIpProt.SIP.callee.sourcePreference • voIpProt.SIP.caller.sourcePreference • voIpProt.SIP.CID.sourcePreference NULL (default) - The remote party caller ID information from the last SIP response is used. 100, 180, 183, 200 This parameter does not apply to shared lines.	No

Connected Line Identification

You can view the identity of the callee on the caller's phone screen.

If the contact details are stored on your phone, the saved contact name and number will be displayed.

Calling Line Identification

The Calling Line Identity Presentation (CLIP) displays the phone number of the caller on the phone screen.

You can configure this feature by using the parameters in the following table.

Calling Line Identification Parameters

Calling Line Identification Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	voIpProt.SIP.CID.sourcePreference	Specify the priority order for the sources of caller ID information. The headers can be in any order. Null (default) - Caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order. From,P-Asserted-Identity, Remote-Party-ID P-Asserted-Identity,From,Remote-Party-ID Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From Note: By default callee and caller will take identity order from voIpProt.SIP.CID.sourcePreference. If voIpProt.SIP.Caller.SourcePreference or voIpProt.SIP.Callee.SourcePreference are configured then the order set by voIpProt.SIP.CID.sourcePreference is ignored.	No
features.cfg	voIpProt.SIP.caller.sourcePreference	Set the priority order to display the caller's identity for incoming calls. Null (default) Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From String	No
features.cfg	voIpProt.SIP.callee.sourcePreference	Set the priority order to display the callee's identity for outgoing calls. Null (default) Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From String	No

SIP Header Warnings

You can configure the warning field from a SIP header to display a pop-up message on the phone, for example, when a call transfer failed due to an invalid extension number.

You can display pop-up messages in any language supported by the phone. The messages display for three seconds unless overridden by another message or action.

For a list of supported SIP header warnings, see the article ‘Supported SIP Request Headers' in Polycom Knowledge Base.

SIP Header Warning Parameters

You can use the parameters in the following table to enable the warning display or specify which warnings to display.

SIP Header Warning Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.header.warning.enable`	0 (default) - The warning header is not displayed. 1 - The warning header is displayed if received.	No
`sip-interop.cfg`	`voIpProt.SIP.header.warning.codes.accept`	Specify a list of accepted warning codes. Null (default) - All codes are accepted. Only codes between 300 and 399 are supported. For example, if you want to accept only codes 325 to 330: `voIpProt.SIP.header.warning.codes.accept=325,326,327,328,329,330`	No

Accessing URLs in SIP Messages

When this feature is enabled, the server attaches a URL to incoming and active calls.

The web browser or microbrowser can read this URL and present it as web content that displays on the phone screen. This feature is supported on VVX 500/501 and 1500 phones.

This feature is flexible and can be used in some of the following ways:

- In a Call Center environment, the phone displays extended information about a customer before the agent takes the call. The phone can also display a script of questions for the agent to ask during the call.
- In a hotel, a guest can view the restaurant menu on the phone.

Access URL in SIP Messages Parameters

You can configure the retrieval method for web content and enable users to choose to retrieve web content using either Active or Passive mode.

If your call server supports access URLs, you can also specify active or passive retrieval in the SIP header. If parameters in the SIP signal conflict with the file configuration, parameters in the SIP signaling take precedence.

You can also enable new web content to be added to the Settings menu on the phone, and users can set the default display mode for individual URLs to active or passive from the phone's menu.

Access URL in SIP Messages Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`mb.ssawc.enabled`	0 (default) - Spontaneous display of web content is disabled. 1 - Spontaneous web content display is enabled.	No
`features.cfg`	`mb.ssawc.call.mode`	passive (default) - Web content is displayed only when requested by the user. Passive mode is recommended when the microbrowser is used for other applications. When passive mode is enabled, an icon displays beside a call appearance indicating that web content is available, and the user can press Select to view the content. Active - Web content is retrieved spontaneously and displayed immediately.	No

Distinctive Incoming Call Treatment

You can apply distinctive treatment to specific calls and contacts in the contact directory.

You can set up distinctive treatment for each of your contacts by specifying a Divert Contact, enabling Auto-Reject, or enabling Auto-Divert for a specific contact in the local contact directory. You can also apply distinctive treatment to calls and contacts through the phone's user interface.

If you enable both the auto divert and auto reject features, auto divert has precedence over auto reject.

Related Links

Local Contact Directory Parameters on page 277

Distinctive Call Waiting

You can use the alert-info values and class fields in the SIP header to map calls to distinct call-waiting types.

You can apply three call waiting types: beep, ring, and silent. The following table shows you the parameters you can configure for this feature. This feature requires call server support.

Distinctive Call Waiting Parameters

You can use the alert-info values and class fields in the SIP header to map calls to distinct call-waiting types.

You can apply three call waiting types: beep, ring, and silent. The following table lists available parameters. This feature requires call server support.

Distinctive Call Waiting Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.alertInfo.x.class`	Alert-Info fields from INVITE requests are compared as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class is applied. default (default)	No
`sip-interop.cfg`	`voIpProt.SIP.alertInfo.x.value`	Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE. NULL (default)	No

Presence Status

You can enable users to monitor the status of other remote users and phones.

By adding remote users to a buddy list, users can monitor changes in the status of remote users in real time or they can monitor remote users as speed-dial contacts. Users can also manually specify their status in order to override or mask automatic status updates to others and can receive notifications when the status of a remote line changes.

Polycom phones support a maximum of 64 buddies for Open SIP server platforms and 200 contacts on the Skype for Business server. For information on Skype for Business contacts, refer to the *Polycom UC Software with Skype for Business - Deployment Guide* on Polycom Voice Support.

Related Links

Local Contact Directory Parameters on page 277

Presence Status Parameters

Use the parameters in the following table to enable the presence feature and display the **MyStatus** and **Buddies** soft keys on the phone.

Presence Status Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.presence.enabled`	0 (default) - Disable the presence feature—including buddy managements and user status. 1 - Enable the presence feature with the buddy and status options.	No
`features.cfg`	`pres.idleSoftkeys`	1 (default) - The MyStat and Buddies presence idle soft keys display. 0 - The MyStat and Buddies presence idle soft keys do not display.	No
`features.cfg`	`pres.reg`	The valid line/registration number that is used for presence. This registration sends a SUBSCIRBE for presence. If the value is not a valid registration, this parameter is ignored. 1 (default) 1 - 34	No

Do Not Disturb

You can enable Do Not Disturb (DND) locally on the phone or on the server.

The local DND feature is enabled by default, and users can enable or disable DND for all or individual registered lines on the phone. When enabled, users are not notified of incoming calls placed to their line.

Server-Based Do Not Disturb

If you want to enable server-based DND, you must enable the feature on both a registered phone and on the server.

The following conditions apply for server-based DND:

- Server-based DND can be applied to multiple registered lines on a phone; however, applying DND to individual registrations is not supported.
- Server-based DND cannot be enabled on a phone configured as a shared line.

- If server-based DND is enabled but not turned on when the DND feature is enabled on the phone, the “Do Not Disturb” message displays on the phone, but incoming calls continue to ring.
- Server-based DND disables local Call Forward and DND, however, if an incoming is not routed through the server, an audio alert still plays on the phone.

Do Not Disturb Parameters

Use the parameters in the following table to configure the local DND feature.

Do Not Disturb Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.doNotDisturb.enable`	1(default) - Enable Do Not Disturb (DND). 0 - Disable Do Not Disturb (DND).	Yes
`sip-interop.cfg`	`voIpProt.SIP.serverFeatureControl.dnd`	0 (default) - Disable server-based DND. 1 - Server-based DND is enabled. Server and local phone DND are synchronized.	No
`sip-interop.cfg`	`voIpProt.SIP.serverFeatureControl.localProcessing.dnd`	This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.dnd` . If set to 1 (default) and `voIpProt.SIP.serverFeatureControl.dnd` is set to 1, the phone and the server perform DND. If set to 0 and `voIpProt.SIP.serverFeatureControl.dnd` is set to 1, DND is performed on the server-side only, and the phone does not perform local DND. If both `voIpProt.SIP.serverFeatureControl.localProcessing.dnd` and `voIpProt.SIP.serverFeatureControl.dnd` are set to 0, the phone performs local DND and the `localProcessing` parameter is not used.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.rejectBusyOnDnd`	If 1 (default), and DND is turned on, the phone rejects incoming calls with a busy signal. If 0, and DND is turned on, the phone gives a visual alert of incoming calls and no audio ringtone alert. Note: This parameter does not apply to shared lines since not all users may want DND enabled.	No
`reg-advanced.cfg`	`call.donotdisturb.perReg`	This parameter determines if the do-not-disturb feature applies to all registrations on the phone or on a per-registration basis. 0 (default) - DND applies to all registrations on the phone. 1 - Users can activate DND on a per-registration basis. Note: If `voIpProt.SIP.serverFeatureControl.dnd` is set to 1 (enabled), this parameter is ignored.	No

Remote Party Disconnect Alert Tone

Remote Party Disconnect Alert Tone alerts users when the call has been disconnected by a remote party or network.

When a remote party or network on an active call gets disconnected, an alert is played to notify the user about the lost connection. The tone is played only for an active call.

Remote Party Disconnect Alert Tone Parameters

You can configure this feature by using the parameter in the following table.

Remote Party Disconnect Alert Tone Parameters

Template	Parameter	Permitted Values
`features.cfg`	`call.remoteDisconnect.toneType`	Choose an alert tone to play when the remote party disconnects call. Silent (Default) messageWaiting, instantMessage, remoteHoldNotification, localHoldNotification, positiveConfirm, negativeConfirm, welcome, misc1, misc2, misc3, misc4, misc5, misc6, misc7, custom1, custom2, custom3, custom4, custom5, custom6, custom7, custom8, custom9, custom10

Call Waiting Alerts

By default, the phone alerts users to incoming calls while a user is in an active call.

You can choose to disable these call waiting alerts and specify ringtones for incoming calls.

In addition, you can configure the phone to display the Call Waiting menu under the Preferences option on the phone.

Call Waiting Alert Parameters

Use the parameters in the following table to configure call waiting alerts.

Call Waiting Alert Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.callWaiting.enable`	Enable or disable call waiting. 1 (default) - The phone alerts you to an incoming call while you are in an active call. If 1, and you end the active call during a second incoming call, you are alerted to the second incoming call. 0 - You are not alerted to incoming calls while in an active call and the incoming call is treated as if you did not answer it.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.callWaiting.ring`	Specifies the ringtone of incoming calls when another call is active. If no value is set, the default value is used. beep (default) - A beep tone plays through the selected audio output mode on the active call. ring - The configured ringtone plays on the speaker. silent - No ringtone.	No

Missed Call Notifications

By default, a counter with the number of missed calls displays on the Recent Calls icon on the phone.

You can configure the phone to record all missed calls or to display only missed calls that arrive through the SIP server. You can also enable missed call notifications for each registered line on a phone.

Missed Call Notification Parameters

Use the following table to configure options for missed call notifications.

Missed Call Notification Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`call.missedCallTracking.x.enabled`	1 (default) - Missed call tracking for a specific registration is enabled. If `call.missedCallTracking.x.enabled` is set to 0, then the missed call counter is not updated regardless of what `call.serverMissedCalls.x.enabled` is set to (and regardless of how the server is configured) and the missed call list does not display in the phone menu. If `call.missedCallTracking.x.enabled` is set to 1 and `call.serverMissedCalls.x.enabled` is set to 0, then the number of missed calls is incremented regardless of how the server is configured. If `call.missedCallTracking.x.enabled` is set to 1 and `call.serverMissedCalls.x.enabled` is set to 1, then the handling of missed calls depends on how the server is configured.	Yes
`reg-advanced.cfg`	`call.serverMissedCall.x.enabled`	0 (default) - All missed-call events increment the counter for a specific registration. 1 - Only missed-call events sent by the server will increment the counter. Note: This feature is supported only with the BroadSoft Synergy call server (previously known as Sylantro).	Yes

Last Call Return

The phone supports redialing the last received call.

This feature requires support from a SIP server. With many SIP servers, this feature is implemented using a particular star code sequence. With some SIP servers, specific network signaling is used to implement this feature. When enabled, the phone displays an LCR soft key that users can select to place a call to the phone address that last called them.

Last Call Return Parameters

The last call return string value that you enter for parameter `call.lastCallReturnString` depends on the call server you use. Consult with your call server provider for the last call return string.

Last Call Return Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.lastCallReturn.enabled`	0 (default) - Disable last call return feature. 1 - Enable last call return.	No
`sip-interop.cfg`	`call.lastCallReturnString`	Specify the string sent to the server when the user selects the last call return action. The string is usually a star code. *69 (default) string - maximum 32 characters	No

Call Hold

Call hold enables users to pause activity on an active call so that they can use the phone for another task, such as searching the phone's menu for information.

When an active call is placed on hold, a message displays informing the held party that they are on hold.

If supported by the call server, you can enter a music-on-hold URI. For more information, see RFC Music on Hold draft-worley-service-example.

Call Hold Parameters

See the following table for a list of available parameters you can configure for this feature.

Call Hold Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.useRFC2543hold`	0 (default) - SDP media direction parameters (such as a=sendonly) per RFC 3264 when initiating a call. 1 - the obsolete c=0.0.0.0 RFC2543 technique is used when initiating a call.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.useSendonlyHold	1 (default) - The phone will send a reinvite with a stream mode parameter of “ sendonly ” when a call is put on hold. 0 - The phone will send a reinvite with a stream mode parameter of “ inactive ” when a call is put on hold Note: The phone will ignore the value of this parameter if set to 1 when the parameter voIpProt.SIP.useRFC2543hold is also set to 1 (default is 0).	No
sip-interop.cfg	call.hold.localReminder.enabled	0 (default) - Users are not reminded of calls that have been on hold for an extended period of time. 1 - Users are reminded of calls that have been on hold for an extended period of time.	Yes
sip-interop.cfg	call.hold.localReminder.period	Specify the time in seconds between subsequent hold reminders. 60 (default)	Yes
sip-interop.cfg	call.hold.localReminder.startDelay	Specify a time in seconds to wait before the initial hold reminder. 90 (default)	Yes
sip-interop.cfg	voIpProt.SIP.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. This parameter is used if reg.x.musicOnHold.uri is Null. Null (default) SIP URI	No

Hold Implementation

The phone supports two currently accepted means of signaling hold.

The phone can be configured to use either hold signaling method. The phone supports both methods when signaled by the remote endpoint.

Supported Hold Methods

Method	Notes
Signal the media directions with the “a” SDP media attributes sendonly, recvonly, inactive, or sendrecv.	Preferred method.
Set the “c” destination addresses for the zmedia streams in the SDP to zero. For example, c=0.0.0.0	No longer recommended due to RTCP problems associated with this method. Receiving sendrecv, sendonly, or inactive from the server causes the phone to revert to the other hold method.

Call Park and Retrieve

This feature enables users to park an active call to a call orbit and retrieve parked calls from the call orbit on any phone.

Whereas call hold keeps the held call on the same line, call park moves the call to a separate address where the call can be retrieved by any phone. This feature requires support from a SIP server and setup of this feature depends on the SIP server. For example, while some SIP servers implement group call pick-up using a particular star-code sequence, others implement the feature using network signaling.

You can also restrict the user to park an active call to a park orbit which already has a call parked. You can configure this feature using configuration parameter.

Call Park and Retrieve Parameters

The configuration parameters for the call park and retrieve feature are located in two template files.

You can enable the feature using the features.cfg template file or the `sip-interop.cfg` file.

Use the parameters in the following table to configure this feature.

Call Park and Retrieve Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`attendant.resourceList.x.rejectParkOnBusy`	0 (default) - Parks the call even when the park orbit already has a call parked to it. 1 – Rejects the call park when the park orbit already has a call parked and alerts the user with a popup message.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	call.parkedCallRetrieveMethod	The method the phone uses to retrieve a BLF resource's call which has dialog state confirmed. legacy (default) - Indicates that the phone uses the method specified in call.parkedCallRetrieveString . native - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header).	No
sip-interop.cfg, site.cfg	call.parkedCallRetrieveString	The star code that initiates retrieval of a parked call. Null (default) Permitted values are star codes.	No
features.cfg	feature.callPark.enabled	0 (default) - Disables the call park and call retrieve features. 1 - Enables the call park and call retrieve features.	Yes

Call Transfer

The call transfer feature enables users to transfer an existing active call to a third-party address.

You can configure the call transfer feature and set the default transfer type.

Users can perform the following types of call transfers:

- Blind Transfer—Users complete a call transfer without speaking with the other party first.
- Consultative Transfer—Users speak with the other party before completing the transfer.

  By default, users can complete a call transfer without waiting for the other party to answer the call first, which is a Blind Transfer. In this case, Party A can transfer Party B's call to Party C before Party C answers the transferred call. You can disable the blind transfer feature so that users must wait for the other party to answer before completing the transfer.

Call Transfer Parameters

Use the following table to specify call transfer behavior.

Call Transfer Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.allowTransferOnProceeding`	1 (default) - Transfer during the proceeding state of a consultation call is enabled. 0 - Transfer during the proceeding state of a consultation call is enabled 2 - Phones will accept an INVITE with replaces for a dialog in early state. This is needed when using transfer on proceeding with a proxy call server such as openSIPS, reSIProcate or SipXecs.	No
`features.cfg`	`call.defaultTransferType`	Set the transfer type the phone uses when transferring a call. Generic Base Profile: Consultative (default) - Users can immediately transfer the call to another party. Skype Base Profile: Blind (default) - The call is placed on hold while a new call is placed to the other party.	No

Call Forwarding

Polycom phones support a flexible call forwarding feature that enables users to forward incoming calls to another contact or phone line.

Users can enable call forwarding in the following ways:

- To all calls
- To incoming calls from a specific caller or extension
- During an incoming call
- When the phone is busy
- When do not disturb is enabled
- After a set number of rings before the call is answered
- To a predefined destination chosen by the user

If you are registering phones with the Skype for Business Server, the following call forwarding options are available on Skype for Business-enabled phones:

- Forward to a contact
- Forward to voicemail
- Forward to Delegates

- Simultaneously Ring Delegates
- Simultaneously Ring Group Contacts

Call Forward on Shared Lines

You can enable server-based call forwarding for shared lines.

If using BroadWorks R20 server, note the following:

- Local call-forwarding is not supported on shared lines.
- Dynamic call forwarding—forwarding incoming calls without answering the call—is not supported.

Note: The server-based and local call forwarding features do not work with the shared call appearance (SCA) and bridged line appearance (BLA) features. In order to enable users to use call forwarding, disable SCA or BLA enabled.

Call Forwarding Parameters

Use the parameters in the following table to configure feature options for call forwarding.

No parameters are needed to enable call forwarding on Skype for Business-enabled phones.

Call Forwarding Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.forward.enable`	1 (default) - Enables call forwarding. 0 - Disables call forwarding. Users cannot use Call Forward and the option is removed from the phone's Features menu.	No
`sip-interop.cfg`	`voIpProt.SIP.serverFeatureControl.cf`	0 (default) - The server-based call forwarding is not enabled. 1 - The server-based call forwarding is enabled.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.serverFeatureControl.localProcessing.cf`	This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.cf` . 1 (default) - If set to 1 and `voIpProt.SIP.serverFeatureControl.cf` is set to 1, the phone and the server perform call forwarding. 0 - If set to 0 and `voIpProt.SIP.serverFeatureControl.cf` is set to 1, call forwarding is performed on the server side only, and the phone does not perform local call forwarding. If both `voIpProt.SIP.serverFeatureControl.localProcessing.cf` and `voIpProt.SIP.serverFeatureControl.cf` are set to 0, the phone performs local call forwarding and the `localProcessing` parameter is not used.	No
`sip-interop.cfg`	`voIpProt.SIP.header.diversion.enable`	0 (default) - If set to 0, the diversion header is not displayed. 1 - If set to 1, the diversion header is displayed if received.	Yes
`sip-interop.cfg`	`voIpProt.SIP.header.diversion.list.useFirst`	1 (default) - If set to 1, the first diversion header is displayed. 0 - If set to 0, the last diversion header is displayed.	Yes
`site.cfg`	`divert.x.contact`	All automatic call diversion features uses this forward-to contact. All automatically forwarded calls are directed to this contact. The contact can be overridden by a busy contact, DND contact, or no-answer contact as specified by the `busy` , `dnd` , and `noAnswer` parameters that follow.Null (default) string - Contact address that includes ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or6416@polycom.com).	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	divert.x.sharedDisabled	1 (default) - Disables call diversion features on shared lines. 0 - Enables call diversion features on shared lines.	Yes
site.cfg	divert.x.autoOnSpecificCaller	1 (default) - Enables the auto divert feature of the contact directory for calls on registration x. You can specify to divert individual calls or divert all calls.0 - Disables the auto divert feature of the contact directory for registration x.	Yes
site.cfg	divert.busy.x.enabled	1 (default) - Diverts calls registration x is busy. 0 - Does not divert calls if the line is busy.	Yes
site.cfg	divert.busy.x.contact	Calls are sent to the busy contact's address if it is specified; otherwise calls are sent to the default contact specified by divert.x.contact .Null (default)string - contact address.	Yes
site.cfg	divert.dnd.x.enabled	0 (default) - Divert calls when DND is enabled on registration x. 1 - Does not divert calls when DND is enabled on registration x.	Yes
site.cfg	divert.dnd.x.contact	Calls are sent to the DND contact's address if it is specified; otherwise calls are sent to the default contact specified by divert.x.contact . Null (default)string - contact address.	Yes
site.cfg	divert.fwd.x.enabled	1 (default) - Users can forward calls on the phone's Home screen and use universal call forwarding. 0 - Users cannot enable universal call forwarding (automatic forwarding for all calls on registration x).	Yes
site.cfg	divert.noanswer.x.enabled	1 (default) - Unanswered calls after the number of seconds specified by timeout are sent to the no-answer contact .0 - Unanswered calls are diverted if they are not answered.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	divert.noanswer.x.contact	Null (default) - The call is sent to the default contact specified by divert.x.contact . string - contact address	Yes
site.cfg	divert.noanswer.x.timeout	55 (default) - Number of seconds for timeout. positive integer	Yes
reg-advanced.cfg	reg.x.fwd.busy.contact	The forward-to contact for calls forwarded due to busy status. Null (default) - The contact specified by divert.x.contact is used. string - The contact specified by divert.x.contact is not used	No
reg-advanced.cfg	reg.x.fwd.busy.status	0 (default) - Incoming calls that receive a busy signal is not forwarded 1 - Busy calls are forwarded to the contact specified by reg.x.fwd.busy.contact .	No
reg-advanced.cfg	reg.x.fwd.noanswer.contact	Null (default) - The forward-to contact specified by divert.x.contact is used. string - The forward to contact used for calls forwarded due to no answer.	No
reg-advanced.cfg	reg.x.fwd.noanswer.ringCount	The number of seconds the phone should ring for before the call is forwarded because of no answer. The maximum value accepted by some call servers is 20. 0 - (default) 1 to 65535	No
reg-advanced.cfg	reg.x.fwd.noanswer.status	0 (default) - The calls are not forwarded if there is no answer. 1 - The calls are forwarded to the contact specified by reg.x.noanswer.contact after ringing for the length of time specified by reg.x.fwd.noanswer.ringCount .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.serverFeatureControl.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.cf . 0 (default) - The server-based call forwarding is disabled. 1 - server based call forwarding is enabled.	Yes
site.cfg	divert.x.sharedDisabled	1 (default) - Disables call diversion features on shared lines. 0 - Enables call diversion features on shared lines.	Yes
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.cf	0 (default) - Disable server-based call forwarding. 1 - Enable server-based call forwarding. This parameter overrides reg.x.serverFeatureControl.cf .	Yes
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.localProcessing.cf	1 (default) - Allows to use the value for voIpProt.SIP.serverFeatureControl.cf. 0 - Does not use the value for This parameter depends on the value of voIpProt.SIP.serverFeatureControl.cf .	No
sip-interop.cfg	reg.x.serverFeatureControl.localProcessing.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.cf . 0 (default) - If reg.x.serverFeatureControl.cf is set to 1 the phone does not perform local Call Forward behavior. 1 - The phone performs local Call Forward behavior on all calls received.	No
sip-interop.cfg	call.shared.disableDivert	1 (default) - Enable the diversion feature for shared lines. 0 - Disable the diversion feature for shared lines. Note that this feature is disabled on most call servers.	Yes

Automatic Off-Hook Call Placement

You can configure the phone to automatically place a call to a specified number when the phone goes off-hook, which is sometimes referred to as Hot Dialing.

The phone goes off-hook when a user lifts the handset, selects New Call, or presses the headset or speakerphone buttons on the phone.

Automatic Off-Hook Call Placement Parameters

As shown in the following table, you can specify an off-hook call contact, enable or disable the feature for each registration, and specify a protocol for the call.

If you are provisioning the VVX 500 series, 600 series, or 1500 phones, you can specify whether the automatic call uses the SIP (audio only) protocol or the H.323 (video) protocol.

Automatic Off-Hook Call Placement Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`call.autoOffHook.x.contact`	Enter a SIP URL contact address. The contact must be an ASCII-encoded string containing digits, either the user part of a SIP URL (for example, 6416), or a full SIP URL (for example, `6416@polycom.com`). NULL (default)	No
`reg-advanced.cfg`	`call.autoOffHook.x.enabled`	0 (default) - No call is placed automatically when the phone goes off hook, and the other parameters are ignored. 1 - When the phone goes off hook, a call is automatically placed to the contact you specify in call.autoOffHook.x.contact and using the protocol you specify in call.autoOffHook.x.protocol. Only the VVX 500/501, 600/601, and 1500 phones use the `protocol` parameter. If no protocol is specified, the phone uses the protocol specified by `call.autoRouting.preferredProtocol` . If a line is configured for a single protocol, the configured protocol is used.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	call.autoOffHook.x.protocol	Specify the calling protocol. Only the VVX 500/501, 600/601, and 1500 business media phones use the protocol parameter. If no protocol is specified, the phone uses the protocol specified by call.autoRouting.preferredProtocol . If a line is configured for a single protocol, the configured protocol is used. NULL (default) SIP H323	No

Directed Call Pickup

Directed call pickup enables users to pick up incoming calls to another phone by dialing the extension of that phone.

This feature requires support from a SIP server and setup of this feature depends on the SIP server. For example, while some SIP servers implement directed call pick-up using a star-code sequence, others implement the feature using network signaling.

Directed Call Pickup Parameters

You can enable directed call pickup in the features.

cfg template file and the sip-interop.cfg file.

The parameters you use to configure this feature depends on your call server. To enable or disable this feature for Sylantro call servers, set the parameter feature.directedCallPickup.enabled to 1.

To configure this feature for all other call servers, use the following parameters:

- call.directedCallPickupMethod
- call.directedCallPickupString

Note that the pickup string can be different for different call servers, so check with your call server provider if you configure legacy mode for directed call pickup.

The following table lists the configuration parameters for the directed call pick-up feature.

Directed Call Pickup Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.directedCallPickup.enabled	0 (default) - Disables the directed call pickup feature. 1 - Enables the directed call pickup feature.	Yes
sip-interop.cfg	call.directedCallPickupMethod	Specifies how the phone performs a directed call pick-up from a BLF contact. legacy (default) - Indicates that the phone uses the method specified in call.directedCallPickupString . native - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header.	No
sip-interop.cfg, site.cfg	call.directedCallPickupString	The star code to initiate a directed call pickup. *97 (default) Note: The default value supports the BroadWorks calls server only. You must change the value if your organization uses a different call server.	No
sip-interop.cfg	voIpProt.SIP.strictReplacesHeader	This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources. 1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when call.directedCallPickupMethod is configured as native. 0 - Call pick-up requires a call id only.	No

Group Call Pickup

This feature enables users to pick up incoming calls to any phone within a predefined group of phones, without dialing the extension of another phone.

Group Call Pickup Parameters

This feature requires support from a SIP server and setup of this feature depends on the SIP server.

For example, while some SIP servers implement group call pick-up using a particular star-code sequence, others implement the feature using network signaling.

Use the parameter in the following table to enable this feature.

Group Call Pickup Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.groupCallPickup.enabled`	0 (default) - Disable SIP-B Group Call Pickup feature. 1 - Enable SIP-B Group Call Pickup feature.	Yes

Multiple Line Registrations

Polycom phones can have multiple line registrations.

Each registration requires an address or phone number. Phones registered with Microsoft Skype for Business Server support only one Skype for Business registration.

When multiple registrations are available, users can select which registration to use for certain features, including which registration to use for outgoing calls or when initiating new instant messages.

Note: You must use a unique address or a phone number for each registration. Using the same address or phone number for multiple registrations might cause unexpected behavior.

Maximum Number of Registrations

The maximum number of registrations vary by phone and are listed in the following table.

In addition to the maximum registrations listed in the table, you can also add up to three VVX Expansion Modules to a single VVX 300 series, 400 series, 500 series, or 600 series phone to increase the total number of registrations to 34.

Maximum Number of Registrations Per Phone

Phone Model Name	Maximum Registrations
VVX 101	One (1)
VVX 150, 201	Two (2)
VVX 250	Thirty four (34)
VVX 300/301/310/311/350	Thirty four (34)
VVX 400/401/410/411/450	Thirty four (34)
VVX 500/501	Thirty four (34)
VVX 600/601	Thirty four (34)

Phone Model Name	Maximum Registrations
VVX 1500	Twenty four (24)

Multiple Line Registrations Parameters

Each registration can be mapped to one or more line keys, however, a line key can be used for only one registration.

The maximum number of call appearances you can set varies by phone model.

Multiple Registrations Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.acd-agent-available	0 (default) - The ACD feature is disabled for registration. 1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.	No
reg-advanced.cfg reg-advanced.cfg	reg.x.acd-login-logout reg.x.acd-agent-available	0 (default) - The ACD feature is disabled for registration. 1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.	No
reg-advanced.cfg reg-advanced.cfg	reg.x.acd-login-logout reg.x.acd-agent-available	0 (default) - The ACD feature is disabled for registration. 1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.	No
reg-basic.cfg	reg.x.address	The user part (for example, 1002) or the user and the host part (for example, 1002@polycom.com) of the registration SIP URI or the H.323 ID/ extension. Null (default) string address	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.advancedConference.maxParticipants	Sets the maximum number of participants allowed in a push to conference for advanced conference calls. The number of participants configured must match the number of participants allowed on the ALU CTS. 3 (default) 0 - 25	No
reg-advanced.cfg	reg.x.advancedConference.pushToConference	0 (default) - Disable push-to-conference functionality. 1 - Enable push-to-conference functionality.	No
reg-advanced.cfg	reg.x.advancedConference.subscribeForConfEvents	1 (default) - Conference participants to receive notifications for conference events is enabled. 0 - Conference participants to receive notifications for conference events is disabled.	No
reg-advanced.cfg	reg.x.advancedConference.subscribeForConfEventsOnCCPE	1 (default) - Enable the conference host to receive notifications for conference events. 0 - Disable the conference host to receive notifications for conference events.	No
reg-advanced.cfg	reg.x.auth.domain	The domain of the authorization server that is used to check the user names and passwords. Null (default)string	No
reg-advanced.cfg	reg.x.auth.optimizedInFailover	The destination of the first new SIP request when failover occurs. 0 (default) - The SIP request is sent to the server with the highest priority in the server list. 1 - The SIP request is sent to the server which sent the proxy authentication request.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-basic.cfg	reg.x.auth.password	The password to be used for authentication challenges for this registration. Null (default) string - It overrides the password entered into the Authentication submenu on the Settings menu of the phone.	No
reg-advanced.cfg	reg.x.auth.useLoginCredentials	0 - (default) The Login credentials are not used for authentication to the server on registration x. 1 - The login credentials are used for authentication to the server.	No
reg-basic.cfg	reg.x.auth.userId	User ID to be used for authentication challenges for this registration. Null (default) string - If the User ID is non-Null, it overrides the user parameter entered into the Authentication submenu on the Settings menu of the phone.	No
reg-advanced.cfg	reg.x.bargeInEnabled	0 (default) - barge-in is disabled for line x. 1 - barge-in is enabled (remote users of shared call appearances can interrupt or barge in to active calls).	No
	reg.x.bridgeInEnabled	0 (default) - Bridge In feature is disabled. 1 - Bridge In feature is enabled.	No
features.cfg	reg.x.broadsoft.userId	Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. Null (default) string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	reg.x.broadsoft.useXspCredentials	If this parameter is disabled, the phones use standard SIP credentials to authenticate. 1 (default) - Use this value, if phone lines are registered with a server running BroadWorks R19 or earlier. 0 - Set to 0, if phone lines are registered with a server running BroadWorks R19 SP1 or later.	No
features.cfg	reg.x.broadsoft.xsp.password	Enter the password associated with the BroadSoft user account for the line. Required only when reg.x.broadsoft.useXspCredentials=1 . Null (default) string	No
reg-advanced.cfg	reg.x.callsPerLineKey	Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration. This per-registration parameter overrides call.callsPerLineKey . 24 (default) 1-24 VVX 101, 201 8 (default) 1 - 8	No
reg-advanced.cfg	reg.x.displayName	The display name used in SIP signaling and/or the H.323 alias used as the default caller ID. Null (default) UTF-8 encoded string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	reg.x.enablePvtHoldSoftKey	This parameter applies only to shared lines. 0 (default) - To disable user on a shared line to hold calls privately. 1 - To enable users on a shared line to hold calls privately.	No
features.cfg	reg.x.enablePvtHoldSoftKey	This parameter applies only to shared lines. 0 (default) - To disable user on a shared line to hold calls privately. 1 - To enable users on a shared line to hold calls privately.	No
reg-advanced.cfg	reg.x.enhancedCallPark.enabled	0 (default) - To disable the BroadWorks Enhanced Call Park feature. 1 - To enable the BroadWorks Enhanced Call Park feature.	No
	reg.x.filterReflectedBlaDialogs	1 (default) - bridged line appearance NOTIFY messages are ignored. 0 - bridged line appearance NOTIFY messages is not ignored	No
reg-advanced.cfg	reg.x.fwd.busy.contact	The forward-to contact for calls forwarded due to busy status. Null (default) - The contact specified by divert.x.contact is used. string - The contact specified by divert.x.contact is not used	No
reg-advanced.cfg	reg.x.fwd.busy.contact	The forward-to contact for calls forwarded due to busy status. Null (default) - The contact specified by divert.x.contact is used. string - The contact specified by divert.x.contact is not used	No
reg-advanced.cfg	reg.x.fwd.busy.status	0 (default) - Incoming calls that receive a busy signal is not forwarded 1 - Busy calls are forwarded to the contact specified by reg.x.fwd.busy.contact .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.fwd.busy.status	0 (default) - Incoming calls that receive a busy signal is not forwarded 1 - Busy calls are forwarded to the contact specified by `reg.x.fwd.busy.contact` .	No
reg-advanced.cfg	reg.x.fwd.noanswer.contact	Null (default) - The forward-to contact specified by `divert.x.contact` is used. string - The forward to contact used for calls forwarded due to no answer.	No
reg-advanced.cfg	reg.x.fwd.noanswer.contact	Null (default) - The forward-to contact specified by `divert.x.contact` is used. string - The forward to contact used for calls forwarded due to no answer.	No
reg-advanced.cfg	reg.x.fwd.noanswer.ringCount	The number of seconds the phone should ring for before the call is forwarded because of no answer. The maximum value accepted by some call servers is 20. 0 - (default) 1 to 65535	No
reg-advanced.cfg	reg.x.fwd.noanswer.ringCount	The number of seconds the phone should ring for before the call is forwarded because of no answer. The maximum value accepted by some call servers is 20. 0 - (default) 1 to 65535	No
reg-advanced.cfg	reg.x.fwd.noanswer.status	0 (default) - The calls are not forwarded if there is no answer. 1 - The calls are forwarded to the contact specified by `reg.x.noanswer.contact` after ringing for the length of time specified by `reg.x.fwd.noanswer.ringCount` .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.fwd.noanswer.status	0 (default) - The calls are not forwarded if there is no answer. 1 - The calls are forwarded to the contact specified by reg.x.noanswer.contact after ringing for the length of time specified by reg.x.fwd.noanswer.ringCount .	No
sip-interop.cfg	reg.x.gruu	1 - The phone sends sip.instance in the REGISTER request. 0 (default) - The phone does not send sip.instance in the REGISTER request.	No
debug.cfg	reg.x.gruu	Specify if the phone sends sip.instance in the REGISTER request. 0 (default) 1	No
reg-advanced.cfg	reg.x.header.pearlymedia.support	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.	No
reg-basic.cfg	reg.X.insertOBPAddressInRoute	1 (Default) - The outbound proxy address is added as the topmost route header. 0 - The outbound proxy address is not added to the route header.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-basic.cfg`	`reg.x.label`	The text label that displays next to the line key for registration x. The maximum number of characters for this parameter value is 256; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters you use. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The rules for parameter up.cfgLabelElide determine how the label is truncated. Null (default) - the label is determined as follows: • If `reg.1.useteluriAsLineLabel=1` , then the tel URI/phone number/ address displays as the label. • If *`reg.1.useteluriAsLineLabel=0`*, then the value for `reg.x.displayName` , if available, displays as the label. If `reg.x.displayName` is unavailable, the user part of `reg.x.address` is used. UTF-8 encoded string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`reg.x.line.y.label`	Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when `up.cfgUniqueLineLabel=1` . If `reg.x.linekeys=1` , this parameter does not have any effect. x = the registration index number starting from 1. y = the line index from 1 to the value set by `reg.x.linekeys` . Specifying a string sets the label used for the line key registration on phones with multiple line keys. If no parameter value is set for `reg.x.line.y.label` , the phone automatically numbers multiple lines by prepending "<y>_" where <y> is the line index from 1 to the value set by `reg.x.linekeys` . • The following examples show labels for line 1 on a phone with user registration 1234, where `reg.x.linekeys=2` : ◦ If no label is configured for registration, the labels are "1_1234" and "2_1234". ◦ If `reg.1.line.1.label=Polycom` and `reg.1.line.2.label=VVX` , the labels display as 'Polycom' and 'VVX'.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`reg.x.line.y.label`	Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when `up.cfgUniqueLineLabel=1` . If `reg.x.linekeys=1` , this parameter does not have any effect. x = the registration index number starting from 1. y = the line index from 1 to the value set by `reg.x.linekeys` . Specifying a string sets the label used for the line key registration on phones with multiple line keys. If no parameter value is set for `reg.x.line.y.label` , the phone automatically numbers multiple lines by prepending “<y>_” where <y> is the line index from 1 to the value set by `reg.x.linekeys` . • The following examples show labels for line 1 on a phone with user registration 1234, where `reg.x.linekeys=2` : ◦ If no label is configured for registration, the labels are “1_1234” and “2_1234”. ◦ If `reg.1.line.1.label=Polycom` and `reg.1.line.2.label=VVX` , the labels display as ‘Polycom' and ‘VVX'.	No
`reg-basic.cfg`	`reg.x.lineAddress`	The line extension for a shared line. This parameter applies to private lines and BroadSoft call park and retrieve. If there is no extension provided for this parameter, the call park notification is ignored for the shared line. Null (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.lineKeys	Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model. 1 (default) 1 to max	No
reg-advanced.cfg	reg.x.lineKeys	Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model. 1 (default) 1 to max	No
lync.cfg	reg.x.lisdisclaimer	This parameter sets the value of the location policy disclaimer. For example, the disclaimer may be “Warning: If you do not provide a location, emergency services may be delayed in reaching your location should you need to call for help.” Null (default) string, 0 to 256 characters	No
reg-advanced.cfg	reg.x.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. Null (default) - This parameter does not overrides voIpProt.SIP.musicOnHold.uri . a SIP URI - This parameter overrides voIpProt.SIP.musicOnHold.uri .	No
reg-advanced.cfg	reg.x.offerFullCodecListUponResume	1 (default) - The phone sends full audio and video capabilities after resuming a held call irrespective of the audio and video capabilities negotiated at the initial call answer. 0 - The phone does not send full audio and video capabilities after resuming a held call.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-basic.cfg	reg.x.outboundProxy.address	The IP address or hostname of the SIP server to which the phone sends all requests. Null (default) IP address or hostname	No
sip-interop.cfg	reg.x.outboundProxy.failOver.failBack.mode	The mode for failover failback (overrides reg.x.server.y.failOver.failBack.mode). duration - (default) The phone tries the primary server again after the time specified by reg.x.outboundProxy.failOver.failBack.timeout expires. newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.failBack.timeout	3600 (default) -The time to wait (in seconds) before failback occurs (overrides reg.x.server.y.failOver.failBack.timeout). 0, 60 to 65535 - The phone does not fail back until a failover event occurs with the current server.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.failRegistrationOn	1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration. 0 - The reRegisterOn parameter is enabled, existing registrations remain active.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.onlySignalWithRegistered	1 (default) - The reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. 0 - The reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.outboundProxy.failOver.reRegisterOn	This parameters overrides reg.x.server.y.failOver.reRegisterOn . 0 (default) - The phone won't attempt to register with the secondary server. 1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server.	No
reg-advanced.cfg	reg.x.outboundProxy.port	The port of the SIP server to which the phone sends all requests. 0 - (default) 1 to 65535	No
reg-advanced.cfg	reg.x.outboundProxy.transport	The transport method the phone uses to communicate with the SIP server. DNSnaptr (default) DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly	No
features.cfg	reg.x.path	0 (Default) - The path extension header field in the Register request message is not supported for the specific line registration. 1 - The phone supports and provides the path extension header field in the Register request message for the specific line registration.	No
sip-interop.cfg	reg.x.protocol.H323	You can use this parameter for the VVX 500/501, 600/601, and 1500. 0 (default) - H.323 signaling is not enabled for registration x. 1 - H.323 signaling is enabled for registration x.	No
sip-interop.cfg	reg.x.protocol.H323	You can use this parameter for the VVX 500/501, 600/601, and 1500. 0 (default) - H.323 signaling is not enabled for registration x. 1 - H.323 signaling is enabled for registration x.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	reg.x.protocol.SIP	You can use this parameter for the VVX 500/501, 600/601, and 1500. 1 (default) - SIP signaling is enabled for this registration. 0 - SIP signaling is not enabled for this registration.	No
sip-interop.cfg	reg.x.proxyRequire	Null (default) - No Proxy-Require is sent. string - Needs to be entered in the Proxy-Require header.	No
features.cfg	reg.x.regevent	0 (default) - The phone is not subscribed to registration state change notifications for the specific phone line. 1 - The phone is subscribed to registration state change notifications for the specific phone line. This parameter overrides the global parameter voIpProt.SIP.regevent.	No
reg-advanced.cfg	reg.x.rejectNDUBInvite	Specify whether or not 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) - If an NDUB event occurs, the phone does not reject the call. 1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code.	No
reg-advanced.cfg	reg.x.ringType	The ringer to be used for calls received by this registration. The default is the first non-silent ringer. If you use the configuration parameters ringer13 and ringer14 on a single registered line, the phone plays SystemRing.wav. default (default) ringer1 to ringer24	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.ringType`	The ringer to be used for calls received by this registration. ringer2 (default) - Is the first non-silent ringer. ringer1 to ringer24 - To play ringer on a single registered line.	No
`site.cfg`	`reg.x.server.H323.y.address`	Address of the H.323 gatekeeper. Null (default) IP address or hostname	No
`site.cfg`	`reg.x.server.H323.y.address`	Address of the H.323 gatekeeper. Null (default) IP address or hostname	No
`site.cfg`	`reg.x.server.H323.y.address`	Address of the H.323 gatekeeper. Null (default) IP address or hostname	No
`site.cfg`	`reg.x.server.H323.y.expires`	Desired registration period. 3600 positive integer	No
`site.cfg`	`reg.x.server.H323.y.expires`	Desired registration period. 3600 positive integer	No
`site.cfg`	`reg.x.server.H323.y.expires`	Desired registration period. 3600 positive integer	No
`site.cfg`	`reg.x.server.H323.y.port`	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. 0 (default) 0 to 65535	No
`site.cfg`	`reg.x.server.H323.y.port`	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. 0 (default) 0 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.H323.y.port	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. 0 (default) 0 to 65535	No
site.cfg	reg.x.server.y.address	If this parameter is set, it takes precedence even if the DHCP server is available. Null (default) - SIP server does not accepts registrations. IP address or hostname - SIP server that accepts registrations. If not Null, all of the parameters in this table override the parameters specified in voIpProt.server.*	No
reg-advanced	reg.x.server.y.expires	The phone's requested registration period in seconds. The period negotiated with the server may be different. The phone attempts to re-register at the beginning of the overlap period. 3600 - (default) positive integer, minimum 10	No
reg-advanced	reg.x.server.y.expires.lineSeize	Requested line-seize subscription period. 30 - (default) 0 to 65535	No
reg-advanced	reg.x.server.y.expires.overlap	The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone tries to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value. 60 (default) 5 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.failOver.failBack.mode	duration (default) - The phone tries the primary server again after the time specified by reg.x.server.y.failOver.failBack.timeout . newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to. registration - The phone tries the primary server again when the registration renewal signaling begins. This parameter overrides voIpProt.server.x.failOver.failBack.mode)	No
site.cfg	reg.x.server.y.failOver.failBack.timeout	3600 (default) - The time to wait (in seconds) before failback occurs. 0 - The phone does not fail back until a failover event occurs with the current server. 60 to 65535 - If set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again.	No
site.cfg	reg.x.server.y.failOver.failRegistrationOn	1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists) at the point of failing over. 0 - The reRegisterOn parameter is disabled, existing registrations remain active.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`reg.x.server.y.failOver.onlySignalWithRegistered`	1 (default) - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. 0 - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).	No
`site.cfg`	`reg.x.server.y.failOver.reRegisterOn`	0 (default) - The phone does not attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information. 1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server. This parameter overrides `voIpProt.server.x.failOver.reRegisterOn` .	No
`site.cfg`	`reg.x.server.y.port`	Null (default) - The port of the SIP server does not specifies registrations. 0 - The port used depends on `reg.x.server.y.transport` . 1 to 65535 - The port of the SIP server that specifies registrations.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.register	1 (default) - Calls can not be routed to an outbound proxy without registration. 0 - Calls can be routed to an outbound proxy without registration. See voIpProt.server.x.register for more information, see *SIP Server Fallback Enhancements on Polycom Phones - Technical Bulletin 5844* on Polycom Engineering Advisories and Technical Notifications.	No
sip-interop.cfg	reg.x.server.y.registerRetry.baseTimeOut	For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server.Used in conjunction with reg.x.server.y.registerRetry.maxTimeOut to determine how long to wait. 60 (default) 10 - 120 seconds	No
sip-interop.cfg	reg.x.server.y.registerRetry.maxTimeout	For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Use in conjunction with r eg.x.server.y.registerRetry.baseTimeOut to determine how long to wait. The algorithm is defined in RFC 5626. 180 - (default) 60 - 1800 seconds	No
reg-advanced.cfg	reg.x.server.y.retryMaxCount	The number of retries attempted before moving to the next available server. 3 - (default) 0 to 20 - 3 is used when the value is set to 0.	No
reg-advanced.cfg	reg.x.server.y.retryTimeOut	0 (default) - Use standard RFC 3261 signaling retry behavior. 0 to 65535 - The amount of time (in milliseconds) to wait between retries.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.server.y.specialInterop`	Specify the server-specific feature set for the line registration. Standard (Default) VVX 101: Standard GENBAND ALU-CTS DT VVX 201: Standard, GENBAND ALU-CTS ocs2007r2 lync2010 All other phones: Standard GENBAND ALU-CTS ocs2007r2 lync2010 lcs2005	
`reg-advanced.cfg`	`reg.x.server.y.subscribe.expires`	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 seconds - (default) 10 - 2147483647 (seconds) You can use this parameter in conjunction with `reg.x.server.y.subscribe.expires.overlap` .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.server.y.subscribe.expires	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 seconds - (default) 10 - 2147483647 (seconds) You can use this parameter in conjunction with reg.x.server.y.subscribe.expires.overlap .	No
reg-advanced.cfg	reg.x.server.y.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 seconds (default) 5 - 65535 seconds	No
reg-advanced.cfg	reg.x.server.y.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 seconds (default) 5 - 65535 seconds	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.transport	The transport method the phone uses to communicate with the SIP server. DNSnaptr (default) - If reg.x.server.y.address is a hostname and reg.x.server.y.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If reg.x.server.y.address is an IP address, or a port is given, then UDP is used. TCPpreferred - TCP is the preferred transport; UDP is used if TCP fails. UDPOnly - Only UDP is used. TLS - If TLS fails, transport fails. Leave port field empty (defaults to 5061) or set to 5061 . TCPOnly - Only TCP is used.	No
site.cfg	reg.x.server.y.useOutboundProxy	1 (default) - Enables to use the outbound proxy specified in reg.x.outboundProxy.address for server x. 0 - Disable to use the outbound proxy specified in reg.x.outboundProxy.address for server x.	No
reg-advanced.cfg	reg.x.serverFeatureControl.callRecording	1 (default) - BroadSoft BroadWorks v20 call recording feature for individual phone lines is enabled. 0 - BroadSoft BroadWorks v20 call recording feature for individual phone lines is disabled.	No
reg-advanced.cfg	reg.x.serverFeatureControl.callRecording	1 (default) - BroadSoft BroadWorks v20 call recording feature for individual phone lines is enabled. 0 - BroadSoft BroadWorks v20 call recording feature for individual phone lines is disabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.serverFeatureControl.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.cf . 0 (default) - The server-based call forwarding is disabled. 1 - server based call forwarding is enabled.	Yes
reg-advanced.cfg	reg.x.serverFeatureControl.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.cf . 0 (default) - The server-based call forwarding is disabled. 1 - server based call forwarding is enabled.	Yes
reg-advanced.cfg	reg.x.serverFeatureControl.dnd	This parameter overrides voIpProt.SIP.serverFeatureControl.dnd. 0 (default) - server-based do-not-disturb (DND) is disabled. 1 - server-based DND is enabled and the call server has control of DND.	Yes
sip-interop.cfg	reg.x.serverFeatureControl.localProcessing.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.cf . 0 (default) - If reg.x.serverFeatureControl.cf is set to 1 the phone does not perform local Call Forward behavior. 1 - The phone performs local Call Forward behavior on all calls received.	No
sip-interop.cfg	reg.x.serverFeatureControl.localProcessing.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.cf . 0 (default) - If reg.x.serverFeatureControl.cf is set to 1 the phone does not perform local Call Forward behavior. 1 - The phone performs local Call Forward behavior on all calls received.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	reg.x.serverFeatureControl.localProcessing.dnd	This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.dnd . 0 (default) - If reg.x.serverFeatureControl.dnd is set to 1, the phone does not perform local DND call behavior. 1 - The phone performs local DND call behavior on all calls received.	No
reg-advanced.cfg	reg.x.serverFeatureControl.securityClassification	0 (default) - The visual security classification feature for a specific phone line is disabled. 1 - The visual security classification feature for a specific phone line is enabled.	No
reg-advanced.cfg	reg.x.serverFeatureControl.securityClassification	0 (default) - The visual security classification feature for a specific phone line is disabled. 1 - The visual security classification feature for a specific phone line is enabled.	No
reg-advanced.cfg	reg.x.serverFeatureControl.signalingMethod	Controls the method used to perform call forwarding requests to the server. serviceMsForwardContact (default) string	No
sip-interop.cfg	reg.x.srtp.enable	1 (default) - The registration accepts SRTP offers. 0 - The registration always declines SRTP offers.	Yes
sip-interop.cfg	reg.x.srtp.offer	This parameter applies to the registration initiating (offering) a phone call. 0 (default) - No secure media stream is included in SDP of a SIP INVITE. 1 - The registration includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	reg.x.srtp.require	0 (default) - Secure media streams are not required. 1 - The registration is only allowed to use secure media streams.	Yes
sip-interop.cfg	reg.x.srtp.simplifiedBestEffort	This parameter overrides sec.srtp.simplifiedBestEffort . 1 (default) - Negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported. 0 - No SRTP is supported.	No
sip-interop.cfg	reg.x.strictLineSeize	0 (default) - Dial prompt is provided immediately without waiting for a successful OK from the call server. 1 - The phone is forced to wait for 200 OK on registration x when receiving a TRYING notify. This parameter overrides voIpProt.SIP.strictLineSeize for registration x.	No
sip-interop.cfg	reg.x.tcpFastFailover	0 (default) - A full 32 second RFC compliant timeout is used. 1 - failover occurs based on the values of reg.x.server.y.retryMaxCount and voIpProt.server.x.retryTimeOut .	No
reg-advanced.cfg	reg.x.terminationType	Determines the type of termination that is used for the line where the line can be managed automatically on the VVX, the wireless handset, or on both. X = each registration index. NULL (default) VVX, DECT, or VVX-DECT	No
reg-advanced.cfg	reg.x.thirdPartyName	Null (default) - In all other cases. string address -This field must match the reg.x.address value of the registration which makes up the part of a bridged line appearance (BLA).	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.thirdPartyName	Null (default) - In all other cases. string address -This field must match the reg.x.address value of the registration which makes up the part of a bridged line appearance (BLA).	No
reg-advanced.cfg	reg.x.type	private (default) - Use standard call signaling. shared - Use augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.	No
reg-advanced.cfg	reg.x.type	private (default) - Use standard call signaling. shared - Use augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.	No
reg-advanced.cfg	reg.x.useCompleteUriForRetrieve	This parameters overrides voipPort.SIP.useCompleteUriForRetrieve . 1 (default) - The target URI in BLF signaling uses the complete address as provided in the XML dialog document. 0 - Only the user portion of the XML dialog document is used and the current registrar's domain is appended to create the full target URI.	No
sip-basic.cfg	voipProt.server.x.address	The IP address or hostname and port of a SIP server that accepts registrations. Multiple servers can be listed starting with x=1 to 4 for fault tolerance. Null (default), IP address, or hostname	No
sip-interop.cfg	voIpProt.server.x.expires	The phone's requested registration period in seconds. 3600 (default) positive integer, minimum 10 The period negotiated with the server may be different. The phone attempts to re-register at the beginning of the overlap period. For example, if expires="300" and overlap="5", the phone re-registers after 295 seconds (300-5).	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.expires	The phone's requested registration period in seconds. Note: The period negotiated with the server may be different. The phone will attempt to re-register at the beginning of the overlap period. 3600 (default) positive integer, minimum 10	No
sip-interop.cfg	voIpProt.server.x.expires.lineSeize	Requested line-seize subscription period. 30 (default) positive integer, minimum 10	No
sip-interop.cfg	voIpProt.server.x.expires.lineSeize	Requested line-seize subscription period. 30 (default) positive integer, minimum 0 was 10	No
sip-interop.cfg	voIpProt.server.x.expires.overlap	The number of seconds before the expiration time returned by server x at which the phone should try to re-register. If the server value is less than the configured overlap value, the phone tries to re-register at half the expiration time returned by the server. 60 (default) 5 to 65536	No
sip-interop.cfg	voIpProt.server.x.expires.overlap	The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone will try to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value. 60 (default) 5 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.failOver.failBack.mode	Specify the failover failback mode. duration (default) - The phone tries the primary server again after the time specified by voIpProt.server.x.failOver.failBack.timeout newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to. registration - The phone tries the primary server again when the registration renewal signaling begins.	No
sip-interop.cfg	voIpProt.server.x.failOver.failBack.timeout	If voIpProt.server.x.failOver.failBack.mode is set to duration, this is the time in seconds after failing over to the current working server before the primary server is again selected as the first server to forward new requests to. Values between 1 and 59 result in a timeout of 60 and 0 means do not fail-back until a fail-over event occurs with the current server. 3600 (default) 0, 60 to 65535	No
sip-interop.cfg	voIpProt.server.x.failOver.failRegistrationOn	1 (default) - When set to 1, and the reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over. 0 - When set to 0, and the reRegisterOn parameter is enabled, existing registrations remain active. This means that the phone attempts failback without first attempting to register with the primary server to determine if it has recovered.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.failOver.onlySignalWithRegistered	1 (default) - When set to 1, and the reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. 0 - When set to 0, and the reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).	No
sip-interop.cfg	voIpProt.server.x.failOver.reRegisterOn	0 (default) - When set to 0, the phone won't attempt to register with the second. 1 - When set to 1, the phone attempts to register with (or by, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server.	No
sip-basic.cfg	voIpProt.server.x.port	The port of the server that specifies registrations. 0 (default) - If 0, the port used depends on voIpProt.server.x.transport. 1 to 65535	No
	voIpProt.server.x.protocol.SIP	1 (default) - Server is a SIP proxy/registrar 0 - If set to 0, and the server is confirmed to be a SIP server, then the value is assumed to be 1.	No
sip-interop.cfg	voIpProt.server.x.register	1 (default) - Calls can not be routed to an outbound proxy without registration. 0 - Calls can be routed to an outbound proxy without registration. For more information, see *Technical Bulletin 5844: SIP Server Fallback Enhancements on Polycom Phones*.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.registerRetry.baseTimeOut	The base time period to wait before a registration retry. Used in conjunction with voIpProt.server.x.registerRetry.maxTimeOut to determine how long to wait. The algorithm is defined in RFC 5626. If both parameters voIpProt.server.x.registerRetry.baseTimeOut and reg.x.server.y.registerRetry.baseTimeOut are set, the value of reg.x.server.y.registerRetry.baseTimeOut takes precedence. 60 - (default) 10 - 120	No
sip-interop.cfg	voIpProt.server.x.registerRetry.maxTimeOut	The maximum time period to wait before a registration retry. Used in conjunction with voIpProt.server.x.registerRetry.maxTimeOut to determine how long to wait. The algorithm is defined in RFC 5626. If both parameters voIpProt.server.x.registerRetry.maxTimeOut and reg.x.server.y.registerRetry.maxTimeOut are set, the value of reg.x.server.y.registerRetry.maxTimeOut takes precedence. 60 - (default) 10 - 1800	No
sip-interop.cfg	voIpProt.server.x.retryMaxCount	The number of retries that will be attempted before moving to the next available server. 3 (default) 0 to 20 - If set to 0, 3 is used.	No
sip-interop.cfg	voIpProt.server.x.retryTimeOut	0 (default) - Use standard RFC 3261 signaling retry behavior. 0 to 65535 - The amount of time (in milliseconds) to wait between retries.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.specialInterop	Enables server-specific features for all registrations. Standard (default) VVX 101 = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT VVX 201 = Standard, GENBAND, GENBAND-A2, ALU-CTS, ocs2007r2, lync2010 All other phones = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT, ocs2007r2, lync2010, lcs2005	No
sip-interop.cfg	voIpProt.server.x.subscribe.expires	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 - (default) 10 - 2147483647	No
sip-interop.cfg	voIpProt.server.x.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 - (default) 5 - 65535 seconds	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.transport	The transport method the phone uses to communicate with the SIP server. Null or DNSnaptr (default) - If voIpProt.server.x.address is a hostname and voIpProt.server.x.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If voIpProt.server.x.address is an IP address, or a port is given, then UDP is used. TCPpreferred - TCP is the preferred transport; UDP is used if TCP fails. UDPOnly - Only UDP will be used. TLS - If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. TCPOnly - Only TCP will be used.	No
sip-interop.cfg	voIpProt.server.x.useOutboundProxy	1 (default) - Enables to use the outbound proxy specified in voIpProt.SIP.outboundProxy.address for server x. 0 - Enables not to use the outbound proxy specified in voIpProt.SIP.outboundProxy.address for server x.	No
sip-interop.cfg	voIpProt.SIP.acd.signalingMethod	0 (default) - The ‘SIP-B' signaling is supported. (This is the older ACD functionality.) 1 - The feature synchronization signaling is supported. (This is the new ACD functionality.)	Yes
sip-interop.cfg	voIpProt.SIP.acd.signalingMethod	0 (default) - The ‘SIP-B' signaling is supported. (This is the older ACD functionality.) 1 - The feature synchronization signaling is supported. (This is the new ACD functionality.)	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.alertInfo.x.class	Alert-Info fields from INVITE requests are compared as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class is applied. default (default) See the list of ring classes in Ringtone Parameters.	No
sip-interop.cfg	voIpProt.SIP.alertInfo.x.class	Alert-Info fields from INVITE requests are compared as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class is applied. default (default)	No
sip-interop.cfg	voIpProt.SIP.alertInfo.x.class	Alert-Info fields from INVITE requests are compared as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class is applied. default (default) See the list of ring classes in Ringtone Parameters.	No
sip-interop.cfg	voIpProt.SIP.alertInfo.x.value	Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE. NULL (default)	No
sip-interop.cfg	voIpProt.SIP.alertInfo.x.value	Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE. NULL (default)	No
sip-interop.cfg	voIpProt.SIP.alertInfo.x.value	Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE. NULL (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.allowTransferOnProceeding`	1 (default) - Transfer during the proceeding state of a consultation call is enabled. 0 - Transfer during the proceeding state of a consultation call is enabled 2 - Phones will accept an INVITE with replaces for a dialog in early state. This is needed when using transfer on proceeding with a proxying call server such as openSIPS, reSIProcate or SipXecs.	No
`sip-interop.cfg`	`voipProt.SIP.anat.enabled`	Enables or disables Alternative Network Address Types (ANAT). 0 (default) - ANAT is disabled. 1 - ANAT is enabled.	No
`sip-interop.cfg`	`voIpProt.SIP.authOptimizedInFailover`	0 (default) - The first new SIP request is sent to the server with the highest priority in the server list when failover occurs. 1 - The first new SIP request is sent to the server that sent the proxy authentication request when failover occurs.	No
`features.cfg`	`voIpProt.SIP.callee.SourcePreference`	Set priority order to display the callee's identity for outgoing calls. Null (default) Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From String	`features.cfg`
`features.cfg`	`voIpProt.SIP.Caller.SourcePreference`	Set priority order to display the caller's identity for incoming calls. Null (default) Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From String	`features.cfg`

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.callinfo.precedence.overAlertinfo	0 (default) - Give priority to call-info header with answer-after string over alert-info feature is disabled. 1 - Give priority to call-info header with answer-after string over alert-info feature is enabled.	No
sip-interop.cfg	voIpProt.SIP.callinfo.precedence.overAlertinfo	0 (default) - The alert-info is given priority over call-info header. 1 - The call-info header with answer-after string is given priority over alert-info header.	No
sip-interop.cfg	voIpProt.SIP.CID.request.sourceSipMessage	Specify which header in the SIP request to retrieve remote party caller ID from. You can use: • voIpProt.SIP.callee.sourcePreference • voIpProt.SIP.caller.sourcePreference • voIpProt.SIP.CID.sourcePreference UPDATE takes precedence over the value of this parameter. NULL (default) - Remote party caller ID information from INVITE is used. INVITE PRACK ACK This parameter does not apply to shared lines.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.CID.response.sourceSipMessage	Specify which header in the SIP request to retrieve remote party caller ID from. You can use: • voIpProt.SIP.callee.sourcePreference • voIpProt.SIP.caller.sourcePreference • voIpProt.SIP.CID.sourcePreference NULL (default) - The remote party caller ID information from the last SIP response is used. 100, 180, 183, 200 This parameter does not apply to shared lines.	No
sip-interop.cfg	voIpProt.SIP.CID.sourcePreference	Specify the priority order for the sources of caller ID information. The headers can be in any order. Null (default) - Caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order. From,P-Asserted-Identity, Remote-Party-ID P-Asserted-Identity,From,Remote-Party-ID Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From Note: By default callee and caller will take identity order from voIpProt.SIP.CID.sourcePreference. If voIpProt.SIP.Caller.SourcePreference or voIpProt.SIP.Callee.SourcePreference are configured then the order set by voIpProt.SIP.CID.sourcePreference is ignored.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.compliance.RFC3261.validate.contentLanguage	1 (default) - Validation of the SIP header content language is enabled. 0 - Validation of the SIP header content language is disabled	No
sip-interop.cfg	voIpProt.SIP.compliance.RFC3261.validate.contentLength	1 (default) - Validation of the SIP header content length is enabled. 0 - Validation of the SIP header content length is disabled	No
sip-interop.cfg	voIpProt.SIP.compliance.RFC3261.validate.uriScheme	1 (default) - Validation of the SIP header URI scheme is enabled. 0 - Validation of the SIP header URI scheme is disabled	No
sip-interop.cfg	voIpProt.SIP.conference.address	Null (default) - Conferences are set up on the phone locally. String 128 max characters - Enter a conference address. Conferences are set up by the server using the conferencing agent specified by this address. Acceptable values depend on the conferencing server implementation policy.	No
sip-interop.cfg	voIpProt.SIP.conference.parallelRefer	0(deafult) - A parallel REFER is not sent to the call server. 1 - A parallel REFER is not sent to the call server. Note: This parameter must be set for Siemens OpenScape Centralized Conferencing.	No
sip-interop.cfg	voIpProt.SIP.connectionReuse.useAlias	0 (default) - The alias parameter is not added to the via header 1 - The phone uses the connection reuse draft which introduces "alias".	No
sip-interop.cfg	voIpProt.SIP.dialog.strictXLineID	0 (default) - The phone will not look for x-line-id (call appearance index) in a SIP INVITE message. 1 - The phone will look for x-line-id (call appearance index) in a SIP INVITE message	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.dialog.usePvalue	0 (default) - Phone uses a pval field name in Dialog. 1 - Phone uses a pvalue field name in Dialog.	No
sip-interop.cfg	voIpProt.SIP.dialog.useSDP	0 (default) - A new dialog event package draft is used (no SDP in dialog body). 1 - Use this setting to send SDP in the dialog body for backwards compatibility	No
sip-interop.cfg	voIpProt.SIP.dtmfViaSignaling.rfc2976	Enable or disable DTMF relays for active SIP calls. Not supported for H.323 calls. 0 (default) - DTMF digit information is not sent 1 - DTMF digit information is sent in RFC2976 SIP INFO packets during a call.	Yes
sip-interop.cfg	voIpProt.SIP.dtmfViaSignaling.rfc2976.nonLegacyEncoding	Controls the behavior of the Star and Pound keys used for DTMF relays for active SIP calls. Not supported for H.323 calls. 0 (default) - The phone sends 10 when the Star key (*) is pressed and 11 when the Pound key (#) is pressed. 1 - The phone sends an asterisk (*) when the Star key is pressed and a hashtag (#) when the Pound key is pressed.	Yes
sip-basic.cfg	voIpProt.SIP.enable	A flag to determine if the SIP protocol is used for call routing, dial plan, DTMF, and URL dialing. 1 (default) - The SIP protocol is used. 0 - The SIP protocol is not used.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.failoverOn503Response	A flag to determine whether or not to trigger a failover if the phone receives a 503 response. You must use a registration expiry of 66 seconds or greater for failover with a 503 response to work properly. This rule applies both to the phone configuration (reg.x.server.y.expires and voIpProt.server.x.expires) as well as the 200 OK register response from the server. 1 (default) 0	No
sip-interop.cfg	voIpProt.SIP.header.diversion.enable	0 (default) - If set to 0, the diversion header is not displayed. 1 - If set to 1, the diversion header is displayed if received.	Yes
sip-interop.cfg	voIpProt.SIP.header.diversion.list.useFirst	1 (default) - If set to 1, the first diversion header is displayed. 0 - If set to 0, the last diversion header is displayed.	Yes
sip-interop.cfg	voIpProt.SIP.header.pEarlyMedia.support	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.	
sip-interop.cfg	voIpProt.SIP.header.warning.codes.accept	Specify a list of accepted warning codes. Null (default) - All codes are accepted only codes between 300 and 399 are supported. comma separated list	No
sip-interop.cfg	voIpProt.SIP.header.warning.codes.accept	Specify a list of accepted warning codes. Null (default) - All codes are accepted. Only codes between 300 and 399 are supported. For example, if you want to accept only codes 325 to 330: voIpProt.SIP.header.warning.codes.accept=325,326,327,328,329,330	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.header.warning.enable	0 (default) - The warning header is not displayed. 1 - The warning header is displayed if received.	No
sip-interop.cfg	voIpProt.SIP.IM.autoAnswerDelay	The time interval from receipt of the instant message invitation to automatically accepting the invitation. 10 (default) 0 to 40	No
sip-interop.cfg	voIpProt.SIP.IMS.enable	This parameter applies to all registered or unregistered SIP lines on the phone. 0 (Default) - The phone does not support IMS features introduced in UC Software 5.5.0. 1 - The phone supports IMS features introduced in UC Software 5.5.0.	
sip-interop.cfg	voIpProt.SIP.intercom.alertInfo	The string you want to use in the Alert-Info header. You can use the following characters: '@', '-' ,'_' , '.' . If you use any other characters, NULL, or empty spaces, the call is sent as normal without the Alert-Info header. Intercom (default) Alpha - Numeric string	No
sip-interop.cfg	voIpProt.SIP.keepalive.sessionTimers	0 (default) - The session timer is disabled. 1 - The session timer is enabled.	No
sip-interop.cfg	voIpProt.SIP.lineSeize.retries	Controls the number of times the phone will retry a notify when attempting to seize a line (BLA). 10 (default) 3 to 10	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.local.port	The local port for sending and receiving SIP signaling packets. 5060 - The value is used for the local port but is not advertised in the SIP signaling. 0 to 65535 - If set to 0,the 5060 value is used for the local port but is not advertised in the SIP signaling. For other values, that value is used for the local port and it is advertised in the SIP signaling	Yes
sip-interop.cfg	voIpProt.SIP.looseContact	0 (default) - The port parameter is added to the contact header in TLS case. 1 - The port parameter is not added to the contact header or SIP messages.	No
sip-interop.cfg	voIpProt.SIP.ms-forking	This parameter is applies when installing Microsoft Live Communications Server. 0 (default) - Support for MS-forking is disabled. 1 - Support for MS-forking is enabled. Note: If any endpoint registered to the same account has MS-forking disabled, all other endpoints default back to non-forking mode. Windows Messenger does not use MS-forking so be aware of this behavior if one of the endpoints is using Windows Messenger.	No
sip-interop.cfg	voIpProt.SIP.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. This parameter is used if reg.x.musicOnHold.uri is Null. Null (default) SIP URI	No
sip-interop.cfg	voIpProt.SIP.newCallOnUnRegister	1 (default) - The phone generate new Call-ID and From tag during re-registration. 0 - The phone does not generate new Call-ID and From tag during re-registration.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-basic.cfg	voIpProt.SIP.outboundProxy.address	The IP address or hostname of the SIP server to which the phone sends all requests. Null (default) IP address or hostname	No
sip-interop.cfg	voIpProt.SIP.outboundProxy.failOver.failBack.mode	Duration (default) - The phone tries the primary server again after the time specified by reg.x.outboundProxy.failOver.failBack.timeout expires. newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to. registration - The phone tries the primary server again when the registration renewal signaling begins.	No
sip-interop.cfg	voIpProt.SIP.outboundProxy.failOver.failBack.timeout	The time to wait (in seconds) before failback occurs (overrides voIpProt.server.x.failOver.failBack.timeout). 3600 (default) -If the fail back mode is set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again. 0, 60 to 65535 -If set to 0, the phone will not fail-back until a fail-over event occurs with the current server.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.outboundProxy.failOver.failRegistrationOn	1 (default) - When set to 1, and the reRegisterOn parameter is enabled, the phone will silently invalidate an existing registration (if it exists), at the point of failing over. 0 - When set to 0, and the reRegisterOn parameter is enabled, existing registrations will remain active. This means that the phone will attempt failback without first attempting to register with the primary server to determine if it has recovered. Note: voIpProt.SIP.outboundProxy.failOver.reRegisterOn must be enabled.	No
sip-interop.cfg	voIpProt.SIP.outboundProxy.failOver.onlySignalWithRegistered	1 (default) - No signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. 0 - signaling will be accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred). This parameter overrides voIpProt.server.x.failOver.onlySignalWithRegistered . Note: reRegisterOn and failRegistrationOn parameters must be enabled	No
sip-interop.cfg	voIpProt.SIP.outboundProxy.failOver.reRegisterOn	This parameter overrides the voIpProt.server.x.failOver.reRegisterOn . 0 (default) - The phone won't attempt to register with the secondary server, since the phone will assume that the primary and secondary servers share registration information. 1 - The phone will attempt to register with the secondary server. If the registration succeeds signaling will proceed with the secondary server.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.outboundProxy.port	The port of the SIP server to which the phone sends all requests. 0 (default) 0 to 65535	No
sip-interop.cfg	voIpProt.SIP.outboundProxy.transport	DNSnaptr (default) - If reg.x.outboundProxy.address is a hostname and reg.x.outboundProxy.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If reg.x.outboundProxy.address is an IP address, or a port is given, then UDP is used. TCPpreferred - TCP is the preferred transport, UDP is used if TCP fails. UDPOnly - Only UDP will be used. TLS - If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. TCPOnly - Only TCP will be used.	No
sip-interop.cfg	voIpProt.SIP.pingInterval	The number in seconds to send PING message. 0 (default) - This feature is disabled. 0 to 3600 - This feature is enabled.	No
sip-interop.cfg	voIpProt.SIP.pingMethod	The ping method to be used. PING (default) OPTIONS	No
sip-interop.cfg	voIpProt.SIP.presence.nortelShortMode	This parameter is required when using the Presense feature with an Avaya or GENBAND server. 0 (default) 1 - Different headers are sent in SUBSCRIBE when used feature with an Avaya or GENBAND server. Support is indicated by adding a header Accept-Encoding: x-nortel-short. A PUBLISH is sent to indicate the status of the phone.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	voIpProt.SIP.regevent	0 (default) - The phone is not subscribed to registration state change notifications for all phone lines. 1 - The phone is subscribed to registration state change notifications for all phone lines. This parameter is overridden by the per-phone parameter reg.x.regevent.	
sip-interop.cfg	voIpProt.SIP.rejectNDUBInvite	Specify whether or not 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) - If an NDUB event occurs, the phone does not reject the call for all line registrations. 1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code for all line registrations.	
sip-interop.cfg	voIpProt.SIP.renewSubscribeOnTLSRefresh	1 (default) - For an as-feature-event, the SUBSCRIBE message is sent along with the RE-REGISTER when Transport Layer Security (TLS) breaks. 0 - The SUBSCRIBE and RE-REGISTER messages is sent at different times.	No
sip-interop.cfg	voIpProt.SIP.rport	0 (default) – The phone does not insert the rport parameter into the Via header of its requests. 1 – The phone inserts the rport parameter, as defined by RFC 3581, into the Via header of its requests.	No
sip-interop.cfg	voIpProt.SIP.requestURI.E164.addGlobalPrefix	0 (default) - ‘+' global prefix is not added to the E.164 user parts in sip: URIs. 1 - ‘+' global prefix is added to the E.164 user parts in sip: URIs.	No
sip-interop.cfg	voIpProt.SIP.requestValidation.digest.realm	Determines the string used for Realm. PolycomSPIP (default) string	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.requestValidation.x.method	Null (default) - no validation is made. Source - ensure request is received from an IP address of a server belonging to the set of target registration servers. digest: challenge requests with digest authentication using the local credentials for the associated registration (line). both or all: apply both of the above methods.	Yes
sip-interop.cfg	voIpProt.SIP.requestValidation.x.method	Null (default) - no validation is made. Source - ensure request is received from an IP address of a server belonging to the set of target registration servers. digest: challenge requests with digest authentication using the local credentials for the associated registration (line). both or all: apply both of the above methods.	Yes
sip-interop.cfg	voIpProt.SIP.requestValidation.x.request	Sets the name of the method for which validation will be applied. Null (default) INVITE, ACK, BYE, REGISTER, CANCEL, OPTIONS, INFO, MESSAGE, SUBSCRIBE, NOTIFY, REFER, PRACK, UPDATE Note: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases.	Yes
sip-interop.cfg	voIpProt.SIP.requestValidation.x.request	Sets the name of the method for which validation will be applied. Null (default) INVITE, ACK , BYE, REGISTER, CANCEL, OPTIONS, INFO, MESSAGE, SUBSCRIBE, NOTIFY, REFER, PRACK, UPDATE Note: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.requestValidation.x.request.y.event	Determines which events specified with the Event header should be validated; only applicable when voIpProt.SIP.requestValidation.x.request is set to SUBSCRIBE or NOTIFY . Null (default) - all events will be validated. A valid string - specified event will be validated.	Yes
sip-interop.cfg	voIpProt.SIP.requestValidation.x.request.y.event	Determines which events specified with the Event header should be validated; only applicable when voIpProt.SIP.requestValidation.x.request is set to SUBSCRIBE or NOTIFY . Null (default) - all events will be validated. A valid string - specified event will be validated.	Yes
sip-interop.cfg	voIpProt.SIP.RFC3261TimerI	0 (default) - Timer I for reliable transport will be fired at five seconds. This parameter does not cause any change for unreliable transport. 1 - Timer I for reliable transport will be fired at zero seconds.	No
sip-interop.cfg	voIpProt.SIP.sendCompactHdrs	0 (default) - SIP header names generated by the phone use the long form, for example From . 1 - SIP header names generated by the phone use the short form, for example f .	No
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.callRecording	0 (default) - The BroadSoft BroadWorks v20 call recording feature for multiple phones is disabled. 1 - The BroadSoft BroadWorks v20 call recording feature for multiple phones is enabled.	No
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.cf	0 (default) - The server-based call forwarding is not enabled. 1 - The server-based call forwarding is enabled.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.cf	0 (default) - Disable server-based call forwarding. 1 - Enable server-based call forwarding.	Yes
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.dnd	0 (default) - Disable server-based DND. 1 - Server-based DND is enabled. Server and local phone DND are synchronized.	No
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.localProcessing.cf	This parameter depends on the value of voIpProt.SIP.serverFeatureControl.cf . 1 (default) - If set to 1 and voIpProt.SIP.serverFeatureControl.cf is set to 1, the phone and the server perform call forwarding. 0 - If set to 0 and voIpProt.SIP.serverFeatureControl.cf is set to 1, call forwarding is performed on the server side only, and the phone does not perform local call forwarding. If both voIpProt.SIP.serverFeatureControl.localProcessing.cf and voIpProt.SIP.serverFeatureControl.cf are set to 0, the phone performs local call forwarding and the localProcessing parameter is not used.	No
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.localProcessing.cf	1 (default) - Allows to use the value for voIpProt.SIP.serverFeatureControl.cf. 0 - Does not use the value for This parameter depends on the value of voIpProt.SIP.serverFeatureControl.cf .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.localProcessing.dnd	This parameter depends on the value of voIpProt.SIP.serverFeatureControl.dnd . If set to 1 (default) and voIpProt.SIP.serverFeatureControl.dnd is set to 1, the phone and the server perform DND. If set to 0 and voIpProt.SIP.serverFeatureControl.dnd is set to 1, DND is performed on the server-side only, and the phone does not perform local DND. If both voIpProt.SIP.serverFeatureControl.localProcessing.dnd and voIpProt.SIP.serverFeatureControl.dnd are set to 0, the phone performs local DND and the localProcessing parameter is not used.	No
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.missedCalls	0 (default) - Server-based missed calls is not enabled. 1 - Server-based missed calls is enabled. The call server has control of missed calls.	Yes
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.securityClassification	0 (default) - The visual security classification feature for all lines on a phone is disabled. 1 - The visual security classification feature for all lines on a phone is enabled.	No
sip-interop.cfg	voIpProt.SIP.serverFeatureControl.securityClassification	0 (default) - The visual security classification feature for all lines on a phone is disabled. 1 - The visual security classification feature for all lines on a phone is enabled.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	v oIpProt.SIP.specialEvent.checkSync.alwaysReboot	0 (default) - The phone will only reboot if necessary. Many configuration parameter changes can be applied dynamically without the need for a reboot. 1 - The phone always reboot when a NOTIFY message is received from the server with event equal to check-sync even if there has not been a change to software or configuration.	No
site.cfg	voIpProt.SIP.specialEvent.checkSync.downloadCallList	0 (default) - The phone does not download the call list for the logged-in user when a check sync event's NOTIFY message is received from the server. 1 - The phone downloads the call list for the logged-in user when a check sync event's NOTIFY message is received from the server.	No
site.cfg	voIpProt.SIP.specialEvent.checkSync.downloadCallList	0 (default) - The phone does not download the call list for the user after receiving a checksync event in the NOTIFY. 1 - The phone downloads the call list for the user after receiving a checksync event in the NOTIFY.	No
site.cfg	voIpProt.SIP.specialEvent.checkSync.downloadDirectory	0 (default) - The phone downloads updated directory files after receiving a checksync NOTIFY message. 1 - The phone downloads the updated directory files along with any software and configuration updates after receiving a checksync NOTIFY message. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates. Note: The parameter hotelingMode.type set to 2 or 3 overrides this parameter.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.specialEvent.lineSeize.nonStandard`	Controls the response for a line-seize event SUBSCRIBE. 1 (default) - This speeds up the processing of the response for line-seize event. 0 - This will process the response for the line seize event normally	Yes
`sip-interop.cfg`	`voIpProt.SIP.strictLineSeize`	0 (default) - Dial prompt is provided immediately when you attempt to seize a shared line without waiting for a successful OK from the call server. 1 - The phone is forced to wait for a 200 OK response when receiving a TRYING notify.	No
`sip-interop.cfg`	`voIpProt.SIP.strictReplacesHeader`	This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources. 1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when call.directedCallPickupMethod is configured as native. 0 - Call pick-up requires a call id only.	No
`sip-interop.cfg`	`voIpProt.SIP.strictReplacesHeader`	This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources. 1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when call.directedCallPickupMethod is configured as native. 0 - Call pick-up requires a call id only.	No
`sip-interop.cfg`	`voIpProt.SIP.strictUserValidation`	0 (default) - The phone is forced to match the user portion of signaling exactly. 1 - The phone will use the first registration if the user part does not match any registration.	No
`sip-interop.cfg`	`voIpProt.SIP.supportFor100rel`	1 (default) - The phone advertises support for reliable provisional responses in its offers and responses. 0 - The phone will not offer 100rel and will reject offers requiring 100rel.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-basic.cfg	voIpProt.SIP.supportFor199	Determine support for the 199 response code. For details on the 199 response code, see RFC 6228. 0 (Default) - The phone does not support the 199 response code. 1- The phone supports the 199 response code.	
sip-interop.cfg	voIpProt.SIP.tcpFastFailover	0 (default) - A full 32 second RFC compliant timeout is used. 1 - A failover occurs based on the values of reg.x.server.y.retryMaxCount and voIpProt.server.x.retryTimeOut.	No
sip-interop.cfg	voIpProt.SIP.tlsDsk.enable	0 (default) - TLS DSK is disabled. 1 - TLS DSK is enabled.	No
sip-interop.cfg	voIpProt.SIP.turnOffNonSecureTransport	0 (default) - Stop listening to port 5060 when using AS-SIP feature is disabled. 1 - Stop listening to port 5060 when using AS-SIP feature is enabled.	Yes
sip-interop.cfg	voIpProt.SIP.use486forReject	0 (default) - The phone will not transmit 486 response. 1 - The phone will not transmit 486 response.	No
sip-interop.cfg	voIpProt.SIP.useContactInReferTo	0 (default) - The “To URI” is used in the REFER. 1 - The “Contact URI” is used in the REFER.	No
sip-interop.cfg	voIpProt.SIP.useLocalTargetUriforLegacyPickup	1 (default) - The target URI in BLF signaling uses the complete address as provided in the XML dialog document. 0 - Only the user portion of the target URI in the XML dialog document is used and the current registrar's domain is appended to create the address for pickup or retrieval.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.useRFC2543hold`	0 (default) - SDP media direction parameters (such as a=sendonly) per RFC 3264 when initiating a call. 1 - the obsolete c=0.0.0.0 RFC2543 technique is used when initiating a call.	No
`sip-interop.cfg`	`voIpProt.SIP.useRFC2543hold`	0 (default) - SDP media direction parameters (such as a=sendonly) per RFC 3264 when initiating a call. 1 - the obsolete c=0.0.0.0 RFC2543 technique is used when initiating a call.	No
`sip-interop.cfg`	`voIpProt.SIP.useRFC3264HoldOnly`	0 (default) - When set to 0, and no media direction is specified, the phone enters backward compatibility mode when negotiating SDP and responds using the c=0.0.0.0 RFC 2543 signaling method. 1 - When set to 1, and no media direction is specified, the phone uses `sendrecv` compliant with RFC 3264 when negotiating SDP and generates responses containing RFC 3264-compliant media attributes for calls placed on and off hold by either end. Note: `voIpProt.SIP.useSendonlyHold` applies only to calls on phones that originate the hold.	No
`sip-interop.cfg`	`voIpProt.SIP.useSendonlyHold`	1 (default) - The phone will send a reinvite with a stream mode parameter of “sendonly” when a call is put on hold. 0 - The phone will send a reinvite with a stream mode parameter of “inactive” when a call is put on hold Note: The phone will ignore the value of this parameter if set to 1 when the parameter voIpProt.SIP.useRFC2543hold is also set to 1 (default is 0).	No

Related Links

Flexible Call Appearances on page 381
Multiple Line Keys Per Registration on page 380

Multiple Line Keys Per Registration

You can assign a single registered phone line address to multiple line keys on Polycom phones.

This feature can be useful for managing a high volume of calls to a single line. This feature is not supported when registered with Microsoft Skype for Business Server.

Related Links

Multiple Line Registrations Parameters on page 327

Multiple Line Keys Per Registration Parameters

Use the parameter in the following table to configure this feature.

This feature is one of several features associated with Call Appearances.

Multiple Line Keys Per Registration Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.lineKeys`	Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model. 1 (default) 1 to max	No

Multiple Call Appearances

You can enable each registered phone line to support multiple concurrent calls and have each concurrent call display on the phone's user interface.

For example, with multiple call appearances, users can place one call on hold, switch to another call on the same registered line, and have both calls display on the phone.

This feature is one of several features associated with flexible call appearances. If you assign a registered line to multiple line keys, the default number of concurrent calls applies to all line keys.

Multiple Call Appearance Parameters

Use the parameters in the following table to set the maximum number of concurrent calls per registered line and the default number of calls per line key.

Note that you can set the value for the `reg.1.callsPerLineKey` parameter to a value higher than 1, for example, 3. After you set the value to 3, for example, you can have three call appearances on line 1. By default, any additional incoming calls are automatically forwarded to voicemail. If you set more than two call appearances, a call appearance counter displays at the top-right corner on the phone.

Multiple Call Appearances Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-basic.cfg	call.callsPerLineKey	Set the maximum number of concurrent calls per line key. This parameter applies to all registered lines. Note that this parameter can be overridden by the per-registration parameter reg.x.callsPerLineKey . The maximum number of concurrent calls per line key varies by phone model and is listed for each phone in the column Calls Per Line Key in the table Flexible Call Appearances. 24 1 - 24 VVX 101, 201 8 (default) 1- 8	No
reg-advanced.cfg	reg.x.callsPerLineKey	Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration. This per-registration parameter overrides call.callsPerLineKey . 24 (default) 1-24 VVX 101, 201 8 (default) 1 - 8	No

Flexible Call Appearances

A number of features are associated with flexible call appearances, including Multiple Line Registrations, Multiple Line Keys Per Registration, and Multiple Call Appearances.

Use the following table to understand how you can organize registrations, line keys per registration, and concurrent calls per line key.

The following table includes the following types of call appearances:

- Registrations—The maximum number of user registrations
- Line Keys—The maximum number of line keys
- Line Keys Per Registration—The maximum number of line keys per user registration
- Calls Per Line Key—The maximum number of concurrent calls per line key
- Concurrent Calls (including Conference Legs)—The runtime maximum number of concurrent calls, and the number of conference participants minus the conference initiator.

Phone Model	Registrations	Line Keys	Line keys Per Registration	Calls Per Line Key	Concurrent Calls[4]
VVX 101, 150, 201	1	2	2	8	8 (2)
VVX 300/301/310/311/250/350	34	48	48	24	24 (2)
VVX 400/401/410/411/450	34	48	48	24	24 (2)
VVX 500/501	34	48	48	24	24 (2)
VVX 600/601	34	48	48	24	24 (2)
VVX 1500	24	24	24	24	24 (2)
SoundStructure VOIP Interface [5]	12	12	12	24	24 (2)

Related Links

Calling Party Identification Parameters on page 299
Calling Party Identification Parameters on page 300
Multiple Line Registrations Parameters on page 327
Per-Registration Call Parameters on page 617
Per-Registration Dial Plan Parameters on page 623

Bridged Line Appearance

Bridged line appearance connects calls and lines to multiple phones.

With bridged line appearance enabled, an active call displays simultaneously on multiple phones in a group. By default, the answering phone has sole access to the incoming call, which is called line seize. If

[4] * Note that each conference leg counts as one call. The total number of concurrent calls in a conference indicated in this table includes all conference participants minus the moderator.

[5] ** For more information on using line and call appearances with the SoundStructure VOIP Interface, refer to the SoundStructure Design Guide, available at Polycom Support.

the answering phone places the call on hold, that call becomes available to all phones of that group. All call states—active, inactive, on hold—are displayed on all phones of a group.

Important:	Shared call appearances and bridged line appearances are similar signaling methods that enable more than one phone to share the same line or registration. The methods you use vary with the SIP call server you are using. In the configuration files, bridged lines are configured by shared line parameters. The barge-in feature is not available with bridged line appearances; it is available only with shared call appearances.

Bridged Line Appearance Signaling

A bridged line is an address of record managed by a server.

The server allows multiple endpoints to register locations against the address of record.

The phone supports Bridged Line Appearances (BLA) using the SUBSCRIBE-NOTIFY method in the SIP Specific Event Notification framework (RFC 3265). The event used is ‘dialog' for bridged line appearance subscribe and notify.

Bridged Line Appearance Parameters

To begin using bridged line appearance, you must get a registered address dedicated for use with bridged line appearance from your call server provider.

This dedicated address must be assigned to a phone line in the `reg.x.address` parameter of the reg-basic.cfg template.

Use the parameters in the following table to configure this feature.

Bridged Line Appearance Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.shared.disableDivert`	1 (default) - Enable the diversion feature for shared lines. 0 - Disable the diversion feature for shared lines. Note that this feature is disabled on most call servers.	Yes
`reg-advanced.cfg`	`reg.x.type`	private (default) - Use standard call signaling. shared - Use augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.	No
`reg-advanced.cfg`	`reg.x.thirdPartyName`	Null (default) - In all other cases. string address -This field must match the `reg.x.address` value of the registration which makes up the part of a bridged line appearance (BLA).	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`divert.x.sharedDisabled`	1 (default) - Disables call diversion features on shared lines. 0 - Enables call diversion features on shared lines.	Yes

Voicemail

When you configure Polycom phones with a SIP URL that integrates with a voicemail server contact, users receive a visual and audio alert when they have new voicemail messages available on their phone.

Voicemail Parameters

Use the parameters in the following table to configure voicemail and voicemail settings.

Voicemail Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-basic.cfg`	`msg.mwi.x.callBackMode`	The message retrieval mode and notification for registration x. registration (default) - The registration places a call to itself (the phone calls itself). contact - a call is placed to the contact specified by `msg.mwi.x.callback.` disabled - Message retrieval and message notification are disabled.	No
`sip-interop.cfg`	`msg.mwi.x.callBack`	The contact to call when retrieving messages for this registration if `msg.mwi.x.callBackMode` is set to `contact .` ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com) NULL (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	msg.mwi.x.subscribe	Specify the URI of the message center server. ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com) If non-Null, the phone sends a SUBSCRIBE request to this contact after bootup. NULL (default)	
site.cfg	mwi.backLight.disable	Specify if the phone screen backlight illuminates when you receive a new voicemail message. 0 (default) - Disable the back light message alert. 1 - Enable the back light message alert.	Yes
features.cfg	up.mwiVisible	Specify if message waiting indicators (MWI) display or not. 0 (default) - If msg.mwi.x.callBackMode=0, MWI do not display in the message retrieval menus. 1 - MWI display.	Yes
sip-interop.cfg	up.oneTouchVoiceMail	1 (default) - Lync Base Profile 0 (default) - Generic Base Profile 0 (default) - The phone displays a summary page with message counts. 1 - You can call voicemail services directly from the phone, if available on the call server, without displaying the voicemail summary.	Yes

Local Call Recording

Local call recording enables you to record audio calls to a USB device connected to the phone.

You can play back recorded audio on the phone or devices that run applications like Windows Media Player® or iTunes® on a Windows® or Apple® computer. To use this feature, ensure that the USB port is enabled.

Audio calls are recorded in .wav format and include a date/time stamp. The phone displays the recording time remaining on the attached USB device, and users can browse all recorded files using the phone's menu.

Important: Federal, state, and/or local laws may legally require that you notify some or all of the call parties when a call recording is in progress.

This feature is available on the following devices:

- VVX 401, 411 business media phones
- VVX 5xx and 6xx series business media phones
- VVX 250, 350, and 450 business IP phones
- SoundStructure VoIP Interface

Local Call Recording Parameters

Use the parameters in the following table to configure local call recording.

Local Call Recording Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.callRecording.enabled`	0 (default) - Disable audio call recording. 1 - Enable audio call recording.	Yes

Centralized Call Recording

This feature enables users to record audio and video calls and control call recording directly from phones registered with BroadSoft BroadWorks r20 server.

Users can manage recorded audio and video files on a third-party call recording server.

By default, far-side participants are not alerted when calls are being recorded. The BroadWorks r20server provides administrators with the option to enable an announcement to play at the beginning of a call when a call is being recorded. If a call recorded is in progress when the call is transferred, the recording continues for the new call.

Note: You can record calls using a central server or locally using the phone's USB call recording feature - you cannot use both at the same time. By default, both features are disabled. If you enable one call recording feature, ensure that the other is disabled. Use either centralized or the local call recording; do not use both.

Centralized Call Recording Parameters

You must enable this feature on the BroadSoft BroadWorks r20 server and on the phones using the configuration parameters listed in the following table.

On the BroadSoft server, assign phone users one of several call recording modes listed in Call Recording Modes.

Use the configuration parameters in the following table to enable this feature on the phone.

Centralized Call Recording Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.serverFeatureControl.callRecording`	0 (default) - The BroadSoft BroadWorks v20 call recording feature for multiple phones is disabled. 1 - The BroadSoft BroadWorks v20 call recording feature for multiple phones is enabled.	Yes
`reg-advanced.cfg`	`reg.x.serverFeatureControl.callRecording`	1 (default) - BroadSoft BroadWorks v20 call recording feature for individual phone lines is enabled. 0 - BroadSoft BroadWorks v20 call recording feature for individual phone lines is disabled.	No

Call Recording Modes

Set the call recording modes on the BroadSoft BroadWorks R20 server using the following call recording modes:

- **Never Mode** Call recording is never initiated and the phone never displays call recording soft keys.
- **Always Mode** The entire incoming or outgoing call is recorded and no control options are available to users. During active calls, the phone displays a Record symbol. Call recording stops when the call ends and the call is stored on the server.
- Always with Pause/Resume Support Mode Call recording starts automatically when the call connects and the Pause and Resume soft key are available. The phone display indicates the status of the call recording state. Call recording stops when the call ends and the recorded part of the call is stored on the server.
- **On Demand Mode** Call recording starts on the server when the call connects, but the recorded file is not saved until the user initiates the recording. When the user presses the Start soft key, the recording is saved to the server and the phone displays the Pause and Resume soft keys.
- **On Demand Mode with User-Initiated Start Mode** Call recording does not begin automatically and a Record soft key displays. If users want to record an active call, they need to press Record > Start to start recording and save the recording to the server. While recording, the phone displays the Pause, Resume, and Stop soft keys.
- **Recording two separate calls and creating a conference** This mode enables users to record two participants as separate call sessions when connected in a conference call. The server stores the conference call as two separate recording sessions.

Busy Lamp Field (BLF)

The busy lamp field (BLF)/attendant console feature enhances support for phone-based monitoring.

The Busy Lamp Field (BLF) feature enables the following functions for users:

- Monitor the status of lines on remote phones
- Display remote party information
- Answer incoming calls to remote phones (called directed call pickup)
- Park and retrieve calls

When BLF is enabled, a BLF line key icon displays on the phone screen for users monitoring remote phones. The BLF line key displayed indicates that BLF related features are available.

BLF Types

VVX phones support the following types of BLF:

- An enhanced BLF, supported on all VVX phones except the VVX 1500 business media phone, notifies users of all states of monitored phones, including the active, idle, and ringing states.
- A basic version of BLF, available only on VVX 1500 business media phone, enables users to be monitored and to monitor idle and active phone states. The basic version of BLF enables VVX 1500 phones registered to users to fully monitor other VVX 1500 phones. However, VVX 1500 phones monitoring other VVX phones are notified of the idle and active states of monitored phones only, and are not notified of incoming calls to the monitored phones.

Note that BLF is not available with Polycom phones registered with Skype for Business Server.

BLF Icons

The following table shows the BLF key icons that display on the phone.

States	Line Icons
Monitored line is idle	👤
Monitored line is busy	👤⛔

Note: For information on how to manage calls to monitored phones, see the section "Handling Remote Calls on Attendant Phones" in *Technical Bulletin 62475: Using Statically Configured Busy Lamp Field with Polycom SoundPoint IP and VVX Phones* at Polycom Profiled UC Software Features.

BLF Feature Options

The BLF feature must be supported by a call server and the specific functions vary with the call server you use.

You may need to consult your SIP server partner or Polycom channel partner to find out how to configure BLF feature options.

You can configure the following feature options for BLF:

- Line key labels
- Enhanced feature keys
- Call appearances display
- Call waiting audio notifications
- Caller ID information display
- One-touch call park and retrieve
- One-touch directed call pickup

BLF Configuration Methods

Typically, call servers support one of two methods of BLF configuration.

Using the first method, you subscribe to a BLF resource list that is set up on your call server. Using the second method, you enter BLF resources to a configuration file and the call server directs the requests to those BLF resources. If you are unsure which method to use, consult your SIP server partner or Polycom Channel partner. This section shows you how to set up BLF using both methods.

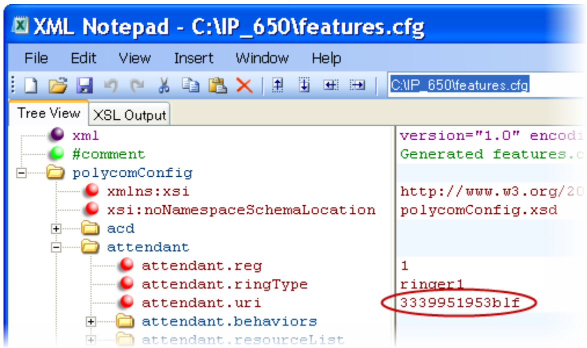
Use this feature with TCP preferred transport. When using BLF with BroadSoft, the initial subscription to BLF can receive very large responses as the number of monitored resources increases. This requires packet fragmentation which may be unreliable in its transmission across the network. In such cases, it is recommended to use TCP for BLF either by changing all SIP services to

TCP or by adding the TCP transport attribute to your attendant.uri parameter. For example: attendant.uri=1234blf@example

BLF Resource List Subscription on a Call Server

To subscribe to a BLF list on a call server, you must access the call server and set up a list of monitored resources.

The call server provides you with an address for that BLF resource list. To subscribe to that list, enter the address and any other information specific to your call server in the `attendant.uri` field located in the features.cfg template file.

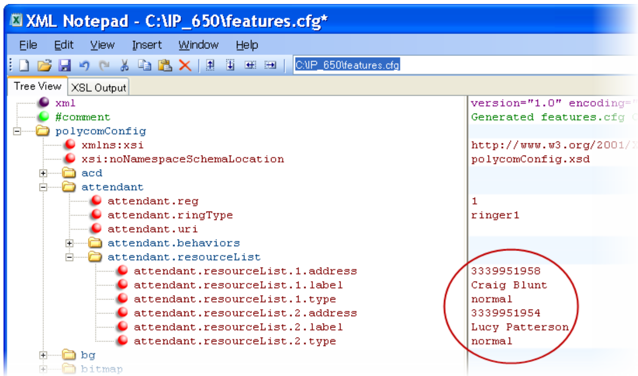


```
xml                                   version="1.0" encodi
#comment                              Generated features.c
polycomConfig
  xmlns:xsi                           http://www.w3.org/20
  xsi:noNamespaceSchemaLocation       polycomConfig.xsd
  acd
  attendant
    attendant.reg                     1
    attendant.ringType                ringer1
    attendant.uri                     3339951953blf
    attendant.behaviors
    attendant.resourceList
```

BLF Resource Specification in the Configuration File

To specify BLF resources in the configuration file, use the features.

cfg template file and enter the address (phone number) of the BLF resource of the monitored contact, the label that displays beside the line key on the phone, and the type of resource being monitored. Multiple registrations are available for a single SIP server. Your call server must support dialog even package defined in RFC 4235 in order to configure BLF using this method. In the following example, the phone is monitoring Craig Blunt and Lucy Patterson.


XML Notepad - C:\IP_650\features.cfg*
File Edit View Insert Window Help
C:\IP_650\features.cfg
Tree View
XSL Output

```
xml                                             version="1.0" encoding="
#comment                                        Generated features.cfg
polycomConfig
  xmlns:xsi                                     http://www.w3.org/2001/
  xsi:noNamespaceSchemaLocation                 polycomConfig.xsd
  acd
  attendant
    attendant.reg                               1
    attendant.ringType                          ringer1
    attendant.uri
    attendant.behaviors
    attendant.resourceList
      attendant.resourceList.1.address          3339951958
      attendant.resourceList.1.label            Craig Blunt
      attendant.resourceList.1.type             normal
      attendant.resourceList.2.address          3339951954
      attendant.resourceList.2.label            Lucy Patterson
      attendant.resourceList.2.type             normal
  bg
  bitmap
```

Specifying the type of monitored resource as `normal` or `automata` changes the default actions of key presses. Enter `normal` as the resource type if the monitored resource type is a phone and `automata` as the resource type if the monitored resource type is, for example, a call orbit. If you select `normal` , pressing the BLF line key places an active call on hold before dialing the selected BLF phone. If you select `automata` , pressing the BLF line key immediately transfers active calls to that resource.

Busy Lamp Field Configuration Parameters

The maximum number of BLF entries for phones is 50.

In the following table, x in a parameter is the number of the BLF entry in the list. If you are using static BLF, you need to configure the number of each entry.

Busy Lamp Field Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`attendant.behaviors.display.remoteCallerID.automata`	These parameters depend on the value set for the parameter `attendant.resourceList.x.type` . If the parameter `attendant.resourceList.x.type` is set to automata, use the parameter `attendant.behaviors.display.remoteCallerID.automata` . 1 (default) - Automata remote party caller ID information is presented to the attendant. 0 - The string `unknown` is substituted for both name and number information.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	attendant.behaviors.display.remoteCallerID.normal	These parameters depend on the value set for the parameter attendant.resourceList.x.type . If the parameter attendant.resourceList.x.type is set to normal, use the parameter attendant.behaviors.display.remoteCallerID.normal . 1 (default) - Normal remote party caller ID information is presented to the attendant. 0 - The string unknown is substituted for both name and number information.	No
features.cfg	attendant.behaviors.display.spontaneousCallAppearances.automata	0 (default) - The call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter. Note that the values of these call appearance parameters depend on the values applied to attendant.resourceList.x.type . When this parameter is set to 0, the ringtone ‘Ring Splash' does not play when attendant.ringType=ringer14 . 1 - The normal or automatic call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played).	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`attendant.behaviors.display.spontaneousCallAppearances.normal`	1 (default) - The normal or automatic call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played). 0 - The call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter. Note that the values of these call appearance parameters depend on the values applied to `attendant.resourceList.x.type` . When this parameter is set to 0, the ringtone ‘Ring Splash' does not play when `attendant.ringType=ringer14` .	No
`features.cfg`	`attendant.behaviours.display.spontaneousCallAppearances.automata`	Specifies how call appearances display on the attendant phone. 0 (default) - The call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter. 1 - The automata call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played). Note that the values of these call appearance parameters depend on the values applied to `attendant.resourceList.x.type` .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	attendant.behaviours.display.spontaneousCallAppearances.normal	Specifies how call appearances display on the attendant phone. 1 (default) - The normal call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played). 0 - The call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter. Note that the values of these call appearance parameters depend on the values applied to attendant.resourceList.x.type .	No
features.cfg	attendant.callWaiting.enable	0 (default) - The phone does not generate acoustic indication of call waiting for attendant calls monitored by BLF. 1 - The phone generates an acoustic indication of call waiting for attendant calls monitored by BLF.	No
features.cfg	attendant.callWaiting.ring	This parameter is valid only if attendant.callWaiting.enable is set to 1. Specifies the ring type to be used for notifying an attendant call if there is an active call already present on the phone. Silent - No acoustic indication is provided. beep - Beep tone is played when there is an active call on the phone and an attendant call is received. ring - Ring tone configured in attendant.ringType is used to alert the user when there is an active call on the phone and an attendant call is received.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`attendant.reg`	Specifies an index number for the BLF resource. The index of the registration is used to send a SUBSCRIBE to the list SIP URI specified in `attendant.uri` . For example, `attendant.reg = 2` means the second registration is used. 1 (default) Permitted value is any positive integer.	No
`features.cfg`	`attendant.resourceList.x.address`	The user referenced by `attendant.reg=""` subscribes to this URI for dialog. If a user part is present, the phone subscribes to a sip URI constructed from the user part and domain of the user referenced by `attendant.reg` . Transport for BLF subscriptions may be modified by including a transport parameter into the subscription address. For example: `sip:blf12345@domain.com;transport=tcp` Permitted value is a string that constitutes a valid SIP URI (`sip:6416@polycom.com`) or contains the user part of a SIP URI (`6416`). Null (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	attendant.resourceList.x.bargeInMode	Enable or disable barge-in and choose the default barge-in mode. This parameter applies to the Alcatel-Lucent CTS only. Null (default) - If no value is entered, the Barge In feature is disabled. All - Press and hold the BLF line to display all barge-in options. Quick press to barge-in as Normal. Normal - Barge-in plays an audio tone to indicate the arrival of a new participant to the call and all call participants can interact. Listen - The user barging in can listen on the call only. Their outbound audio is not transmitted to either party. Whisper - The user barging in can hear all parties but their audio is only transmitted to the user they are monitoring.	No
features.cfg	attendant.resourceList.x.callAddress	Use this parameter when the call signaling address for the BLF line is different than the address set by attendant.resourceList.x.address. Null (default) Maximum 255 characters	No
features.cfg	attendant.resourceList.x.label	The text label displays adjacent to the associated line key. If set to Null, the label is derived from the user part of attendant.resourceList.x.address . Null (default) Permitted value is a UTF-8 encoded string.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	attendant.resourceList.x.proceedingIsRecipient	A flag to determine if pressing the associated line key for the monitored user picks up the call. 1 - If the call server does not support inclusion of the direction attribute in its dialog XML. 0 (default)	No
features.cfg	attendant.resourceList.x.requestSilentBargeIn	0 (default) - A tone plays when a contact barges in on a call. 1 - No tone is played when a contact barges in on a call.	No
features.cfg	attendant.resourceList.x.type	The type of resource being monitored and the default action to perform when pressing the line key adjacent to monitored user x. normal (default) -The default action is to initiate a call if the user is idle or busy and to perform a directed call pickup if the user is ringing. Any active calls are first placed on hold. Note that the value normal applies the call appearance setting attendant.behaviors.display.*.normal . automata -The default action is to perform a park/blind transfer of any currently active call. If there is no active call and the monitored user is ringing/busy, an attempt to perform a directed call pickup/park retrieval is made. Note that the value automata applies the call appearance setting attendant.behaviors.display.*.automata=0 .	No
features.cfg	attendant.restrictPickup	0 (default) - The attendant can pick up calls to monitored users while they show as ringing. 1 - The attendant cannot pick up the monitored call.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`attendant.ringType`	The ringtone that plays when a BLF dialog is in the offering state. ringer1 (default) ringer1 - ringer 24	No
`features.cfg`	`attendant.uri`	The list SIP URI on the server. If this is just a user part, the URI is constructed with the server hostname/IP. Note: If this parameter is set, then the individually addressed users configured by `attendant.resourceList` and `attendant.behaviors` are ignored. Null (default) Strings are permitted.	No
`sip-interop.cfg`	`call.directedCallPickupMethod`	Specifies how the phone performs a directed call pick-up from a BLF contact. legacy (default) - Indicates that the phone uses the method specified in `call.directedCallPickupString .` `native` - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header.	No
`sip-interop.cfg, site.cfg`	`call.directedCallPickupString`	The star code to initiate a directed call pickup. *97 (default) Note: The default value supports the BroadWorks calls server only. You must change the value if your organization uses a different call server.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.parkedCallRetrieveMethod`	The method the phone uses to retrieve a BLF resource's call which has dialog state confirmed. legacy (default) - Indicates that the phone uses the method specified in `call.parkedCallRetrieveString` . `native` - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header).	No
`sip-interop.cfg, site.cfg`	`call.parkedCallRetrieveString`	The star code that initiates retrieval of a parked call. Null (default) Permitted values are star codes.	No
`sip-interop.cfg`	`voipPort.SIP.useCompleteUriForRetrieve`	1 (default) - The target URI in BLF signaling uses the complete address as provided in the XML dialog document. 0 - Only the user portion of the target URI in the XML dialog document is used and the current registrar's domain is appended to create the address for retrieval.	No
`sip-interop.cfg`	`voIpProt.SIP.strictReplacesHeader`	This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources. 1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when call.directedCallPickupMethod is configured as native. 0 - Call pick-up requires a call id only.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.useLocalTargetUriforLegacyPickup	1 (default) - The target URI in BLF signaling uses the complete address as provided in the XML dialog document. 0 - Only the user portion of the target URI in the XML dialog document is used and the current registrar's domain is appended to create the address for pickup or retrieval.	No

Instant Messaging

All VVX phones can send and receive instant text messages.

When instant messaging is enabled, the phone's message waiting indicator (MWI) visually alerts users new instant messages; you can also set audio alerts.

Support for Instant Messaging varies by call server. Consult your SIP server partner to find out if this feature is supported. Instant Messaging is not with Skype for Business.

Related Links

Local Contact Directory Parameters on page 277

Instant Messaging Parameters

Use the parameters in the following table to configure this feature.

Instant Messaging Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.messaging.enabled	0 (default) - Disable instant messaging. 1 - Enable instant messaging.	Yes

Local and Centralized Conference Calls

You can set up local or centralized audio and video conferences.

Local conferences require a host phone to process the audio and video of all parties. Alternatively, users can use an external audio bridge, available via a central server, to create a centralized conference call. All Polycom phones support local- and server-based centralized conferencing. Polycom recommends using centralized conferencing for conferences with four or more parties. The availability of centralized conferencing and features can vary by the call platform you use.

VVX phones and SoundStructure VoIP Interface support a maximum of three participants in local conference calling.

Local and Centralized Conference Call Parameters

The following table lists available call management parameters.

Use the parameters in the following table to set up a conference type and the options available for each type of conference.

You can specify whether, when the host of a three-party local conference leaves the conference, the other two parties remain connected or disconnected. If you want the other two parties remain connected, the phone performs a transfer to keep the remaining parties connected. If the host of four-party local conference leaves the conference, all parties are disconnected and the conference call ends. If the host of a centralized conference leaves the conference, each remaining party remains connected. For more ways to manage conference calls, see Conference Management.

Local and Centralized Conference Call Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	call.localConferenceCallHold	0 (default) - The host cannot place parties on hold. 1 - During a conference call, the host can place all parties or only the host on hold.	No
sip-interop.cfg	call.transferOnConferenceEnd	1 (default) - After the conference host exits the conference, the remaining parties can continue. 0 - After the conference host exits the conference, all parties are exited and the conference ends.	No
sip-interop.cfg	call.singleKeyPressConference	Specify whether or not all parties hear sound effects while setting up a conference. 0 (default) - Phone sound effects are heard only by the conference initiator. 1 - A conference is initiated when a user presses Conference the first time. Also, all sound effects (dial tone, DTMF tone while dialing and ringing back) are heard by all participants in the conference.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.conference.address	Null (default) - Conferences are set up on the phone locally. String 128 max characters - Enter a conference address. Conferences are set up by the server using the conferencing agent specified by this address. Acceptable values depend on the conferencing server implementation policy.	No

Conference Management

This feature enables users to add, hold, mute, and remove conference participants, as well as obtain additional information about participants.

VVX phone users can also choose which conference call participants to exchange video with.

When you enable conference management, a Manage soft key displays on the phone during a conference, and users can use the soft key to access conference management options.

Conference Management Parameters

Use the parameters in the following table to enable this feature.

Conference Management Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.nWayConference.enabled	0 (default) - Disable the n-way conferencing managing feature. You can hold three-way conferences but the options to manage the conference do not display. 1 - Enable n-way conferencing. You can hold conferences with the maximum number of parties, and the options to manage the conference display.	No

Local Digit Map

The local digit map feature allows the phone to automatically call a dialed number when configured.

Dial plans apply on-hook when no Skype for Business line is registered or when line switching is enabled and at least one line has a non-empty dial plan.

Digit maps are defined by a single string or a list of strings. If a dialed number matches any string of a digit map, the call is automatically placed. If a dialed number matches no string—an impossible match—you can specify the phone's behavior. If a number ends with #, you can specify the phone's behavior, called trailing # behavior. You can also specify the digit map timeout, the period of time after you dial a number that the call is placed. The configuration syntax of the digit map is based on recommendations in section 2.1.5 of RFC 3435.

Note: For instructions on how to modify the local digit map, see *Technical Bulletin 11572: Changes to Local Digit Maps on SoundPoint IP, SoundStation IP, and Polycom VVX 1500 Phones* at Polycom Engineering Advisories and Technical Notifications.

Local Digit Maps Parameters

Polycom support for digit map rules varies for open SIP servers and Microsoft Skype for Business Server.

Use the parameters in the following table to configure this feature.

Configure the Local Digit Map

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.applyToCallListDial`	Choose whether the dial plan applies to numbers dialed from the received call list or missed call list, including sub-menus. 1 (default) 0	Yes
`site.cfg`	`dialplan.applyToDirectoryDial`	Lync Base Profile – 1 (default) Generic Base Profile – 0 (default) 0— The dial plan is not applied to numbers dialed from the directory or speed dial, including auto-call contact numbers. 1—The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call contact numbers.	Yes
`site.cfg`	`dialplan.applyToForward`	Lync Base Profile – 1 (default) Generic Base Profile – 0 (default) 0—The dial plan does not apply to forwarded calls. 1—The dial plan applies to forwarded calls.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dialplan.applyToTelUriDial	Choose whether the dial plan applies to URI dialing. 1 (default) 0	Yes
site.cfg	dialplan.applyToUserDial	Choose whether the dial plan applies to calls placed when the user presses Dial. 1 (default) 0	Yes
site.cfg	dialplan.applyToUserSend	Choose whether the dial plan applies to calls placed when the user presses Send. 1 (default) 0	Yes
site.cfg	dialplan.conflictMatchHandling	0 (default for Generic Profile) 1 (default for Skype Profile)	
site.cfg	dialplan.digitmap.timeOut	Specify a timeout in seconds for each segment of the digit map using a string of positive integers separated by a vertical bar (|). After a user presses a key, the phone waits this many seconds before matching the digits to a dial plan and dialing the call. (Default) 3 | 3 | 3 | 3 | 3| 3 If there are more digit maps than timeout values, the default value 3 is used. If there are more timeout values than digit maps, the extra timeout values are ignored.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.digitmap`	Specify the digit map used for the dial plan using a string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. This parameter enables the phone to automatically initiate calls to numbers that match a digit map pattern. Generic Base Profile (default) – `[2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT` Lync Base Profile (default) – NULL `[2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT` (default) The string is limited to 2560 bytes and 100 segments of 64 bytes, and the following characters are allowed in the digit map • A comma (,), which turns dial tone back on. • A plus sign (+) is allowed as a valid digit. • The extension letter 'R' indicates replaced string. • The extension letter 'Pn' indicates precedence, where 'n' range is 1-9. 1—Low precedence 9—High precedence	Yes
`debug.cfg`	`dialplan.filterNonDigitUriUsers`	Determine whether to filter out (+) from the dial plan. 0 (default) 1	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.impossibleMatchHandling`	0 (default)—The digits entered up to and including the point an impossible match occurred are sent to the server immediately. 1—The phone gives a reorder tone. 2—Users can accumulate digits and dispatch the call manually by pressing Send. If a call orbit number begins with pound (#) or asterisk (*), you need to set the value to 2 to retrieve the call using off-hook dialing.	Yes
`site.cfg`	`dialplan.removeEndOfDial`	Sets if the trailing # is stripped from the digits sent out. 1 (default) 0	Yes
`site.cfg`	`dialplan.routing.emergency.outboundIdentity`	Choose how your phone is identified when you place an emergency call. NULL (default) 10-25 digit number SIP TEL URI If using a URI, the full URI is included verbatim in the P-A-I header. For example: • `dialplan.routing.emergency.outboundIdentity` = 5551238000 • `dialplan.routing.emergency.outboundIdentity` = sip:john@emergency.com • `dialplan.routing.emergency.outboundIdentity` = tel: +16045558000	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dialplan.routing.emergency.preferredSource	Set the precedence of the source of emergency outbound identities. ELIN (default)— the outbound identity used in the SIP P-Asserted-Identity header is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN). Config— the parameter dialplan.routing.emergency.outboundIdentity has priority when enabled, and the LLDP-MED ELIN value is used if dialplan.routing.emergency.outboundIdentity is NULL.	No
site.cfg	dialplan.routing.emergency.x.description	Set the label or description for the emergency contact address. x=1: Emergency, Others: NULL (default) string x is the index of the emergency entry description where x must use sequential numbering starting at 1.	Yes
site.cfg	dialplan.routing.emergency.x.server.y	Set the emergency server to use for emergency routing (dialplan.routing.server.x.address where x is the index). x=1: 1, Others: Null (default) positive integer x is the index of the emergency entry and y is the index of the server associated with emergency entry x. For each emergency entry (x), one or more server entries (x,y) can be configured. x and y must both use sequential numbering starting at 1.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.routing.emergency.x.value`	Set the emergency URL values that should be watched for. When the user dials one of the URLs, the call is directed to the emergency server defined by `dialplan.routing.server.x.address` . x=15: 911, others: Null (default) SIP URL (single entry) x is the index of the emergency entry description where x must use sequential numbering starting at 15.	No
`site.cfg`	`dialplan.routing.server.x.address`	Set the IP address or hostname of a SIP server to use for routing calls. Multiple servers can be listed starting with x=1 to 3 for fault tolerance. Null (default) IP address hostname Blind transfer for 911 or other emergency calls may not work if registration and emergency servers are different entities.	Yes
`site.cfg`	`dialplan.routing.server.x.port`	Set the port of a SIP server to use for routing calls. 5060 (default) 1 to 65535	Yes
`site.cfg`	`dialplan.routing.server.x.transport`	Set the DNS lookup of the first server to use and dialed if there is a conflict with other servers. DNSnaptr (default) TCPpreferred UDPOnly TLS TCPOnly For example, if `dialplan.routing.server.1.transport` = "UDPOnly" and `dialplan.routing.server.2.transport` = "TLS", then UDPOnly is used.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.userDial.timeOut`	Specify the time in seconds that the phone waits before dialing a number entered while the phone is on hook. Generic Base Profile (default) – 0 Lync Base Profile (default) – 4 0-99 seconds You can apply `dialplan.userDial.timeOut` only when its value is lower than `up.IdleTimeOut` .	No

Open SIP Digit Map

If you are using a list of strings, each string in the list can be specified as a set of digits or timers, or as an expression which the gateway uses to find the shortest possible match.

In addition, the digit map feature allows SIP URI dialing to match the URIs based on dial plan.

The following is a list of digit map string rules for open SIP environments.

- The following letters are case sensitive: x, T, R, S, and H.
- You must use only *, #, +, or 0-9 between the second and third R.
- If a digit map does not comply, it is not included in the digit plan as a valid map. That is, no match is made.
- There is no limit to the number of R triplet sets in a digit map. However, a digit map that contains less than a full number of triplet sets (for example, a total of 2 Rs or 5 Rs) is considered an invalid digit map.
- Digit map extension letter R indicates that certain matched strings are replaced. Using an RRR syntax, you can replace the digits between the first two Rs with the digits between the last two Rs. For example, *R555R604R* would replace 555 with 604. Digit map timer letter T indicates a timer expiry. Digit map protocol letters S and H indicate the protocol to use when placing a call.
- If you use T in the left part of RRR's syntax, the digit map will not work. For example, R0TR322R will not work.

The following examples illustrate the semantics of the syntax:

- R9R604Rxxxxxxx-Replaces 9 with 604
- xxR601R600Rxx-When applied to 1160122 gives 1160022
- R9RRxxxxxxx-Remove 9 at the beginning of the dialed number (replace 9 with nothing)
    - For example, if you dial 914539400, the first 9 is removed when the call is placed.
- RR604Rxxxxxxx-Prepend 604 to all seven-digit numbers (replace nothing with 604)
    - For example, if you dial 4539400, 604 is added to the front of the number, so a call to 6044539400 is placed.
- xR60xR600Rxxxxxxx-Replace any 60x with 600 in the middle of the dialed number that matches.

  For example, if you dial 16092345678, a call is placed to 16002345678.

- 911xxx.T-A period (.) that matches an arbitrary number, including zero, of occurrences of the preceding construct. For example:
- 911123 with waiting time to comply with T is a match
- 9111234 with waiting time to comply with T is a match
- 91112345 with waiting time to comply with T is a match and the number can grow indefinitely given that pressing the next digit takes less than T.
- `sip\:764xxxxxRR@registrar.polycomcsn.comR` - appends `@registrar.polycomcsn.com` to any URI calls matching with "764xxxxx".

  For example, if you make a SIP URI call with 76412345 then `@registrar.polycomcsn.com` is appended to the string such that the SIP URI call INVITE becomes `sip:: 76412345@vc.polycom.com` . Here, `@domain` string is required only for SIP URI calls from unregistered lines.
- `sip\:xxxx\@registrar\.polycomcsn\.com` - This will match with any four digit URI calls having the domain `@registrar.polycomcsn.com` .

  For example, if you configure three lines and has dial plan based line switching enabled. Now, if the third line's dial plan has `sip\:xxxx\@registrar\.polycomcsn\.com` then call will be initiated from the third line if user dial `1234@registrar.polycomcsn.com` because it matches with the third line's dial plan.
- `0xxxS|33xxH` —All four digit numbers starting with a 0 are placed using the SIP protocol, whereas all four digit numbers starting with 33 are placed using the H.323 protocol.

Note: Only VVX 500/510, 600/611, and 1500 phones support the H. On all other phones, the H is ignored and users need to perform the Send operation to complete dialing. For example, if the digit map is 33xxH, the result is as follows: If a VVX 1500 user dials 3302 on an H.323 or dual protocol line, the call is placed after the user dials the last digit.

Generating Secondary Dial Tone with Digit Maps

You can regenerate a dial tone by adding a comma "," to the digit map.

You can dial seven-digit numbers after dialing "8" as shown next in the example rule `8,[2-9]xxxxxxT` :

```
[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|8,[2-9]xxxxxxT|[2-9]xx.T
```

By adding the digit "8", the dial tone plays again, and users can complete the remaining seven-digit number. In this example, if users also have a 4-digit extension that begins with "8", then users will hear dial tone after the first "8" was dialed because "8" matches the "8" in the digit map.

If you want to generate dial tone without the need to send the "8", replace one string with another using the special character "R" as shown next in the rule *R8RR*. In the following example, replace "8" with an empty string to dial the seven-digit number:

```
[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|R8RR,[2-9]xxxxxxT|[2-9]xx.T
```

Enhanced 911 (E.911)

This E.911 feature allows you to configure one of three sources the phone obtains location information from:

- LLDP-MED
- DHCP via option 99

- LIS compliant with RFC 5985

Configuring the source of location information allows the phone to share its location details in the invite sent when a 911 call is made to ensure the 911 operator dispatches emergency services to the correct address.

Enhanced 911 (E.911) Parameters

Use the following parameters to configure E.911.

E.911 Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`feature.E911.HELD.server`	NULL (default) Set the IP address or hostname of the Location Information Server (LIS) address. For example, host.domain.com or https://xxx.xxx.xxx.xxx.	No
`site.cfg`	`feature.E911.HELD.username`	NULL (default) Set the user name used to authenticate to the Location Information Server.	No
`site.cfg`	`feature.E911.HELD.password`	NULL (default) Set the password used to authenticate to the Location Information Server.	No
`site.cfg`	`feature.E911.HELD.identity`	Set the vendor-specific element to include in a location request message. For example, ‘companyID’. NULL (default) String 255 character max	No
`site.cfg`	`feature.E911.HELD.identityValue`	Set the value for the vendor-specific element to include in a location request message. NULL (default) String 255 character max	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	feature.E911.locationRetryTimer	Specify the retry timeout value in seconds for the location request sent to the Location Information Server (LIS). The phone does not retry after receiving location information received through the LIS. 60 seconds (default) 60 - 86400 seconds	No
site.cfg	feature.E911.HELD.nai.enable	You can include or omit the Network Access Identifier (NAI) containing the SIP user information used to subscribe to the Location Information Server (LIS). 0 (default) – The NAI is omitted as a device identity in the location request sent to the LIS. 1 - The NAI is included as a device identity in the location request sent to the LIS.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`locInfo.source`	Specify the source of phone location information. This parameter is useful for locating a phone in environments that have multiple sources of location information. LLDP (default for Generic Base Profile) – Use the network switch as the source of location information. MS_E911_LIS (default for Lync Base Profile)– Use the Skype for Business Server as the source of location information. CONFIG – You can manually configure the source of location information. Skype only. LIS – Use the location information server as the source of location information. Generic Base Profile only. DHCP – Use DHCP as the source of location information. Generic Base Profile only. If location information is not available from a default or configured source, the fallback priority is as follows: Generic Base Profile: No fallback supported for Generic Base Profile Lync Base Profile: MS_E911_LIS > CONFIG > LLDP	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`feature.E911.enabled`	0 (default) – Disable the E.911 feature. The INVITE sent for emergency calls from the phone does not include the geolocation header, geolocation option in supported header, geolocation-routing header, or the GEOPRIV location object. 1 – Enable the E.911 feature. The INVITE sent for emergency calls from the phone includes the geolocation header defined in RFC 6442 and PIDF presence element as specified in RFC3863 with a GEOPRIV location object specified in RFC4119 for in Open SIP environments. This parameter is mutually exclusive of the GENBAND E.911 feature and if this parameter and feature.genband.E911.enabled are enabled, this parameter takes precedence.	No
`site.cfg`	`feature.E911.HELD.requestType`	Any (default) - Send a request to the Location Information Server (LIS) to return either 'Location by Reference' or 'Location by Value'. Note this is not the 'Any' value referred to in RFC 5985. Civic – Send a request to the LIS to return a location by value in the form of a civic address for the device as defined in RFC 5985. RefID – Send a request to the LIS to return a set of Location URIs for the device as defined in RFC 5985.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`voIpProt.SIP.header.priority.enable`	0 (default) – Do not include a priority header in the E.911 INVITE message. 1 - Include a priority header in the E.911 INVITE message.	No
`site.cfg`	`voIpProt.SIP.header.geolocation-routing.enable`	0 (default) – Do not include the geolocation-routing header in the E. 911 INVITE message. 1 - Include the geolocation-routing header in the E.911 INVITE message.	No
`site.cfg`	`feature.E911.HELD.secondary.server`	Set the IP address or hostname of the secondary Location Information Server (LIS) address. For example, host.domain.com or https://xxx.xxx.xxx.xxx. NULL (default) Dotted-decimal IP address Hostname Fully-qualified domain name (FQDN)	No
`site.cfg`	`feature.E911.HELD.secondary.username`	Set a user name to authenticate to the secondary Location information Server (LIS). NULL (default) String	No
`site.cfg`	`feature.E911.HELD.secondary.password`	Set a password to authenticate to the secondary LIS. NULL (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	feature.E911.usagerule.retransmission	0 (default) - The recipient of this Location Object is not permitted to share the enclosed Location Information, or the object as a whole, with other parties. 1 - Distributing this Location is permitted.	No

Assured Services - Session Initiation Protocol (AS-SIP)

The Assured Services-Session Initiation Protocol (AS-SIP) feature provides the mechanism that allows outgoing precedence (priority) calls to be created.

Use dialplan.digitmap parameter to configure an outgoing call's precedence. You can create Multi-Level Precedence and Preemption (MLPP) for outgoing calls. The precedence levels are automatically assigned to the calls in the initial outgoing signaling. You can configure a call’s precedence level to be changed by the Session Initiation Protocol (SIP) server in subsequent signaling.

When all call appearances are occupied for a line and a new incoming call having a higher priority than the other calls in the same line is in place, the call appearance with the lowest priority is removed for the new call. When the calls are of the same precedence level, the calls in progress are selected for preemption in the following order: alerting, held then active. For calls of the same precedence level and the same state, more recently created alerting calls held for shorter or longer duration are selected first. The conference calls are treated as a single call for the purposes of preemption as they occupy only one call appearance until they are split.

The precedence level assigned to an incoming call alters the following behavior:

- **Ring Type:** The Resource-Priority header contents map to one or more custom ring types which may in turn map to a unique ringer sound effect pattern. When the Resource-Priority header is present, the feature by which the ring type assignment is mapped to Alert-Info header content will be disabled. The default behavior for precedence calls will be the normal ringer pattern with an accelerated tone. The ring type and subsequent ring pattern assigned to precedence calls via the Resource-Priority header mapping has higher priority than other types of distinctive incoming call. This includes per-line ring pattern assignment and local contact directory-based treatment such as auto-divert or auto-reject.
- **Call Waiting:** When the call.callWaiting.ring parameter is set to beep, a beep tone plays through the selected audio output mode on the active call. If the parameter is set to ring, the call waiting tone pattern is derived from the ring type assigned to the alerting call (se.rt.xxx.callWait) and plays through the speaker. The default behavior for precedence calls will be the normal call waiting pattern with three short busts rather than one 300 ms.
- **Precedence Handling:** When a user receives multiple precedence calls, they are displayed on the call alerting list according to their precedence level. The high precedence calls are listed at the top and the low precedence calls are listed at the bottom of the call alerting list.

- **Visual Indication:** The priority string displays as follows on the call screen of the phone for priority calls:
  - **VVX 101, VVX 201, VVX 3xx business media phones:** P-1,P-2,P-3
  - **VVX 4xx, VVX 5xx, VVX 6xx, VVX 15xx business media phones:** Priority-1,Priority-2,Priority-3
  - **VVX 150 business IP phone:** P-1,P-2,P-3
  - **VVX 250, 350, 450 business IP phones:** Priority-1,Priority-2,Priority-3

Preemption Behavior on Low Priority Calls

A 180 ringing response is sent to the far end only when a call appearance is allocated for the incoming precedence call.

The following table illustrates the preemption behavior of the low priority call's status.

Preemption Behavior on Low Priority Calls

Low Priority Call's Status for Preemption	Behavior
Connected	The call is terminated with a BYE request containing a preemption Reason header, and a local preemption tone is played for a configurable duration or until the user hangs up, whichever comes first.
Locally Held	The call may be terminated with a BYE request containing a preemption Reason header.
Alerting	A 486 Busy Here response is sent to the far end containing a preemption Reason header.
Dial Tone or Setup	When the final call appearance is in the dial tone or setup (digit collection) state (including consultation calls) and a precedence call arrives, no action is taken until the new outgoing call is of higher priority or is not is determined. If the call is of lower priority, then the call is not placed and a preemption tone is played for a configurable duration or until the user hangs up, whichever is less. If the call is of the same or higher priority, then the incoming call is terminated by sending a 486 Busy Here response to the far end containing a preemption Reason header.

Low Priority Call's Status for Preemption	Behavior
Preceding	If the final call appearance is in the dial tone or setup (digit collection) state (including consultation calls) when a precedence call arrives, no action is taken until it can be determined whether the new outgoing call is of higher priority or not. If the call is determined to be of lower priority, then the call is not placed and a preemption tone should be played for a configurable duration or until the user hangs up, whichever is less. If the call is determined to be of the same or higher priority, then the incoming call is terminated by sending a 486 Busy Here response to the far end containing a preemption Reason header.

AS-SIP Parameters

The following table lists the parameters to configure the AS-SIP feature.

AS-SIP Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.assuredService.enable	0 (default) - Disables the AS-SIP feature. 1 - Enables the AS-SIP feature	No
sip-interop.cfg	voIpProt.SIP.assuredService.defaultPriority	Default priority assigned to an outgoing call. 1 (default) 1 to 10 This value will be overriden if priority is assigned from dialplan for that number.	No
sip-interop.cfg	voIpProt.SIP.assuredService.precedenceThreshold	The minimum call priority required for a call to be considered a precedence call. 2 (default) 1 to 10	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.assuredService.serverControlled	1 (default) - The precedence level for an outgoing call is owned by the server or non-EI equipment. The precedence level may appear in SIP responses and it may differ from the level originally set by the phone; 0 - The precedence level is owned by the phone and must not change.	No
sip-interop.cfg	voIpProt.SIP.assuredService.preemptionAutoTerminationDelay.local	Duration in seconds after a local preemption event before a call appearance is automatically cleared. 0 (default) 0- 3600	No
sip-interop.cfg	voIpProt.SIP.assuredService.preemptionAutoTerminationDelay.remote	Duration in seconds after a remote preemption event before a call appearance is automatically cleared. 3 (default) 0-3600	No
reg-advanced.cfg	voIpProt.SIP.assuredService.namespace	The name space scheme to use in SIP signaling. UCRdsn (default) dsn drsn UCRdrsn custom ets	No
sip-interop.cfg	voIpProt.SIP.assuredService.namespace.custom.name	The name for the custom namespace label. Null (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.assuredService.namespace.custom.priority.x`	The namespace priority values, lowest to highest. Null (default) String	No

Bluetooth Support for VVX Business Media Phones

You can enable VVX 600 and 601 business media phones to pair and connect with Bluetooth devices such as smartphones and tables.

When enabled, users can set a device name for supported VVX phones so that users can identify the VVX phone while scanning or connecting to it over Bluetooth. Users can also manage calls and enter DTMF digits from the VVX phone by setting the phone as the audio device for their Bluetooth device.

Bluetooth Device Parameters

The following table includes the configuration parameters you can use to configure Bluetooth discovery and set the phone Bluetooth device name.

Bluetooth Device Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`bluetooth.device.discoverable`	Specify the discovery mode to make VVX 600/601 phones visible to other Bluetooth devices. 1 (Default) 0	No
`features.cfg`	`bluetooth.device.name`	Specifies the Bluetooth device name of the device. String (default) 1 - Minimum 20 - Maximum	No

International Dialing Prefix

Enter a '+' symbol before you dial an international phone numbers to identify to the switch that the phone number you are dialing is international.

International Dialing Prefix Parameters

The following parameters configure the international dialing prefixes.

International Dialing Prefix Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`call.internationalDialing.enabled`	This parameter applies to all numeric dial pads on the phone, including for example, the contact directory. Changes you make to this parameter cause a restart or reboot. 1 (default) - Disable the key tap timer that converts a double tap of the asterisk "*" symbol to the "+" symbol to indicate an international call. By default, this parameter is enabled so that a quick double tap of "*" converts immediately to "+". To enter a double asterisk "**", tap "*" once and wait for the key tap timer to expire to enter a second "*". 0 - When you disable this parameter, you cannot dial"+" and you must enter the international exit code of the country you are calling from to make international calls.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	call.internationalPrefix.key	The phone supports international call prefix (+) with both '0' and '*'. 0 (default) - Set the international prefix with *. 1 - Set the international prefix with 0.	No

Shared Lines

Topics:

- Shared Call Appearances
- Private Hold on Shared Lines
- Intercom Calls
- Push-to-Talk
- Group Paging
- SIP-B Automatic Call Distribution

This section shows you how to configure shared line features.

Shared Call Appearances

Shared call appearance enables an active call to display simultaneously on multiple phones in a group.

All call states of a call—active, inactive, on hold—are displayed on all phones of a group.

By default, the answering phone has sole access to the incoming call, which is called line seize. If the answering phone places the call on hold, that call becomes available for pickup to all phones in that group. You can enable other phones in the group the ability to enter a conversation on one of the group phones, which is referred to as a barge in.

Note: Shared call appearances and bridged line appearances are similar signaling methods that enable more than one phone to share the same line or registration. The method you use varies with the SIP call server you are using.

Shared Call Appearances Parameters

This feature is dependent on support from a SIP call server.

To enable shared call appearances on your phone, you must obtain a shared line address from your SIP service provider.

Use the parameters in the following table to configure options for this feature.

.

Shared Call Appearances Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-basic.cfg`	`reg.x.address`	The user part (for example, 1002) or the user and the host part (for example, `1002@polycom.com`) of the registration SIP URI or the H.323 ID/ extension. Null (default) string address	No
`reg-advanced.cfg`	`reg.x.type`	private (default) - Use standard call signaling. shared - Use augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.	No
`sip-interop.cfg`	`call.shared.reject`	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.	No
`sip-interop.cfg`	`call.shared.exposeAutoHolds`	0 (default) - No re-INVITE is sent to the server when setting up a conference on a shared line. 1 - A re-INVITE is sent to the server when setting up a conference on a shared line.	Yes
`sip-interop.cfg`	`call.shared.oneTouchResume`	0 (default) - Selecting the shared line opens all current calls that the user can choose from. 1 - All users on a shared line can resume held calls by pressing the shared line key. If more than one call is on hold, the first held call is selected and resumed. A quick press and release of the line key resumes a call whereas pressing and holding down the line key shows a list of calls on that line.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	call.shared.preferCallInfoCID	0 (default) - The Caller-ID information received in the 200 OK status code is not ignored if the NOTIFY message received with caller information includes display information. 1 - The Caller-ID information received in the 200 OK status code is ignored if the NOTIFY message received with caller information includes display information.	No
sip-interop.cfg, site.cfg	call.shared.remoteActiveHoldAsActive	1 (default) - Shared remote active/hold calls are treated as a active call on the phone. 0 - Shared remote active/hold calls are not treated as a active call on the phone.	No
sip-interop.cfg	call.shared.seizeFailReorder	1 (default) - Play a re-order tone locally on shared line seize failure. 0 - Do not play a re-order tone locally on shared line seize failure.	Yes
sip-interop.cfg	voIpProt.SIP.specialEvent.lineSeize.nonStandard	Controls the response for a line-seize event SUBSCRIBE. 1 (default) - This speeds up the processing of the response for line-seize event. 0 - This will process the response for the line seize event normally	Yes
reg-advanced.cfg	reg.x.ringType	The ringer to be used for calls received by this registration. The default is the first non-silent ringer. If you use the configuration parameters ringer13 and ringer14 on a single registered line, the phone plays SystemRing.wav. default (default) ringer1 to ringer24	No
sip-interop.cfg	reg.x.protocol.H323	You can use this parameter for the VVX 500/501, 600/601, and 1500. 0 (default) - H.323 signaling is not enabled for registration x. 1 - H.323 signaling is enabled for registration x.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`reg.x.server.H323.y.address`	Address of the H.323 gatekeeper. Null (default) IP address or hostname	No
`site.cfg`	`reg.x.server.H323.y.port`	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. 0 (default) 0 to 65535	No
`site.cfg`	`reg.x.server.H323.y.expires`	Desired registration period. 3600 positive integer	No
`site.cfg`	`reg.x.line.y.label`	Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when `up.cfgUniqueLineLabel=1` . If `reg.x.linekeys=1` , this parameter does not have any effect. x = the registration index number starting from 1. y = the line index from 1 to the value set by `reg.x.linekeys` . Specifying a string sets the label used for the line key registration on phones with multiple line keys. If no parameter value is set for `reg.x.line.y.label` , the phone automatically numbers multiple lines by prepending "<y>_" where <y> is the line index from 1 to the value set by `reg.x.linekeys` . • The following examples show labels for line 1 on a phone with user registration 1234, where `reg.x.linekeys=2` : ◦ If no label is configured for registration, the labels are "1_1234" and "2_1234". ◦ If `reg.1.line.1.label=Polycom` and `reg.1.line.2.label=VVX` , the labels display as 'Polycom' and 'VVX'.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.callsPerLineKey	Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration. This per-registration parameter overrides call.callsPerLineKey . 24 (default) 1-24 VVX 101, 201 8 (default) 1 - 8	No
reg-advanced.cfg	reg.x.header.pearlymedia.support	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.	No
reg-basic.cfg	reg.X.insertOBPAddressInRoute	1 (Default) - The outbound proxy address is added as the topmost route header. 0 - The outbound proxy address is not added to the route header.	No
features.cfg	reg.x.path	0 (Default) - The path extension header field in the Register request message is not supported for the specific line registration. 1 - The phone supports and provides the path extension header field in the Register request message for the specific line registration.	No
features.cfg	reg.x.regevent	0 (default) - The phone is not subscribed to registration state change notifications for the specific phone line. 1 - The phone is subscribed to registration state change notifications for the specific phone line. This parameter overrides the global parameter voIpProt.SIP.regevent.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.rejectNDUBInvite	Specify whether or not 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) - If an NDUB event occurs, the phone does not reject the call. 1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code.	No
reg-advanced.cfg	reg.x.server.y.specialInterop	Specify the server-specific feature set for the line registration. Standard (Default) VVX 101: Standard GENBAND ALU-CTS DT VVX 201: Standard, GENBAND ALU-CTS ocs2007r2 lync2010 All other phones: Standard GENBAND ALU-CTS ocs2007r2 lync2010 lcs2005	
sip-interop.cfg	reg.x.gruu	1 - The phone sends sip.instance in the REGISTER request. 0 (default) - The phone does not send sip.instance in the REGISTER request.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.serverFeatureControl.securityClassification	0 (default) - The visual security classification feature for a specific phone line is disabled. 1 - The visual security classification feature for a specific phone line is enabled.	No
reg-advanced.cfg	reg.x.terminationType	Determines the type of termination that is used for the line where the line can be managed automatically on the VVX, the wireless handset, or on both. X = each registration index. NULL (default) VVX, DECT, or VVX-DECT	No
reg-advanced.cfg reg-advanced.cfg	reg.x.acd-login-logout reg.x.acd-agent-available	0 (default) - The ACD feature is disabled for registration. 1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.	No
reg-advanced.cfg	reg.x.advancedConference.maxParticipants	Sets the maximum number of participants allowed in a push to conference for advanced conference calls. The number of participants configured must match the number of participants allowed on the ALU CTS. 3 (default) 0 - 25	No
reg-advanced.cfg	reg.x.advancedConference.pushToConference	0 (default) - Disable push-to-conference functionality. 1 - Enable push-to-conference functionality.	No
reg-advanced.cfg	reg.x.advancedConference.subscribeForConfEvents	1 (default) - Conference participants to receive notifications for conference events is enabled. 0 - Conference participants to receive notifications for conference events is disabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.advancedConference.subscribeForConfEventsOnCCPE	1 (default) - Enable the conference host to receive notifications for conference events. 0 - Disable the conference host to receive notifications for conference events.	No
reg-advanced.cfg	reg.x.auth.domain	The domain of the authorization server that is used to check the user names and passwords. Null (default)string	No
reg-advanced.cfg	reg.x.auth.optimizedInFailover	The destination of the first new SIP request when failover occurs. 0 (default) - The SIP request is sent to the server with the highest priority in the server list. 1 - The SIP request is sent to the server which sent the proxy authentication request.	No
reg-basic.cfg	reg.x.auth.password	The password to be used for authentication challenges for this registration. Null (default) string - It overrides the password entered into the Authentication submenu on the Settings menu of the phone.	No
reg-basic.cfg	reg.x.auth.userId	User ID to be used for authentication challenges for this registration. Null (default) string - If the User ID is non-Null, it overrides the user parameter entered into the Authentication submenu on the Settings menu of the phone.	No
reg-advanced.cfg	reg.x.auth.useLoginCredentials	0 - (default) The Login credentials are not used for authentication to the server on registration x. 1 - The login credentials are used for authentication to the server.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.bargeInEnabled	0 (default) - barge-in is disabled for line x. 1 - barge-in is enabled (remote users of shared call appearances can interrupt or barge in to active calls).	No
	reg.x.bridgeInEnabled	0 (default) - Bridge In feature is disabled. 1 - Bridge In feature is enabled.	No
features.cfg	reg.x.broadsoft.userId	Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. Null (default) string	No
features.cfg	reg.x.broadsoft.useXspCredentials	If this parameter is disabled, the phones use standard SIP credentials to authenticate. 1 (default) - Use this value, if phone lines are registered with a server running BroadWorks R19 or earlier. 0 - Set to 0, if phone lines are registered with a server running BroadWorks R19 SP1 or later.	No
features.cfg	reg.x.broadsoft.xsp.password	Enter the password associated with the BroadSoft user account for the line. Required only when reg.x.broadsoft.useXspCredentials=1 . Null (default) string	No
reg-advanced.cfg	reg.x.displayName	The display name used in SIP signaling and/or the H.323 alias used as the default caller ID. Null (default) UTF-8 encoded string	No
features.cfg	reg.x.enablePvtHoldSoftKey	This parameter applies only to shared lines. 0 (default) - To disable user on a shared line to hold calls privately. 1 - To enable users on a shared line to hold calls privately.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.enhancedCallPark.enabled	0 (default) - To disable the BroadWorks Enhanced Call Park feature. 1 - To enable the BroadWorks Enhanced Call Park feature.	No
	reg.x.filterReflectedBlaDialogs	1 (default) - bridged line appearance NOTIFY messages are ignored. 0 - bridged line appearance NOTIFY messages is not ignored	No
reg-advanced.cfg	reg.x.fwd.busy.contact	The forward-to contact for calls forwarded due to busy status. Null (default) - The contact specified by divert.x.contact is used. string - The contact specified by divert.x.contact is not used	No
reg-advanced.cfg	reg.x.fwd.busy.status	0 (default) - Incoming calls that receive a busy signal is not forwarded 1 - Busy calls are forwarded to the contact specified by reg.x.fwd.busy.contact .	No
reg-advanced.cfg	reg.x.fwd.noanswer.contact	Null (default) - The forward-to contact specified by divert.x.contact is used. string - The forward to contact used for calls forwarded due to no answer.	No
reg-advanced.cfg	reg.x.fwd.noanswer.ringCount	The number of seconds the phone should ring for before the call is forwarded because of no answer. The maximum value accepted by some call servers is 20. 0 - (default) 1 to 65535	No
reg-advanced.cfg	reg.x.fwd.noanswer.status	0 (default) - The calls are not forwarded if there is no answer. 1 - The calls are forwarded to the contact specified by reg.x.noanswer.contact after ringing for the length of time specified by reg.x.fwd.noanswer.ringCount .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`debug.cfg`	`reg.x.gruu`	Specify if the phone sends sip.instance in the REGISTER request. 0 (default) 1	No
`reg-basic.cfg`	`reg.x.label`	The text label that displays next to the line key for registration x. The maximum number of characters for this parameter value is 256; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters you use. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The rules for parameter up.cfgLabelElide determine how the label is truncated. Null (default) - the label is determined as follows: • If `reg. 1.useteluriAsLineLabel=1` , then the tel URI/phone number/ address displays as the label. • If *`reg. 1.useteluriAsLineLabel=0`*, then the value for `reg.x.displayName` , if available, displays as the label. If `reg.x.displayName` is unavailable, the user part of `reg.x.address` is used. UTF-8 encoded string	No
`reg-basic.cfg`	`reg.x.lineAddress`	The line extension for a shared line. This parameter applies to private lines and BroadSoft call park and retrieve. If there is no extension provided for this parameter, the call park notification is ignored for the shared line. Null (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.lineKeys	Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model. 1 (default) 1 to max	No
lync.cfg	reg.x.lisdisclaimer	This parameter sets the value of the location policy disclaimer. For example, the disclaimer may be “Warning: If you do not provide a location, emergency services may be delayed in reaching your location should you need to call for help.” Null (default) string, 0 to 256 characters	No
reg-advanced.cfg	reg.x.musicOnHold.uri	A URI that provides the media stream to play for the remote party on hold. Null (default) - This parameter does not overrides voIpProt.SIP.musicOnHold.uri . a SIP URI - This parameter overrides voIpProt.SIP.musicOnHold.uri .	No
reg-advanced.cfg	reg.x.offerFullCodecListUponResume	1 (default) - The phone sends full audio and video capabilities after resuming a held call irrespective of the audio and video capabilities negotiated at the initial call answer. 0 - The phone does not send full audio and video capabilities after resuming a held call.	No
reg-basic.cfg	reg.x.outboundProxy.address	The IP address or hostname of the SIP server to which the phone sends all requests. Null (default) IP address or hostname	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	reg.x.outboundProxy.failOver.failBack.mode	The mode for failover failback (overrides reg.x.server.y.failOver.failBack.mode). duration - (default) The phone tries the primary server again after the time specified by reg.x.outboundProxy.failOver.failBack.timeout expires. newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.failBack.timeout	3600 (default) -The time to wait (in seconds) before failback occurs (overrides reg.x.server.y.failOver.failBack.timeout). 0, 60 to 65535 - The phone does not fail back until a failover event occurs with the current server.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.failRegistrationOn	1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration. 0 - The reRegisterOn parameter is enabled, existing registrations remain active.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.onlySignalWithRegistered	1 (default) - The reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. 0 - The reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.outboundProxy.failOver.reRegisterOn`	This parameters overrides `reg.x.server.y.failOver.reRegisterOn` . 0 (default) - The phone won't attempt to register with the secondary server. 1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server.	No
`reg-advanced.cfg`	`reg.x.outboundProxy.port`	The port of the SIP server to which the phone sends all requests. 0 - (default) 1 to 65535	No
`reg-advanced.cfg`	`reg.x.outboundProxy.transport`	The transport method the phone uses to communicate with the SIP server. DNSnaptr (default) DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly	No
`sip-interop.cfg`	`reg.x.protocol.SIP`	You can use this parameter for the VVX 500/501, 600/601, and 1500. 1 (default) - SIP signaling is enabled for this registration. 0 - SIP signaling is not enabled for this registration.	No
`sip-interop.cfg`	`reg.x.proxyRequire`	Null (default) - No Proxy-Require is sent. string - Needs to be entered in the Proxy-Require header.	No
`reg-advanced.cfg`	`reg.x.ringType`	The ringer to be used for calls received by this registration. ringer2 (default) - Is the first non-silent ringer. ringer1 to ringer24 - To play ringer on a single registered line.	No
`reg-advanced.cfg`	`reg.x.serverFeatureControl.callRecording`	1 (default) - BroadSoft BroadWorks v20 call recording feature for individual phone lines is enabled. 0 - BroadSoft BroadWorks v20 call recording feature for individual phone lines is disabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.serverFeatureControl.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.cf . 0 (default) - The server-based call forwarding is disabled. 1 - server based call forwarding is enabled.	Yes
reg-advanced.cfg	reg.x.serverFeatureControl.dnd	This parameter overrides voIpProt.SIP.serverFeatureControl.dnd. 0 (default) - server-based do-not-disturb (DND) is disabled. 1 - server-based DND is enabled and the call server has control of DND.	Yes
sip-interop.cfg	reg.x.serverFeatureControl.localProcessing.cf	This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.cf . 0 (default) - If reg.x.serverFeatureControl.cf is set to 1 the phone does not perform local Call Forward behavior. 1 - The phone performs local Call Forward behavior on all calls received.	No
sip-interop.cfg	reg.x.serverFeatureControl.localProcessing.dnd	This parameter overrides voIpProt.SIP.serverFeatureControl.localProcessing.dnd . 0 (default) - If reg.x.serverFeatureControl.dnd is set to 1, the phone does not perform local DND call behavior. 1 - The phone performs local DND call behavior on all calls received.	No
reg-advanced.cfg	reg.x.serverFeatureControl.securityClassification	0 (default) - The visual security classification feature for a specific phone line is disabled. 1 - The visual security classification feature for a specific phone line is enabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.serverFeatureControl.signalingMethod	Controls the method used to perform call forwarding requests to the server. serviceMsForwardContact (default) string	No
sip-interop.cfg	reg.x.srtp.enable	1 (default) - The registration accepts SRTP offers. 0 - The registration always declines SRTP offers.	Yes
sip-interop.cfg	reg.x.srtp.offer	This parameter applies to the registration initiating (offering) a phone call. 0 (default) - No secure media stream is included in SDP of a SIP INVITE. 1 - The registration includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE.	Yes
sip-interop.cfg	reg.x.srtp.require	0 (default) - Secure media streams are not required. 1 - The registration is only allowed to use secure media streams.	Yes
sip-interop.cfg	reg.x.srtp.simplifiedBestEffort	This parameter overrides sec.srtp.simplifiedBestEffort . 1 (default) - Negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported. 0 - No SRTP is supported.	No
sip-interop.cfg	reg.x.strictLineSeize	0 (default) - Dial prompt is provided immediately without waiting for a successful OK from the call server. 1 - The phone is forced to wait for 200 OK on registration x when receiving a TRYING notify. This parameter overrides voIpProt.SIP.strictLineSeize for registration x.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	reg.x.tcpFastFailover	0 (default) - A full 32 second RFC compliant timeout is used. 1 - failover occurs based on the values of reg.x.server.y.retryMaxCount and voIpProt.server.x.retryTimeOut .	No
reg-advanced.cfg	reg.x.thirdPartyName	Null (default) - In all other cases. string address -This field must match the reg.x.address value of the registration which makes up the part of a bridged line appearance (BLA).	No
reg-advanced.cfg	reg.x.useCompleteUriForRetrieve	This parameters overrides voipPort.SIP.useCompleteUriForRetrieve . 1 (default) - The target URI in BLF signaling uses the complete address as provided in the XML dialog document. 0 - Only the user portion of the XML dialog document is used and the current registrar's domain is appended to create the full target URI.	No
site.cfg	reg.x.server.H323.y.address	Address of the H.323 gatekeeper. Null (default) IP address or hostname	No
site.cfg	reg.x.server.H323.y.port	Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used. 0 (default) 0 to 65535	No
site.cfg	reg.x.server.H323.y.expires	Desired registration period. 3600 positive integer	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`reg.x.server.y.address`	If this parameter is set, it takes precedence even if the DHCP server is available. Null (default) - SIP server does not accepts registrations. IP address or hostname - SIP server that accepts registrations. If not Null, all of the parameters in this table override the parameters specified in `voIpProt.server.*`	No
`reg-advanced`	`reg.x.server.y.expires`	The phone's requested registration period in seconds. The period negotiated with the server may be different. The phone attempts to re-register at the beginning of the overlap period. 3600 - (default) positive integer, minimum 10	No
`reg-advanced`	`reg.x.server.y.expires.lineSeize`	Requested line-seize subscription period. 30 - (default) 0 to 65535	No
`reg-advanced`	`reg.x.server.y.expires.overlap`	The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone tries to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value. 60 (default) 5 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.failOver.failBack.mode	duration (default) - The phone tries the primary server again after the time specified by reg.x.server.y.failOver.failBack.timeout . newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to. registration - The phone tries the primary server again when the registration renewal signaling begins. This parameter overrides voIpProt.server.x.failOver.failBack.mode)	No
site.cfg	reg.x.server.y.failOver.failBack.timeout	3600 (default) - The time to wait (in seconds) before failback occurs. 0 - The phone does not fail back until a failover event occurs with the current server. 60 to 65535 - If set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again.	No
site.cfg	reg.x.server.y.failOver.failRegistrationOn	1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over. 0 - The reRegisterOn parameter is disabled, existing registrations remain active.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.failOver.onlySignalWithRegistered	1 (default) - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. 0 - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).	No
site.cfg	reg.x.server.y.failOver.reRegisterOn	0 (default) - The phone does not attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information. 1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server. This parameter overrides voIpProt.server.x.failOver.reRegisterOn .	No
site.cfg	reg.x.server.y.port	Null (default) - The port of the SIP server does not specifies registrations. 0 - The port used depends on reg.x.server.y.transport . 1 to 65535 - The port of the SIP server that specifies registrations.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.register	1 (default) - Calls can not be routed to an outbound proxy without registration. 0 - Calls can be routed to an outbound proxy without registration. See voIpProt.server.x.register for more information, see *SIP Server Fallback Enhancements on Polycom Phones - Technical Bulletin 5844* on Polycom Engineering Advisories and Technical Notifications.	No
sip-interop.cfg	reg.x.server.y.registerRetry.baseTimeOut	For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server.Used in conjunction with reg.x.server.y.registerRetry.maxTimeOut to determine how long to wait. 60 (default) 10 - 120 seconds	No
sip-interop.cfg	reg.x.server.y.registerRetry.maxTimeout	For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Use in conjunction with r eg.x.server.y.registerRetry.baseTimeOut to determine how long to wait. The algorithm is defined in RFC 5626. 180 - (default) 60 - 1800 seconds	No
reg-advanced.cfg	reg.x.server.y.retryMaxCount	The number of retries attempted before moving to the next available server. 3 - (default) 0 to 20 - 3 is used when the value is set to 0.	No
reg-advanced.cfg	reg.x.server.y.retryTimeOut	0 (default) - Use standard RFC 3261 signaling retry behavior. 0 to 65535 - The amount of time (in milliseconds) to wait between retries.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.server.y.subscribe.expires	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 seconds - (default) 10 - 2147483647 (seconds) You can use this parameter in conjunction with reg.x.server.y.subscribe.expires.overlap .	No
reg-advanced.cfg	reg.x.server.y.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 seconds (default) 5 - 65535 seconds	No
site.cfg	reg.x.server.y.transport	The transport method the phone uses to communicate with the SIP server. DNSnaptr (default) - If reg.x.server.y.address is a hostname and reg.x.server.y.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If reg.x.server.y.address is an IP address, or a port is given, then UDP is used. TCPpreferred - TCP is the preferred transport; UDP is used if TCP fails. UDPOnly - Only UDP is used. TLS - If TLS fails, transport fails. Leave port field empty (defaults to 5061) or set to 5061 . TCPOnly - Only TCP is used.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.useOutboundProxy	1 (default) - Enables to use the outbound proxy specified in reg.x.outboundProxy.address for server x. 0 - Disable to use the outbound proxy specified in reg.x.outboundProxy.address for server x.	No
site.cfg	divert.x.sharedDisabled	1 (default) - Disables call diversion features on shared lines. 0 - Enables call diversion features on shared lines.	Yes

A shared line is an address of record managed by a call server. The server allows multiple endpoints to register locations against the address of record.

Polycom devices support Shared Call Appearance (SCA) using the SUBSCRIBE-NOTIFY method specified in RFC 6665. The events used are:

- call-info for call appearance state notification
- line-seize for the phone to ask to seize the line

Private Hold on Shared Lines

Enable the private hold feature to enable users to hold calls without notifying other phones registered with the shared line.

When you enable the feature, users can hold a call, transfer a call, or initiate a conference call and the shared line displays as busy to others sharing the line.

Private Hold on Shared Lines Parameters

You can configure private hold only using configuration files; you cannot configure the feature on the Web Configuration Utility or from the local phone interface.

Use the parameters in the following table to configure this feature.

Private Hold Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.shared.exposeAutoHolds`	0 (default) - No re-INVITE is sent to the server when setting up a conference on a shared line. 1 - A re-INVITE is sent to the server when setting up a conference on a shared line.	Yes
`features.cfg`	`reg.x.enablePvtHoldSoftKey`	This parameter applies only to shared lines. 0 (default) - To disable user on a shared line to hold calls privately. 1 - To enable users on a shared line to hold calls privately.	No

Intercom Calls

The Intercom feature enables users to place an intercom call that is answered automatically on the dialed contact's phone.

This is a server-independent feature provided the server does not alter the Alert-Info header sent in the INVITE.

Creating a Custom Intercom Soft Key

By default, an Intercom soft key displays on the phone, but you have the option to provide users the ability to initiate intercom calls directly to a specified contact using enhanced feature keys (EFKs).

You do not need to disable the default Intercom soft key to create a custom soft key.

For example, you can create an intercom action string for a custom soft key in one of the following ways:

- $FIntercom$

  This is an F type macro that behaves as a custom Intercom soft key. Pressing the soft key opens the Intercom dial prompt users can use to place an Intercom call by entering the destination's digits and using a speed dial or BLF button.

- <number>$Tintercom$

  This is a T type macro that enables you to specify a Direct intercom button that always calls the number you specify in <number>. No other input is necessary.

Intercom Calls Parameters

Use the parameters in the table to configure the behavior of the calling and answering phone.

Intercom Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.intercom.enable`	0 (default) - Disable the Intercom feature. 1 - Enable the Intercom feature.	No
`features.cfg`	`homeScreen.intercom.enable`	1 (default) - Enable the Intercom icon on the phone Home screen. 0 - Disable the Intercom icon on the phone Home screen.	No
`features.cfg`	`softkey.feature.intercom`	1 (default) - Enables the Intercom soft key. 0 - Disables the Intercom soft key.	No
`sip-interop.cfg`	`voIpProt.SIP.intercom.alertInfo`	The string you want to use in the Alert-Info header. You can use the following characters: '@', '-' ,'_' , '.' . If you use any other characters, NULL, or empty spaces, the call is sent as normal without the Alert-Info header. Intercom (default) Alpha - Numeric string	No
`sip-interop.cfg`	`voIpProt.SIP.alertInfo.x.value`	Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE. NULL (default)	No
`sip-interop.cfg`	`voIpProt.SIP.alertInfo.x.class`	Alert-Info fields from INVITE requests are compared as many of these parameters as are specified (x=1, 2, ..., N) and if a match is found, the behavior described in the corresponding ring class is applied. default (default) See the list of ring classes in Ringtone Parameters.	No

Push-to-Talk

The push-to-talk (PTT) is a collaborative tool that enables users to exchange broadcasts to users subscribed to any of the 25 PTT channels, much like a walkie-talkie.

Users can transmit pages and PTT broadcasts using their handset, headset, or speakerphone. PTT broadcasts can be received on the speakerphone, handset, and headset.

PTT mode is intended primarily for Wi-Fi phones. In PTT mode, the phone behaves like a walkie-talkie. Users can broadcast audio to a PTT channel and recipients subscribed to that channel can respond to messages.

You can specify the same IP multicast address in the parameter `ptt.address` for both PTT and group paging. Use the parameters in the following table to configure this feature.

Note: The push-to-talk and group paging features use an IP multicast address. If you want to change the default IP multicast address, ensure that the new address does not already have an official purpose as specified in the IPv4 Multicast Address Space Registry.

Push-to-Talk Parameters

Administrators must enable group paging and PTT before users can subscribe to a PTT channel.

PTT works in conjunction with group paging, and you can enable PTT or group paging, or enable both to operate simultaneously.

Push-To-Talk Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`ptt.address`	The multicast IP address to send page audio to and receive page audio from. 224.0.1.116 (default) multicast IP address.	
`site.cfg`	`ptt.allowOffHookPages`	0 (default) - PTT messages do not play out on the phone during an active call and the user must accept incoming PTT messages to play out. 1 - PTT messages play out even when there is an active call on the phone.	
`site.cfg`	`ptt.callWaiting.enable`	0 (default) - Incoming PTT sessions do not produce standard call waiting. 1 - Incoming PTT sessions produce standard call waiting behavior on the active audio channel.	
`features.cfg`	`ptt.channel.x.allowReceive`	1 (default) - The channel x receive incoming PTT messages. 0 - The channel x does not receive incoming PTT messages.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	ptt.channel.x.allowTransmit	1 (default) - Outgoing PTT messages are allowed on channel x. 0 - Outgoing PTT messages are not allowed on channel x.	
features.cfg	ptt.channel.x.available	1 (default) - Channel x is available. 0 - Channel x is not available.	
features.cfg	ptt.channel.x.label	Specify a label for channel x.Null (default) string	
features.cfg	ptt.channel.x.subscribed	0 (default) - The PPT is not subscribed for channel x. 1 - 25 - The PTT is subscribed for channel x.	
site.cfg	ptt.codec	Specify codec to use for PTT. G.722 (default)G.711Mu, G.726QI, G.722	
site.cfg	ptt.compatibilityMode	0 (default) - The PTT codec used is controlled by the ptt.codec and ptt.pageMode.codec parameters.1 - The codec used for PTT will be G726QI and payload size used will be 30.	
site.cfg	ptt.defaultChannel	Specify the default channel number used for PTT transmissions.1 (default)1 - 25	
site.cfg	ptt.emergencyChannel	Specify the channel to use for emergency PTT transmissions. 25 (default)1 - 25	
site.cfg	ptt.emergencyChannel.volume	The volume of emergency pages relative to the maximum speakerphone volume of the phone. Positive values are louder than the maximum and negative values are quieter. The gain to use for emergency page/PTT is the maximum termination gain plus this parameter. Note: To enter a negative number, press the * key first. -10 (default)-57 - 0	
site.cfg	ptt.port	Specify the port values to send and receive audio. 5001 (default)0 - 65535	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	ptt.displayName	This display name is shown in the caller ID field of outgoing group pages. If Null, the value from reg.1.displayName is used. NULL (default) up to 64 octet UTF-8 string	
site.cfg	ptt.payloadSize	Specify the payload size for PTT transmissions. 20 (default) 10, 20, 30, 40, 50, 60, 70, 80	
site.cfg	ptt.priorityChannel	Specify the channel number to use for priority PTT transmissions. 24 (default) 1 - 25	
site.cfg	ptt.pttMode.enable	0 (default) - PTT is disabled1 - PTT is enabled.	
site.cfg	ptt.volume	Controls the volume level for pages without changing the volume level for incoming calls. -20 (default) -57 to 0	
techsupport.cfg	voice.handsetHeadset.rxdg.offset	This parameter allows a digital Rx boost for the handset and headset. 0 (default) 9 to -12 - Specify the number of decibels to Offset the RxDg range of the handset and headset.	
techsupport.cfg	voice.handsfreePtt.rxdg.offset	This parameter allows a digital Rx boost for Push-to-Talk. 0 (default) 9 to -12 - Specify the number of decibels to offsets the RxDg range of the handsfree and handsfree Push-to-Talk (PTT).	

Group Paging

The group paging feature is available on VVX phones and Polycom Trio solution.

Group Paging enables users to make pages —one-way audio announcements—to users subscribed to a page group. There are 25 groups/channels users can subscribe to. If you are using Group Paging with

Polycom Trio solution, you can only receive incoming pages. You cannot use Polycom Trio solution to send outgoing pages.

Group paging users can send announcements to recipients subscribed to any of the 25 paging groups. Any announcements sent to the paging group play through the phone's speakerphone.

Administrators must enable paging before users can subscribe to a page group. You can specify the same IP multicast address in the parameter `ptt.address` for both PTT and paging mode.

Note: The push-to-talk and group paging features use an IP multicast address. If you want to change the default IP multicast address, ensure that the new address does not already have an official purpose as specified in the IPv4 Multicast Address Space Registry.

Group Paging Parameters

Administrators must enable paging and PTT before users can subscribe to a page group.

Use the parameters in the following table to configure this feature.

Group Paging Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`ptt.address`	The multicast IP address to send page audio to and receive page audio from. 224.0.1.116 (default) multicast IP address.	
`site.cfg`	`ptt.pageMode.allowOffHookPages`	0 (default) - Group pages do not play out on the phone during an active call except for Priority and Emergency pages. 1 - Group pages play out on the handset during an active call.	
`site.cfg`	`ptt.pageMode.defaultGroup`	The paging group used to transmit an outgoing page if the user does not explicitly specify a group. 1 (default) 1 to 25	
`site.cfg`	`ptt.pageMode.transmit.timeout.continuation`	The time (in seconds) to add to the initial timeout (`ptt.pageMode.transmit.timeout.initial`) for terminating page announcements. If this value is non-zero, Extend displays on the phone. Pressing Extend continues the initial timeout for the time specified by this parameter. If 0, announcements cannot be extended. 60 (default) 0 to 65535	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	ptt.pageMode.transmit.timeout.initial	The number of seconds to wait before automatically terminating an outgoing page announcement 0 (default) -The page announcements do not automatically terminate. 0 to 65535 - The page announcements automatically terminate.	
site.cfg	ptt.pageMode.priorityGroup	The paging group to use for priority pages. 24 (default) 1 to 25	
site.cfg	ptt.pageMode.payloadSize	The page mode audio payload size. 20 (default) 10, 20, ..., 80 milliseconds	
site.cfg	ptt.pageMode.emergencyGroup	The paging group used for emergency pages. 25 (default) 1 to 25	
site.cfg	ptt.pageMode.codec	The audio codec to use for outgoing group pages. Incoming pages are decoded according to the codec specified in the incoming message. G.722 (default) G.711Mu, G.726QI, or G.722	
site.cfg	ptt.pageMode.displayName	This display name is shown in the caller ID field of outgoing group pages. If Null, the value from reg.1.displayName is used. NULL (default) up to 64 octet UTF-8 string	
site.cfg	ptt.pageMode.enable	0 (default) - The group paging is disabled.1 - The group paging is enabled.	
features.cfg	ptt.pageMode.group.x.available	Make the group available to the user. 1 (default) - Group available to the user is enabled. 0 - Group available to the user is disabled.	
features.cfg	ptt.pageMode.group.x.allowReceive	1 (default) - The phone can receive pages on the specified group. 0 -The phone cannot receive pages on the specified group.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	ptt.pageMode.group.x.allowTransmit	Allows outgoing announcements to the group 1 (default) 0	
features.cfg	ptt.pageMode.group.x.label	The label to identify the group ch24: Priority,ch25: Emergency, others:Null ch1, 24, 25: 1, others: 0 (default) string	
features.cfg	ptt.pageMode.group.x.subscribed	Subscribe the phone to the group. A page mode group x, where x= 1 to 25. The `label` is the name used to identify the group during pages. If `available` is disabled (0), the user cannot access the group or subscribe and the other page mode group parameters is ignored. If enabled, the user can access the group and choose to subscribe. If `allowTransmit` is disabled (0), the user cannot send outgoing pages to the group. If enabled, the user may send outgoing pages. 1 (default) - If enabled, the phone subscribes to the group. 0 - If disabled, the phone does not subscribe to the group.	
techsupport.cfg	voice.ringerPage.rxdg.offset	Use this parameter for handsfree paging Rx in high noise environments. 0 (default) 9 to -12 - Raise or lower the volume of the ringer and handsfree page by the specified number of decibels.	

SIP-B Automatic Call Distribution

SIP-B Automatic Call Distribution enables you to use VVX business media phones and VVX business IP phones in a call center agent/supervisor role on a supported call server.

This feature supports ACD agent availability, which depends on support from a SIP server.

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

SIP-B Automatic Call Distribution Parameters

Use the parameters in the following table to configure this feature.

SIP-B Automatic Call Distribution

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.acdLoginLogout.enabled`	0 (default) - Disables the ACD login/ logout feature. 1 - Enables the ACD login/logout feature.	Yes
`reg-advanced.cfg reg-advanced.cfg`	`reg.x.acd-login-logout reg.x.acd-agent-available`	0 (default) - The ACD feature is disabled for registration. 1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.	No
`sip-interop.cfg`	`voIpProt.SIP.acd.signalingMethod`	0 (default) - The ‘SIP-B' signaling is supported. (This is the older ACD functionality.) 1 - The feature synchronization signaling is supported. (This is the new ACD functionality.)	Yes
`features.cfg`	`acd.simplifiedAgentStateControl`	0 (default) - Displays menu items. 1 - Hides ASignIN and associated soft keys. Also hides menu items under Menu > Settings > Feature > ACD.	No

Customizing Devices

Topics:

This section provides information on customizing Polycom phones.

Microbrowser and Web Browser

The web browser and a microbrowser, also known as the idle browser, include a Server Name Indication add-on allowing multiple secure websites to use the same IP address to support the Polycom phone browser. This allows a server to present multiple certificates on the same IP address and TCP port.

The following phones support the web browser and idle browser:

- VVX 250, 350, and 450 business IP phones
- VVX 3xx, 4xx, 5xx, 6xx, and 1500 business media phones

Note that the exact functions and performance of the microbrowser and web browser vary by phone model.

For more information on creating applications for the phones, see the *Polycom Web Application Developer's Guide* at Polycom UC Software Support Center.

Note: The browser restarts in the following situations:

- The browser uses over 30MB of memory.
- The amount of free memory on the phone is below 6MB.
- The real time is between 1am to 5am.

After the browser restarts, the last displayed web page is restored.

Microbrowser and Web Browser Parameters

You can configure the microbrowser and web browser to display a non-interactive web page on the phone's idle screen, and you can specify an interactive home web page that users can launch in a web browser.

The next table lists parameters that configure the home page, proxy, and size limits used by the microbrowser and browser when selected to provide services.

Use the Microbrowser and the Web Browser

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	apps.push	Specify the push server settings, including message type, port, tunnel, and a user name and password.	
applications.cfg	apps.push.alertSound	0 (default) - There is no sound when an alert is pushed. 1 - There is sound when an alert is pushed.	No
applications.cfg	apps.push.messageType	Choose a priority level for push messages from the application server to the phone. 0 (None) - (default) - Discard push messages 1 (Normal) Allows only normal push messages 2 (Important) Allows only important push messages 3 (High) Allows only priority push messages 4 (Critical) Allows only critical push 5 (All) Allows all push messages	No
applications.cfg	apps.push.password	The password to access the push server URL. NULL (default) string	No
applications.cfg	apps.push.secureTunnelEnabled	1 (default) - The web server is connected through a secure tunnel. 0 - The web server is not connected through a secure tunnel.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	apps.push.secureTunnelPort	Specify the port the phone uses to communicate to the web server when the secure tunnel is used. 443 (default) 1 - 65535	No
applications.cfg	apps.push.secureTunnelRequired	1 (default) - Communications to the web server require a secure tunnel. 0 - Communications to the web server do not require a secure tunnel.	No
applications.cfg	apps.push.serverRootURL	The URL of the application server you enter here is combined with the phone address and sent to the phone's browser. For example, if the application server root URL is http://172.24.128.85:8080/sampleapps and the relative URL is / examples/sample.html , the URL sent to the microbrowser is http://172.24.128.85:8080/sampleapps/examples/sample.html . You can use HTTP or HTTPS. NULL (default) URL	No
applications.cfg	apps.push.username	The user name to access the push server URL. To enable the push functionality, you must set values for the parameters apps.push.username and apps.push.password (not null). NULL (default) string	No
applications.cfg	apps.statePolling	Specify phone state polling settings, such as response mode, the poll URL, and a user name and password.	
applications.cfg	apps.statePolling.password	Enter the password that the phone requires to authenticate phone state polling. NULL (default) string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	apps.statePolling.responseMode	1 (default) - Polled data you request is sent to a configured URL. 0 - Polled data is sent in the HTTP response.	No
applications.cfg	apps.statePolling.URL	The URL to which the phone sends call processing state/device/network information. The protocol used can be either HTTP or HTTPS. Note: To enable state polling, the parameters apps.statePolling.URL , apps.statePolling.username , and apps.statePolling.password must be set to non-null values. NULL (default) string	No
applications.cfg	apps.statePolling.username	Enter the user name that the phone requires to authenticate phone state polling. NULL (default) string	No
applications.cfg	apps.telNotification.appInitializationEvent	0 (default) - No telephony notification event is sent. 1 - An XML telephony notification event is sent to report that the phone has completed initialization of its primary UC Software application. This event typically means that the phone is available and ready to receive network requests even if the phone user interface is not yet available.	No
applications.cfg	apps.telNotification.callStateChangeEvent	0 (default) - Call state change notification is disabled. 1 - Call state notification is enabled.	No
applications.cfg	apps.telNotification.incomingEvent	0 (default) - Incoming call notification is disabled. 1 - Incoming call notification is enabled.	No
applications.cfg	apps.telNotification.lineRegistrationEvent	0 (default) - Line registration notification is disabled. 1 - Line registration notification is enabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	apps.telNotification.networkUpEvent	0 (default) - No telephony notification event is sent. 1 – An XML telephony notification event is sent to report that the phone has received link up state from its LAN port and that an IP address was assigned.	No
applications.cfg	apps.telNotification.offhookEvent	0 (default) - Disable off-hook notification. 1 - Enable off-hook notification.	No
applications.cfg	apps.telNotification.onhookEvent	0 (default) - Disable on-hook notification. 1 - Enable on-hook notification.	No
applications.cfg	apps.telNotification.outgoingEvent	0 (default) - Disable outgoing call notification. 1 - Enable outgoing call notification.	No
applications.cfg	apps.telNotification.taInitializationEvent	0 (default) – No telephony notification event is sent. 1 - An XML telephony notification event is sent to report that the phone has started its test automation server and is ready to receive API commands.	No
applications.cfg	apps.telNotification.uiInitializationEvent	0 (default) - No telephony notification event is sent. 1 - An XML telephony notification event is sent to report that the phone has completed start up of the phone user interface and is ready to receive physical key or touch inputs.	No
applications.cfg	apps.telNotification.URL	The URL to which the phone sends notifications of specified events. You can use HTTP or HTTPS. NULL (default) string	No
applications.cfg	apps.telNotification.userLogInOutEvent	0 (default) - Disable user login/logout notification. 1 - Enable user login/logout notification.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	apps.telNotification.x.URL	The URL to which the phone sends notifications of specified events, where x 1 to 9. You can use HTTP or HTTPS. NULL (default) string	No
applications.cfg	mb.idleDisplay.home	Displays the URL of the microbrowser home page when the microbrowser Home page screen is idle. Null (default) valid HTTP URL, String (maximum 255 characters) For example: http://www.example.com/xhtml/frontpage . The microbrowser idle display displaces the idle display indicator.	No
applications.cfg	mb.idleDisplay.refresh	0 (default) - The microbrowser's idle display does not refresh Integer > 5 - Displays the microbrowser's idle display refresh time period in seconds. If an HTTP Refresh header is detected, it is respected, even if this parameter is set to 0. The refresh parameter is respected only in the event that a refresh fails. Once a refresh is successful, the value in the HTTP refresh header, if available, is used.	No
applications.cfg	mb.idleRefresh.onFailure	Helps reduce the requests from the phone when the idle display server is unavailable and specifies a delay in seconds when the phone sends refresh requests to the idle browser. This delay applies only when the server returns HTTP 5xx errors. 60 seconds (default) 60 - 655350 seconds To control the refresh times when the server is functioning, use mb.idleDisplay.refresh .	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
applications.cfg	mb.main.autoBackKey	1 (default) - The phone automatically supplies a **Back** soft key in all main browser screens. 0- The phone does not provide a **Back** soft key.	Yes
applications.cfg	mb.main.home	Specifies the URL of the microbrowser's home page. For example: `http://www.example.com/xhtml/frontpage/home` . Null (default) valid HTTP URL, String (maximum 255 characters)	No
applications.cfg	mb.main.idleTimeout	Specifies the timeout in seconds for the interactive browser. If the interactive browser remains idle for a defined period of time, the phone returns to the idle browser. If set to 0, there is no timeout. 40 (default) 0 - 600	No
applications.cfg	mb.main.loadWebImages	1 (default) - Enables the loading of images in a browser. 0 - Disables the loading of images in a browser.	No
applications.cfg	mb.main.proxy	Specifies the address of the HTTP proxy to be used by the microbrowser. Null (port: 8080) (default) domain name or IP address in the format <address>:<port>	No
applications.cfg	mb.main.reloadPage	0 (default) - The microbrowser displays the content of the most recently viewed web page 1 - The microbrowser loads the URL configured in `mb.main.home` each time the browser is launched.	No
applications.cfg	mb.main.statusbar	0 (default) - The status bar does not gets displayed. 1 - The status bar and status messages are displayed.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`mb.main.toolbar.autoHide.enabled`	1 (default) - The toolbar is not displayed. 0 - The toolbar displays continuously.	No
`applications.cfg`	`mb.proxy`	Specify the Application browser home page, a proxy to use, and size limits.	
`features.cfg`	`mb.ssawc.call.mode`	passive (default) - Web content is displayed only when requested by the user. Passive mode is recommended when the microbrowser is used for other applications. When passive mode is enabled, an icon displays beside a call appearance indicating that web content is available, and the user can press Select to view the content. Active - Web content is retrieved spontaneously and displayed immediately.	No
`features.cfg`	`mb.ssawc.enabled`	0 (default) - Spontaneous display of web content is disabled. 1 - Spontaneous web content display is enabled.	No

Support for REST API

VVX phones support REST APIs that enable you to execute certain functions and retrieve information.

For more information on these APIs, see *REST API Reference Manual for Polycom VVX Business Media Phones* at Polycom Engineering Advisories and Technical Notifications.

The REST API feature is disabled by default. You can configure the REST API feature on your phone using parameters.

REST API Parameters

The following table includes parameters to configure the REST API feature.

REST API Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
`applications.cfg`	`apps.restapi.enabled`	Enable or disable the REST API feature. 0 (default) 1	No

Soft Keys

You can create custom soft keys on all VVX phones to enable users to access frequently used functions, create menu shortcuts to frequently used phone settings, or create a soft key in place of a hard key not available on the phone.

For example, if the phone does not have a Do Not Disturb hard key, you can create a Do Not Disturb soft key.

You can create custom soft keys as any of the following:

- An enhanced feature key sequence
- A speed dial contact directory entry
- An enhanced feature key macro
- A URL
- A chained list of actions

Related Links

Macro Definitions on page 479
Macro Definitions on page 479

Call State for Custom Soft Keys

You can configure soft keys to display certain functions depending on the phone's menu level or call state.

For example, you can make a Call Park soft key available when the phone is in an active call state.

You can configure custom soft keys to display for the following call states:

- Idle—There are no active calls.
- Active—This state starts when a call is connected. It stops when the call stops or changes to another state (like hold or dial tone).
- Alerting (or ringing or incoming proceeding)—The phone is ringing.
- Dial tone—You can hear a dial tone.
- Proceeding (or outgoing proceeding)—This state starts when the phone sends a request to the network. It stops when the call is connected.

- Setup—This state starts when the user starts keying in a phone number. This state ends when the Proceeding state starts.
- Hold—The call is put on hold locally.

Soft Key Parameters

You can create up to 10 custom soft keys.

If you configure more soft keys than what can fit on the phone's screen, a More soft key displays. Users can use the More soft key to display any additional soft keys available.

If you want the phone to display both default and custom soft keys, you can configure them in any order. However, the order in which soft keys display depends on the phone's menu level and call state. If you have configured custom soft keys to display with the default soft keys, the order of the soft keys may change.

Note: The Hold, Transfer, and Conference soft keys are grouped together to avoid usability issues. You may experience errors if you try to insert a soft key between these three grouped soft keys.

The following table shows you the parameters for configuring soft keys. Note that this feature is part of enhanced feature keys (EFK), and you must enable the EFK parameters to configure soft keys. See the Enhanced Feature Keys section for details about configuring soft keys and line keys.

Configure Soft Keys

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.enhancedFeatureKeys.enabled`	0 (default) - Disables the enhanced feature keys feature. 1 - Enables the enhanced feature keys feature.	No
`features.cfg`	`softkey.x.action`	Controls the action or function for the custom soft key x. Null (default) macro action string, 2048 characters This value uses the same macro action string syntax as an Enhanced Feature Key.	No
`features.cfg`	`softkey.x.enable`	0 (default) - The x soft key is disabled. 1 - The x soft key is enabled.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.x.insert	0 (default) - The phone places the soft key in the first available position. 0 to 10 - The phone places the soft key in the corresponding position and moves the following soft keys by one position to the right. For example, if the soft key is set to 3, the soft key is displayed in the third position from the left. If the soft key already exists in the third position, it is moved to fourth position and the following soft keys are moved to right by one space. If softkey.x.precede is configured, this value is ignored. If the insert location is greater than the number of soft keys, the key is positioned last after the other soft keys.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.x.label	The text displayed on the soft key label. If Null, the label is determined as follows: • If the soft key performs an Enhanced Feature Key macro action, the label of the macro defined using efk.efklist is used. • If the soft key calls a speed dial, the label of the speed dial contact is used. • If the soft key performs chained actions, the label of the first action is used. • If the soft key label is Null and none of the preceding criteria are matched, the label is blank. Null (default) String Note that the maximum number of characters for this parameter value is 16; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters used. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The phone truncates the beginning of numerical labels (for example, …4567) and truncates the end of alphabetical labels (for example, Abcd…).	No
features.cfg	softkey.x.precede	0 (default) - The phone locates the soft key in the first available position from left. 1 - The phone locates the soft key before the default soft key position.	No
features.cfg	softkey.x.use	Specify which call states the soft key displays in.	
features.cfg	softkey.x.use.active	0 (default) - Does not display the soft key x during an active call. 1 - Displays the soft key x during an active call.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.x.use.alerting	0 (default) - Does not display the soft key x in an alerting state during an active call. 1 - Displays the soft key x in an alerting state during an active call.	No
features.cfg	softkey.x.use.dialtone	0 (default) - Does not display the soft key in the dial tone state during an active call. 1 - Displays the soft key x in the dial tone state during an active call.	No
features.cfg	softkey.x.use.hold	0 (default) - Does not display the soft key x in the hold state during an active call. 1 - Displays the soft key x in the hold state during an active call.	No
features.cfg	softkey.x.use.idle	0 (default) - Does not display the soft key x in the idle state during an active call. 1 - Displays the soft key x in the idle state during an active call.	No
features.cfg	softkey.x.use.park	0 (default) - Does not display the soft key x in the parked state during an active call. 1 - Displays the soft key x in the parked state during an active call.	No
features.cfg	softkey.x.use.proceeding	0 (default) - Does not display the soft key x in the proceeding state during an active call. 1 - Displays the soft key x in the proceeding state during an active call.	No
features.cfg	softkey.x.use.setup	0 (default) - Does not display the soft key x in the setup state during an active call. 1 - Displays the soft key x in the setup state during an active call.	No
features.cfg	softkey.feature.intercom	1 (default) - Enables the Intercom soft key. 0 - Disables the Intercom soft key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.feature.doNotDisturb	1 (default) - Enables the DND soft key on the phone. 0 - Disables the DND soft key on the phone.	
features.cfg	softkey.feature.basicCallManagement.redundant	1 (default) - Displays the Hold, Transfer, and Conference soft keys. 0 - Does not display the Hold, Transfer, and Conference soft keys.	No
features.cfg	softkey.feature.buddies	1 (default) - Displays the Buddies soft key. 0 - Does not display the Buddies soft key.	No
features.cfg	softkey.feature.callers	0 (default) - Displays the Callers soft key for all platforms. 1 - Does not display the Callers soft key for all platforms.	No
features.cfg	softkey.feature.directories	1 (default) - Displays the Directories (Dir) soft key. 0 - Does not display the Directories (Dir) soft key.	Yes
features.cfg	softkey.feature.doNotDisturb	1 (default) - Enables the DND soft key. 0 - Disables the DND soft key.	No
features.cfg	softkey.feature.endcall	1 (default) - Displays the End Call soft key. 0 - Does not display the End Call soft key.	No
features.cfg	softkey.feature.forward	1 (default) - Displays the Forward soft key. 0 - Does not display the Forward soft key.	No
features.cfg	softkey.feature.join	1 (default) - Displays the Join soft key. 0 - Does not display the Join soft key.	No
features.cfg	softkey.feature.mystatus	1 (default) - Displays the MyStatus soft key (if pres.idleSoftKeys is set to 1). 0 - Does not display the MyStatus soft key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.feature.newcall	1 (default) - Displays the New Call soft key is displayed. 0 - Does not display the New Call soft key.	No
features.cfg	softkey.feature.redial	0 (default) - Displays the Redial soft key. 1 - Does not display the Redial soft key. The parameter feature.enhancedFeatureKeys.enabled must be set to 1 first to configure this feature, and the parameter efk.softkey.alignleft must be set to 1 to move enabled soft keys into the positions of disabled soft keys.	No
features.cfg	softkey.feature.split	1 (default) - Displays the Split soft key to split the conference call to individual calls. 0 - Does not display the Split soft key.	No

Soft Key Customization Parameters

You can use the soft key parameters to customize soft keys on the phone interface.

Note that feature.enhancedFeatureKeys.enabled must be enabled (set to 1) to use the Configurable Soft Key feature.

In the following table listing soft key configuration parameters, x=1 to a maximum number of 10 soft keys.

Soft Key Customization Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.feature.basicCallManagement.redundant	1 (default) - Displays the Hold, Transfer, and Conference soft keys. 0 - Does not display the Hold, Transfer, and Conference soft keys.	No
features.cfg	softkey.feature.buddies	1 (default) - Displays the Buddies soft key. 0 - Does not display the Buddies soft key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	softkey.feature.callers	0 (default) - Displays the Callers soft key for all platforms. 1 - Does not display the Callers soft key for all platforms.	No
features.cfg	softkey.feature.directories	1 (default) - Displays the Directories (Dir) soft key. 0 - Does not display the Directories (Dir) soft key.	Yes
features.cfg	softkey.feature.doNotDisturb	1 (default) - Enables the DND soft key. 0 - Disables the DND soft key.	No
features.cfg	softkey.feature.endcall	1 (default) - Displays the End Call soft key. 0 - Does not display the End Call soft key.	No
features.cfg	softkey.feature.forward	1 (default) - Displays the Forward soft key. 0 - Does not display the Forward soft key.	No
features.cfg	softkey.feature.join	1 (default) - Displays the Join soft key. 0 - Does not display the Join soft key.	No
features.cfg	softkey.feature.mystatus	1 (default) - Displays the MyStatus soft key (if pres.idleSoftKeys is set to 1). 0 - Does not display the MyStatus soft key.	No
features.cfg	softkey.feature.newcall	1 (default) - Displays the New Call soft key is displayed. 0 - Does not display the New Call soft key.	No
features.cfg	softkey.feature.redial	0 (default) - Displays the Redial soft key. 1 - Does not display the Redial soft key. The parameter feature.enhancedFeatureKeys.enabled must be set to 1 first to configure this feature, and the parameter efk.softkey.alignleft must be set to 1 to move enabled soft keys into the positions of disabled soft keys.	No
features.cfg	softkey.feature.split	1 (default) - Displays the Split soft key to split the conference call to individual calls. 0 - Does not display the Split soft key.	No

Disabling Default Soft Keys

You can disable the display of any of the following default soft key to make room for custom soft keys:

- New Call
- End Call
- Split
- Join
- Forward
- Directories
- MyStatus and buddies
- Hold, transfer, and conference

Should tasks be named “Example”?

Use the following example configuration to automatically transfer an active call to a BroadSoft voicemail.

In this example, *55 is the star code for BroadSoft voicemail, and 8545 is the extension of the voicemail line the call transfers to. The exact star code to transfer the active call to voicemail depends on your call server.

Enabling the parameter `softkey.1.use.active` causes the soft key to display when a call becomes active on the line. When you press the soft key—labeled VMail in this example—the call is placed on hold and automatically transferred to a BroadSoft voicemail.

Procedure

1. Update the configuration file as follows:
   - `softkey.1.label="VMail"`
   - `softkey.1.action="$FTransfer$$Cpause1$$FDialpadStar$$FDialpad5$ $FDialpad5$$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$$FSoftKey1$"`
   - `softkey.1.enable="1"`
   - `softkey.1.use.active="1"`
2. Reboot the phone.

   When an incoming call connects and becomes active, the VMail soft key displays.

Example: Send-to-Voicemail Prompt

Use the following example to enable users to enter a voicemail extension to transfer an active call to BroadSoft voicemail.

In this example, *55 is the star code used for BroadSoft voicemail. The exact star code to transfer the active call to voicemail depends on your call server.

Enabling the parameter `softkey.1.use.active` causes the soft key to display when a call becomes active on the line. When a user presses the soft key, the call is placed on hold and a field prompts the user to enter the extension of a voicemail line to transfer the call to. The `efk.prompt*` parameters control the numeric prompt field users enter the extension into.

Note that this example works only on line 1 of the phone.

Procedure

1. Update the configuration file as follows:
   - `softkey.1.label="VMail"`
   - `softkey.1.action="^*55$P1N10$$Tinvite$"`
   - `softkey.1.enable="1"`
   - `softkey.1.use.active="1"`
   - `efk.efkprompt.1.label="Voice Mail"`
   - `efk.efkprompt.1.status="1"`
   - `efk.efkprompt.1.type="numeric"`
2. Reboot the phone.

   When an incoming call connects and becomes active, the VMail soft key displays.
3. Press the **VMail** soft key.

   A field displays prompting you to enter an extension.
4. Enter the extension you want to transfer the call to.
5. Press the **Enter** soft key.

Example: Speed Dial Soft Key with a Pause

Use the following example to configure a soft key to automatically dial a number with a pause in the dialing sequence.

In this example, use `$CpauseX$` where `X` is the number of seconds to pause—7 in this example. Adding this pause function enables users to automatically dial into a conference ID that requires an entry code after the conference call is connected.

Procedure

1. Update the configuration file as follows:
   - `softkey.1.label="VMail"`
   - `softkey.1.action="$S1$$Tinvite$$Cwc$$Cpause7$$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$"`
   - `softkey.1.enable="1"`
   - `softkey.1.use.idle="1"`
   - `feature.enhancedFeatureKeys.enabled="1"`

The values for this example are explained as follows:

- `$S1$`— Speed dial line 1
- `$S1$$Tinvite$$` —The phone sends an invite to $S1$
- `$Cwc$` —The phone waits for the call to connect
- `$Cpause7$` —The phone waits for 7 seconds before dialing the remaining numbers
- `$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$` —The phone enters the entry code 8545.

Example: Directory-Linked Speed Dial Soft Key with a Pause

Use the following example to add a speed dial line key linked to a directory file with a pause in the dialing sequence.

Procedure

1. Update the configuration file as follows:
   - `feature.enhancedFeatureKeys.enabled="1"`
   - `efk.efklist.1.action.string="501$Tinvite$$Cwc$$Cpause7$1234#$Tdtmf$"` `efk.efklist.1.label="number"`
   - `efk.efklist.1.mname="number"`
   - `efk.efklist.1.status="1"`
2. In a contact directory file or speed dial file (000000000000-directory.xml or <MACaddress>-directory.xml), add the following:
   - `<fn>Call Number</fn>`
   - `<ct>!number</ct>`
   - `<sd>99</sd>`

The following values are included in the action string: `<ct>"501$Tinvite$$Cwc$$Cpause7$1234#$Tdtmf$"`:

- `501$Tinvite$` —Dial 501
- `$Cwc$` —Wait for the call to connect
- `$Cpause7$` —A seven second pause
- `1234#$Tdtmf$` —Send 1234 dual-tone multi-frequency

The following EFK commands are linked to the directory file:

- The parameter `efk.efklist.1.mname="number"` is linked to the speed dial contact `<ct>!number</ct>` of the directory file
- Use `<fn>Call Number</fn>` to define the name that displays on the key
- Use `<sd>99</sd>` to identify which directory entry to link to the key

Note: For more example configurations, see the two following documents at Polycom Engineering Advisories and Technical Notifications:

- *Using Enhanced Feature Keys and Configurable Soft Keys on Polycom Phones: Technical Bulletin 42250*
- *Using Enhanced Feature Keys (EFK) Macros to Change Soft Key Functions on Polycom Community: Feature Profile 42250*

Enhanced Feature Keys

Enhanced feature keys (EFK) enables you to customize the functions of a phone's line, soft, and hard keys to assign frequently used functions to keys or to create menu shortcuts to frequently used phone settings.

Enhanced feature key functionality is implemented using star code sequences like *89 and SIP messaging. Star code sequences that define EFK functions are written as macros that you apply to line and soft keys. The EFK macro language was designed to follow current configuration file standards and to be extensible (see Macro Definitions).

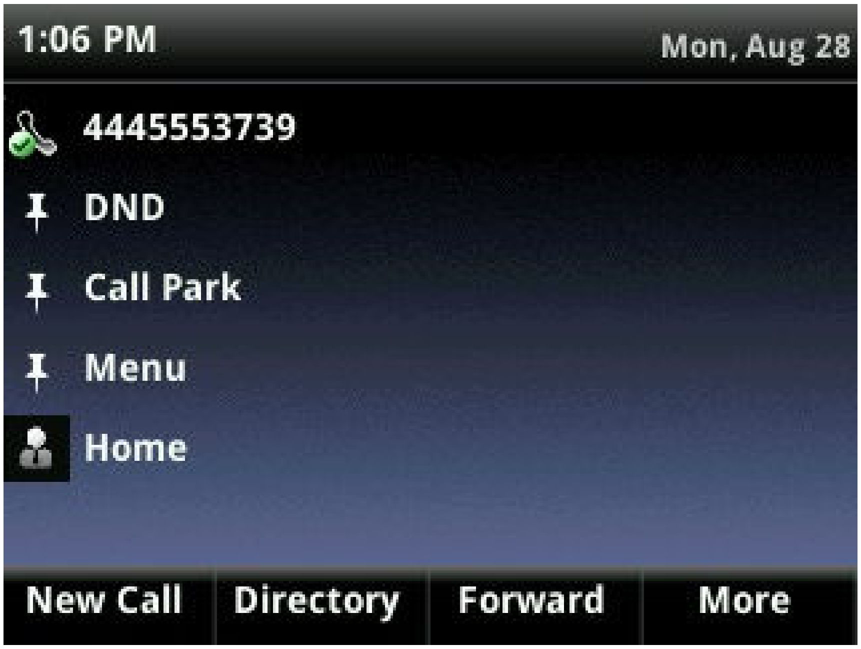
In addition, you can configure an EFK as a line key allowing the users to execute the macro action defined to that line key. When this feature is enabled, all the EFK macros that are configured using `efk.eklist` parameter and has `efk.efklist.x.status=1` will display as a line key. You can enable or disable this feature using configuration parameter or importing the configuration file using the Web Configuration Utility.

For example, configure the phone with the following configuration:

```
feature.enhancedFeatureKeys.enabled="1"

feature.EFKLineKey.enabled="1"

efk.efklist.1.mname="DND"

efk.efklist.1.status="1"

efk.efklist.1.action.string="$FDoNotDisturb$"
```

After you run and update configuration, the DND EFK will display as a line key. When you press the DND line key, Do Not Disturb functionality is executed.

In addition, you can use Flexible Line Keys feature for an EFK and assign to a line key that displays anywhere on the phone’s screen. For more information, see Flexible Line Key Assignments.



Related Links

Macro Definitions on page 479

Macro Definitions on page 479

Phone Keypad Parameters on page 483

Enhanced Feature Keys Parameters

The rules for configuring EFK for line keys, soft keys, and hard keys vary.

Before configuring EFK, refer to Macro Definitions to become familiar with the macro language.

Note that the configuration file changes and the enhanced feature key definitions can be included together in one configuration file. However, Polycom recommends creating a new configuration file to make configuration changes.

See the following table for the parameters you can configure and a brief explanation of how to use the contact directory to configure line keys.

Enhanced Feature Keys Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.callsPerLineKey	Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration. This per-registration parameter overrides call.callsPerLineKey . 24 (default) 1-24 VVX 101, 201 8 (default) 1 - 8	No
features.cfg	feature.enhancedFeatureKeys.enabled	0 (default) - Disables the enhanced feature keys feature. 1 - Enables the enhanced feature keys feature.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.EFKLineKey.enabled	0 (default) – Does not allow to configure EFK as a line key. 1 – Allows to configure EFK as a line key. Before you enable this parameter, make sure the parameter value for feature.enhancedFeatureKeys.enabled is set to 1.	No
features.cfg	efk.efklist.x.action.string	The action string contains a macro definition of the action that the feature key performs. Null (default) String (maximum of 64 characters) If EFK is enabled, this parameter must have a value (it cannot be Null). For a list of macro definitions and example macro strings, see Macro Definitions.	Yes
features.cfg	efk.efklist.x.label	The text string used as a label on any user text entry screens during EFK operation. Null (default) - The Null string is used. String (maximum of 64 characters) If the label does not fit on the screen, the text is shortened and '…' is appended.	Yes
features.cfg	efk.efklist.x.mname	The unique identifier used by the speed dial configuration to reference the enhanced feature key entry. Cannot start with a digit. Note that this parameter must have a value, it cannot be Null. expanded_macro (default) String (maximum of 64 characters)	Yes
features.cfg	efk.efklist.x.status	0 (default) - Disables the key x. Null - Disables the key x. 1 -Enables the key x.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	efk.efklist.x.type	The SIP method to be performed. invite (default) - Performs the required action using the SIP INVITE method. Null - default of INVITE is used. This parameter is included for backwards compatibility. Do not use if possible. If efk.x.action.string contains types, this parameter is ignored.	Yes
features.cfg	efk.efkprompt.x.label	The prompt text on the user prompt screen. Null (default) - No prompt displays. String If the label does not fit on the screen, the label is shortened and '…' is appended.	Yes
features.cfg	efk.efkprompt.x.status	This parameter must have a value, it cannot be Null. 0 (default) - Disables the key. 1 - Enabled the key. If a macro attempts to use a prompt that is disabled or invalid, the macro execution fails.	Yes
features.cfg	efk.efkprompt.x.type	The type of characters entered by the user. text (default) - The characters are interpreted as letters. numeric - The characters are interpreted as numbers. If Null, numeric is used. If this parameter has an invalid value, this prompt, and all parameters depending on this prompt, are invalid. Note: A mix of numeric and text is not supported.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	efk.efkprompt.x.userfeedback	The user input feedback method. visible (default) - The text is visible. masked - The text displays as asterisk characters (*), which can be used to mask password fields. If Null, visible is used. If this parameter has an invalid value, this prompt, and all parameters depending on this prompt, are invalid.	Yes
features.cfg	efk.version	The version of the EFK elements. This parameter is not required if there are no efk.efklist entries. 2 (default) - Supported version for SIP 3.1 and later. 1 - Supported version for or SIP 3.0.x or earlier. Null - Disables the EFK feature.	Yes
features.cfg	efk.softkey.alignleft	Use this parameter to left-align soft keys and remove blank soft keys from the order. 0 (default) 1 - Left-aligns soft keys and removes blank soft keys from the order Note: This parameter does not work with custom soft keys.	Yes

Some Guidelines for Configuring Enhanced Feature Keys

Use the following guidelines to help you to configure enhanced feature keys (EFKs) efficiently:

- Activation of EFK functions requires valid macro construction.
- All failures are logged in the phone's app logs at level 4 (Minor Error).
- If two macros have the same name, the first one is used and the subsequent one is ignored.
- A sequence of characters prefixed with “!” are parsed as a macro name. The exception is the speed dial reference, which starts with “!” and contains digits only.
- A sequence of characters prefixed with “^” is the action string.
- “!” and “^” macro prefixes cannot be mixed in the same macro line.
- The sequence of characters must be prefixed by either “!” or “^” to be processed as an enhanced feature key. All macro references and action strings added to the local directory contact field must be prefixed by either “!” or “^”.
- Action strings used in soft key definitions do not need to be prefixed by “^”. However, the “!” prefix must be used if macros or speed dials are referenced.

- A sequence of macro names in the same macro is supported (for example, “!m1!m2”).
- A sequence of speed dial references is supported (for example, “!1!2”).
- A sequence of macro names and speed dial references is supported (for example, “!m1!2!m2”).
- Macro names that appear in the local contact directory must follow the format “!<macro name>” , where <macro name> must match an <elklist> mname entry. The maximum macro length is 100 characters.
- A sequence of macros is supported, but cannot be mixed with other action types.
- Action strings that appear in the local contact directory must follow the format “^<action string>”. Action strings can reference other macros or speed dial indexes. Protection against recursive macro calls exists (the enhanced feature keys fails after you reach 50 macro substitutions).

Contact Directory Macros

Because line keys and their functions are linked to fields in the contact directory file, you need to match the contact field (ct) in the directory file to the macro name field (mname) in the configuration file that contains the EFK parameters.

When you enter macro names to the contact field (ct) in the directory file, add the ‘!' prefix to the macro name. The template directory configuration file is named 000000000000-directory~.xml. To use this file, remove the tilde (~) from the file name.

Related Links

Local Contact Directory on page 276

Special Characters

Macro names and macro labels cannot contain these special characters.

If they do, you may experience unpredictable behavior.

The following special characters are used to implement the enhanced feature key functionality:

- ! The characters following it are a macro name.
- ' or ASCII (0x27) This character delimits the commands within the macro.
- $ This character delimits the parts of the macro string. This character must exist in pairs, where the $ delimits the characters to be expanded.
- ^ This character indicates that the following characters represent the expanded macro (as in the action string).
- Macro names and macro labels cannot contain these special characters. If they do, you may experience unpredictable behavior.

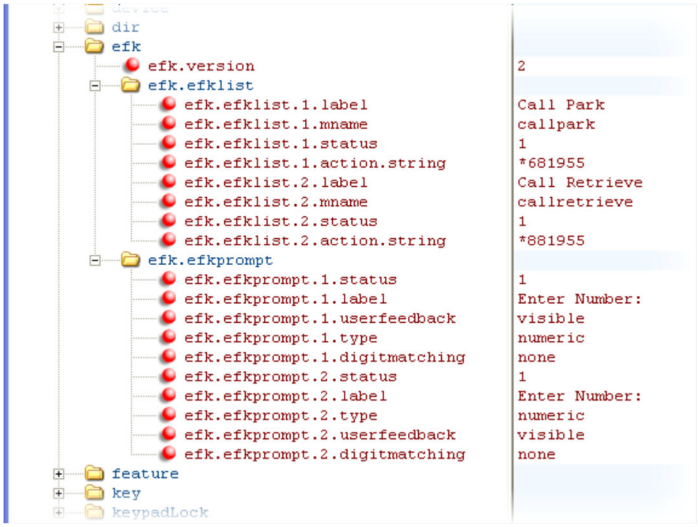
Enhanced Feature Key Example Configurations

The following configurations shown in the below illustration were set in the features.

cfg file:

- For the `efk.efklist.x.*` parameters, the following configurations were applied:
    - Line key 1 has been assigned a Call Park address (1955) and line key 2 a call retrieve function.
    - The parameter `acton.string` shows the macro definition for these two functions.
    - Status is enabled and a label has been specified to display next to the line key.
    - The entry in the `mname` parameter corresponds to the `contact (ct)` field in the contact directory.

- For the `efk.prompt.*` parameters, the following configurations were applied:
    - `Status` is enabled.
    - The label on the user prompt has been defined as Enter Number: and this prompt displays on the phone screen.
    - The `type` parameter has been set to `numeric` to allow only numbers.
    - `userfeedback` is specified as `visible` , which enables users to see the numbers entered into the prompt.



```
device
dir
efk
    efk.version                              2
    efk.efklist
        efk.efklist.1.label                  Call Park
        efk.efklist.1.mname                  callpark
        efk.efklist.1.status                 1
        efk.efklist.1.action.string          *681955
        efk.efklist.2.label                  Call Retrieve
        efk.efklist.2.mname                  callretrieve
        efk.efklist.2.status                 1
        efk.efklist.2.action.string          *881955
    efk.efkprompt
        efk.efkprompt.1.status               1
        efk.efkprompt.1.label                Enter Number:
        efk.efkprompt.1.userfeedback         visible
        efk.efkprompt.1.type                 numeric
        efk.efkprompt.1.digitmatching        none
        efk.efkprompt.2.status               1
        efk.efkprompt.2.label                Enter Number:
        efk.efkprompt.2.type                 numeric
        efk.efkprompt.2.userfeedback         visible
        efk.efkprompt.2.digitmatching        none
feature
key
keypadLock
```

Macro Definitions

The `efk.`

`efklist.x.action.string` can be defined by macro actions, prompt macro substiution or an expanded macro.

Related Links

Enhanced Feature Keys on page 473
Soft Keys on page 462
Enhanced Feature Keys on page 473
Soft Keys on page 462

Macro Actions

The action string is executed in the order it displays.

User input is collected before any action is taken. The action string can contain the fields shown in the following table.

Action String	Description
$L<label>$	This is the label for the entire operation. The value can be any string including the null string (in this case, no label displays). This label is used if no other operation label collection method worked (up to the point where this field is introduced). Make this the first entry in the action string to be sure this label is used; otherwise another label may be used and this one ignored.
digits	The digits to be sent. The appearance of this parameter depends on the action string.
$C<command>$	This is the command. It can appear anywhere in the action string. Supported commands (or shortcuts) include: hangup (`hu`) hold (`h`) waitconnect (`wc`) pause <number of seconds> (`p` *`<num sec>`*) where the maximum value is 10
$T<type>$	The embedded action type. Multiple actions can be defined. Supported action types include: `invite dtmf refer intercom` Polycom recommends that you always define this field. If it is not defined, the supplied digits are dialed using INVITE (if no active call) or DTMF (if an active call). The use of refer method is call server dependent and may require the addition of star codes.
$M<macro>$	The embedded macro. The <macro> string must begin with a letter. If the macro name is not defined, the execution of the action string fails.
$P<prompt num>N<num digits>$	The user input prompt string.
$S<speed dial index>$	The speed dial index. Only digits are valid. The action is found in the contact field of the local directory entry pointed to by the index
$F<internal function>$	An internal key function.
URL	A URL. Only one per action string is supported.

Related Links

Local Contact Directory on page 276

Prompt Macro Substitution

The macros provide a generic and easy way to manage and define the prompt to be displayed to the user, the maximum number of characters that the user can input, and the action that the phone performs after all user input has been collected.

The macros are case sensitive.

If a macro attempts to use a prompt that is disabled, the macro execution fails. A prompt is not required for every macro.

For example, the `efk.efklist.x.action.string` can be defined by a macro substitution string, PnNn, where the following applies:

- Pn is the prompt x as defined by `efk.efkprompt.x` .
- Nn is the number of digits or letters that the user can enter. The value must be between 1 and 32 characters otherwise the macro execution fails. The user must press the **Enter** soft key to complete data entry.

Related Links

Local Contact Directory on page 276

Expanded Macros

Expanded macros are prefixed with the ^ character and are inserted directly into the local directory contact (ct) field.

Related Links

Local Contact Directory on page 276

Example Macros

The action string `$Changup$*444*$P1N4$$Tinvite$$Cwaitconnect$$P2N3$$Cpause2$$Tdtmf$ $Changup$` is executed in order as follows:

1. The user is prompted for 4 digits. For example, 1234.
2. The user is prompted for 3 digits. For example, 567.
3. The user's active call is disconnected.
4. The string *444*1234 is sent using the INVITE method.
5. After connection, there is a two second pause, and then the string 567 is sent using DTMF dialing on the active call.
6. The active call is disconnected.

Because line keys and their functions are linked to fields in the directory file, the macro name you enter in `efk.list.x.mname` must match the name you enter to the `contact (ct)` field in the directory file. The macro name you enter in the `(ct)` field of the directory file must begin with the '!' prefix.

Flexible Line Key Assignment

You can enable users to assign a line key function to any line key on the phone.

By default, functions are assigned to line keys in succession—the order in which the line key displays on the phone. Flexible Line Keys (FLK) enables you to break that ordering and assign a line key function to a line key that displays anywhere on the phone's screen. You can apply this feature to any line key function, including line appearance, speed dial, busy lamp field (BLF), presence, and Enhanced Feature Keys.

This feature is available on the VVX 200 series, VVX 300 series, 400 series, 500 series, 600 series, and VVX Expansion Modules.

Note: Line keys on VVX phones and expansion modules are numbered sequentially, and the line keys on VVX expansion modules depend on how many lines your phone supports. For example, a VVX 600/601 phone supports 16 lines, numbered 1-16. The first line on an expansion module connected to a VVX 600/601 phone is line 17.

Flexible Line Keys Parameters

Line keys that you configure using this feature override the default line key assignments as well as any custom line key configurations you may have made.

To use this feature, you need to specify the function of each line key on the phone. You do this by assigning a category (`lineKey.x.category`) and an index (`lineKey.x.index`) to each line key, both of which are explained in the Enhanced Feature Key Example Configurations.

Use the parameters in the following table to configure this feature.

Flexible Line Keys Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`lineKey.reassignment.enabled`	Specify at least two calls per line key. 0 (default) - Disable the flexible line key assignment. 1 - Enable the flexible line key assignment.	No
`reg-advanced.cfg`	`lineKey.x.category`	Specify the line key category. Unassigned (default) Line BLF EFK SpeedDial Presence	No
`reg-advanced.cfg`	`lineKey.x.index`	Specify the line key number (dependent on category). 0 (default) - The index value for BLF or presence. 0- 9999	No

Assigning BLF and Presence to Line Keys

Specific conditions apply when you assign BLF or presence to line keys.

If you are assigning BLF or presence to a line key, assign that line key to `index=0` to indicate automatic ordering. BLF and presence line keys are self-ordering, meaning that if you have these features assigned to multiple line keys, you can specify the location of the BLF or presence line key but not the order in which they display. For example, you can assign a BLF line key to index 1, 3, and 5 but you cannot specify how the contacts are ordered, which BLF contacts display on line keys 1, 3, and 5.

In addition, to assign BLF and presence to a line key, you need to assign a corresponding registration line. You can configure multiple line keys per registration if each line key has a corresponding `reg.x.lineKeys` parameter.

Flexible Line Key Assignment Categories and Index

The FLK category specifies the function of the line key.

The index specifies the order in which the line keys display on the phone screen. Use the following table to help you assign a category and an index to the line keys on your phone. Note that the category Unassigned leaves the line key blank.

Flexible Line Key Assignment Categories and Index

Category	Index
Unassigned	Null
Line	The Line index number.
BLF	0
Speed Dial	The speed dial index number.
Presence	0
EFK	0

Phone Keypad

You can customize many of the default key functions on the phone's keypad interface.

Polycom recommends that you configure only those phone keys with removable key caps, which includes Directories, Applications, Conference, Transfer, Redial, Menu, Messages, Do Not Disturb, and Call Lists.

Note: Polycom recommends that you remap only those keys with removable key caps. If you remap other keys, your phone may not work properly. You should not remap the following keys: the dial pad, volume control, handsfree, mute, headset, hold, and the navigation arrow keys.

Phone Keypad Parameters

You can configure phone keys in the following ways:

- Assign a function or feature to a key
- Turn a phone key into a speed dial
- Assign enhanced feature key (EFK) operations to a phone key

  For example, you can map a phone menu path to a single key press using a macro code. See Enhanced Feature Keys.

- Delete all functions and features from a phone key

Use the parameters in the following table to change the layout of your phone's keypad.

Phone Keypad Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
features.cfg	key.x.function.prim	Set the primary key function for key y on phone model x. Null (default) String (maximum of 255 characters)	No
features.cfg	key.x.subPoint.prim	Set the secondary key function for key y on phone model x. Null (default) String (maximum of 255 characters)	No

Related Links

Enhanced Feature Keys on page 473

Multiple Key Combinations

You can reboot the phone, reset the phone to factory default values, upload log files from the phone to your provisioning server, set the Base Profile, and view phone details with a multiple key combination (MKC) on your Polycom phones.

Note: For other methods for resetting and rebooting your Polycom phones, refer to *Updating, Troubleshooting, and Resetting SoundPoint IP, SoundStation IP, and VVX 1500 Phones*: *Quick Tip 18298 at* Polycom Engineering Advisories and Technical Notifications.

Rebooting the Phone with a MKC

You can reboot the phones with a multiple key combination (MKC) that varies by phone model.

Rebooting the phone downloads new software and new configuration files if available on the provisioning server.

Depending on your phone model, press and hold the following keys simultaneously until you hear a confirmation tone (for about three seconds).

Phone Reboot Multiple Key Combinations

Phone Model	MKC
VVX 101, 150, 201, 250	0, 1, and 3
VVX 300, 310, 350	0, 1, and 3

Phone Model	MKC
VVX 301, 311	0, 1, and 3
VVX 400, 410, 450	0, 1, and 3
VVX 401, 411	0, 1, and 3
VVX 500, 501	0, 1, and 3
VVX 600, 601	0, 1, and 3
VVX 1500	Delete, Volume-, Volume+, and Select

Resetting the Phone to Defaults with a MKC

You can reset a phone to factory default settings with a multiple key combination (MKC) that varies by phone model.

This is useful when you use more than one method to configure phones and phone features. Resetting the phone to defaults clears the flash parameters, removes log files, user data, and cached data, and resets the administrator password to 456.

Resetting to factory defaults also resets the administrator password (factory default password is 456). Polycom recommends that you change the administrative password from the default value.

Note: After you reset to factory defaults on a Polycom VVX 1500 D phone, you must re-enable the H. 323 protocol through a configuration file change or by using the Web Configuration Utility.

Depending on your phone model, press and hold the following keys simultaneously during the updater/ BootROM countdown process until the administrator password prompt displays.

Factory Default Multiple Key Combinations

Phone Model	MKC
VVX 101, 150, 201, 250	1, 3, and 5
VVX 300, 310, 350	1, 3, and 5
VVX 301, 311	1, 3, and 5
VVX 400, 410, 450	1, 3, and 5
VVX 401, 411	1, 3, and 5
VVX 500, 501	1, 3, and 5
VVX 600, 601	1, 3, and 5
VVX 1500	4, 6, 8, and * dial pad keys

Uploading Log Files with a MKC

You can use a a multiple key combination (MKC) to upload log files to your provisioning server with a multiple key combination that varies by phone model.

Uploading log files copies the log files from the phone to the provisioning server. and creates new files named **<MACaddress** > **-now-xxx.log**.

Depending on your phone model, press and hold one the following keys simultaneously for about three seconds until you hear a confirmation tone.

Log Upload Multiple Key Combinations

Phone Model	MKC
VVX 101, 150, 201, 250	1, 5, and 9
VVX 300, 310, 350	1, 5, and 9
VVX 301, 311	1, 5, and 9
VVX 400, 410, 450	1, 5, and 9
VVX 401, 411	1, 5, and 9
VVX 500, 501	1, 5, and 9
VVX 600, 601	1, 5, and 9
VVX 1500	Up, Down, Left, and Right arrow keys

Set the Base Profile with a MKC

You can set the base profile with a multiple key combination (MKC), which allows for quick setup of Polycom phones with Microsoft Lync Server and Skype for Business Server.

Depending on your phone model, press and hold the following keys simultaneously for about three seconds until you hear a confirmation tone.

Factory Default Multiple Key Combinations

Phone Model	MKC
VVX 101, 150, 201, 250	1, 4, and 9
VVX 300, 310, 350	1, 4, and 9
VVX 301, 311	1, 4, and 9
VVX 400, 410, 450	1, 4, and 9
VVX 401, 411	1, 4, and 9
VVX 500, 501	1, 4, and 9

Phone Model	MKC
VVX 600, 601	1, 4, and 9
VVX 1500	1, 4, and 9

View Phone Details with a MKC

You can use a multiple key combination to view frequently-used administrator phone details including:

- IP Address
- Boot Server Type
- MAC Address
- VLAN
- Boot Server Address
- UC Software version

Procedure

1. Press and hold keys **1**,**4**, and **7**.

Defining the Phone Key Layout

You can redefine certain hard key functions using parameters in the configuration files.

The following figures and tables show the default key layouts for the following phone models:

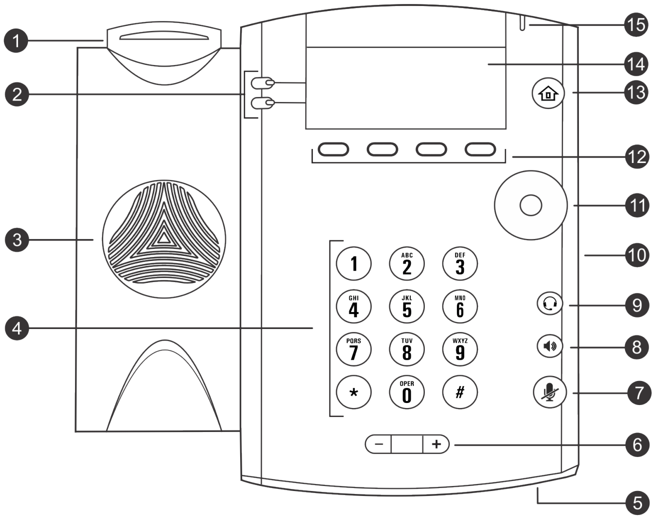
Related Links

Key Mapping Parameters on page 635
System and Model Names on page 552

VVX 101 and 201 Business Media Phones Key Layout

The following figure and table show the available phone key functions.

V



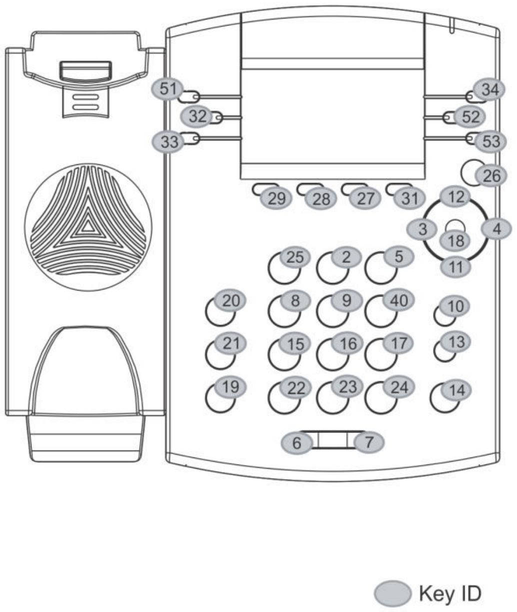
VVX 101 and 201 Default Key Functions

KEY ID	Function	KEY ID	Function
1	Hookswitch	9	Headset key
2	Line keys	10	Security slot (on side)
3	Speaker	11	Navigation keys / Select key
4	Dial pad keys	12	Soft keys
5	Microphone	13	Home key
6	Volume keys	14	Screen
7	Mute key	15	Message Waiting Indicator
8	Speakerphone key		

VVX 300, 301, 310, and 311 Business Media Phones Key Layout

The following figure and table show the available phone key functions.

IDs that have no function are described as n/a.



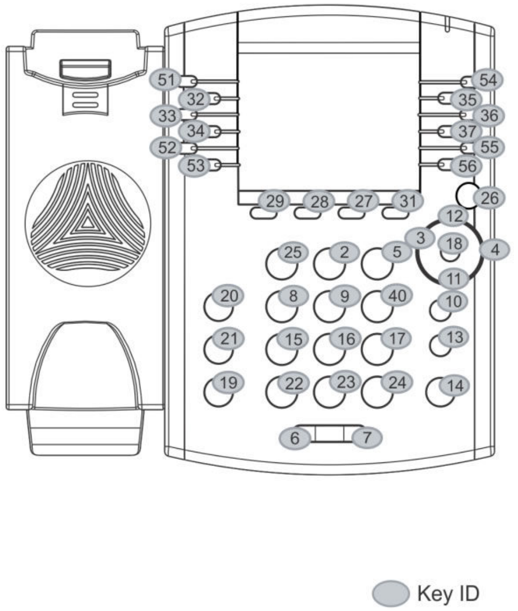
VVX 3xx Default Key Functions

KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
1	n/a	15	Dialpad7	29	SoftKey1	43	n/a
2	Dialpad2	16	Dialpad8	30	n/a	44	n/a
3	ArrowLeft	17	Dialpad9	31	SoftKey4	45	n/a
4	ArrowRight	18	Select	32	Line2	46	n/a
5	Dialpad3	19	Hold	33	Line3	47	n/a
6	VolDown	20	Transfer	34	Line4	48	n/a
7	VolUp	21	Messages	35	n/a	49	n/a
8	Dialpad4	22	DialpadStar	36	n/a	50	n/a
9	Dialpad5	23	Dialpad0	37	n/a	51	Line1
10	Headset	24	DialpadPound	38	n/a	52	Line5
11	ArrowDown	25	Dialpad1	39	n/a	53	Line6
12	ArrowUp	26	Home	40	Dialpad6		
13	Handsfree	27	SoftKey3	41	n/a		
14	MicMute	28	SoftKey2	42	n/a		

VVX 400, 401, 410, and 411 Business Media Phones Key Layout

The following figure and table show the available phone key functions.

IDs that have no function are described as n/a.



VVX 4xx Default Key Functions

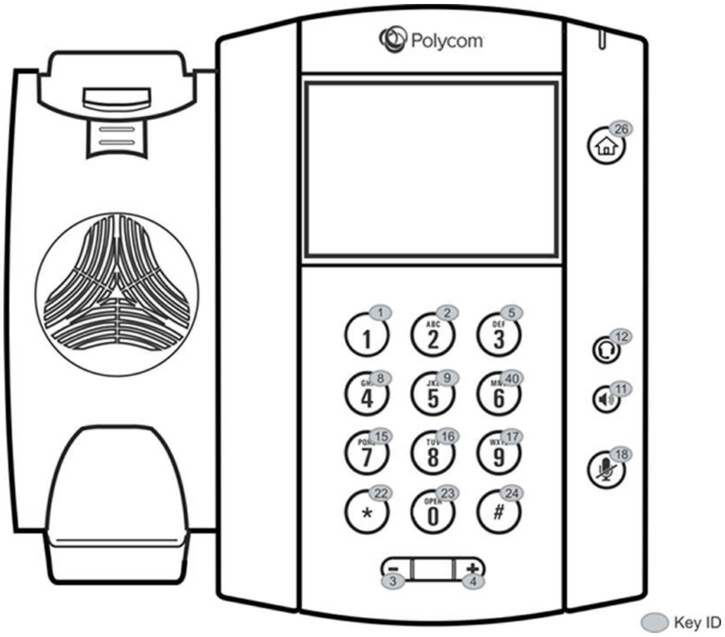
KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
1	n/a	15	Dialpad7	29	SoftKey1	43	n/a
2	Dialpad2	16	Dialpad8	30	n/a	44	n/a
3	ArrowLeft	17	Dialpad9	31	SoftKey4	45	n/a
4	ArrowRight	18	Select	32	Line2	46	n/a
5	Dialpad3	19	Hold	33	Line3	47	n/a
6	VolDown	20	Transfer	34	Line4	48	n/a
7	VolUp	21	Messages	35	Line8	49	n/a
8	Dialpad4	22	DialpadStar	36	Line9	50	n/a
9	Dialpad5	23	Dialpad0	37	Line10	51	Line1
10	Headset	24	DialpadPound	38	n/a	52	Line5
11	ArrowDown	25	Dialpad1	39	n/a	53	Line6

KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
12	ArrowUp	26	Home	40	Dialpad6	54	Line7
13	Handsfree	27	SoftKey3	41	n/a	55	Line11
14	MicMute	28	SoftKey2	42	n/a	56	Line12

VVX 500 and 501 Business Media Phones Key Layout

The following figure and table show the available phone key functions.

IDs that have no function are described as n/a.



VVX 5xx Default Key Functions

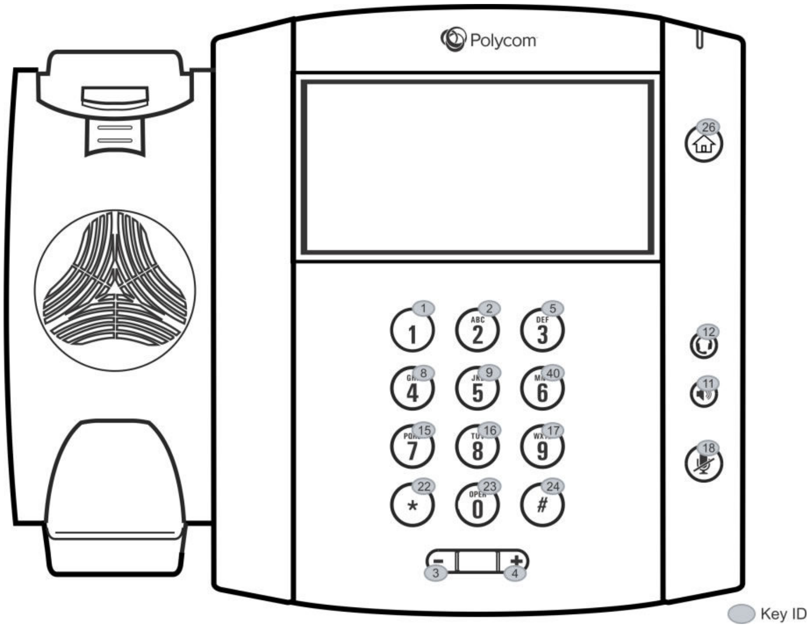
KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
1	Dialpad1	12	Headset	23	Dialpad0	34	n/a
2	Dialpad2	13	n/a	24	DialpadPound	35	n/a
3	VolDown	14	n/a	25	n/a	36	n/a
4	VolUp	15	Dialpad7	26	Home	37	n/a
5	Dialpad3	16	Dialpad8	27	n/a	38	n/a

KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
6	n/a	17	Dialpad9	28	n/a	39	n/a
7	n/a	18	MicMute	29	n/a	40	Dialpad6
8	Dialpad4	19	n/a	30	n/a	41	n/a
9	Dialpad5	20	n/a	31	n/a	42	n/a
10	n/a	21	n/a	32	n/a		
11	Handsfree	22	DialpadStar	33	n/a		

VVX 600 and 601 Business Media Phones Key Layout

The following figure and table show the available phone key functions.

IDs that have no function are described as n/a.



VVX 6xx Default Key Functions

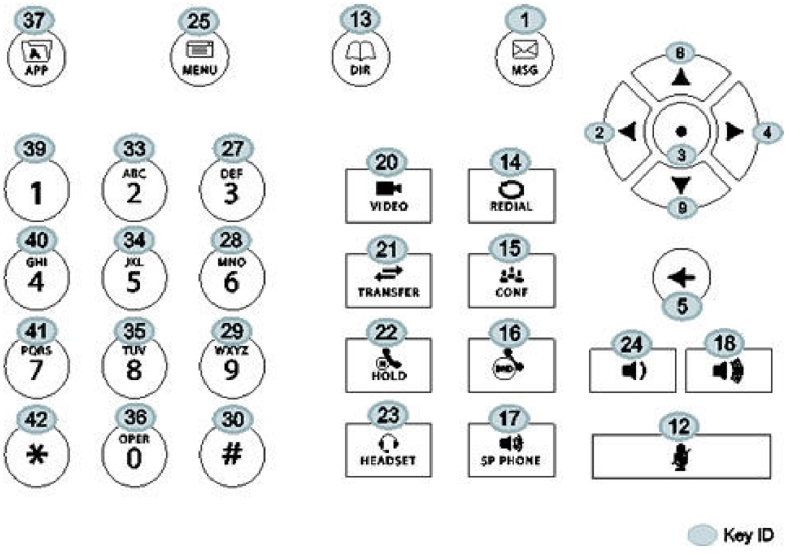
KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
1	Dialpad1	12	Headset	23	Dialpad0	34	n/a
2	Dialpad2	13	n/a	24	DialpadPound	35	n/a

KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
3	VolDown	14	n/a	25	n/a	36	n/a
4	VolUp	15	Dialpad7	26	Home	37	n/a
5	Dialpad3	16	Dialpad8	27	n/a	38	n/a
6	n/a	17	Dialpad9	28	n/a	39	n/a
7	n/a	18	MicMute	29	n/a	40	Dialpad6
8	Dialpad4	19	n/a	30	n/a	41	n/a
9	Dialpad5	20	n/a	31	n/a	42	n/a
10	n/a	21	n/a	32	n/a		
11	Handsfree	22	DialpadStar	33	n/a		

VVX 1500 Business Media Phone Key Layout

The following figure and table show the available phone key functions.

IDs that have no function are described as n/a.



VVX 1500 Default Key Functions

KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
1	Messages	12	MicMute	23	Headset	34	Dialpad5

KEY ID	Function	KEY ID	Function	KEY ID	Function	KEY ID	Function
2	ArrowLeft	13	Directories	24	VolDown	35	Dialpad8
3	Select	14	Redial	25	Menu	36	Dialpad0
4	ArrowRight	15	Conference	26	n/a	37	Applications
5	Delete	16	DoNotDisturb	27	Dialpad3	38	n/a
6	n/a	17	Handsfree	28	Dialpad6	39	Dialpad1
7	n/a	18	VolUp	29	Dialpad9	40	Dialpad4
8	ArrowUp	19	n/a	30	DialpadPound	41	Dialpad7
9	ArrowDown	20	Video	31	n/a	42	DialpadStar
10	n/a	21	Transfer	32	n/a		
11	n/a	22	Hold	33	Dialpad2		

Mapping Internal Key Functions

A complete list of internal key functions for enhanced feature keys and hard key mappings is shown in the table Key Labels and Internal Functions.

Note the following guidelines:

- The **Function** value is case sensitive.
- Some functions are dependent on call state. Generally, if the soft key displays on a call screen, the soft key function is executable.
- Some functions depend on the feature being enabled. For example, the BuddyStatus and MyStatus soft keys require the presence feature to be enabled.
- Hard key remappings do not require the enhanced feature key feature to be enabled. This includes the speed dial function on older platforms. On newer platforms, use line key functions.

The table below shows only line1 to line 6 functions.

Key Labels and Internal Functions

Function	Description	Notes
ACDAvailable	Status for Automatic Call Distribution when available.	
ACDLogin	Login to Automatic Call Distribution.	
ACDLogout	Logout from Automatic Call Distribution.	

Function	Description	Notes
ACDUnavailable	Status for Automatic Call Distribution when unavailable.	
Answer	Answer an incoming call.	Call screen only
Applications	Main Browser	
ArrowDown	Move arrow down	
ArrowLeft	Move arrow left	
ArrowRight	Move arrow right	
ArrowUp	Move arrow up	
BargeIn	Barge In to show appearances, Barge In	Call screen only
BuddyStatus	Status of the contacts added to Buddy list.	
Callers	Displays the list of callers.	
CallList	Displays the call logs.	
CallPark	Park an active call.	Call screen only
CallPickup	Call pick-up on the phone.	Call screen only
Conference	Begin a conference call.	Call screen only
Delete	Delete the selected item.	
Dialpad0	Dialpad 0	
Dialpad1	Dialpad 1	
Dialpad2	Dialpad 2	
Dialpad3	Dialpad 3	
Dialpad4	Dialpad 4	
Dialpad5	Dialpad 5	
Dialpad6	Dialpad 6	
Dialpad7	Dialpad 7	
Dialpad8	Dialpad 8	
Dialpad9	Dialpad 9	

Function	Description	Notes
DialpadPound	Dialpad pound sign	
DialpadStar	Dialpad star sign	
DialpadURL	Navigate to a specific address or location.	Call screen only
DirectedPickup	Directed call pick-up on the phone.	Call screen only
Directories	Displays the directory items.	
Divert	Forward a call.	
DoNotDisturb	Do Not Disturb menu	
EnterRecord	Enter a call record.	Call screen only
Exit	Exit existing menu.	Menu only
GroupPickup	Group call pick-up on the phone.	
Handsfree	Use handsfree	
Headset	Use headset	Desktop phones only
Hold	Toggle hold	
Join	Joins a call to an active call to make a conference.	Call screen only
LCR	Last Call Return	
Line1	Line Key 1	
Line2	Line Key 2	
Line3	Line Key 3	
Line4	Line Key 4	
Line5	Line Key 5	
Line6	Line Key 6	
ListenMode	Turn on speaker to listen only.	
LockPhone	Lock the phone.	
Menu	Displays the main menu.	
Messages	Messages menu	

Function	Description	Notes
MicMute	Mute the microphone.	
MyStatus	View my status.	
NewCall	Place a new call.	Call screen only
Null	Do nothing	
Offline	Offline for presence	
Page	Group Paging	
ParkedPickup	Specifies how the phone performs a parked call pick-up.	Call screen only
QuickSetup	Quick Setup feature	Call screen only
Redial	Redial the last dialed number.	Call screen only
Select	Select an item.	
ServerACDAgentAvailable	Status for server-based Automatic Call Distribution agent when available.	
ServerACDAgentUnavailable	Status for server-based Automatic Call Distribution agent when unavailable.	
ServerACDSignIn	Login to a server-based Automatic Call Distribution.	
ServerACDSignOut	Logout from a server-based Automatic Call Distribution.	
Setup	Settings menu	
Silence	Silence the call ringer.	Call screen only
SoftKey1	SoftKey 1	
SoftKey2	SoftKey 2	
SoftKey3	SoftKey 3	
SoftKey4	SoftKey 4	
Softkey5	Softkey 5	
SpeedDial	Place a call to a number assigned to the SpeedDial.	

Function	Description	Notes
Split	Split a conference call.	Call screen only
Talk	Push-to-Talk	
Transfer	Transfer a call	Call screen only
Video	Enables the video in a call.	Polycom VVX 500/501, 600/601, and 1500 business media phones.
VolDown	Set volume down	
VolUp	Set volume up	

Network

Topics:

Polycom's Open SIP UC Software enables you to make custom network configurations.

Related Links

Supported Network Configurations on page 23

Two-Way Active Measurement Protocol

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

TWAMP defines the following protocols:

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

TWAMP Limitations

TWAMP includes the following limitations:

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

Two-Way Active Measurement Protocol Configuration Parameters

The following table includes the new or modified parameters for the two-way active measurement protocol feature.

Two-Way Active Measurement Protocol Configuration Parameters

Template	Parameters	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`feature.twamp.enabled`	0 (default) - Disable TWAMP protocol support. 1 - Enable TWAMP protocol support.	No
`site.cfg`	`twamp.port.udp.PortRangeEnd`	Set the TWAMP UDP session max port range value. 60000 (default) 1024 - 65486	No
`site.cfg`	`twamp.port.udp.PortRangeStart`	Set the TWAMP UDP session start port range value. 40000 (default) 1024 - 65485	No
`site.cfg`	`twamp.udp.maxSession`	Set the maximum UDP session supported by TWAMP. 1 (default) 1 - 10	No

3GPP Technical Specifications

For an IP Multimedia Subsystem (IMS) environment, Polycom has introduced support for a subset of the 3rd Generation Partnership Project technical specifications (3GPP TS) 24.229, 24.615, and 24.629.

In addition, Polycom phones provide partial or complete support for the following RFCs:

- RFC 3327
- RFC 3608
- RFC 3680
- RFC 6665
- RFC 6228

- RFC 3261
- RFC 5009
- RFC 7462
- RFC 7329
- RFC 6026
- RFC 3581
- RFC 6947

VVX phones support the following IMS feature enhancements:

- The call waiting ring-back tone plays to inform users that a 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 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 ring back tone.
- The VQMon messages generated by the phone can contain service route information in SIP route headers.
- In a NAT network, a phone may need to send keep-alive messages to maintain the IP addresses mapping in the NAT table.

3GPP Technical Specifications Parameters

Use the 3GPP parameters in the following table to configure IP Multimedia Subsystem (IMS) features.

3GPP Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	nat.keepalive.tcp.payload	Configure a customizable string as the payload of a TCP keep-alive message. The string value cannot be blank. CRLFCRLFCRLFCRLFCRLFCRLFCRLFCRLF (default)	No
sip-interop.cfg	nat.keepalive.udp.payload	Configure a customizable string as the payload of a UDP keep-alive message. You can leave the string value blank to configure an empty payload. CRLFCRLF (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.header.pearlymedia.support	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.	No
reg-basic.cfg	reg.X.insertOBPAddressInRoute	1 (Default) - The outbound proxy address is added as the topmost route header. 0 - The outbound proxy address is not added to the route header.	No
features.cfg	reg.x.path	0 (Default) - The path extension header field in the Register request message is not supported for the specific line registration. 1 - The phone supports and provides the path extension header field in the Register request message for the specific line registration.	No
features.cfg	reg.x.regevent	0 (default) - The phone is not subscribed to registration state change notifications for the specific phone line. 1 - The phone is subscribed to registration state change notifications for the specific phone line. This parameter overrides the global parameter voIpProt.SIP.regevent.	No
reg-advanced.cfg	reg.x.rejectNDUBInvite	Specify whether or not 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) - If an NDUB event occurs, the phone does not reject the call. 1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.server.y.specialInterop	Specify the server-specific feature set for the line registration. VVX 101: Standard (default), GENBAND, ALU-CTS, DT VVX 201: Standard (default), GENBAND, ALU-CTS, ocs2007r2, lync2010 All other phones: Standard (default), GENBAND, ALU-CTS, ocs2007r2, lync2010, lcs2005	
features.cfg	voice.qualityMonitoring.processServiceRoute.enable	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 the phone contain service route information, if available, in SIP route headers.	Yes
sip-interop.cfg	voIpProt.SIP.header.pEarlyMedia.support	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.	
sip-interop.cfg	voIpProt.SIP.IMS.enable	This parameter applies to all registered or unregistered SIP lines on the phone. 0 (Default) - The phone does not support IMS features introduced in UC Software 5.5.0. 1 - The phone supports IMS features introduced in UC Software 5.5.0.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`voIpProt.SIP.regevent`	0 (default) - The phone is not subscribed to registration state change notifications for all phone lines. 1 - The phone is subscribed to registration state change notifications for all phone lines. This parameter is overridden by the per-phone parameter reg.x.regevent.	
`sip-interop.cfg`	`voIpProt.SIP.rejectNDUBInvite`	Specify whether or not 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) - If an NDUB event occurs, the phone does not reject the call for all line registrations. 1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code for all line registrations.	
`reg-basic.cfg`	`voIpProt.SIP.supportFor199`	Determine support for the 199 response code. For details on the 199 response code, see RFC 6228. 0 (Default) - The phone does not support the 199 response code. 1- The phone supports the 199 response code.	

Technical Report-069

Technical Report-069 (TR-069) enables you to remotely manage end-user devices.

As a bidirectional SOAP/HTTP-based protocol, TR-069 enables secure communication between Auto Configuration Servers (ACS) and Polycom phones. Using TR-069, you can remotely configure and manage Polycom phones by provisioning systems that comply with TR-069 technical specification.

TR-069 Parameters

Polycom provides parameters for the TR-104 and TR-106 data models that support provisioning of TR-069-enabled devices by an Auto-Configuration Server (ACS).

TR-104 is a parameter data model for VoIP-only devices, and TR-106 is a parameter data model for all TR-069-enabled devices.

Use the parameters in the following table to configure this feature.

TR-069 Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg; tr069.cfg`	`device.feature.tr069.enabled`	0 (default) - Disables TR-069 feature 1 - Enables TR-069 feature	No
`device.cfg; tr069.cfg`	`device.feature.tr069.enabled.set`	0 (default) 1	No
`tr069.cfg`	`device.tr069.acs.password`	Sets the TR-069 ACS server password used to authenticate the phone. `Null` (default) String (256 maximum characters)	No
`tr069.cfg`	`device.tr069.acs.url`	Sets the URL for the TR-069 ACS server. `Null` (default) URL (256 maximum characters)	No
`tr069.cfg`	`device.tr069.acs.username`	Sets the TR-069 ACS server user name used to authenticate the phone. `PlcmSpip` (default) String (256 maximum characters)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
tr069.cfg	device.tr069.cpe.password	Specifies the TR-069 CPE password, which authenticates a connection request from the ACS server. Null (default) String (256 maximum characters)	No
tr069.cfg	device.tr069.cpe.username	Specifies the TR-069 CPE user name, which authenticates a connection request from the ACS server. PlcmSpip (default) String (256 maximum characters)	No
tr069.cfg	device.tr069.periodicInform.enabled	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.	No
tr069.cfg	device.tr069.periodicInform.interval	Specifies the time interval in seconds in which the CPE must attempt to connect with the ACS to send CPE information if device.tr069.periodicInform.enabled =”1”. 18000 (default) 0 to 36000	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`tr069.cfg`	`device.tr069.upgradesManaged.enabled`	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.	No
`tr069.cfg`	`log.level.change.tr069`	Sets the log levels for the TR-069 feature. 4 (default) 0 - 6	No

Configuring TR-069

You can configure the TR-069 feature through the phone menu, Web Configuration Utility, or configuration parameters on a central server.

You can configure Polycom phones with an ACS server, including user name and password, using DHCP Option 43 for IPv4 and DHCP Option 17 for IPv6.

Configure TR-069 Settings on the Phone Menu

You can configure TR-069 settings on the phone menu.

Procedure

1. Go to **Settings** > **Advanced** > **Administration Settings** > **Network Configuration.**
2. Select **TR-069**, and select **Enabled**.
3. In the**TR069 Menu**, select **ACS Configuration** and enter values for the following settings:
   - URL
   - Username
   - Password
   - Periodic Inform
   - Inform Interval
4. In **Phone/CPE Configuration**, configure a user name and password.
5. In **Upgrade Management**, select **Enable** or **Disable**.

Configure TR-069 from the Web Configuration Utility

You can configure TR-069 from the Web Configuration Utility.

Procedure

1. In the Web Configuration Utility, navigate to **Settings** > **Provisioning Server** > **TR-069 Menu**.

Map TR-106 Parameters to Polycom Parameters

The data model TR-106 defines the TR-069 ACS parameter details.

The parameters listed as ‘Internal Value' are not directly mapped to a configuration parameter on the phone, and the phone generates these values dynamically to provide to the ACS server.

The following table lists the TR-106 parameters and their corresponding Polycom parameters.

TR-106 Parameters to Polycom Parameters

TR-106 ACS parameter names	Parameter (Polycom parameter names)	Writable
Device		
Device.DeviceInfo		
Manufacturer	Internal Value	No
ManufacturerOUI	Internal Value	No
ModelName	Internal Value	No
ProductClass	Internal Value	No
SerialNumber	Internal Value	No
HardwareVersion	Internal Value	No
SoftwareVersion	Internal Value	No
UpTime	Internal Value	No
Device.ManagementServer.		
URL	`device.tr069.acs.url`	Yes
Username	`device.tr069.acs.username`	Yes
Password	`device.tr069.acs.password`	Yes
PeriodicInformEnable	`device.tr069.periodicInform.enabled`	Yes
PeriodicInformInterval	`device.tr069.periodicInform.interval`	Yes

TR-106 ACS parameter names	Parameter (Polycom parameter names)	Writable
ConnectionRequestURL	Internal Value	No
ConnectionRequestUsername	`device.tr069.cpe.username`	Yes
ConnectionRequestPassword	`device.tr069.cpe.password`	Yes
UpgradesManaged	`device.tr069.upgradesManaged.enabled`	Yes
STUNServerAddress	`tcpIpApp.ice.stun.server`	Yes
STUNServerPort	`tcpIpApp.ice.stun.udpPort`	Yes
STUNUsername	`tcpIpApp.ice.username`	Yes
STUNPassword	`tcpIpApp.ice.password`	Yes
Device.LAN.		
IPAddress	Internal Value	No
SubnetMask	Internal Value	No
DNSServers	Internal Value	No
MACAddress	Internal Value	No
MACAddressOverride	Internal Value	No

Map TR-104 Parameters to Polycom Parameters

The data model TR-104 defines the TR-069 ACS parameter details.

The parameters listed as ‘Internal Value' are not directly mapped to a configuration parameter on the phone and the phone generates these values dynamically to provide to the ACS server.

The following table list the TR-104 parameters and their corresponding Polycom parameters.

TR-104 Parameters to Polycom Parameters

TR-104 ACS parameter names	CPE Parameter (Polycom parameter names)	Writable
VoiceService.{i}.VoiceProfile.{i}.		
DigitMap	`dialplan.digitmap`	Yes
VoiceService.{i}.VoiceProfile.{i}.SIP.		
RegistrarServer	`voIpProt.server.X.address`	Yes
RegistrarServerPort	`voIpProt.server.X.port`	Yes

TR-104 ACS parameter names	CPE Parameter (Polycom parameter names)	Writable
OutboundProxy	voIpProt.SIP.outboundProxy.address	Yes
OutboundProxyPort	voIpProt.SIP.outboundProxy.port	Yes
RegisterExpires	voIpProt.server.X.expires	Yes
RegistersMinExpires	voIpProt.server.X.expires.overlap	Yes
RegisterRetryInterval	voIpProt.server.X.retryTimeOut	Yes
VoiceService.{i}.VoiceProfile.{i}.SIP.EventSubscribe.{i}.		
ExpireTime	voIpProt.server.X.subscribe.expires	Yes
VoiceService.{i}.VoiceProfile.{i}.H323.		
Gatekeeper	voIpProt.server.H323.X.address	Yes
GatekeeperPort	voIpProt.server.H323.X.port	Yes
VoiceService.{i}.VoiceProfile.{i}.RTP.		
LocalPortMin	tcpIpApp.port.rtp.mediaPortRangeStart	Yes
LocalPortMax	tcpIpApp.port.rtp.mediaPortRangeEnd	Yes
VoiceService.{i}.VoiceProfile.{i}.RTP.SRTP.		
Enable	sec.srtp.enable	Yes
VoiceService.{i}.VoiceProfile.{i}.ButtonMap.Button.{i}.		
ButtonName	softkey.X.label	Yes
FacilityAction	softkey.X.action	Yes
UserAccess	softkey.X.enable	Yes
VoiceService.{i}.VoiceProfile.{i}.Line.{i}.		
DirectoryNumber	reg.X.address	Yes
VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP.		
AuthUserName	reg.X.auth.userId	Yes
AuthPassword	reg.X.auth.password	Yes

TR-104 ACS parameter names	CPE Parameter (Polycom parameter names)	Writable
VoiceService.{i}.VoiceProfile.{i}.Line.{i}.CallingFeatures.		
CallForwardUnconditionalEnable	`reg.X.fwdStatus`	Yes
CallForwardUnconditionalNumber	`reg.X.fwdContact`	Yes
CallForwardOnBusyEnable	`reg.X.fwd.busy.status`	Yes
CallForwardOnBusyNumber	`reg.X.fwd.busy.contact`	Yes
CallForwardOnNoAnswerEnable	`reg.X.fwd.noanswer.status`	Yes
CallForwardOnNoAnswerNumber	`reg.X.fwd.noanswer.contact`	Yes
CallForwardOnNoAnswerRingCount	`reg.X.fwd.noanswer.ringCount`	Yes
DoNotDisturbEnable	`divert.dnd.X.enabled`	Yes

Supported TR-069 Remote Procedure Call (RPC) Methods

The following table lists the supported RPC methods.

RPC Methods

RPC Method	Description
GetRPCMethods	Discovers the set of methods supported by the phone.
SetParameterValues	Modifies the value of one or more phone parameters.
GetParameterValues	Obtains the value of one or more phone parameters.
GetParameterNames	Discovers the parameters accessible on a particular phone.
GetParameterAttributes	Reads the attributes associated with one or more phone parameters.
SetParameterAttributes	Modifies attributes associated with one or more phone parameters.
Reboot	Reboots the phone.
Download	Causes the phone to download a specified file from the designated location. Supported file types for download: Firmware Image Configuration File
FactoryReset	Resets the phone to its factory default state.
TransferComplete	Informs the ACS of the completion (either successful or unsuccessful) of a file transfer initiated by an earlier Download or Upload method call.

RPC Method	Description
AddObject	Adds a new instance of an object defined on the phone.
DeleteObject	Removes a particular instance of an object.

Advice of Charge

In an IP Multimedia Subsystem (IMS) environment, Polycom phones support the Advice of Charge (AoC) feature as defined in Technical Specification (TS) 24.647 version 9.1.0 Release 9.

You can enable Polycom phones to display call charges information, which can include:

- Call setup charge and call tariff information - Displayed at the beginning of a call.
- Cumulative call cost - Displayed on an ongoing call.
- Complete call cost - Displayed after a call ends.

Advice of Charge Parameters

The following parameters configure the Advice of Charge (AoS) feature.

Before configuring AoS parameters, you must set `voIpProt.SIP.IMS.enable to 1.`

Advice of Charge Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.aoc.enable`	0 (Default) - The phone does not display call charge information. 1 - The phone displays call charge information.	No
`features.cfg`	`feature.adviceOfCharge.allowAudioNotification`	0 (Default) - There is no audio beep sound when the call charges information is updated on the phone display. 1 - The phone gives an audio beep when the call charges information is updated on the phone display.	No

Enhanced IPv4 ICMP Management

VVX phones support IPv4 by enabling the phone to ignore Internet Control Message Protocol (ICMP) redirect requests for an alternate path from the router or gateway.

IPv4 Parameters

You can configure IPv4using parameters listed in the following table.

IPv4 Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`device.cfg`	`device.icmp.ipv4IcmpIgnoreRedirect`	1 (default) - The phone ignores ICMP redirect requests for an alternate path from the router or gateway. 0 - The phone allows ICMP redirects.	No

IPv6 Protocol Support

VVX phones support IPv6 and you can configure the phones to operate in IPv4, IPv6, or dual stack (IPv4/ IPv6) mode.

You can enable and configure IPv6 support from the phone menu, the Web Configuration Utility, or with centralized provisioning.

IPv6 Parameters

Use the parameters in the following table to enable and configure IPv6.

IPv6 Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.dhcp.bootSrvUseOpt	Specifies the source for the boot server address for the phone. It can take values from 0 to 9. In IPv4 mode, the following values are applicable: • 0 (Default) - The phone gets the boot server address from Option 66. • 1 - The phone gets the boot server details from the custom option number provided through DHCP. • 2 - The phone uses the boot server configured through the Server Menu. • 3 - The phone uses the custom option first or uses Option 66 if the custom option is not present 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.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg ; wireless.cfg	device.ipv6.icmp.echoReplies	NULL (default) 0 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.echoReplies.set	0 (default) 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.genDestUnreachable	0 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.genDestUnreachable.set	0 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.ignoreRedirect	0 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.ignoreRedirect.set	0 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.txRateLimiting	0 1	No
device.cfg ; wireless.cfg	device.ipv6.icmp.txRateLimiting.set	0 - 6000	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.net.ipStack	Configures the IPv4, IPv6, or dual stack mode for the phone. 0 (Default) - IPv4 is enabled and IPv6 is disabled. 1 - IPv6 is enabled and IPv4 is disabled. 2 - Dual stack is enabled and phone operates on both IPv4 and IPv6 networks at the same time.	No
device.cfg, site.cfg	device.net.ipv6AddrDisc	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.	No
device.cfg, site.cfg	device.net.ipv6Address	Specify a valid global IPv6 unicast address for the phone. Null (default)	No
device.cfg, site.cfg	device.net.ipv6Gateway	Specify theIPv6 address of the default gateway for the phone. Null (default)	No
device.cfg, site.cfg	device.net.ipv6LinkAddress	Specifies a valid Link Local IPv6 address for the phone. Null (default)	No
device.cfg, site.cfg	device.net.ipv6PrivacyExtension	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.	No
device.cfg, site.cfg	device.net.ipv6ULAAddress	Specifies a valid Unique Local IPv6 address (ULA) for the phone. Null (default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, site.cfg	device.net.preferredNetwork	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.	No
device.cfg, site.cfg	ipv6.mldVersion	2 (default) 1	No
sip-interop.cfg	voipProt.SIP.anat.enabled	Enables or disables Alternative Network Address Types (ANAT). 0 (default) - ANAT is disabled. 1 - ANAT is enabled.	No

Real-Time Transport Protocol (RTP) Ports

You can configure RTP ports for your environment in the following ways:

- Filter incoming packets by IP address or port.
- Reject packets arriving from a non-negotiated IP address, an unauthorized source, or non-negotiated port for greater security.
- Enforce symmetric port operation for RTP packets. When the source port is not set to the negotiated remote sink port, arriving packets are rejected.
- Fix the phone's destination transport port to a specified value regardless of the negotiated port.

  This is useful for communicating through firewalls. When you use a fixed transport port, all RTP traffic is sent to and arrives on that specified port. Incoming packets are sorted by the source IP address and port, which allows multiple RTP streams to be multiplexed.
- Specify the phone's RTP port range.

  Since the phone supports conferencing and multiple RTP streams, the phone can use several ports concurrently. Consistent with RFC 1889, 3550, and 3551, the next-highest odd-numbered port is used to send and receive RTP.

RTP Ports Parameters

Use the parameters in the following table to configure RTP packets and ports.

Real-Time Transport Protocol Port Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.port.rtp.feccPortRange.enable	0 (default) – Use the Open SIP far-end camera control media port range. 1 - Use the far-end camera control port range configuration for Open SIP-registered lines.	No
site.cfg	tcpIpApp.port.rtp.feccPortRangeEnd	Specify the far-end camera control port range end port for Open SIP registrations. 2419 (default) 1024 - 65486	No
site.cfg	tcpIpApp.port.rtp.feccPortRangeStart	Specify the far-end camera control port range start port for Open SIP registrations. 2372 (default) 1024 – 65486	No
site.cfg	tcpIpApp.port.rtp.filterByIp[1]	IP addresses can be negotiated through the SDP or H.323 protocols. 1 (Default) - Phone rejects RTP packets that arrive from non-negotiated IP addresses. The H.323 protocol is supported on the VVX 500/501, 600/601, and 1500 phones.	Yes
site.cfg	tcpIpApp.port.rtp.filterByPort[1]	Ports can be negotiated through the SDP protocol. 0 (Default) 1 - Phone rejects RTP packets arriving from (sent from) a non-negotiated port.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.port.rtp.forceSend[1]	Send all RTP packets to, and expect all RTP packets to arrive on, this port. Range is 0 to 65535. 0 (Default) - RTP traffic is not forced to one port. Both tcpIpApp.port.rtp.filterByIp and tcpIpApp.port.rtp.filterByPort must be set to 1.	Yes
site.cfg	tcpIpApp.port.rtp.mediaPortRangeEnd	Determines the maximum supported end range of audio ports. Range is 1024 to 65485. 2269 (Default)	Yes
site.cfg	tcpIpApp.port.rtp.mediaPortRangeStart[1]	Set the starting port for RTP port range packets. Use an even integer ranging from 1024 to 65440. 2222 (Default) Each call increments the port number +2 to a maximum of 24 calls after the value resets to the starting point. Because port 5060 is used for SIP signaling, ensure that port 5060 is not within this range when you set this parameter. A call that attempts to use port 5060 has no audio.	Yes
site.cfg	tcpIpApp.port.rtp.videoPortRange.enable	Specifies the range of video ports. 0 - Video ports are chosen within the range specified by tcpIpApp.port.rtp.mediaPortRangeStart and tcpIpApp.port.rtp.mediaPortRangeEnd . 1 - Video ports are chosen from the range specified by tcpIpApp.port.rtp.videoPortRangeStart and tcpIpApp.port.rtp.videoPortRangeEnd . Base profile (Default) Skype = 1 (Default) Generic = 0 (Default)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`tcpIpApp.port.rtp.videoPortRangeEnd`	Determines the maximum supported end range of video ports. Range is 1024 to 65535. 2319 (Default)	Yes
`site.cfg`	`tcpIpApp.port.rtp.videoPortRangeStart`	Determines the start range for video ports. Range is 1024 to 65486. 2272 (Default) Used only if value of `tcpIpApp.port.rtp.videoPortRange.enable` is 1.	Yes

Network Address Translation (NAT)

Network Address Translation (NAT) enables a local area network (LAN) to use one set of IP addresses for internal traffic and another set for external traffic.

The phone's signaling and RTP traffic use symmetric ports. Note that the source port in transmitted packets is the same as the associated listening port used to receive packets.

Network Address Translation Parameters

You can configure the external IP addresses and ports used by the NAT on the phone's behalf on a per-phone basis.

Use the parameters in the following table to configure NAT.

Network Access Translation Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`nat.ip`	Specifies the IP address to advertise within SIP signaling. This should match the external IP address used by the NAT device. Null (default) IP address	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	nat.keepalive.interval	The keep-alive interval in seconds. Sets the interval at which phones sends a keep-alive packet to the gateway/NAT device to keep the communication port open so that NAT can continue to function. If Null or 0, the phone does not send out keep-alive messages. 0 (default) 0 - 3600	No
sip-interop.cfg	nat.mediaPortStart	The initially allocated RTP port. Overrides the value set for `tcpIpApp.port.rtp.mediaPortRangeStart` parameter. 0 (default) 0 - 65440	Yes
sip-interop.cfg	nat.signalPort	The port used for SIP signaling. Overrides the `voIpProt.local.port` parameter. 0 (default) 1024 - 65535	No

Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of phone service if, for example, the call server is taken offline for maintenance, the server fails, or the connection between the phone and the server fails.

Polycom phones support failover and fallback server redundancy types. In some cases, you can deploy a combination of the two server redundancy types. Consult your SIP server provider for recommended methods of configuring phones and servers for failover configuration.

Note: The concurrent failover/fallback feature is not compatible with Microsoft environments.

For more information, see *Technical Bulletin 5844: SIP Server Fallback Enhancements on Polycom Phones* and *Technical Bulletin 66546: Configuring Optional Re-Registration on Failover Behavior*.

Server Redundancy Parameters

Use the parameters in the following table to set up server redundancy for your environment.

Server Redundancy Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.auth.optimizedInFailover`	Set the destination for the first new SIP request when failover occurs. 0 (default) - The SIP request is sent to the server with the highest priority in the server list. 1 - The SIP request is sent to the server that sent the proxy authentication request.	No
`sip-interop.cfg`	`reg.x.outboundProxy.failOver.failBack.mode`	The mode for failover failback (overrides `reg.x.server.y.failOver.failBack.mode`). duration (default) - The phone tries the primary server again after the time specified by `reg.x.outboundProxy.failOver.failBack.timeout` expires. newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL you configured for the server the phone is registered to.	No
`reg-advanced.cfg`	`reg.x.outboundProxy.failOver.failBack.timeout`	3600 (default) - The time to wait (in seconds) before failback occurs (overrides `reg.x.server.y.failOver.failBack.timeout`). 0, 60 to 65535 - The phone does not fail back until a failover event occurs with the current server.	No
`reg-advanced.cfg`	`reg.x.outboundProxy.failOver.failRegistrationOn`	1 (default) - The global and per-line `reRegisterOn` parameter is enabled and the phone silently invalidates an existing registration. 0 - The global and per-line `reRegisterOn` parameter is enabled and existing registrations remain active.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.outboundProxy.failOver.onlySignalWithRegistered	1 (default) - The global and per-line reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. 0 - The global and per-line reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed.	No
reg-advanced.cfg	reg.x.outboundProxy.failOver.reRegisterOn	This parameter overrides reg.x.server.y.failOver.reRegisterOn . 0 (default) - The phone won't attempt to register with the secondary server. 1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server.	No
reg-advanced.cfg	reg.x.outboundProxy.port	The port of the SIP server to which the phone sends all requests. 0 - (default) 1 to 65535	No
reg-advanced.cfg	reg.x.outboundProxy.transport	The transport method the phone uses to communicate with the SIP server. DNSnaptr (default) DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.failOver.failBack.mode	Specify the failover failback mode. duration (default) - The phone tries the primary server again after the time specified by voIpProt.server.x.failOver.failBack.timeout newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to. registration - The phone tries the primary server again when the registration renewal signaling begins.	No
sip-interop.cfg	voIpProt.server.x.failOver.failBack.timeout	If voIpProt.server.x.failOver.failBack.mode is set to duration, this is the time in seconds after failing over to the current working server before the primary server is again selected as the first server to forward new requests. Values between 1 and 59 result in a timeout of 60. 0 means do not fail-back until a fail-over event occurs with the current server. 3600 (default) 0, 60 to 65535	No
sip-interop.cfg	voIpProt.server.x.failOver.failRegistrationOn	1 (default) - When set to 1, and the global or per-line reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over. 0 - When set to 0, and the global or per-line reRegisterOn parameter is enabled, existing registrations remain active. This means that the phone attempts failback without first attempting to register with the primary server to determine if it has recovered.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.failOver.onlySignalWithRegistered	1 (default) - When set to 1, and the global or per-line reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. 0 - When set to 0, and the global or per-line reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).	No
sip-interop.cfg	voIpProt.server.x.failOver.reRegisterOn	0 (default) - When set to 0, the phone won't attempt to register with the second. 1 - When set to 1, the phone attempts to register with (or by, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server.	No

DNS SIP Server Name Resolution

If a DNS name is given for a proxy/registrar address, the IP addresses associated with that name is discovered as specified in RFC3263.

If a port is given, the only lookup is an A record. If no port is given, NAPTR and SRV records are tried before falling back on A records if NAPTR and SRV records return no results. If no port is given, and none is found through DNS, port 5060 is used. If the registration type is TLS, port 5061 is used.

Caution: Failure to resolve a DNS name is treated as signaling failure that causes a failover.

The following configuration causes the phone to build an SRV request based on the address you provide, including all subdomains. Use the format:

- `voIpProt.SIP.outboundProxy.address` ="*sip.example.com*”
- `voIpProt.SIP.outboundProxy.port` ="0"

This SRV request produces a list of servers ordered by weight and priority, enabling you to specify sub-domains for separate servers, or you can create partitions of the same system. Please note that while making SRV queries and transport is configured as TCP, the phone adds the prefix `<_service._proto.>` to the configured address/FQDN but does not remove the sub-domain prefix, for example `sip.example.com` becomes `_sip._tcp.sip.example.com` . A single SRV query can be resolved into many different servers, session border controllers (SBCs), or partitions ordered by weight and priority, for example, `voice.sip.example.com` and `video.sip.example.com` . Alternatively, use DNS NAPTR to discover what services are available at the root domain.

Customer Phone Configuration

The phones at the customer site are configured as follows:

- Server 1 (the primary server) is configured with the address of the service provider call server. The IP address of the server(s) is provided by the DNS server, for example: `reg.1.server.1.address=voipserver.serviceprovider.com` .
- Server 2 (the fallback server) is configured to the address of the router/gateway that provides the fallback telephony support and is on-site, for example: `reg.1.server.2.address=172.23.0.1` .

Caution: Be careful when using multiple servers per registration. It is possible to configure the phone for more than two servers per registration but ensure that the phone and network load generated by registration refresh of multiple registrations does not become excessive. This is of particular concern when a phone has multiple registrations with multiple servers per registration and some of these servers are unavailable.

For Outgoing Calls (INVITE Fallback)

When the user initiates a call, the phone completes the following steps to connect the call:

1. The phone tries to call the working server.
2. If the working server does not respond correctly to the INVITE, the phone tries and makes a call using the next server in the list (even if there is no current registration with these servers). This could be the case if the Internet connection has gone down, but the registration to the working server has not yet expired.
3. If the second server is also unavailable, the phone tries all possible servers (even those not currently registered) until it either succeeds in making a call or exhausts the list at which point the call fails.

At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used:

- If TCP is used, then the signaling fails if the connection fails or the Send fails.
- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted through all servers in the list and this is the last server, then the signaling fails after the complete UDP timeout defined in RFC 3261. If it is not the last server in the list, the maximum number of retries using the configurable retry timeout is used.

Caution: If DNS is used to resolve the address for Servers, the DNS server is unavailable, and the TTL for the DNS records has expired, the phone attempts to contact the DNS server to resolve the address of all servers in its list before initiating a call. These attempts timeout, but the timeout mechanism can cause long delays (for example, two minutes) before the phone call proceeds using the working server. To prevent this issue, long TTLs should be used. Polycom recommends deploying an on-site DNS server as part of the redundancy solution.

VoIP Server Parameters

The next table describes VoIP server configuration parameters.

VoIP Server Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`voIpProt.server.dhcp.available`	0 (default) - Do not check with the DHCP server for the SIP server IP address. 1 - Check with the server for the IP address.	Yes
`site.cfg`	`voIpProt.server.dhcp.option`	The option to request from the DHCP server if `voIpProt.server.dhcp.available` = 1. 128 (default) to 254 If `reg.x.server.y.address` is non-Null, it takes precedence even if the DHCP server is available.	Yes
`site.cfg`	`voIpProt.server.dhcp.type`	Type to request from the DHCP server if `voIpProt.server.dhcp.available` is set to 1. 0 (default) - Request IP address 1 - Request string	Yes
`site.cfg`	`voIpProt.OBP.dhcpv4.type`	Define the type of Outbound Proxy address. 0 (default) - IP address 1 - String	Yes
`site.cfg`	`voIpProt.OBP.dhcpv4.option`	Phone requests for DHCP option 120 and applies the outbound proxy obtained in DHCP to `voIpProt.SIP.outboundProxy.address` 120 (default)	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voIpProt.OBP.dhcpv6.option	Define the type of Outbound Proxy address from DHCPv6. 21 (default) - list of domain name 22 - list of IP address	Yes

Phone Operation for Registration

After the phone has booted up, it registers to all configured servers.

Server 1 is the primary server and supports greater SIP functionality than other servers. For example, SUBSCRIBE/NOTIFY services used for features such as shared lines, presence, and BLF is established only with Server 1.

Upon the registration timer expiry of each server registration, the phone attempts to re-register. If this is unsuccessful, normal SIP re-registration behavior (typically at intervals of 30 to 60 seconds) proceeds and continues until the registration is successful (for example, when the Internet link is again operational). While the primary server registration is unavailable, the next highest priority server in the list serves as the working server. As soon as the primary server registration succeeds, it returns to being the working server.

Note: If `reg.x.server.y.register` is set to 0, the phone does not register to that server. However, the INVITE fails over to that server if all higher priority servers are down.

Recommended Practices for Fallback Deployments

In situations where server redundancy for fallback purpose is used, the following measures should be taken to optimize the solution:

- Deploy an on-site DNS server to avoid long call initiation delays that can result if the DNS server records expire.
- Do not use `OutBoundProxy` configurations on the phone if the `OutBoundProxy` could be unreachable when the fallback occurs.
- Avoid using too many servers as part of the redundancy configuration as each registration generates more traffic.
- Educate users as to the features that are not available when in fallback operating mode.

Note: The concurrent/registration failover/fallback feature is not compatible with Microsoft environments.

Static DNS Cache

Failover redundancy can be used only when the configured IP server hostname resolves (through SRV or A record) to multiple IP addresses.

Unfortunately, the DNS cache cannot always be configured to take advantage of failover redundancy.

You can statically configure a set of DNS NAPTR SRV and/or A records into the phone. You can enter a maximum of 12 record entries for DNS-A, DNS-NAPTR, and DNS-SRV. records.

Support for negative DNS caching as described in RFC 2308 is also provided to allow faster failover when prior DNS queries have returned no results from the DNS server. For more information, see RFC2308.

Configuring Static DNS

Phones configured with a DNS server behave as follows:

1. The phone makes an initial attempt to resolve a hostname that is within the static DNS cache. For example, a query is made to the DNS if the phone registers with its SIP registrar.
2. If the initial DNS query returns no results for the hostname or cannot be contacted, then the values in the static cache are used for their configured time interval.
3. After the configured time interval has elapsed, a resolution attempt of the hostname again results in a query to the DNS.
4. If a DNS query for a hostname that is in the static cache returns a result, the values from the DNS are used and the statically cached values are ignored.

If a phone is not configured with a DNS server, when the phone attempts to resolve a hostname within the static DNS cache, it always returns the results from the static cache.

Static DNS Parameters

Use the following table to configure static DNS settings.

Static DNS Cache Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-basic.cfg`	`reg.x.address`	The user part (for example, 1002) or the user and the host part (for example, `1002@polycom.com`) of the registration SIP URI or the H.323 ID/ extension. Null (default) string address	No
`sip-interop.cfg`	`reg.x.server.y`	Specify the call server used for this registration.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.server.y.specialInterop	Specify the server-specific feature set for the line registration. VVX 101: Standard (default), GENBAND, ALU-CTS, DT VVX 201: Standard (default), GENBAND, ALU-CTS, ocs2007r2, lync2010 All other phones: Standard (default), GENBAND, ALU-CTS, ocs2007r2, lync2010, lcs2005	
site.cfg	reg.x.server.y.address	If this parameter is set, it takes precedence even if the DHCP server is available. Null (default) - SIP server does not accepts registrations. IP address or hostname - SIP server that accepts registrations. If not Null, all of the parameters in this table override the parameters specified in voIpProt.server.*	No
reg-advanced	reg.x.server.y.expires	The phone's requested registration period in seconds. The period negotiated with the server may be different. The phone attempts to re-register at the beginning of the overlap period. 3600 - (default) positive integer, minimum 10	No
reg-advanced	reg.x.server.y.expires.lineSeize	Requested line-seize subscription period. 30 - (default) 0 to 65535	No
reg-advanced	reg.x.server.y.expires.overlap	The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone tries to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value. 60 (default) 5 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.failOver.failBack.mode	duration (default) - The phone tries the primary server again after the time specified by reg.x.server.y.failOver.failBack.timeout . newRequests - All new requests are forwarded first to the primary server regardless of the last used server. DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to. registration - The phone tries the primary server again when the registration renewal signaling begins. This parameter overrides voIpProt.server.x.failOver.failBack.mode	No
site.cfg	reg.x.server.y.failOver.failBack.timeout	3600 (default) - The time to wait (in seconds) before failback occurs. 0 - The phone does not fail back until a failover event occurs with the current server. 60 to 65535 - If set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again.	No
site.cfg	reg.x.server.y.failOver.failRegistrationOn	1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over. 0 - The reRegisterOn parameter is disabled, existing registrations remain active.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.failOver.onlySignalWithRegistered	1 (default) - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server. 0 - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).	No
site.cfg	reg.x.server.y.failOver.reRegisterOn	0 (default) - The phone does not attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information. 1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server. This parameter overrides voIpProt.server.x.failOver.reRegisterOn .	No
site.cfg	reg.x.server.y.port	Null (default) - The port of the SIP server does not specify registrations. 0 - The port used depends on reg.x.server.y.transport . 1 to 65535 - The port of the SIP server that specifies registrations.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.register	1 (default) - Calls can not be routed to an outbound proxy without registration. 0 - Calls can be routed to an outbound proxy without registration. See voIpProt.server.x.register for more information, see *SIP Server Fallback Enhancements on Polycom Phones - Technical Bulletin 5844* on Polycom Engineering Advisories and Technical Notifications.	No
sip-interop.cfg	reg.x.server.y.registerRetry.baseTimeOut	For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server.Used in conjunction with reg.x.server.y.registerRetry.maxTimeOut to determine how long to wait. 60 (default) 10 - 120 seconds	No
sip-interop.cfg	reg.x.server.y.registerRetry.maxTimeout	For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Use in conjunction with r eg.x.server.y.registerRetry.baseTimeOut to determine how long to wait. The algorithm is defined in RFC 5626. 180 - (default) 60 - 1800 seconds	No
reg-advanced.cfg	reg.x.server.y.retryMaxCount	The number of retries attempted before moving to the next available server. 3 - (default) 0 to 20 - 3 is used when the value is set to 0.	No
reg-advanced.cfg	reg.x.server.y.retryTimeOut	0 (default) - Use standard RFC 3261 signaling retry behavior. 0 to 65535 - The amount of time (in milliseconds) to wait between retries.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.server.y.subscribe.expires	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 seconds - (default) 10 - 2147483647 (seconds) You can use this parameter in conjunction with reg.x.server.y.subscribe.expires.overlap .	No
reg-advanced.cfg	reg.x.server.y.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 seconds (default) 5 - 65535 seconds	No
site.cfg	reg.x.server.y.transport	The transport method the phone uses to communicate with the SIP server. DNSnaptr (default) - If reg.x.server.y.address is a hostname and reg.x.server.y.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If reg.x.server.y.address is an IP address, or a port is given, then UDP is used. TCPpreferred - TCP is the preferred transport; UDP is used if TCP fails. UDPOnly - Only UDP is used. TLS - If TLS fails, transport fails. Leave port field empty (defaults to 5061) or set to 5061. TCPOnly - Only TCP is used.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	reg.x.server.y.useOutboundProxy	1 (default) - Enables to use the outbound proxy specified in reg.x.outboundProxy.address for server x. 0 - Disable to use the outbound proxy specified in reg.x.outboundProxy.address for server x.	No
site.cfg	divert.x.sharedDisabled	1 (default) - Disables call diversion features on shared lines. 0 - Enables call diversion features on shared lines.	Yes
site.cfg	dns.cache.A.x.	Specify the DNS A address, hostname, and cache time interval.	
site.cfg	dns.cache.A.x.address	Null (default) IP version 4 address	No
site.cfg	dns.cache.A.x.name	Null (default) valid hostname	No
site.cfg	dns.cache.A.x.ttl	The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and it resets TTL timer again. 300 (default) 300 to 536870912 (2^29), seconds	No
site.cfg	dns.cache.NAPTR.x.	Specify the DNS NAPTR parameters, including: name, order, preference, regexp, replacement, service, and ttl.	
site.cfg	dns.cache.NAPTR.x.flags	The flags to control aspects of the rewriting and interpretation of the fields in the record. Characters are case-sensitive. At this time, only ‘S', ‘A', ‘U', and ‘P' are defined as flags. See RFC 2915 for details of the permitted flags. Null (default) A single character from [A-Z, 0-9]	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dns.cache.NAPTR.x.name	Null (default) domain name string - The domain name to which this resource record refers.	No
site.cfg	dns.cache.NAPTR.x.order	0 (default) 0 to 65535 - An integer that specifies the order in which the NAPTR records must be processed to ensure the correct ordering of rules.	No
site.cfg	dns.cache.NAPTR.x.preference	0 (default) 0 to 65535 - A 16-bit unsigned integer that specifies the order in which NAPTR records with equal "order" values should be processed. Low numbers are processed before high numbers.	No
site.cfg	dns.cache.NAPTR.x.regexp	This parameter is currently unused. Applied to the original string held by the client. The substitution expression is applied in order to construct the next domain name to look up. The grammar of the substitution expression is given in RFC 2915. Null (default)string containing a substitution expression	No
site.cfg	dns.cache.NAPTR.x.replacement	The next name to query for NAPTR records depending on the value of the flags field. It must be a fully qualified domain-name. Null (default) domain name string with SRV prefix	No
site.cfg	dns.cache.NAPTR.x.service	Specifies the service(s) available down this rewrite path. For more information, see RFC 2915. Null (default) string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dns.cache.NAPTR.x.ttl	The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and it resets TTL timer again.300 (default) 300 to 536870912 (2^29), seconds	No
site.cfg	dns.cache.A.networkOverride	0 (default) - Does not allow the static DNS A record entry to take priority over dynamic network DNS. 1 – Allows the static DNS cached A record entry to take priority over dynamic network DNS. Moreover, the DNS TTL value is ignored.	No
site.cfg	dns.cache.SRV.x.	Specify DNS SRV parameters, including: name, port, priority, target, ttl, and weight.	
site.cfg	dns.cache.SRV.x.name	Null (default) Domain name string with SRV prefix	No
site.cfg	dns.cache.SRV.x.port	The port on this target host of this service. For more information, see RFC 2782. 0 (default) 0 to 65535	No
site.cfg	dns.cache.SRV.x.priority	The priority of this target host. For more information, see RFC 2782. 0 (default) 0 to 65535	No
site.cfg	dns.cache.SRV.x.target	Null (default) domain name string - The domain name of the target host. For more information, see RFC 2782.	No

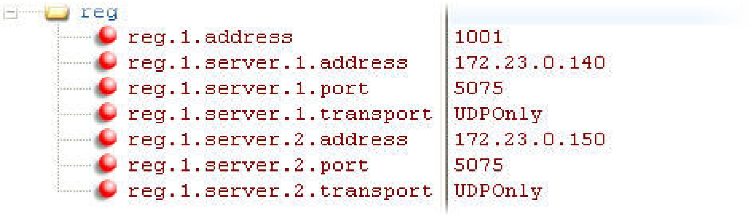
Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dns.cache.SRV.x.ttl	The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and it resets TTL timer again. 300 (default) 300 to 536870912 (2^29), seconds	No
site.cfg	dns.cache.SRV.x.weight	A server selection mechanism. For more information, see RFC 2782. 0 (default) 0 to 65535	No
site.cfg	tcpIpApp.dns.address.overrideDHCP	Specifies how DNS addresses are set. 0 (default) - DNS address requested from the DHCP server. 1 - DNS primary and secondary address is set using the parameters tcpIpApp.dns.server and tcpIpApp.dns.altServer .	Yes
site.cfg	tcpIpApp.dns.domain.overrideDHCP	Specifies how the domain name is retrieved or set. 0 (default) - Domain name retrieved from the DHCP server, if one is available. 1 - DNS domain name is set using the parameter tcpIpApp.dns.domain .	Yes

Example Static DNS Cache Configuration

The following example shows how to configure static DNS cache using A records IP addresses in SIP server address fields.

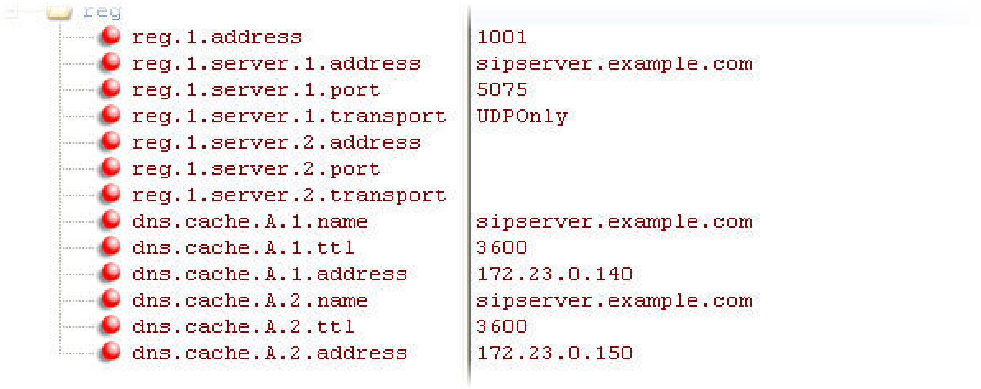
The addresses listed in this example are read by Polycom UC Software in the order listed.

When the static DNS cache is not used, the site.cfg configuration looks as follows:



```
reg
  reg.1.address                 1001
  reg.1.server.1.address        172.23.0.140
  reg.1.server.1.port           5075
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address        172.23.0.150
  reg.1.server.2.port           5075
  reg.1.server.2.transport      UDPOnly
```

When the static DNS cache is used, the `site.cfg` configuration looks as follows:

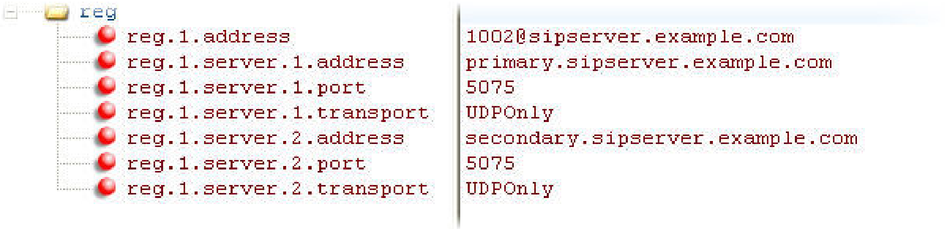


```
reg
  reg.1.address                 1001
  reg.1.server.1.address        sipserver.example.com
  reg.1.server.1.port           5075
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address
  reg.1.server.2.port
  reg.1.server.2.transport
  dns.cache.A.1.name            sipserver.example.com
  dns.cache.A.1.ttl             3600
  dns.cache.A.1.address         172.23.0.140
  dns.cache.A.2.name            sipserver.example.com
  dns.cache.A.2.ttl             3600
  dns.cache.A.2.address         172.23.0.150
```

Example: Static DNS Cache with A Records

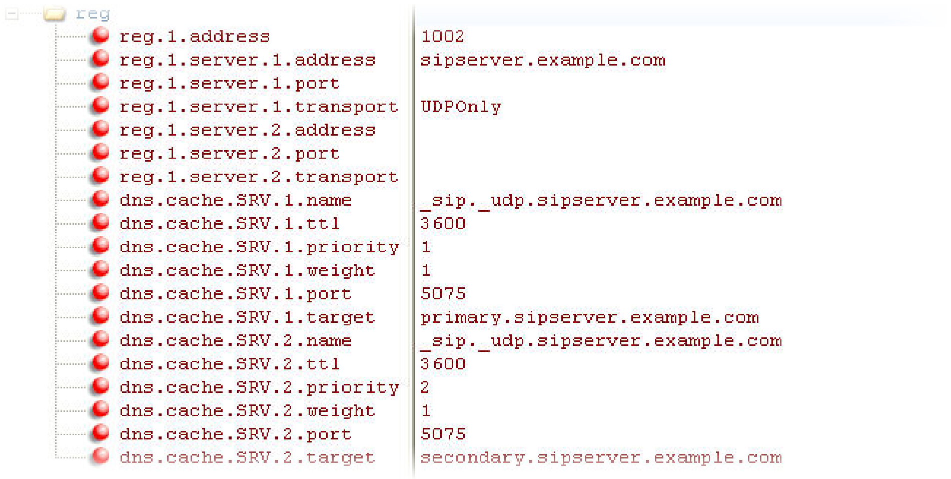
This example shows how to configure static DNS cache where your DNS provides A records for `reg.x.server.x.address` but not SRV. In this case, the static DNS cache on the phone provides SRV records. For more information, see RFC 3263.

When the static DNS cache is not used, the `site.cfg` configuration looks as follows:



```
reg
  reg.1.address                 1002@sipserver.example.com
  reg.1.server.1.address        primary.sipserver.example.com
  reg.1.server.1.port           5075
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address        secondary.sipserver.example.com
  reg.1.server.2.port           5075
  reg.1.server.2.transport      UDPOnly
```

When the static DNS cache is used, the `site.cfg` configuration looks as follows:



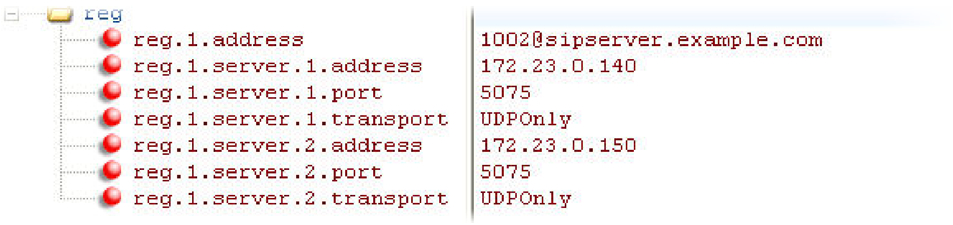
```
reg
  reg.1.address                 1002
  reg.1.server.1.address        sipserver.example.com
  reg.1.server.1.port
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address
  reg.1.server.2.port
  reg.1.server.2.transport
  dns.cache.SRV.1.name          _sip._udp.sipserver.example.com
  dns.cache.SRV.1.ttl           3600
  dns.cache.SRV.1.priority      1
  dns.cache.SRV.1.weight        1
  dns.cache.SRV.1.port          5075
  dns.cache.SRV.1.target        primary.sipserver.example.com
  dns.cache.SRV.2.name          _sip._udp.sipserver.example.com
  dns.cache.SRV.2.ttl           3600
  dns.cache.SRV.2.priority      2
  dns.cache.SRV.2.weight        1
  dns.cache.SRV.2.port          5075
  dns.cache.SRV.2.target        secondary.sipserver.example.com
```

Note: The `reg.1.server.1.port` and `reg.1.server.2.port` values in this example are set to null to force SRV lookups.

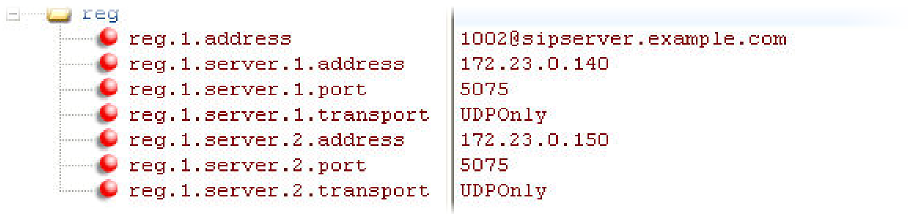
Example: Static DNS Cache with NAPTR and SRV Records

This example shows how to configure static DNS cache where your DNS provides NAPTR and SRV records for `reg.x.server.x.address` .

When the static DNS cache is not used, the `site.cfg` configuration looks as follows:

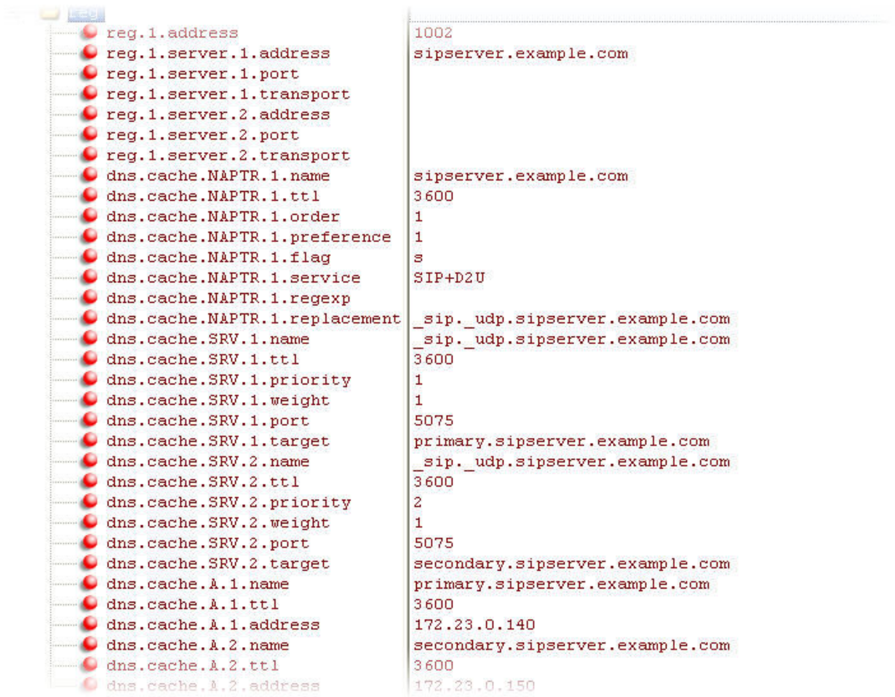


```
reg
  reg.1.address                 1002@sipserver.example.com
  reg.1.server.1.address        172.23.0.140
  reg.1.server.1.port           5075
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address        172.23.0.150
  reg.1.server.2.port           5075
  reg.1.server.2.transport      UDPOnly
```



```
reg
  reg.1.address                 1002@sipserver.example.com
  reg.1.server.1.address        172.23.0.140
  reg.1.server.1.port           5075
  reg.1.server.1.transport      UDPOnly
  reg.1.server.2.address        172.23.0.150
  reg.1.server.2.port           5075
  reg.1.server.2.transport      UDPOnly
```

When the static DNS cache is used, the `site.cfg` configuration looks as follows:



reg

reg.1.address	1002
reg.1.server.1.address	sipserver.example.com
reg.1.server.1.port	
reg.1.server.1.transport	
reg.1.server.2.address	
reg.1.server.2.port	
reg.1.server.2.transport	
dns.cache.NAPTR.1.name	sipserver.example.com
dns.cache.NAPTR.1.ttl	3600
dns.cache.NAPTR.1.order	1
dns.cache.NAPTR.1.preference	1
dns.cache.NAPTR.1.flag	s
dns.cache.NAPTR.1.service	SIP+D2U
dns.cache.NAPTR.1.regexp	
dns.cache.NAPTR.1.replacement	_sip._udp.sipserver.example.com
dns.cache.SRV.1.name	_sip._udp.sipserver.example.com
dns.cache.SRV.1.ttl	3600
dns.cache.SRV.1.priority	1
dns.cache.SRV.1.weight	1
dns.cache.SRV.1.port	5075
dns.cache.SRV.1.target	primary.sipserver.example.com
dns.cache.SRV.2.name	_sip._udp.sipserver.example.com
dns.cache.SRV.2.ttl	3600
dns.cache.SRV.2.priority	2
dns.cache.SRV.2.weight	1
dns.cache.SRV.2.port	5075
dns.cache.SRV.2.target	secondary.sipserver.example.com
dns.cache.A.1.name	primary.sipserver.example.com
dns.cache.A.1.ttl	3600
dns.cache.A.1.address	172.23.0.140
dns.cache.A.2.name	secondary.sipserver.example.com
dns.cache.A.2.ttl	3600
dns.cache.A.2.address	172.23.0.150

Note: The `reg.1.server.1.port` , `reg.1.server.2.port` , `reg.1.server.1.transport` , and `reg.1.server.2.transport` values in this example are set to null to force NAPTR lookups.

IP Type-of-Service

The type-of-service field in an IP packet header consists of four type-of-service (TOS) bits and a 3-bit precedence field.

Each TOS bit can be set to either 0 or 1. The precedence field can be set to a value from 0 through 7. The type of service can be configured specifically for RTP packets and call control packets, such as SIP signaling packets.

IP Type-of-Service Parameters

You can configure the IP TOS feature specifically for RTP and call control packets, such as SIP signaling packets.

Type of Service (ToS) and the Differentiated Services Code Point (DSCP) allows specification of a datagrams desired priority and routing through low-delay, high-throughput, or highly-reliable networks.

The IP ToS header consists of four ToS bits and a 3-bit precedence field.

DSCP replaces the older ToS specification and uses a 6-bit DSCP in the 8-bit differentiated services field (DS field) in the IP header.

The parameters listed in the table configure the following Quality of Service (QoS) options:

- The 802.1p/Q user_priority field RTP, call control, and other packets

- The “type of service” field RTP and call control packets

IP Type of Service Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`qos.ethernet.callControl.user_priority`	Set the user-priority for call control packets. 5 (default) 0 - 7	Yes
`site.cfg`	`qos.ethernet.other.user_priority`	Set the user-priority for packets that do not have a per-protocol setting. 2 (default) 0 - 7	Yes
`site.cfg`	`qos.ethernet.rtp.user_priority`	Set the priority of voice Real-Time Protocol (RTP) packets. 5 (default) 0 - 7	Yes
`site.cfg`	`qos.ethernet.rtp.video.user_priority`	User-priority used for Video RTP packets. 5 (default) 0 - 7	Yes
`site.cfg`	`qos.ethernet.tcpQosEnabled`	0 (default) - The phone does not send configured QoS priorities for SIP over TCP transport. 1 - The phone sends configured QoS priorities for SIP over TCP transport.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	qos.ip.callControl.dscp	Specify the DSCP of packets. If the value is set to the default NULL the phone uses qos.ip.callControl.* parameters. If the value is not NULL, this parameter overrides qos.ip.callControl.* parameters. ◦ NULL (default) ◦ 0 to 63 ◦ EF ◦ Any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Yes
site.cfg	qos.ip.callControl.max_reliability	Set the max reliability bit in the IP ToS field of the IP header used for call control. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.callControl.max_throughput	Set the throughput bit in the IP ToS field of the IP header used for call control. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.callControl.min_cost	Set the min cost bit in the IP ToS field of the IP header used for call control. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	qos.ip.callControl.min_delay	Set the min delay bit in the IP ToS field of the IP header used for call control. 1 (default) - The bit is set. 0 - The bit in the IP ToS field of the IP header is not set.	Yes
site.cfg	qos.ip.callControl.precedence	Set the min delay bit in the IP ToS field of the IP header used for call control. 5 (default) 0 - 7	Yes
site.cfg	qos.ip.rtp.dscp	Specify the DSCP of packets. If the value is set to the default NULL the phone uses quality.ip.rtp.* parameters. If the value is not NULL, this parameter overrides quality.ip.rtp.* parameters. ◦ Null (default) ◦ 0 to 63 ◦ EF ◦ Any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Yes
site.cfg	qos.ip.rtp.max_reliability	Set the max reliability bit in the IP ToS field of the IP header used for RTP. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.rtp.max_throughput	Set the throughput bit in the IP ToS field of the IP header used for RTP. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	qos.ip.rtp.min_cost	Set the min cost bit in the IP ToS field of the IP header used for RTP. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.rtp.min_delay	Set the min delay bit in the IP ToS field of the IP header used for RTP. 1 (default) - The bit is set. 0 - The bit in the IP ToS field of the IP header is not set.	Yes
site.cfg	qos.ip.rtp.precedence	Set the precedence bit in the IP ToS field of the IP header used for RTP. 5 (default) 0 - 7	Yes
site.cfg	qos.ip.rtp.video.dscp	Allows you to specify the DSCP of packets. If the value is set to the default NULL the phone uses qos.ip.rtp.video.* parameters. If the value is not NULL, this parameter overrides qos.ip.rtp.video.* parameters. ◦ NULL (default) ◦ 0 to 63 ◦ EF ◦ Any of AF11,AF12, AF13,AF21, AF22,AF23, AF31,AF32, AF33,AF41, AF42,AF43	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	qos.ip.rtp.video.max_reliability	Set the reliability bits in the IP ToS field of the IP header used for RTP video. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.rtp.video.max_throughput	Set the throughput bits in the IP ToS field of the IP header used for RTP video. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.rtp.video.min_cost	Set the min cost bits in the IP ToS field of the IP header used for RTP video. 0 (default) - The bit in the IP ToS field of the IP header is not set. 1 - The bit is set.	Yes
site.cfg	qos.ip.rtp.video.min_delay	Set the min delay bits in the IP ToS field of the IP header used for RTP video. 1 (default) - The bit is set. 0 - The bit in the IP ToS field of the IP header is not set.	Yes
site.cfg	qos.ip.rtp.video.precedence	Set the precedence bits in the IP ToS field of the IP header used for RTP video. 5 (default) 0 - 7	Yes

SIP Instance Support

In environments where multiple phones are registered using the same address of record (AOR), the phones are identified by their IP address.

However, firewalls set up in these environments can regularly change the IP addresses of phones for security purposes. You can configure SIP instance to identify individual phones instead of using IP addresses. This feature complies with RFC 3840.

This feature is not available on:

- VVX 150 business IP phone
- VVX 101 and 201 business media phones

SIP Instance Parameters

The parameter `reg.x.gruu` provides a contact address to a specific user agent (UA) instance, which helps to route the request to the UA instance and is required in cases in which the REFER request must be routed to the correct UA instance. Refer to the following table for information on configuring this feature.

			Change Causes
Template	Parameter	Permitted Values	Restart or Reboot
`sip-interop.cfg`	`reg.x.gruu`	1 - The phone sends sip.instance in the REGISTER request. 0 (default) - The phone does not send sip.instance in the REGISTER request.	No

Provisional Polling of Polycom Phones

You can configure phones to poll the server for provisioning updates automatically, and you can set the phone's automatic provisioning behavior to one of the following:

- **Absolute**—The phone polls at the same time every day.
- **Relative**—The phone polls every x seconds, where x is a number greater than 3600.
- **Random**—The phone polls randomly based on a set time interval.
    - If the time period is less than or equal to one day, the first poll is at a random time between when the phone starts up and the polling period. Afterwards, the phone polls every x seconds.
    - If you set the polling period to be greater than one day with the period rounded up to the nearest day, the phone polls on a random day based on the phone's MAC address and within a random time set by the start and end polling time.

Provisional Polling Parameters

Use the parameters in the following table to configure provisional polling.

Note that If `prov.startupCheck.enabled` is set to 0, then Polycom phones do not look for the sip.ld or the configuration files when they reboot, lose power, or restart. Instead, they look only when receiving a checksync message, a polling trigger, or a manually started update from the menu or web UI.

Some files such as bitmaps, .wav, the local directory, and any custom ringtones are downloaded each time as they are stored in RAM and lost with every reboot.

Provisional Polling of Polycom Phones

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`prov.polling`	To enable polling and set the mode, period, time, and time end parameters.	
`site.cfg`	`prov.polling.enabled`	0 (default) - Disables the automatic polling for upgrades. 1 - Initiates the automatic polling for upgrades.	No
`site.cfg`	`prov.polling.mode`	The polling modes for the provisioning server. `abs` (default) – The phone polls every day at the time specified by `prov.polling.time` . `rel` – The phone polls after the number of seconds specified by `prov.polling.period` . `random` – The phone polls at random between a starting time set in `prov.polling.time` and an end time set in `prov.polling.timeRandomEnd` . If you set the polling period in `prov.polling.period` to a time greater than 86400 seconds (one day) polling occurs on a random day within that polling period and only between the start and end times. The day within the period is decided based upon the phones MAC address and does not change with a reboot whereas the time within the start and end is calculated again with every reboot	No
`site.cfg`	`prov.polling.period`	The polling period is calculated in seconds and is rounded up to the nearest number of days in an absolute and random mode. If this is set to a time greater than 86400 (one day) polling occurs on a random day based on the phone's MAC address. 86400 (default) - Number of seconds in a day. Integer - An integer value greater than 3600 seconds.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.polling.time	The start time for polling on the provisioning server. 03:00 (default) hh:mm	No
site.cfg	prov.polling.timeRandomEnd	The stop time for polling on the provisioning server. Null (default) hh:mm	No

Example Provisional Polling Configuration

The following are examples of polling configurations you can set up:

- If prov.polling.mode is set to rel and prov.polling.period is set to *7200*, the phone polls every two hours.
- If prov.polling.mode is set to abs and prov.polling.timeRandomEnd is set to *04:00*, the phone polls at 4am every day.
- If prov.polling.mode is set to random, prov.polling.period is set to *604800 (7 days)*, prov.polling.time is set to 01:00 , prov.polling.timeRandomEnd is set to *05:00*, and you have 25 phones, a random subset of those 25 phones, as determined by the MAC address, polls randomly between 1am and 5am every day.
- If prov.polling.mode is set to *abs* and prov.polling.period is set to *2328000*, the phone polls every 20 days.

SIP Subscription Timers

You can configure a subscription expiry independently of the registration expiry.

You can also configure an overlap period for a subscription independently of the overlap period for the registration, and a subscription expiry and subscription overlap for global SIP servers and per-registration SIP servers. Note that per-registration configuration parameters override global parameters. If you have not explicitly configured values for any user features, the default subscription values are used.

SIP Subscription Timers Parameters

Use the parameters in the following table to configure when a SIP subscription expires and when expirations overlap.

SIP Subscription Timers

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.server.x.subscribe.expires	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 - (default) 10 - 2147483647	No
sip-interop.cfg	voIpProt.server.x.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 - (default) 5 - 65535 seconds	No
reg-advanced.cfg	reg.x.server.y.subscribe.expires	The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period. 3600 seconds - (default) 10 - 2147483647 (seconds) You can use this parameter in conjunction with reg.x.server.y.subscribe.expires.overlap .	No
reg-advanced.cfg	reg.x.server.y.subscribe.expires.overlap	The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server. 60 seconds (default) 5 - 65535 seconds	No

Incoming Network Signaling Validation

You can choose from the following optional levels of security for validating incoming network signaling:

- Source IP address validation
- Digest authentication
- Source IP address validation and digest authentication

Network Signaling Validation Parameters

The following table includes the parameters you can use to specify the validation type, method, and the events for validating incoming network signaling.

Network Signaling Validation Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`voIpProt.SIP.requestValidation.x.method`	Null (default) - no validation is made. Source - ensure request is received from an IP address of a server belonging to the set of target registration servers. digest: challenge requests with digest authentication using the local credentials for the associated registration (line). both or all: apply both of the above methods.	Yes
`sip-interop.cfg`	`voIpProt.SIP.requestValidation.x.request`	Sets the name of the method for which validation will be applied. Null (default) INVITE, ACK , BYE, REGISTER, CANCEL, OPTIONS, INFO, MESSAGE, SUBSCRIBE, NOTIFY, REFER, PRACK, UPDATE Note: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.requestValidation.x.request.y.event	Determines which events specified with the Event header should be validated; only applicable when voIpProt.SIP.requestValidation.x.request is set to SUBSCRIBE or NOTIFY . Null (default) - all events will be validated. A valid string - specified event will be validated.	Yes

System and Model Names

The following table outlines the system and model names that Polycom phones transmit with network protocols.

If you need to customize your network for a specific phone model, you can parse the network packets for these strings.

Polycom VVX System and Model Names

Model	System Name	Model Name
VVX 101	Polycom VVX 101	VVX-VVX_101
VVX 150	Polycom VVX 150	VVX-VVX_150
VVX 201	Polycom VVX 201	VVX-VVX_201
VVX 250	Polycom VVX 250	VVX-VVX_250
VVX 300	Polycom VVX 300	VVX-VVX_300
VVX 301	Polycom VVX 301	VVX-VVX_301
VVX 310	Polycom VVX 310	VVX-VVX_310
VVX 311	Polycom VVX 311	VVX-VVX_311
VVX 350	Polycom VVX 350	VVX-VVX_350
VVX 400	Polycom VVX 400	VVX-VVX_400
VVX 401	Polycom VVX 401	VVX-VVX_401
VVX 410	Polycom VVX 410	VVX-VVX_410
VVX 411	Polycom VVX 411	VVX-VVX_411

Model	System Name	Model Name
VVX 450	Polycom VVX 450	VVX-VVX_450
VVX 500	Polycom VVX 500	VVX-VVX_500
VVX 501	Polycom VVX 501	VVX-VVX_501
VVX 600	Polycom VVX 600	VVX-VVX_600
VVX 601	Polycom VVX 601	VVX-VVX_601
VVX 1500	Polycom VVX 1500	VVX-VVX_1500
SoundStructure	SoundStructure VoIP Interface	SoundStructure VoIP Interface

Related Links

Defining the Phone Key Layout on page 487

Configuring Wireless Network Settings on VVX Phones

Polycom UC Software supports wireless network connectivity using the Polycom® Wi-Fi wireless network adapter with VVX phones that support USB.

The user plugs in the adapter and enable the wireless network. You can manually configure a VVX phone to connect to a wireless network by selecting an enterprise network for better security. This option is useful when a wireless network doesn't broadcast its SSID.

Configure a Wireless Network

You must configure an enterprise-based network by selecting EAP method. VVX phones support EAP-PEAP/MSCHApv2, EAP-FAST, and EAP-TLS methods for an enterprise-based security profile.

Procedure

1. Go to **Settings** > **Advance** > **Administrator Settings** > **Network Configuration** > **Wi-Fi**.

   Ensure **Enabled** is set to **Yes**.
2. Enter the **IP Configuration**, if not set via DHCP.
3. Enter the wireless network's SSID.
4. Select a WPA2-Enterprise network security and configure the following settings:
   a. Select **EAP-Method** as the authentication type.
   b. Enter the **User ID** and **Password**.

Wireless Network Parameters

The following table lists the parameters to configure your wireless network.

Wireless Network Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, wireless.cfg	device.wifi.enabled	0 (default) - Disable the wireless interface. 1 - Enable the wireless interface.	Yes
device.cfg, wireless.cfg	device.wifi.dhcpEnabled	0 (default) - Disable DHCP on the wireless interface. 1 - Enable DHCP on the wireless interface.	Yes
device.cfg, debug.cfg	device.wifi.ipAddress	Set the network address of the wireless device if not using DHCP. 0 to 255 characters.	No
device.cfg, site.cfg	device.wifi.subnetMask	Set the network mask address of the wireless device if not using DHCP. 0 to 255 characters.	No
device.cfg, site.cfg	device.wifi.ipGateway	Set the IP gateway address for the wireless device if not using DHCP. 0 to 255 characters.	No
device.cfg, wireless.cfg	device.wifi.ssid	Set the SSID of the wireless network. 0 to 32 characters.	No
device.cfg, wireless.cfg	device.wifi.securityMode	Set the wireless security mode. Open WPA-PSK WPA(2)-PSK WPA2-Enterprise	No
device.cfg, wireless.cfg	device.wifi.psk.keyType	Set the Pre-Shared Key (PSK) type 0 (default) - Passphrase. 1 - Hexadecimal key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, wireless.cfg	device.wifi.psk.key	Set the Pre-Shared Key. 0 to 128 characters.	No
device.cfg, wireless.cfg	device.wifi.wpa2Ent.method	Set the EAP type used for 802.1x authentication. PEAPv0/MSCHAPv2 (default) TLS FAST The security profile for PEAPv0/MSCHAPv2, TLS is device.sec.TLS.profileSelection.dot1x	No
device.cfg, wireless.cfg	device.wifi.wpa2Ent.user	Set the WPA2-Enterprise Security user name. 0 to 128 characters.	No
device.cfg, wireless.cfg	device.wifi.wpa2Ent.password	Set the WPA2-Enterprise Security password. 0 to 128 characters.	No
device.cfg, wireless.cfg	device.wifi.wpa2Ent.anonid	EAP-FAST only. Set the anonymous identity (user name) for 802.1x authentication. 0 to 128 characters.	No
device.cfg, wireless.cfg	device.wifi.wpa2Ent.eapFast.inBandProv	0 (default) - Disable in-band provisioning. 1 - Enable in-band provisioning.	No
device.cfg, wireless.cfg	device.wifi.radio.regulatoryDomain	Set the regulatory domain to appropriate value for your location. 0 (default) - Global. 1 - US 2 - EU	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg, wireless.cfg	device.wifi.radio.band2_4GHz.enable	1 (default) - Enable the 2.4 GHz wireless band. 0 - Disable the 2.4 GHz wireless band.	No
device.cfg, wireless.cfg	device.wifi.radio.band5GHz.enable	0 (default) - Disable the 5 GHz wireless band. 1 - Enable the 5 GHz wireless band.	No

Third-Party Servers

Topics:

This section provides information on configuring phones and features with third-party servers.

Alcatel-Lucent Converged Telephony Server

This section shows you how to configure Polycom phones with Alcatel-Lucent (ALU) Converged Telephony Server (CTS).

Advanced Conferences

When users are signed into the ALU CTS on VVX phones, they can initiate ad-hoc conference calls with two or more contacts.

Users can also create a participant list and manage conference participants. This feature is not supported on:

- VVX 150 business IP phone
- VVX 101 and 201 business media phones

Advance Conference includes the following features:

- **Roster** Provides a list of participants in the conference
- **Conference Controller** The person who creates the conference and can add or drop participants, and mute and unmute participants.
- **Push-to-Conference** Enables users to create a list of participants when initiating a conference call.
- Join two calls into a conference call
- Join a call to an active call to make a conference call

Advanced Conferences Parameters

Use the parameters in the following table to configure this feature.

When you configure the number of participants in a conference using the parameter `reg.x.advancedConference.maxParticipants` , make sure the number of participants you configure matches the number of participants allowed on the ALU CTS.

ALU Advanced Conferences

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.advancedConference.enabled	0 (default) - Disables and does not display advanced conferences and conference controls for ALU advanced conferences. 1 - Enables and displays advanced conferences and conference controls for ALU advanced conferences.	No
reg-advanced.cfg	reg.x.advancedConference.pushToConference	0 (default) - Disable push-to-conference functionality. 1 - Enable push-to-conference functionality.	No
reg-advanced.cfg	reg.x.advancedConference.maxParticipants	Sets the maximum number of participants allowed in a push to conference for advanced conference calls. The number of participants configured must match the number of participants allowed on the ALU CTS. 3 (default) 0 - 25	No
reg-advanced.cfg	reg.x.advancedConference.subscribeForConfEvents	1 (default) - Conference participants to receive notifications for conference events is enabled. 0 - Conference participants to receive notifications for conference events is disabled.	No
reg-advanced.cfg	reg.x.advancedConference.subscribeForConfEventsOnCCPE	1 (default) - Enable the conference host to receive notifications for conference events. 0 - Disable the conference host to receive notifications for conference events.	No

Shared Call Appearance

The Shared Call Appearance feature enables users who share a line to monitor and bridge into calls on the shared line.

Each line supports up to 21 call appearances. This feature is disabled by default. You can enable the feature and configure the hold request for the line. This feature is supported on:

- VVX 300 series, VVX 400 series, VVX 500 series, and VVX 600 series business media phones
- VVX 250, 450, and 450 business IP phones

Note the following when using shared call appearance with ALU CTS:

- Members of the SCA group cannot resume remotely held calls.
- The phones support 21 shared call appearances per line.
- The maximum number of calls associated with a shared call appearance group is the same as the number of calls provisioned for that shared line.
- An incoming call to a shared call appearance group can be presented to the group as long as there is one available idle call appearance.
- All shared call appearances are able to receive and originate calls, regardless of the call activity on the other shared call appearances.
- Users can bridge into an active SCA call that is in shared mode.

Shared Call Appearance Parameters

Use the parameters in the following table to configure this feature.

Shared Call Appearance

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.scap.defCallTypeExclusive`	1 (default) - An outgoing call from the call group is private. After the call is answered, the user must press the **Share** soft key to make the call public so that other people on the line can bridge in.	No
`features.cfg`	`feature.scap.HoldRequestUriUserPart`	Specifies the Hold request for Shared Call Appearance calls to the ALU server. This value must match the value configured on ALU server for SCA hold request. SCAP-Hold (default) string	No

Bridge In for Shared Call Appearance

Bridge In is for Shared Call Appearance lines registered with the ALU CTS.

This feature enables multiple users in a Shared Call Appearance group to view and bridge into active calls on a shared line. By default, group members can bridge into active calls only. Users cannot bridge into held or incoming calls. Multiple people can bridge into one active call.

This feature is not supported on VVX 1500 business media phones.

Bridge In Parameters

Use the parameter in the following table to enable this feature.

Barge In and Bridge In

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	`reg.x.bridgeInEnabled`	0 (default) - Bridge In feature is disabled. 1 - Bridge In feature is enabled.	No

Barge-In for Busy Lamp Field Lines

This feature enables users to barge in on active and held calls on Busy Lamp Field (BLF) lines and supports three barge-in modes: Normal, Whisper and Silent.

The Barge In feature for BLF lines is disabled by default.

This feature is not supported on VVX 1500 business media phones.

Barge In Parameters

Use the parameters in the following table to enable this feature.

Barge In for BLF

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	attendant.resourceList.x.bargeInMode	Enable or disable barge-in and choose the default barge-in mode. This parameter applies to the Alcatel-Lucent CTS only. Null (default) - If no value is entered, the Barge In feature is disabled. All - Press and hold the BLF line to display all barge-in options. Quick press to barge-in as Normal. Normal - Barge-in plays an audio tone to indicate the arrival of a new participant to the call and all call participants can interact. Listen - The user barging in can listen on the call only. Their outbound audio is not transmitted to either party. Whisper - The user barging in can hear all parties but their audio is only transmitted to the user they are monitoring.	No
features.cfg	attendant.resourceList.x.requestSilentBargeIn	0 (default) - A tone plays when a contact barges in on a call. 1 - No tone is played when a contact barges in on a call.	No

Dual Tone Multi Frequency (DTMF) Relay

This feature enables users to press DTMF commands during active SIP audio calls and conference calls to perform actions.

This feature is not supported for H.323 calls.

DTMF Relay Parameters

Use the parameters in the following table to configure this feature.

Configure DTMF Relay

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SIP.dtmfViaSignaling.rfc2976	Enable or disable DTMF relays for active SIP calls. Not supported for H. 323 calls. 0 (default) - DTMF digit information is not sent 1 - DTMF digit information is sent in RFC2976 SIP INFO packets during a call.	Yes
sip-interop.cfg	voIpProt.SIP.dtmfViaSignaling.rfc2976.nonLegacyEncoding	Controls the behavior of the Star and Pound keys used for DTMF relays for active SIP calls. Not supported for H. 323 calls. 0 (default) - The phone sends 10 when the Star key (*) is pressed and 11 when the Pound key (#) is pressed. 1 - The phone sends an asterisk (*) when the Star key is pressed and a hashtag (#) when the Pound key is pressed.	Yes

Visitor Desk Phone

Visitor desk phone (VDP) enables users registered with the ALU CTS to access personal settings on a shared phone after logging in.

After the user logs in, the user profile configuration file is downloaded to the phone, and the user can access any enabled services, such as message-waiting indicator, busy lamp field, or shared call appearance. VDP is not supported on VVX 1500 business media phones.

If a user logs into a second phone when already logged into a first phone, the user is automatically logged out of the first phone. When logged in or out, users can dial an access code to play a message indicating if that user is logged in to a phone and the remaining time in a session.

On the server, you can configure the duration of a login period after which the user must re-enter credentials to the phone. When the time is nearing expiration, the server calls the phone and plays a message indicating the remaining time and prompts the user to re-enter credentials to extend the session.

You can configure a common setting for all phones and any user can make calls, including emergency calls, from a phone without having to log in. After the user logs in to the shared phone, personal settings are available as a user profile in <user> phones.cfg and any changes the user makes to phone settings are stored to this file.

The file <user>-directory.xml contains the user's contact list; the phone displays directory updates to the user at each login. Calls a user makes when logged into a phone are stored in call logs <user>-calls.xml. Calls a user makes when not logged in are not stored.

Visitor Desk Phone Parameters

Use the parameters in the following table to configure this feature.

Visitor Desk Phone

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.VDP.enabled`	0 (default) - Disable VDP and the phone does not display the Visitor Login soft key. 1 - Enable VDP and the phone displays the Visitor Login soft key.	Yes
`features.cfg`	`prov.vdp.accessCode.login`	Specifies the VDP login service access code. *771 (default) string	No
`features.cfg`	`prov.vdp.accessCode.logout`	Specifies the VDP logout service access code. *772 (default) string	No

GENBAND Server

GENBAND's application server, also called EXPERiUS™ A2, provides full-featured, IP-based multimedia communications applications for business and consumers.

You can deploy EXPERiUS A2 as a standalone server or in combination with a GENBAND CONTiNUUM™ C20 server; features vary depending on your deployment.

Polycom has performed interoperability tests with GENBAND C20 with Polycom VVX 300 series, 400 series, 500 series, 600 series, and 1500 phones.

The following features are available for phones registered with the GENBAND servers:

- MADN-SCA—A shared group feature that provides support for conference barge in, privacy, and remote call appearance. MADN-SCA requires you to deploy EXPERiUS A2 and CONTiNUUM C20 server.
- Global Address Book—The global address book (GAB) feature is a corporate directory application managed by the GENBAND server.
- Personal Address Book—The personal address book (PAB) feature is managed by the GENBAND server and allows multiple clients (phones, computer software) to read and modify a user's personal directory of contacts. When one client changes a contact all other clients are immediately notified of the change by the GENBAND server.

- E.911—Enhanced 911 services specific to GENBAND C20 server implementation.

Multiple Appearance Directory Number - Single Call Appearance (MADN-SCA)

Multiple appearance directory number—single call appearance (MADN-SCA) enables a group of users to share a single directory number that displays as a single line to each member of the group.

When this feature is enabled, users can initiate or receive calls on this shared line. MADN-SCA requires you to deploy EXPERiUS A2 and CONTiNUUM C20 server.

Only one call can be active on the line at a time on the MADN-SCA shared line. When a call is in progress, any incoming calls to the line receive a busy tone.

MADN-SCA Parameters

The following table lists all parameters available for configuring MADN-SCA and feature options.

Note: If you configure the line-specific parameter `reg.x.server.y.address` , you must also configure values in the line-specific parameter `reg.x.server.y.specialInterop` .

If you configure the global parameter `voIpProt.server.x.address` , you must also configure values in the global parameter `voIpProt.server.x.specialInterop` .

For all deployments, including GENBAND, line-specific configuration parameters override global configuration parameters. If you set values in both line-specific and global parameters, line-specific parameters are applied and global parameters are not applied.

MADN-SCA Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-basic.cfg	reg.x.address	The user part (for example, 1002) or the user and the host part (for example, 1002@polycom.com) of the registration SIP URI or the H.323 ID/ extension. Null (default) string address	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.server.y.specialInterop	Specify the server-specific feature set for the line registration. Standard (Default) VVX 101: Standard GENBAND ALU-CTS DT VVX 150, 201: Standard, GENBAND ALU-CTS ocs2007r2 lync2010 All other phones: Standard GENBAND ALU-CTS ocs2007r2 lync2010 lcs2005	
sip-interop.cfg	voIpProt.server.x.specialInterop	Enables server-specific features for all registrations. Standard (default) VVX 101 = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT VVX 201 = Standard, GENBAND, GENBAND-A2, ALU-CTS, ocs2007r2, lync2010 All other phones = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT, ocs2007r2, lync2010, lcs2005	No
reg-advanced.cfg	reg.x.type	Private (default) - Use standard call signaling. Shared - Use augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.bargeInEnabled	0 (default) - barge-in is disabled for line x. 1 - barge-in is enabled (remote users of shared call appearances can interrupt or barge in to active calls).	No
reg-advanced.cfg	reg.x.callsPerLineKey	Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration. This per-registration parameter overrides call.callsPerLineKey . 24 (default) 1-24 VVX 101, 201 8 (default) 1 - 8	No
reg-basic.cfg	reg.x.auth.userId	User ID to be used for authentication challenges for this registration. Null (default) string - If the User ID is non-Null, it overrides the user parameter entered into the Authentication submenu on the Settings menu of the phone.	No
reg-basic.cfg	reg.x.auth.password	The password to be used for authentication challenges for this registration. Null (default) string - It overrides the password entered into the Authentication submenu on the Settings menu of the phone.	No
reg-basic.cfg	reg.x.outboundProxy.address	The IP address or hostname of the SIP server to which the phone sends all requests. Null (default) IP address or hostname	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
reg-advanced.cfg	reg.x.auth.domain	The domain of the authorization server that is used to check the user names and passwords. Null (default)string	No
reg-advanced.cfg	reg.x.thirdPartyName	Null (default) - In all other cases. string address -This field must match the reg.x.address value of the registration which makes up the part of a bridged line appearance (BLA).	No

Configuring Privacy on a MADN-SCA Line

In the UC Software download, Polycom provides the following two sample enhanced feature key (EFK) macros that you can configure to display on the phone to change privacy states: `privacyReleaseRestoreESK.cfg` and `privacyEnableESK.cfg` .

When you set the line to shared, an incoming call alerts all the members of the group simultaneously, and the call can be answered by any group member. On the server, you can configure a privacy setting that determines whether or not, after the call is answered, other members of the group can barge in to the same call and whether or not a call on hold can be picked up by other members of the group.

Optionally, you can configure star codes on the server that you can dial on the phone to toggle the privacy setting during a single active call. Note the following call behavior. If the line is configured for privacy by default, you can use a star code to toggle privacy on and off during an active call. When the call ends, the line resets to privacy settings. If the line is configured on the server with privacy off, you can use a star code to toggle to privacy on during an active call but you cannot toggle back to privacy off during the call. When the call ends, the line resets to privacy off.

Example MADN-SCA Configuration

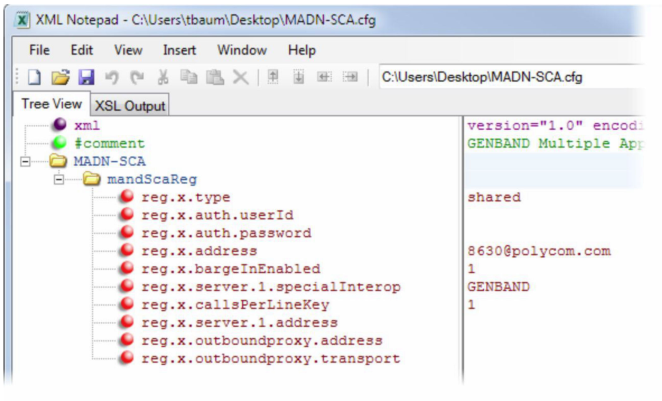
The following example configuration shows the minimum configuration you need to enable MADN-SCA on the phone.

You can use the parameters in the template configuration files or create your own configuration file from the parameters.

Procedure

1. Enter values for the following parameters in a configuration file and save.

   The value `8630@polycom.com` is an example registration address.


XML Notepad - C:\Users\tbaum\Desktop\MADN-SCA.cfg
File
Edit
View
Insert
Window
Help
C:\Users\Desktop\MADN-SCA.cfg
Tree View
XSL Output

```
xml                                    version="1.0" encodi
#comment                               GENBAND Multiple App
MADN-SCA
  mandScaReg
    reg.x.type                         shared
    reg.x.auth.userId
    reg.x.auth.password
    reg.x.address                      8630@polycom.com
    reg.x.bargeInEnabled               1
    reg.x.server.1.specialInterop      GENBAND
    reg.x.callsPerLineKey              1
    reg.x.server.1.address
    reg.x.outboundproxy.address
    reg.x.outboundproxy.transport
```

2. Enter the name of the configuration file to the CONFIG_FILES field of the master configuration file and save.

Global Address Book (GAB)

GENBAND's global address book (GAB) is a read-only global directory set up by an administrator and can co-exist with other corporate directories on the phone.

Global Address Book Parameters

Use the parameters in the following table to configure this feature.

GAB Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.corporateDirectory.alt.enabled	0 (default) - Disables the global address book service. 1 - Enables the global address book service.	No
features.cfg	dir.corp.alt.address	Enter the URL address of the GAB service provided by the server. Null (default) Hostname FQDN	No
features.cfg	dir.corp.alt.port	Set the port that connects to the server if a full URL is not provided. 0 (default) Null 1 to 65535	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.alt.user	Enter the user name used to authenticate to the GENBAND server. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.viewPersistence	Determine if the results from the last address directory search displays on the phone. 0 (default) 1	No
features.cfg	dir.corp.alt.attribute.x.filter	Enter a filter to use to set a predefined search string through configuration files. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.attribute.x.sticky	0 (default) —the filter string criteria for attribute x is reset after a reboot. 1—the filter string criteria is retained through a reboot. If you set an attribute to be sticky (set this parameter to 1), a ‘*' displays before the label of the attribute on the phone.	No
features.cfg	dir.corp.alt.attribute.x.label	Enter a label to identify a user. Null (default) UTF-8 encoding string	No
features.cfg	dir.corp.alt.attribute.x.name	Enter the name of the parameter to match on the server. Each name must be unique; however, a global address book entry can have multiple parameters with the same name. You can configure up to eight parameters (x = 1 to 8). Null (default) UTF-8 encoding string	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.corp.alt.attribute.x.type	Define how x is interpreted by the phone. Entries can have multiple parameters of the same type. first_name last_name (default) phone_number SIP_address Other—for display purposes only. If the user saves the entry to the local contact directory on the phone, first_name, last_name, and phone_number are copied. The user can place a call to the phone_number and SIP_address from the global address book directory.	No
site.cfg	dir.local.serverFeatureControl.method	Specifies a method for synchronizing the directory and server. None (default) GENBANDSOPI - Enables the GENBANDSOPI protocol on the phone to get the personnel address book service from the GENBAND server.	No

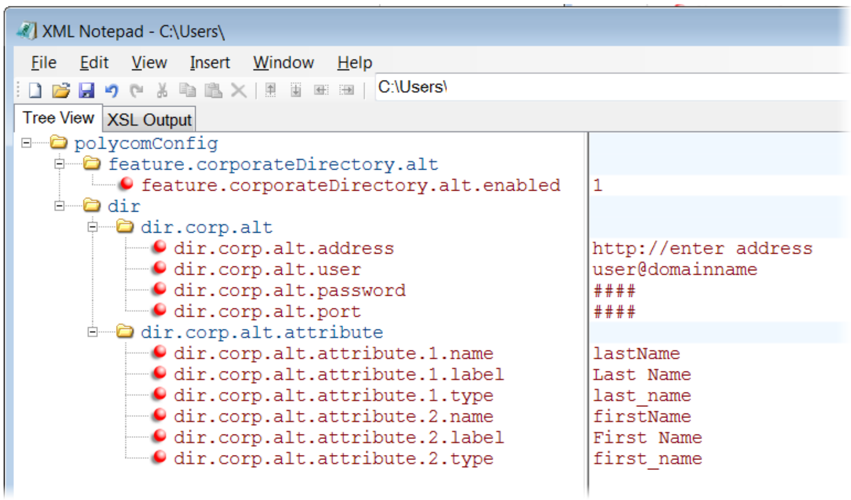
Example GAB Configuration

The following example shows the minimum parameters you need to configure to enable GAB on the phone.

Procedure

1. Enable GAB by configuring the values in `feature.corporateDirectory.alt` and `dir.corp.alt` .

   The following illustration includes an example GAB address book parameters in `dir.corp.alt.attribute` .



```
polycomConfig
  feature.corporateDirectory.alt
    feature.corporateDirectory.alt.enabled    1
  dir
    dir.corp.alt
      dir.corp.alt.address                    http://enter address
      dir.corp.alt.user                       user@domainname
      dir.corp.alt.password                   ####
      dir.corp.alt.port                       ####
    dir.corp.alt.attribute
      dir.corp.alt.attribute.1.name           lastName
      dir.corp.alt.attribute.1.label          Last Name
      dir.corp.alt.attribute.1.type           last_name
      dir.corp.alt.attribute.2.name           firstName
      dir.corp.alt.attribute.2.label          First Name
      dir.corp.alt.attribute.2.type           first_name
```

2. Save the configuration file.
3. Enter the name of the configuration file to the CONFIG_FILES field of the master configuration file and save.

Personal Address Book (PAB)

The personal address book (PAB) enables users to read and modify a personal directory of contacts on their phone.

When users modify contact information using any soft client, desk phone, or mobile client registered to the same line, the change is made on all other clients, and users are notified immediately of the change by the GENBAND server.

Personal Address Book Parameters

Use the parameters in the following table to configure this feature.

Note that when you enable server control, five telephone number fields per contact are available.

PAB Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.corporateDirectory.alt.enabled`	0 (default) - Disables the global address book service. 1 - Enables the global address book service.	No

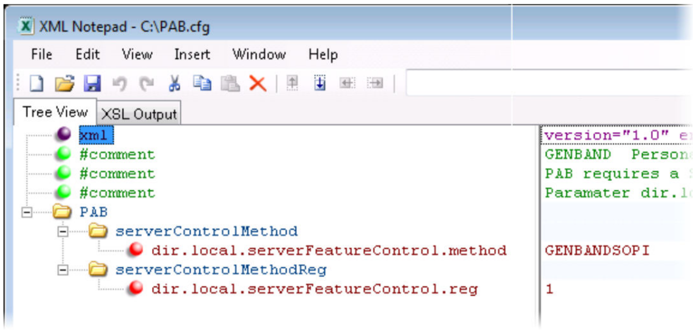
Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dir.local.serverFeatureControl.method`	Specifies a method for synchronizing the directory and server. None (default) GENBANDSOPI - Enables the GENBANDSOPI protocol on the phone to get the personnel address book service from the GENBAND server.	No
`site.cfg`	`dir.local.serverFeatureControl.reg`	Specifies the phone line to enable the personal address book feature on.1 (default) 1 -34	No
`site.cfg`	`dir.genband.local.contacts.maxSize`	Specify the maximum number of contacts available in the GENBAND personnel address book contact directory.	

Example Personal Address Book Configuration

The following example shows an example PAB configuration.

Procedure

1. Enter the values shown for the following parameters and save the configuration file.



```
xml                                         version="1.0" e
#comment                                    GENBAND  Person
#comment                                    PAB requires a
#comment                                    Paramater dir.l
PAB
  serverControlMethod
    dir.local.serverFeatureControl.method   GENBANDSOPI
  serverControlMethodReg
    dir.local.serverFeatureControl.reg      1
```

2. Enter the configuration file to the CONFIG_FILES field of the master configuration file and save.

Enhanced 911 (E.911) Location for GENBAND

With the Enhanced 911 (E.911) feature, you can set the location of the phone for emergency calls on the phone or on the provisioning server. When the phone starts up, the phone prompts users to choose a location, which is stored on the phone. The location that users set for the phone is used to identify the phone location to 911 operators dispatching emergency services. This feature is available for all phones and is disabled by default only in a GENBAND environment.

By default, users can make a 911 call when the phone is locked, regardless of the call state, or when other features are in use. When a 911 call is in progress, the call control option does not display, users cannot use the hard keys to control a call, and DND or call forwarding are disabled.

Enhanced 911 (E.911) Location Parameters for GENBAND

Use the parameters in the following table to configure this feature.

E911 Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.genband.E911.enabled	0 (default) - Disable the GENBAND E.911 feature. 1 - Enable the GENBAND E.911 feature.	Yes
features.cfg	genband.E911.location.description	Enter a description of the location of the phone, for example, cubicle 105. Ensure that the description string you provide here is identical to the description you configure on the location server. Other (default) String up to 256 characters [platform-specific display size limitations apply]	No
features.cfg	genband.E911.location.locationID	Enter the location ID corresponding to the location description you entered in genband.E911.location.description , for example, 112876. Ensure that the location ID you enter here is identical to the one you configure on the location server. 0 (default) string	No
features.cfg	genband.E911.registration.line	Select the registration line to use to retrieve E.911 location information 1 (default) 0 - 100	No
features.cfg	feature.E911.locationInfoSchema	RFC4119 (default) - XML schema is used in Session Initiation Protocol (SIP) invite as per RFC4119 standard. RFC5139- XML schema is used in Session Initiation Protocol (SIP) invite as per RFC5139 standard.	No

Manually Set the Phone's Location

Users can set their location for emergency call on the phone.

Procedure

1. Register the phone.
2. The phone displays a warning message to set your location for 10 seconds.
3. Press the warning message to enter a location.

   If the warning message disappears, on the phone, go to **Settings** > **Status** > **Diagnostics** > **Warnings**.
4. Select Details to enter a location to the location tree navigation menu.
5. Choose a location and press Save.
6. On the phone, go to **Status** > **Location Information**.

   The location information displays in the Status menu.

Emergency Instant Messages

VVX phones can receive emergency instant text messages.

You can configure audio alerts for incoming instant messages and set the duration of time that emergency messages display.

Emergency Instant Message Parameters

Use the following parameters to configure emergency messages for VVX phones registered with GENBAND.

Emergency Instant Message Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.instantMessaging.displayTimeout`	Specify the time in minutes instant messages display. Messages display until one of the following occurs: ▪ Timeout ▪ Another instant message is received ▪ A popup message displays ▪ The phone receives an incoming call ▪ The user presses any key or message on the phone 1 minute (default) 1 – 60 minutes	No
`features.cfg`	`feature.instantMessaging.ring`	instantMessage (default) – The phone plays a configured tone when an emergency instant message is received. Silent – No tone is played.	No
`features.cfg`	`feature.instantMessaging.enabled`	0 (default) – The phone does not display emergency instant messages. 1 - Received emergency instant messages display on the phone.	No

BroadSoft BroadWorks Server

This section shows you how to configure Polycom devices with BroadSoft Server options.

You can use the features available on the BroadWorks R18 server or the BroadWorks R20 or later server on all VVX phones except the following:

- VVX 101, 150, 201 phones

Note that you cannot register lines with the BroadWorks R18 server and the R20 and later server on the same phone. All lines on the phone must be registered to the same BroadWorks server.

Some BroadSoft features require you to authenticate the phone with the BroadWorks XSP service interface as described in the section Authentication with BroadWorks Xtended Service Platform (XSP) Service Interface.

Authentication with BroadWorks Xtended Service Platform (XSP) Service Interface

You can configure Polycom phones to use advanced features available on the BroadSoft BroadWorks server.

The phones support the following advanced BroadSoft features:

- BroadSoft Enhanced Call Park
- Executive-Assistant
- BroadSoft UC-One directory, favorites, and presence
- BroadSoft UC-One personal call control features

To use these features on Polycom devices with a BroadWorks server, you must authenticate the phone with the BroadSoft XSP service interface.

Authentication for BroadWorks XSP Parameters

The authentication method you use depends on which version of BroadWorks you are running.

If your server is running BroadWorks R19 or earlier, enable the following parameters to authenticate on the BroadWorks server using separate XSP credentials:

- `dir.broadsoft.xsp.address`
- `reg.x.broadsoft.userId`
- `reg.x.broadsoft.xsp.password`
- `reg.x.broadsoft.useXspCredentials`

If your server is running BroadWorks R19 Service Pack 1 or later, enable the following parameters to authenticate on the BroadWorks server using the same SIP credentials you used to register the phone lines: dir.broadsoft.xsp.address

- `reg.x.auth.userId`
- `reg.x.auth.password`
- `reg.x.broadsoft.userId`

See the following table for additional details on these parameters.

Configure BroadWorks XSP Service Interface Authentication

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	reg.x.broadsoft.xsp.password	Enter the password associated with the BroadSoft user account for the line. Required only when reg.x.broadsoft.useXspCredentials=1 . Null (default) string	No
features.cfg	reg.x.broadsoft.userId	Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. Null (default) string	No
features.cfg	reg.x.broadsoft.useXspCredentials	If this parameter is disabled, the phones use standard SIP credentials to authenticate. 1 (default) - Use this value, if phone lines are registered with a server running BroadWorks R19 or earlier. 0 - Set to 0, if phone lines are registered with a server running BroadWorks R19 SP1 or later.	No
reg-basic.cfg	reg.x.auth.userId	User ID to be used for authentication challenges for this registration. Null (default) string - If the User ID is non-Null, it overrides the user parameter entered into the Authentication submenu on the Settings menu of the phone.	No
reg-basic.cfg	reg.x.auth.password	The password to be used for authentication challenges for this registration. Null (default) string - It overrides the password entered into the Authentication submenu on the Settings menu of the phone.	No

BroadWorks Call Decline Policy

For shared lines in a BroadSoft BroadWorks environment, you can enable users to reject calls to a shared line.

By default, users cannot reject calls to a shared line on Polycom phones. When this feature is enabled and a user rejects a call to the shared line, the call is rejected on all phones registered with the shared line.

BroadWorks Call Decline Parameters

Use the parameter in the following table to enable users to reject calls on a shared line.

BroadWorks Call Decline Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.shared.reject`	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.	No

Flexible Seating

Flexible Seating enables a user of an assigned primary phone to simultaneously access a registered line as a guest from an alternate host phone.

The user's primary registration is active on the primary and host phone. Users can access the BroadSoft UC-One contact directory and favorites on the host phone, but the Polycom contact directory and favorites are not available.

Note that Flexible Seating is different from the Hoteling feature in that it provides only the primary registration's label on the host phone without any synchronization of features or settings.

The following conditions apply to the Flexible Seating feature:

- The primary phone and host phone do not sync automatically, but you can manually sync the phones on the BroadSoft BroadWorks server.
- The phone configured for the host user cannot accept incoming calls. The host user can make only emergency outgoing calls that are defined by the BroadWorks server.
- If the Phone Lock feature is enabled, numbers defined in the authorized call list are not allowed for outgoing calls except the emergency numbers set on the BroadWorks server.
- The host user account is intended to be used as a placeholder account that supports guest users and is not intended to be assigned to an actual phone user.
- The guest user cannot change the user password when Flexible Seating is enabled for the phone. You can change the host phone's user password from the Web Configuration Utility at any time. You can change the host phone's user password from the phone screen only when the guest user is not logged in.

Flexible Seating is not compatible with the following features:

- Hoteling
- Visitor Desk Phone (VDP)
- User Profile Feature
- Local Call Forwarding
- Local DND

On the BroadWorks server, you can set a period of time when the server automatically logs out a user from a phone in case a user does not log out.

Flexible Seating Parameters

To configure a host phone to support the primary phone's line registration, you must configure a host user profile and a guest user profile on the BroadSoft BroadWorks server.

In the host user profile configuration files, add the configuration parameters shown in the following table and map these parameters to the corresponding BroadSoft BroadWorks configuration tags.

Flexible Seating Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`hoteling.reg`	1 (default) - Specifies the phone line on the host phone which hosts the guest line.	No
	`hotelingMode.type`	-1 (Default): The parameter does not exist on the BroadSoft server. 0 - Both Flexible Seating and Hoteling are disabled on the BroadSoft Device Management Server (DMS). 1 - Hoteling is enabled 2 - Flexible Seating is enabled but guest is not logged in. 3 - Flexible seating location is enabled and guest is logged in. Note: This parameter overrides `voIpProt.SIP.specialEvent.checkSync.downloadDirectory` when set to 2 or 3.	No

Guest Profile PIN

You can configure a PIN for each guest profile, which enables users to access their guest profile on a host phone using a PIN.

The PIN prevents other users from logging into a guest phone without the phone password or guest PIN. The guest profile PIN takes precedence over the local phone password and the guest user must log out of the phone with the PIN before another user can log in with their password.

BroadSoft BroadWorks Configuration Tags

The following table shows the Polycom parameters you can map to the corresponding BroadSoft tags.

Polycom Configuration Parameter	BroadSoft Tag
`hoteling.reg`	%BWHOTELINGLINE-x%
`hotelingMode.type`	%BWHOTELINGMODE-x%

Executive-Assistant

Using configuration files, you can enable the BroadSoft Executive-Assistant feature on lines registered with the BroadWorks R20 or later server, and assign lines as an executive or an assistant.

Note that all corresponding Executive and Assistant lines must be registered to the same server.

After you enable the feature, users set as executives or assistants can set basic filters to control which calls are sent directly to an assistant to answer or sent to the executive first. Executives can also enable screening, which enables the executive's phone to display the incoming call notification for all filtered calls.

To use this feature on Polycom phones registered with a BroadWorks server, you must authenticate the phone with the BroadSoft XSP service interface.

In addition, depending on the role you assign the user, the **Executive** or **Assistant** icon displays on the Home screen of the phone. You can also simplify the Executive and Assistant menus by adding or removing **Pick Call** and **Barge-in** soft keys from the menu.

Enhanced Feature Keys for Executive-Assistant Menus

You can create enhanced feature keys (EFK) to enable users to quickly access the Overview Executives menu for assistants or the Executive Settings menu for executives.

You can create an Executive or Assistant line key, soft key, or speed dial that displays on the Lines screen in addition to the feature icons that display by default on the Home screen.

When a user presses the Executive EFK on the executive's phone, the Executive Settings menu displays, and when a user presses the Assistant EFK on the assistant's phone, the Overview Executives menu displays. You can configure a line or soft key for this feature using the following EFK macro:

- Executive menu: `"$FExecutiveMenu$"`
- Assistant menu: `$FAssistantMenu$`

Executive-Assistant Parameters

In the BroadWorks Web Portal, you must enable the Executive Service for private and shared executive lines, and the Executive-Assistant Service for private and shared assistant lines.

The BroadWorks server allows the following configuration options: Executive private line, Executive-Assistant Service line, and a shared alias line. Administrators can set up executive and assistant lines in the following scenarios:

- A private executive line with an assistant with a private line
- Shared executive line with an assistant with a private line
- Shared executive line with a shared line alias on the assistant's phone

  - The shared line must be created as a shared location of a line with the Executive Service on the BroadWorks server.
  - In this option, the main line registration is a private line for the assistant, and the secondary registration is a shared line for the executive.

The following table includes the configuration parameters you can use to enable and configure the Executive-Assistant feature.

BroadSoft Executive-Assistant Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.BSExecutiveAssistant.enabled`	`0` (default) - Disables the BroadSoft Executive-Assistant feature. `1` - Enables the BroadSoft Executive-Assistant feature.	No
`features.cfg`	`feature.BSExecutiveAssistant.regIndex`	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.	No
`features.cfg`	`feature.BSExecutiveAssistant.userRole`	`ExecutiveRole` (default) - Sets the registered line as an Executive line. `AssistantRole` - Sets the registered line as an Assistant line. Note: A phone can only have a line set as an Executive or an Assistant; an Executive and an Assistant line cannot be on the same phone.	No
`features.cfg`	`feature.BSExecutiveAssistant.SimplifiedAssistant.enabled`	0 (default) - Displays the the Pick Call and Barge-in soft keys in the Assistants menu on the phone. 1 – Removes the Pick Call and Barge-in soft keys from the Assistants menu on the phone.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.BSExecutiveAssistant.SimplifiedExec.enabled`	0 (default) - Displays the the Pick Call and Barge-in soft keys in the Assistants menu on the phone. 1 – Removes the Pick Call and Barge-in soft keys from the Assistants menu on the phone.	No

Enhanced Call Park

You can configure BroadWorks Enhanced Call Park per registered line.

The following features are available for Enhanced Call Park:

- You can configure Enhanced Call Park only using configuration files; you cannot configure the feature on the Web Configuration Utility or from the local phone interface.
- You can configure Enhanced Call Park for private lines and shared lines. No configuration is necessary to enable the call park notification for monitored BLF lines.
- The default star codes set for the `call.parkedCallRetrieveString` is *88.

Enhanced Call Park Parameters

The following table includes the configuration parameters you can use to enable and configure this feature.

Enhanced Call Park

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`reg.x.enhancedCallPark.enabled`	0 (default) - To disable the BroadWorks Enhanced Call Park feature. 1 - To enable the BroadWorks Enhanced Call Park feature.	No
`reg-basic.cfg`	`reg.x.lineAddress`	The line extension for a shared line. This parameter applies to private lines and BroadSoft call park and retrieve. If there is no extension provided for this parameter, the call park notification is ignored for the shared line. Null (default) String	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.enhancedCallPark.allowAudioNotification`	0 (default) - Disables the audio notifications for parked calls on private and shared lines. 1 - Enables the audio notifications for parked calls on private and shared lines.	No
`sip-interop.cfg, site.cfg`	`call.parkedCallRetrieveString`	The star code that initiates retrieval of a parked call. Null (default) Permitted values are star codes.	No

BroadSoft Directory Support

The BroadSoft directories enable users to search and view their personal, group, or enterprise contacts.

When the BroadSoft directories are integrated with Polycom BroadSoft UC-One Application, users can access the different types of directories and search for contacts. There are five types of BroadSoft Directories:

- Enterprise Directory. This directory enables users to search and view Active Directory global address list of an enterprise. Users can query by first name, last name, phone number, extension and mobile number, and access contact information.
- Group Directory. This directory enables users to view the contact details such as work, extension, and mobile numbers of contacts. Users can place a call to anyone in the user's group.
- Group Common Directory. This directory enables users to view the contact details such as names and phone numbers of common contacts listed in the Group Common Directory.
- Enterprise Common Directory. This directory enables users to view the contact details such as names and phone numbers of common contacts listed in the Enterprise Common Directory.
- Personal Directory. This directory enables users to view the contact details such as names and phone numbers of the contacts in the user's personal directory stored on the server. You must enable this feature to allow users to add, delete, or edit the contacts in the BroadSoft Personal Directory.

BroadSoft Directory Parameters

To perform a search and to view contacts on the BroadSoft directories, configure the directories.

You can configure this feature using the parameters in the following table

.

BroadSoft Directory Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.broadsoftGroupDir.enabled`	0 (default) - Disables Group Directory. 1 - Enables Group Directory.	No
`features.cfg`	`feature.broadsoftdir.enabled`	0 (default) - Disables Enterprise Directory. 1 - Enables Enterprise Directory.	Yes
`features.cfg`	`feature.broadsoftPersonalDir.enabled`	0 (default) - Disables Personal Directory. 1 - Enables Personal Directory.	

Polycom BroadSoft UC-One Application

The Polycom BroadSoft UC-One application integrates with BroadSoft Enterprise Directory and BroadCloud services—a set of hosted services by BroadSoft—to provide the following features:

- BroadSoft Directory—Displays information for all users in the enterprise, for example, work and mobile phone numbers.
- BroadCloud Presence—Enables users to share presence information with the BroadTouch Business Communicator (BTBC) client application.
- BroadCloud Favorites—Enables users to mark contacts as favorites with the BroadTouch Business Communicator (BTBC) client application.

These features are available on Polycom VVX 300, 400, 500 and VVX 600 series business media phones and VVX 250, 350, and 450 business IP phones. These features require support from the BroadSoft BroadWorks R18 SP1 platform with patches and BroadSoft BroadCloud services. For details on how to set up and use these features, see the latest *Polycom VVX Business Media Phones - User Guide* at Latest Polycom UC Software Release.

Polycom's BroadSoft UC-One application enables you to:

- Access the BroadSoft Directory
- Search for contacts in BroadSoft Directory
- View BroadSoft UC-One contacts and groups
- View the presence status of BroadSoft UC-One contacts
- View and filter BroadSoft UC-One contacts
- Activate and control BroadSoft UC-One personal call control features.

BroadSoft UC-One Configuration Parameters

The following table lists all parameters available to configure features in the BroadSoft UC-One application.

BroadSoft UC-One Application

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.qml.enabled`	0 (default) - Disable the QML viewer on the phone. Note that the UC-One directory user interface uses QML as the user interface framework and the viewer is used to load the QML applications. 1 - Enable the QML viewer on phone.	Yes
`features.cfg`	`feature.broadsoftdir.enabled`	0 (default) - Disable simple search for Enterprise Directories. 1 - Enable simple search for Enterprise Directories.	Yes
`features.cfg`	`feature.broadsoftUcOne.enabled`	0 (default) - Disables the BroadSoft UC-One feature. 1 - Enables the BroadSoft UC-One feature.	Yes
`features.cfg`	`feature.presence.enabled`	0 (default) - Disable the presence feature—including buddy managements and user status. 1 - Enable the presence feature with the buddy and status options.	No
`features.cfg`	`homeScreen.UCOne.enable`	1 (default) - Enable the UC-One Settings icon to display on the phone Home screen. 0 - Disable the UC-One Settings icon to display on the phone Home screen.	No
`features.cfg`	`dir.broadsoft.xsp.address`	Set the IP address or hostname of the BroadSoft directory XSP home address. Null (default) IP address Hostname FQDN	No
`applications.cfg`	`dir.broadsoft.xsp.username`	To set the BroadSoft Directory XSP home address.	

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.broadsoft.xsp.password	Set the password used to authenticate to the BroadSoft Directory XSP server. Null (default) UTF-8 encoding string	No
features.cfg	xmpp.1.auth.password	Specify the password used for XMPP registration. Null (Default) UTF-8 encoded string	No
features.cfg	xmpp.1.dialMethod	For SIP dialing, the destination XMPP URI is converted to a SIP URI, and the first available SIP line is used to place the call. SIP (default) String min 0, max 256	No
features.cfg	xmpp.1.jid	Enter the Jabber identity used to register with the presence server, for example: presence.test2@polycom-alpha.eu.bc.im . Null (default) String min 0, max 256	No
features.cfg	xmpp.1.roster.invite.accept	Choose how phone users receive the BroadSoft XMPP invitation to be added to a buddy list. prompt (default) - phone displays a list of users who have requested to add you as a buddy and you can accept or reject the invitation. Automatic	No
features.cfg	xmpp.1.server	Sets the BroadSoft XMPP presence server to an IP address, host name, or FQDN, for example: polycom-alpha.eu.bc.im . Null (default) dotted-decimal IP address, host name, or FQDN.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`xmpp.1.verifyCert`	Specifies to enable or disable verification of the TLS certificate provided by the BroadSoft XMPP presence server. 1 (default) 0	No

Configuring BroadSoft UC-One

You can configure the UC-One Call Settings menu and feature options on the phone, in the Web Configuration Utility, and using configuration parameters.

Configure BroadSoft UC-One on the Phone

You can enable the BroadSoft UC-One feature directly from the phone.

Procedure

1. Navigate to **Settings** > **UC-One**.
2. Under General, click **Enable for BroadSoft UC-One**.

   This enables the UC-One Call Settings menu to display on the phone.

Configure BroadSoft UC-One in the Web Configuration Utility

You can enable the BroadSoft UC-One feature and feature options in the Web Configuration Utility.

Procedure

1. In the Web Configuration Utility, navigate to **Settings** > **UC-One**.
2. Under **Call Settings Features**, enable each feature menu you want available on the phone.

BroadSoft UC-One Directory Parameters

Use the parameters listed in the following table with the Polycom BroadSoft UC-One directory.

BroadSoft UC-One Directory Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	dir.broadsoft.regMap	Specify the registration line credentials you want to use for BroadSoft R20 Server or later to retrieve directory information from the BroadSoft UC-One directory when dir.broadsoft.useXspCredentials =0. 1 (default) 0 - Const_NumLineReg	No
features.cfg	dir.broadsoft.useXspCredentials	Specify which method of credentials the phone uses to sign in with the BroadSoft server. 1 (default)—uses BroadSoft XSP credentials. 0—uses SIP credentials from dir.broadsoft.regMap.	No

Enterprise Directory Default Search

You can view an initial list of contacts in the Enterprise directory.

After you enable the feature, users can view a list of contacts by default without the need to enter a name in the search box of the directory.

Enterprise Directory Search Parameters

The following table includes the parameter for the Enterprise Directory Search feature.

Enterprise Directory Search Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.broadsoftdir.showDefaultSearch	0 (default) - No contacts are displayed when the search box field is empty. 1 - Enables the user to view the initial list of contacts for an empty search box	

BroadSoft Server-Based Call Logs

You can configure the phone to view the list of call logs when the user taps the **Recent** soft key on the phone's screen.

When you enable this feature, users can view the call logs retrieved from the server on the phone.

BroadSoft Server-Based Call Logs Parameters

The following table includes the parameter for the BroadSoft server based call logs feature.

BroadSoft Server Based Call Logs Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.broadsoft.callLogs	Disabled (default) - Disable the BroadSoft server call logs feature. Basic - Enable the BroadSoft server call logs feature.	

BroadSoft Server-Based Redial

You can configure the phone to support BroadSoft Server-Based Redial feature, which allows users to redial the last number dialed from any device connected to the same line or registration.

When enabled, the **Redial** soft key displays on the phone screen. Users can select this soft key to place a call to the last dialed number.

BroadSoft Server-Based Redial Parameters

Use the following parameters to configure this feature.

BroadSoft Server-Based Redial Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.broadsoft.basicCallLogs.redial.enabled	0 (default) - Disables the option to redial the last number. 1 - Enables the phone to redial the last number.	

Anonymous Call Rejection

Anonymous Call Rejection enables users to automatically reject incoming calls from anonymous parties who have restricted their caller identification.

After you enable the feature for users, users can turn call rejection on or off from the phone. When a user turns Anonymous Call Rejection on, the phone gives no indication that an anonymous call was received.

You can configure this option in the Web Configuration Utility.

Configure Anonymous Call Rejection using the Web Configuration Utility

You can configure Anonymous Call Rejection in the Web Configuration Utility.

Procedure

1. Navigate to **Settings** > **UC-One**.
2. Under the **Call Setting Features**, click **Enable for Anonymous Call Rejection**.

Anonymous Call Rejection Parameters

You can enable the Anonymous Call Rejection feature using configuration files or the Web Configuration Utility.

Use the parameters in the following table to enable this feature.

Anonymous Call Rejection

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.broadsoft.xsi.AnonymousCallReject.enabled`	0 (default) - Does not display the Anonymous Call Rejection menu to users. 1 - Displays the Anonymous Call Rejection menu and the user can turn the feature on or off from the phone.	No
`features.cfg`	`feature.broadsoftUcOne.enabled`	0 (default) - Disables the BroadSoft UC-One feature. 1 - Enables the BroadSoft UC-One feature.	Yes
`features.cfg`	`reg.x.broadsoft.userId`	Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. Null (default) string	No

Simultaneous Ring Personal

The Simultaneous Ring feature enables users to add phone numbers to a list of contacts whose phones ring simultaneously when the user receives an incoming call.

When you enable the display of the Simultaneous Ring menu option on the phone, users can turn the feature on or off from the phone and define which numbers should be included in the Simultaneous Ring group.

Simultaneous Ring Parameters

You can enable or disable the Simultaneous Ring feature for users using configuration files or the Web Configuration Utility.

Use the parameters in the following table to enable this feature.

Simultaneous Ring

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.broadsoft.xsi.SimultaneousRing.enabled`	0 (default) - Disables and does not display the Simultaneous Ring Personal feature menu on the phone. 1 - Enables the Simultaneous Ring Personal feature menu on the phone.	No
`features.cfg`	`feature.broadsoftUcOne.enabled`	Enable or disable all BroadSoft UC-One features.	

Line ID Blocking

You can enable or disable the display of the Line ID Blocking menu option on the phone.

When you enable the menu for users, users can choose to hide their phone number before making a call.

Line ID Blocking Parameters

You can configure this feature using configuration parameters or the Web Configuration Utility.

Use the parameters in the following table to enable this feature.

Line ID Blocking

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.broadsoft.xsi.LineIdblock.enabled`	0 (default) - Disables and does not display the Line ID Blocking feature menu on the phone. 1 - Enables the Line ID Blocking feature menu on the phone.	No
`features.cfg`	`feature.broadsoftUcOne.enabled`	0 (default) - Disables the BroadSoft UC-One feature. 1 - Enables the BroadSoft UC-One feature.	Yes

BroadWorks Anywhere

BroadWorks Anywhere enables users to use one phone number to receive calls to and dial out from their desk phone, mobile phone, or home office phone.

When you enable this feature, users can move calls between phones and perform phone functions from any phone. When enabled, the BroadWorks Anywhere settings menu displays on the phone and users can turn the feature on or off and add BroadWorks Anywhere locations on the phone.

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

- $FBWSAnyWhere$

BroadWorks Anywhere Parameters

You can configure BroadWorks Anywhere using configuration files or the Web Configuration Utility.

Use the parameters in the following table to enable this feature.

BroadWorks Anywhere

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.broadsoft.xsi.BroadWorksAnywhere.enabled`	0 (default) - Disables and does not display the BroadWorks Anywhere feature menu on the phone. 1 - Enables the BroadWorks Anywhere feature menu on the phone.	No
`features.cfg`	`feature.broadsoftUcOne.enabled`	0 (default) - Disables the BroadSoft UC-One feature. 1 - Enables the BroadSoft UC-One feature.	Yes

BroadSoft Server-based Call Waiting

You can configure the phone to support server-based call waiting, which enables the server to manage incoming calls while a user is in an active call.

When a user changes the call waiting state, the phone sends a request to the server to update to the new state. You can also configure the phone to specify the ringtone for incoming calls, when another call is in progress.

BroadSoft Server-based Call Waiting Parameters

Use the parameters in the following table to configure server-based call waiting alerts.

Server-based Call Waiting Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.broadsoft.xsi.callWaiting.enabled	0 (default) - Disable incoming calls during an active call. 1 - Enable incoming calls during an active call.	No

Remote Office

Remote Office enables users to set up a phone number on their office phone to forward incoming calls to a mobile device or home office number.

When enabled, this feature enables users to answer incoming calls to the office phone on the phone, and any calls placed from that phone show the office phone number.

Remote Office Parameters

Use the parameters in the following table to enable this feature.

Remote Office

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.broadsoft.xsi.RemoteOffice.enabled	0 (default) - Disables the Remote Office feature menu on the phone. 1 - Enables and displays the Remote Office feature menu on the phone.	No
features.cfg	reg.x.broadsoft.userId	Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. Null (default) string	No
features.cfg	feature.broadsoftUcOne.enabled	0 (default) - Disables the BroadSoft UC-One feature. 1 - Enables the BroadSoft UC-One feature.	Yes
features.cfg	dir.broadsoft.xsp.password	Set the password used to authenticate to the BroadSoft Directory XSP server. Null (default) UTF-8 encoding string	No

BroadSoft UC-One Credentials

Enabling this feature allows users to enter their BroadWorks UC-One credentials on the phone instead of in the configuration files.

The parameters `reg.x.broadsoft.useXspCredentials` , and `feature.broadsoftUcOne.enabled` must be enabled to display the UC-One Credentials menu option on the phone.

BroadSoft UC-One Credential Parameters

Use the parameters in the following table to enable this feature.

Configure XSP User Name an Password

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`dir.broadsoft.xsp.address`	Set the IP address or hostname of the BroadSoft directory XSP home address. Null (default) IP address Hostname FQDN	No
`features.cfg`	`reg.x.broadsoft.userId`	Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface. Null (default) string	No
`features.cfg`	`feature.broadsoftUcOne.enabled`	0 (default) - Disables the BroadSoft UC-One feature. 1 - Enables the BroadSoft UC-One feature.	Yes
`applications.cfg`	`dir.broadsoft.xsp.username`	To set the BroadSoft Directory XSP home address.	
`features.cfg`	`dir.broadsoft.xsp.password`	Set the password used to authenticate to the BroadSoft Directory XSP server. Null (default) UTF-8 encoding string	No
`features.cfg`	`feature.broadsoftdir.enabled`	0 (default) - Disable simple search for Enterprise Directories. 1 - Enable simple search for Enterprise Directories.	Yes

BroadSoft Server-Based Call Forwarding

To enable server-based call forwarding, you must enable the feature on both the server and the registered phone.

If you enable server-based call forwarding on one registration, other registrations are not affected.

The following conditions apply for server-based call forwarding:

- If server-based call forwarding is enabled, but inactive, when a user presses the Forward soft key, the ‘moving arrow' icon does not display on the phone and incoming calls are not forwarded.

The call server uses the Diversion field with a SIP header to inform the phone of a call's history. For example, when you enable call forwarding, the Diversion header allows the receiving phone to indicate who the call was from, and the phone number it was forwarded from.

Hoteling

The Hoteling feature enables users to log in to a guest profile to use any available shared phone.

After logging in, users have access to their own guest profile and settings on the shared phone. When hoteling is enabled, the Guest In soft key displays for users to log in to the phone.

Hoteling is not supported on VVX 101, 150, and 201 phones.

Note: For additional details on configuring the hoteling feature, see *Using Hoteling on Polycom Phones*: *Feature Profile 76554* at Polycom Engineering Advisories and Technical Notifications.

Hoteling Parameters

To enable Hoteling, you must configure Polycom phones with the BroadSoft BroadWorks R17 platform.

You cannot use Hoteling in conjunction with the feature-synchronized automatic call distribution (ACD) feature and you must disable all ACD parameters to use the Hoteling feature. If both features are enabled at the same time, ACD take precedence and the Hoteling GuestIn/GuestOut soft keys do not display.

Use the parameters in the following table to configure Hoteling.

Hoteling Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.hoteling.enabled	0 (default) - Enable Hoteling. 1 - Disable Hoteling.	No
features.cfg	hoteling.reg	Specify the line registration to use for Hoteling. You must disable the Automatic Call Distribution (ACD) feature and all ACD parameters to use Hoteling. 1 (default) 1 - 34	No

Related Links

Feature-Synchronized Automatic Call Distribution (ACD) on page 596

Feature-Synchronized Automatic Call Distribution (ACD)

Feature-synchronized automatic call distribution (ACD) assists organizations in handling a large number of incoming phone calls to a call center with users in agent/supervisor roles.

Feature-synchronized ACD is distinct from and provides more advanced ACD functions than the Hoteling feature. This feature is not supported on VVX 101, 150, and 201 phones.

Feature-synchronized ACD is available in the following services.

- Standard—Standard service enables call center agents to sign in to a shared phone. When an agent is signed in, the phone displays the current state of the agent, for example, whether the agent is available or unavailable to take new calls.
- Premium—Premium service offers two additional features: Hoteling and Queue Status Notification.
    - Hoteling enables agents to use their agent credentials to log in to any available phone. If you want to enable the hoteling feature with feature-synchronized ACD, see the section Hoteling.
    - Queue status notification enables agents to view the queue status of a call center so that agents can adjust their call response.

The capabilities of this feature vary with the SIP call server. Consult your call server provider for information and for documentation. The SIP signaling used for this implementation is described in the BroadSoft BroadWorks document Device Key Synchronization Requirements Document; Release R14 sp2; Document version 1.6.

Note: For more information on standard and premium ACD as well as the hoteling and queue status notification enhancements, see *Feature Profile 76179: Using Premium Automatic Call Distribution for Call Centers* at Polycom Engineering Advisories and Technical Notifications.

Related Links

Hoteling Parameters on page 595

ACD Agent Availability Parameters

Use the parameters in this table to configure ACD agent availability for SIP-B Automatic Call Distribution.

ACD Agent Availability

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.acdAgentAvailability.enabled`	0 (default) - Disables the ACD agent available/unavailable feature. 1 - Enables the ACD agent available/ unavailable feature.	No
`reg-advanced.cfg`	`reg.x.acd-agent-available`	0 (default) - The ACD feature is disabled for registration. 1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.	No

Configuring uaCSTA

When you configure Polycom phones to use user agent Computer Supported Telecommunications Applications (uaCSTA) with a CSTA server, you can remotely control the phone and access phone services using a computer telephony integration (CTI) application on your computer.

Note: The Polycom VVX 101 and 1500 business media phones do not support uaCSTA.

Polycom phones support the Minimum and Basic profiles compliant with “ECMA TR/087: Using CSTA for SIP Phone User Agents (uaCSTA).” For information, see ECMA international.

Note: Polycom phones do not support the Network Reached event.

Polycom supports the following CSTA services and events:

CSTA Monitoring Services

- MonitorStart
- MonitorStop

CSTA Call Control Services

- MakeCall Without Prompt
- AnswerCall
- ClearConnection
- DeflectCall in alerting state
- HoldCall
- RetrieveCall
- SingleStepTransferCall

CSTA Call Control Events

- ServiceInitiated
- Originated
- Delivered
- Diverted
- Established
- ConnectionCleared
- Held
- Retrieved
- Failed
- Transferred

CSTA Maintenance Events

- BackInService
- OutOfService

Capability Exchange Service

- GetSwitchingFunctionDevices

Capability Exchange Event

- SwitchingFunctionDevices

Enable uaCSTA

You can configure one CSTA line on each phone. To ensure CSTA works correctly, Polycom recommends that you configure the CSTA line as the last among all registered lines on the phone.

Procedure

1. Set up an account on your CSTA server.
2. Set the server to CSTA.
3. Enable the Polycom per-registration parameter: `reg.x.csta="1"`.

   When you correctly register a CSTA line on a Polycom phone, the CSTA line displays on the phone with an icon and the default label **CSTA**. You can configure the label of the CSTA line. If the CSTA line is not registered, an icon shows that the line is unregistered.

   A CSTA-registered line has no functionality to users. If a user selects a CSTA line on the phone, a message displays stating that no action is available.

uaCSTA Parameters

Use the following parameters to configure the uaCSTA feature.

You can use one CSTA line per phone.

uaCSTA Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg.advanced`	`reg.x.csta`	Set the CSTA line x to the last registered line on the phone. 0 (default) – Disable User Agent Computer Supported Telecommunications Applications (uaCSTA). 1 – Enable uaCSTA. This per-registration parameter overrides the global parameter `voIpProt.SIP.csta`. A CSTA icon displays on the phone when this parameter is set to 1 and `reg.x.server.y.specialInterop =CSTA`.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg.advanced`	`reg.x.server.y.specialInterop`	Specify the server-specific feature for the line registration. When you set this parameter to CSTA and `reg.x.csta=1`, a CSTA icon displays on the phone. standard (Default)	No
`sip-interop.cfg`	`voIpProt.SIP.csta`	0 (default) – Disable uaCSTA. 1 – Enable uaCSTA. When set to 1, `reg.x.csta` overrides `voIpProt.SIP.csta`.	No

Device Parameters

Topics:

- Changing Device Parameters
- Device Parameters

The < device/ > parameters—also known as device settings—contain default values that you can use to configure basic settings for multiple phones within your network.

Polycom provides a global device.set parameter that you must enable to install software and change device parameters. In addition, each <device/> parameter has a corresponding .set parameter that enables or disables the value for that device parameter. You need to enable the corresponding .set parameter for each parameter you want to apply.

After you complete the software installation or configuration changes to device parameters, remove device.set to prevent the phones from rebooting and triggering a reset of device parameters that phone users might have changed after the initial installation.

If you configure any parameter values using the <device/> parameters, any subsequent configuration changes you make from the Web Configuration Utility or phone local interface do not take effect after a phone reboot or restart.

The <device/> parameters are designed to be stored in flash memory and for this reason the phone does not upload <device/> parameters to the <MAC>-web.cfg or <MAC>-phone.cfg override files if you make configuration changes through the Web Configuration Utility or phone interface. This design protects your ability to manage and access the phones using the standard set of parameters on a provisioning server after the initial software installation.

Changing Device Parameters

Keep the following in mind when modifying device parameters:

- Note that some parameters may be ignored. For example, if DHCP is enabled, it will still override the value set with device.net.ipAddress .
- Though individual parameters are checked to see whether they are in range, the interaction between parameters is not checked. If a parameter is out of range, an error message displays in the log file and the parameter is not be used.
- Incorrect configuration can put the phones into a reboot loop. For example, server A has a configuration file that specifies that server B should be used, and server B has a configuration file that specifies that server A should be used.

To detect errors, including IP address conflicts, Polycom recommends that you test the new configuration files on two phones before initializing all phones.

Types of Device Parameters

The following table outlines the three types of `<device/>` parameters, their permitted values, and the default value.

Types of Device Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.set	0 (default)—Do not use any device.xxx fields to set any parameters. Set this to 0 after the initial software installation. 1—Use the device.xxx fields that have device.xxx.set=1 . Set this to 1 only for the initial software installation.	Yes
device.cfg	device.xxx	string	Yes
device.cfg	device.xxx.set	0 (default)—Do not use the device.xxx value. 1—Use the device.xxx value. For example, if device.net.ipAddress.set=1 , then use the value set for device.net.ipAddress .	Yes

Device Parameters

The following table lists each of the `<device/>` parameters that you can configure.

Note: The default values for the `<device/>` parameters are set at the factory when the phones are shipped. For a list of the default values, see the latest Product Shipping Configuration Change Notice at Polycom Engineering Advisories and Technical Notifications.

Device Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.auth.localAdminPassword	Set the phone's local administrative password. The minimum length is defined by sec.pwd.length.admin . string (32 character max)	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg reg-advanced	device.auth.localUserPassword	Set the phone user's local password. The minimum length is defined by sec.pwd.length.user . string (32 character max)	No
device.cfg	device.auxPort.enable	Enable or disable the phone auxiliary port. 0 1 (default)	Yes
device.cfg	device.baseProfile	NULL (default) Generic —Sets the base profile to Generic for OpenSIP environments. Lync —Sets this Base Profile for Skype for Business deployments.	No
device.cfg site.cfg	device.dhcp.bootSrvOpt	When the boot server is set to Custom or Custom+Option66, specify the numeric DHCP option that the phone looks for. Null 128 to 254	Yes
device.cfg site.cfg	device.dhcp.bootSrvOptType	Set the type of DHCP option the phone looks for to find its provisioning server if device.dhcp.bootSrvUseOpt is set to Custom . IP address—The IP address provided must specify the format of the provisioning server. String—The string provided must match one of the formats specified by device.prov.serverName .	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.dhcp.bootSrvUseOpt	Default—The phone looks for option number 66 (string type) in the response received from the DHCP server. The DHCP server should send address information in option 66 that matches one of the formats described for device.prov.serverName . Custom —The phone looks for the option number specified by device.dhcp.bootSrvOpt , and the type specified by device.dhcp.bootSrvOptType in the response received from the DHCP server. Static —The phone uses the boot server configured through the provisioning server device.prov.* parameters. Custom and Default—The phone uses the custom option first or use Option 66 if the custom option is not present.	Yes
device.cfg site.cfg	device.dhcp.dhcpVlanDiscOpt	Set the DHCP private option to use when device.dhcp.dhcpVlanDiscUseOpt is set to Custom . 128 to 254	Yes
device.cfg site.cfg	device.dhcp.dhcpVlanDiscUseOpt	Set how VLAN Discovery occurs. Disabled—no VLAN discovery through DHCP. Fixed—use predefined DHCP vendor-specific option values of 128, 144, 157 and 191 (device.dhcp.dhcpVlanDiscOpt is ignored). Custom—use the number specified by device.dhcp.dhcpVlanDiscOpt .	Yes
device.cfg site.cfg	device.dhcp.enabled	Enable or disable DHCP. 0 1	Yes
device.cfg site.cfg	device.dhcp.option60Type	Set the DHCP option 60 type. Binary—vendor-identifying information is in the format defined in RFC 3925. ASCII—vendor-identifying information is in ASCII format.	Yes
device.cfg site.cfg	device.dns.altSrvAddress	Set the secondary server to which the phone directs domain name system (DNS) queries. Server Address	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.dns.domain	Set the phone's DNS domain. String	Yes
device.cfg site.cfg	device.dns.serverAddress	Set the primary server to which the phone directs DNS queries. Server Address	Yes
device.cfg site.cfg	device.hostname	Specify a hostname for the phone when using DHCP by adding a hostname string to the phone's configuration. If device.host.hostname.set = 1, and device.host.hostname = Null , the DHCP client uses Option 12 to send a predefined hostname to the DHCP registration server using Polycom_<MACaddress> . String —The maximum length of the hostname string is <=255 bytes, and the valid character set is defined in RFC 1035.	Yes
device.cfg site.cfg	device.net.cdpEnabled	Determine if the phone attempts to determine its VLAN ID and negotiate power through CDP. 0 1	Yes
device.cfg site.cfg wireless.cfg	device.net.dot1x.anonid	EAP-TTLS and EAP-FAST only. Set the anonymous identity (user name) for 802.1X authentication. String	Yes
device.cfg site.cfg wireless.cfg	device.net.dot1x.enabled	Enable or disable 802.1X authentication. 0 1	Yes
device.cfg site.cfg wireless.cfg	device.net.dot1x.identity	Set the identity (user name) for 802.1X authentication. String	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg wireless.cfg	device.net.dot1x.method	Specify the 802.1X authentication method, where EAP-NONE means no authentication. EAP-None EAP-TLS EAP-PEAPv0-MSCHAPv2 EAP-PEAPv0-GTC EAP-TTLS-MSCHAPv2 EAP-TTLS-GTC EAP-FAST EAP-MD5	No
device.cfg site.cfg wireless.cfg	device.net.dot1x.password	Set the password for 802.1X authentication. This parameter is required for all methods except EAP-TLS. String	Yes
device.cfg site.cfg	device.net.etherModeLAN	Set the LAN port mode that sets the network speed over Ethernet. Polycom recommends that you do not change this setting. Auto 10HD 10FD 100HD 100FD 1000FD HD means half-duplex and FD means full duplex.	Yes
device.cfg site.cfg	device.net.etherModePC	Set the PC port mode that sets the network speed over Ethernet. Auto (default) Disabled—disables the PC port. 10HD 10FD 100HD 100FD 1000FD HD means half-duplex and FD means full duplex.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.net.etherStormFilter	1—DoS storm prevention is enabled and received Ethernet packets are filtered to prevent TCP/IP stack overflow caused by bad data or too much data. 0— DoS storm prevention is disabled.	Yes
device.cfg site.cfg	device.net.etherStormFilterPpsValue	Set the corresponding packets per second (pps) for storm filter and to control the incoming network traffic. 17 to 40 38 (default)	No
device.cfg site.cfg	device.net.etherStormFilterPpsValue.set	0 (default) - You cannot configure the device.net.etherStormFilterPpsValue parameter. 1 - You can configure the device.net.etherStormFilterPpsValue parameter.	No
device.cfg site.cfg	device.net.etherVlanFilter	VLAN filtering for VVX phones is done by the Linux operating system and it cannot be disabled. 0 1	Yes
device.cfg	device.net.ipAddress	Set the phone's IP address. This parameter is disabled when device.dhcp.enabled is set to 1. String	Yes
device.cfg site.cfg	device.net.IPgateway	Set the phone's default router. IP address	Yes
device.cfg site.cfg	device.net.lldpEnabled	0—The phone doesn't attempt to determine its VLAN ID. 1—The phone attempts to determine its VLAN ID and negotiate power through LLDP.	Yes
device.cfg site.cfg	device.net.lldpFastStartCount	Specify the number of consecutive LLDP packets the phone sends at the time of LLDP discovery, which are sent every one second. 5 (default) 3 to 10	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.net.subnetMask	Set the phone's subnet mask. This parameter is disabled when device.dhcp.enabled is set to 1. subnet mask	Yes
device.cfg site.cfg	device.net.vlanId	Set the phone's 802.1Q VLAN identifier. Null—No VLAN tagging. 0 to 4094	Yes
device.cfg site.cfg	device.prov.maxRedunServers	Set the maximum number of IP addresses to use from the DNS. 1 - 8	Yes
device.cfg site.cfg	device.prov.password	Set the password for the phone to log in to the provisioning server, which may not be required. If you modify this parameter, the phone re-provisions. The phone may also reboot if the configuration on the provisioning server has changed. string	Yes
device.cfg site.cfg	device.prov.redunAttemptLimit	Set the maximum number of attempts to attempt a file transfer before the transfer fails. When multiple IP addresses are provided by DNS, 1 attempt is considered to be a request sent to each server. 1 to 10	Yes
device.cfg site.cfg	device.prov.redunInterAttemptDelay	Set the number of seconds to wait after a file transfer fails before retrying the transfer. When multiple IP addresses are returned by DNS, this delay only occurs after each IP has been tried. 0 to 300	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.prov.serverName	Enter the IP address, domain name, or URL of the provisioning server followed by an optional directory and optional configuration filename. This parameter is used if (device.dhcp.enabled is 0), if the DHCP server does not send a boot server option, or if the boot server option is static (device.dhcp.bootSrvUseOpt is static). IP address Domain name string URL If you modify this parameter, the phone re-provisions. The phone also reboots if the configuration on the provisioning server has changed.	No
device.cfg site.cfg	device.prov.serverType	Set the protocol the phone uses to connect to the provisioning server. Active FTP is not supported for BootROM version 3.0 or later, and only implicit FTPS is supported. FTP (default) TFTP HTTP HTTPS FTPS	Yes
device.cfg site.cfg	device.prov.tagSerialNo	0—The phone's serial number (MAC address) is not included in the User-Agent header of HTTPS/HTTPS transfers and communications to the microbrowser and web browser. 1— the phone's serial number is included.	No
device.cfg site.cfg	device.prov.upgradeServer	Specify the URL or path for a software version to download to the device. On the Web Configuration Utility, the path to the software version you specify displays in the drop-down list on the Software Upgrade page. NULL (default) string 0 -255 characters	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.prov.user	The user name required for the phone to log in to the provisioning server (if required). If you modify this parameter, the phone re-provisions, and it may reboot if the configuration on the provisioning server has changed. string	No
device.cfg site.cfg	device.prov.ztpEnabled	Enable or disable Zero Touch Provisioning (ZTP). 0 1 For information, see Zero-Touch Provisioning: https://support.polycom.com/content/support/North_America/USA/en/support/voice/Zero_Touch_Provisioning/zero_touch_provisioning_solution.html.	No
device.cfg site.cfg	device.sec.configEncryption.key[1]	Set the configuration encryption key used to encrypt configuration files. string For more information, see the sectionConfiguration File Encryption.	Yes
device.cfg site.cfg	device.sec.coreDumpEncryption.enabled	Determine whether to encrypt the core dump or bypass the encryption of the core dump. 0—encryption of the core dump is bypassed. 1 (default)—the core dump is encrypted	No
device.cfg site.cfg	device.sec.TLS.customCaCert1 (TLS Platform Profile 1) device.sec.TLS.customCaCert2 (TLS Platform Profile 2)	Set the custom certificate to use for TLS Platform Profile 1 and TLS Platform Profile 2 and TLS Application Profile 1 and TLS Application Profile 2. The parameter device.sec.TLS.profile.caCertList must be configured to use a custom certificate. Custom CA certificate cannot exceed 4096 bytes total size. string PEM format	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	device.sec.TLS.customDeviceCert1.privateKey device.sec.TLS.customDeviceCert2.privateKey	Enter the corresponding signed private key in PEM format (X.509). Size constraint: 4096 bytes for the private key.	No
debug.cfg	device.sec.TLS.customDeviceCert1.publicCert device.sec.TLS.customDeviceCert2.publicCert	Enter the signed custom device certificate in PEM format (X.509). Size constraint: 8192 bytes for the device certificate.	No
device.cfg site.cfg	device.sec.TLS.customDeviceCert1.set device.sec.TLS.customDeviceCert2.set	Use to set the values for parameters device.sec.TLS.customDeviceCertX.publicCert and device.sec.TLS.customDeviceCertX.privateKey . Size constraints are: 4096 bytes for the private key, 8192 bytes for the device certificate. 0 (default) 1	No
device.cfg	device.sec.TLS.profile.caCertList1 （ TLS Platform Profile 1 ） device.sec.TLS.profile.caCertList2 （ TLS Platform Profile 2 ）	Choose the CA certificate(s) to use for TLS Platform Profile 1 and TLS Platform Profile 2 authentication: Builtin—The built-in default certificate BuiltinAndPlatform—The built-in and Custom #1 certificates BuiltinAndPlatform2—The built-in and Custom #2 certificates All—Any certificate (built in, Custom #1 or Custom #2) Platform1—Only the Custom #1 certificate Platform2—Only the Custom #2 certificate Platform1AndPlatform2—Either the Custom #1 or Custom #2 certificate	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.sec.TLS.profile.cipherSuite1 （ TLS Platform Profile 1 ） device.sec.TLS.profile.cipherSuite2 （ TLS Platform Profile 2 ）	Enter the cipher suites to use for TLS Platform Profile 1 and TLS Platform Profile 2 string	No
device.cfg site.cfg	device.sec.TLS.profile.cipherSuiteDefault1 （ TLS Platform Profile 1 ） device.sec.TLS.profile.cipherSuiteDefault2 （ TLS Platform Profile 2 ）	Determine the cipher suite to use for TLS Platform Profile 1 and TLS Platform profile 2. 0—The custom cipher suite is used. 1—The default cipher suite is used.	No
device.cfg site.cfg	device.sec.TLS.profile.deviceCert1 （ TLS Platform Profile 1 ） device.sec.TLS.profile.deviceCert2 （ TLS Platform Profile 2 ）	Choose the device certificate(s) for TLS Platform Profile 1 and TLS Platform Profile 2 to use for authentication. Builtin Platform1 Platform2	No
device.cfg site.cfg	device.sec.TLS.profileSelection.dot1x	Choose the TLS Platform Profile to use for 802.1X. PlatformProfile1 PlatformProfile2	No
device.cfg site.cfg	device.sec.TLS.profileSelection.provisioning	Set the TLS Platform Profile to use for provisioning. PlatformProfile1 PlatformProfile2	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.sec.TLS.profileSelection.syslog	Set the TLS Platform Profile to use for syslog. PlatformProfile1 PlatformProfile2	Yes
device.cfg site.cfg	device.sec.TLS.prov.strictCertCommonNameValidation	0 1 (default)—Provisioning server always verifies the server certificate for the commonName/SubjectAltName match with the server hostname that the phone is trying to connect.	No
device.cfg site.cfg	device.sec.TLS.syslog.strictCertCommonNameValidation	0 1—Syslog always verifies the server certificate for the commonName/SubjectAltName match with the server hostname that the phone is trying to connect.	No
device.cfg site.cfg	device.sntp.gmtOffset	Set the GMT offset—in seconds—to use for daylight savings time, corresponding to -12 to +13 hours. -43200 to 46800	No
device.cfg site.cfg	device.sntp.gmtOffsetcityID	Sets the correct time zone location description that displays on the phone menu and in the Web Configuration Utility. NULL (default) 0 to 126 For descriptions of all values, refer to Time Zone Location Description.	No
device.cfg site.cfg	device.sntp.serverName	Enter the SNTP server from which the phone obtains the current time. IP address Domain name string	No
device.cfg site.cfg	device.syslog.facility	Determine a description of what generated the log message. 0 to 23 For more information, see RFC 3164.	No
device.cfg site.cfg	device.syslog.prependMac	0 1—The phone's MAC address is prepended to the log message sent to the syslog server.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
device.cfg site.cfg	device.syslog.renderLevel	Specify the logging level for the lowest severity of events to log in the syslog. When you choose a log level, the log includes all events of an equal or greater severity level, but it excludes events of a lower severity level. 0 or 1—SeverityDebug(7). 2 or 3—SeverityInformational(6). 4—SeverityError(3). 5—SeverityCritical(2). 6—SeverityEmergency(0).	Yes
device.cfg site.cfg	device.syslog.serverName	Set the syslog server IP address or domain name string. IP address Domain name string	No
device.cfg site.cfg	device.syslog.transport	Set the transport protocol that the phone uses to write to the syslog server. None—Transmission is turned off but the server address is preserved. UDP TCP TLS	No

Related Links

Assign a VLAN ID Using DHCP on page 25

Configuration Parameters

Topics:

This section is a reference guide for configuration parameters available for UC Software features.

This section provides a description and permitted values of each configuration parameter.

Quick Setup Soft Key Parameters

The following table lists the parameters that configure Quick Setup soft key.

Quick Setup Soft Key Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.quickSetup.enabled	0 (default) - Disables the quick setup feature. 1 - Enables the quick setup feature.	No

Related Links

Test the Provisioning Settings on page 44

Background Image Parameters

The parameters listed in the following table control how background images display on the phones.

Background Image Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	bg.color.selection	Specify which type of background (w) and index (x) for that type is selected on reboot. The default selection is 1,1 the first solid background. Use w=1 and x=1 (1,1) to select the built-in image. Use w=2 and x= 1 to 4 to select one of the four solid backgrounds. Use w=3 and x= 1 to 6 to select one of the six background bm images You can set backgrounds for specific phone models by adding the model name, for example: bg.color.VVX500.selection , bg.color.VVX1500.selection Note that although the VVX 300 series phones use a grayscale background, you can use this parameter to set the background. 1,1 (default) w,x	No
features.cfg	bg.color.bm.x.name	Specify the name of the phone screen background image file including extension with a URL or file path of a BMP or JPEG image. Note: If the file is missing or unavailable, the built-in default solid pattern is displayed.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	bg.color.bm.x.em.name	Specify the name of the Expansion Module (EM) background image file including extension with a URL or file path of a BMP or JPEG image. Note: If the file is missing or unavailable, the built-in default solid pattern is displayed.	No

Bluetooth Parameters

The following table specifies the Bluetooth parameters for the VVX 600/601 phones.

Bluetooth Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
	bluetooth.pairedDeviceMemorySize	10 (default) 0 - 10	No

Per-Registration Call Parameters

Polycom phones support an optional per-registration feature that enables automatic call placement when the phone is off-hook.

The phones also support a per-registration configuration that determines which events cause the missed-calls counter to increment. You can enable/disable missed call tracking on a per-line basis.

In the following table, x is the registration number.

To view the list of maximum registrations for each phone model see Flexible Call Appearances.

Per-Registration Call Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`call.advancedMissedCalls.addToReceivedList`	Applies to calls on that are answered remotely. 0 (default) - Calls answered from the remote phone are not added to the local receive call list. 1 - Calls answered from the remote phone are added to the local receive call list.	No
`sip-interop.cfg`	`call.advancedMissedCalls.enabled`	Use this parameter to improve call handling. 1 (default) - Shared lines can correctly count missed calls. 0 - Shared lines may not correctly count missed calls.	No
`sip-interop.cfg`	`call.advancedMissedCalls.reasonCodes`	Enter a comma-separated list of reason code indexes interpreted to mean that a call should not be considered as a missed call. 200 (default)	No
`reg-advanced.cfg`	`call.autoAnswer.micMute`	1 (default) - The microphone is initially muted after a call is auto-answered. 0 - The microphone is active immediately after a call is auto-answered.	No
`reg-advanced.cfg`	`call.autoAnswer.ringClass`	The ring class to use when a call is to be automatically answered using the auto-answer feature. If set to a ring class with a type other than `answer` or `ring-answer` , the setting are overridden such that a ringtone of `visual` (no ringer) applies. ringAutoAnswer (default)	No
`reg-advanced.cfg`	`call.autoAnswer.ringTone`	Intercom (default) – Auto answer plays the intercom tone. doubleBeep – Auto answer plays the double-beep tone.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`reg-advanced.cfg`	`call.autoAnswer.SIP`	This parameter cannot be used with VVX 101, 150, or 201 phones. 0 (default) - Disable auto-answer for SIP calls. 1 - Enable auto-answer for SIP calls.	No
`reg-advanced.cfg`	`call.autoAnswer.ringTone`	Sets the auto-answer tone on the phone. intercom (default) – While auto answering a call, phone plays an intercom tone. doubleBeep – Phone plays the double beep tone.	No
`featurescfg`	`call.autoAnswerMenu.enable`	1 (default) - The autoanswer menu displays and is available to the user. 0 - The autoanswer menu is disabled and is not available to the user.	No
`sip-interop.cfg`	`call.BlindTransferSpecialInterop`	0 (default) - Do not wait for an acknowledgment from the transferee before ending the call. 1 - Wait for an acknowledgment from the transferee before ending the call.	No
`sip-interop.cfg`	`call.dialtoneTimeOut`	The time is seconds that a dial tone plays before a call is dropped. 60 (default) 0 - The call is not dropped.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	call.internationalDialing.enabled	Use this parameter to enable or disable the key tap timer that converts a double tap of the asterisk “*” symbol to the “+” symbol used to indicate an international call. 1 (default) - A quick double tap of “*” converts immediately to “+”. To enter a double asterisk “**”, tap “*” once and wait for the key tap timer to expire to enter a second “*”. 0 - You cannot dial”+” and you must enter the international exit code of the country you are calling from to make international calls. This parameter applies to all numeric dial pads on the phone including for example, the contact directory.	Yes
sip-interop.cfg, site.cfg	call.internationalPrefix.key	0 (default) 1	No
sip-interop.cfg	call.localConferenceEnabled	1 (default) - The feature to join a conference during an active call is enabled and you can establish conferences on the phone. 0 - The feature to join a conference during an active call is disabled. When you try to join the Conference, an ‘Unavailable' message displays.	Yes
sip-interop.cfg	call.offeringTimeOut	Specify a time in seconds that an incoming call rings before the call is dropped. 60 (default) 0 - No limit. Note that the call diversion, no answer feature takes precedence over this feature when enabled.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	call.playLocalRingBackBeforeEarlyMediaArrival	Determines whether the phone plays a local ring-back after receiving a first provisional response from the far end. 1 (default) - The phone plays a local ringback after receiving the first provisional response from the far end. If early media is received later, the phone stops the local ringback and plays the early media. 0 - No local ringback plays, and the phone plays only the early media received.	No
site.cfg	call.playLocalRingBackBeforeEarlyMediaArrival	0 (default) - URL mode is used for URL calls. 1 - Number mode is used for URL calls.	No
sip-interop.cfg	call.ringBackTimeOut	Specify a time in seconds to allow an outgoing call to remain in the ringback state before dropping the call. 60 (default) 0 - No limit.	Yes
sip-interop.cfg	call.showDialpadOnProceeding	0 (default) – The phone does not show the dialpad button while a placed call is outgoing. 1 – The phone displays the dialpad button while a placed call is outgoing.	No
sip-interop.cfg, site.cfg	call.stickyAutoLineSeize	0 - Dialing through the call list uses the line index for the previous call. Dialing through the contact directory uses a random line index. 1 - The phone uses sticky line seize behavior. This helps with features that need a second call object to work with. The phone attempts to initiate a new outgoing call on the same SIP line that is currently in focus on the LCD. Dialing through the call list when there is no active call uses the line index for the previous call. Dialing through the call list when there is an active call uses the current active call line index. Dialing through the contact directory uses the current active call line index.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg, site.cfg	call.stickyAutoLineSeize.onHookDialing	0 (default) If call.stickyAutoLineSeize is set to 1, this parameter has no effect. The regular stickyAutoLineSeize behavior is followed. If call.stickyAutoLineSeize is set to 0 and this parameter is set to 1, this overrides the stickyAutoLineSeize behavior for hot dial only. (Any new call scenario seizes the next available line.) If call.stickyAutoLineSeize is set to 0 and this parameter is set to 0, there is no difference between hot dial and new call scenarios. A hot dial occurs on the line which is currently in the call appearance. Any new call scenario seizes the next available line.	Yes
sip-interop.cfg	call.switchToLocalRingbackWithoutRTP	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.	No
site.cfg	call.teluri.showPrompt	1 (default) - Phone displays a pop-up box to either call or cancel the number when tel URI is executed. 0 - Phone does not display the pop-up box.	No
sip-interop.cfg	call.urlModeDialing	0 (default) - Disable URL dialing. 1 - Enable URL dialing.	Yes

Related Links

Flexible Call Appearances on page 381

Per-Registration Dial Plan Parameters

All of the parameters listed in the following table are per-registration parameters that you can configure instead of the general equivalent dial plan parameters.

Note that the per-registration parameters override the general parameters where x is the registration number, for example, `dialplan.x.applyToTelUriDial` overrides `dialplan.applyToTelUriDial` for registration x.

The maximum number of registrations (x) for each phone model is listed in the table Flexible Call Appearances under the column Registrations.

Per-Registration Dial Plan (Digit Map) Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`dialplan.userDial.timeOut`	Specify the time in seconds that the phone waits before dialing a number entered while the phone is on hook. Generic Base Profile (default) – 0 Lync Base Profile (default) – 4 0-99 seconds 0-99 seconds You can apply `dialplan.userDial.timeOut` only when its value is lower than `up.IdleTimeOut` .	No
`site.cfg`	`dialplan.x.applyToCallListDial`	Generic Base Profile (default) – 1 Lync Base Profile (default) – 0 0 - The dial plan does not apply to numbers dialed from the received call list or missed call list, including sub-menus for this line. 1 - The dial plan applies to numbers dialed from the received call list or missed call list, including sub-menus for this line.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dialplan.x.applyToDirectoryDial	Generic Base Profile (default) – 1 Lync Base Profile (default) – 0 0 - The dial plan is not applied to numbers dialed from the directory or speed dial, including auto-call contact numbers for this line. 1 - The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call contact numbers for this line.	Yes
site.cfg	dialplan.x.applyToForward	Generic Base Profile (default) – 1 Lync Base Profile (default) – 0 0 - The dial plan applies to forwarded calls for this line. 1 - The dial plan applies to forwarded calls for this line.	No
site.cfg	dialplan.x.applyToTelUriDial	0 1 (default)	Yes
site.cfg	dialplan.x.applyToUserDial	0 1 (default)	Yes
site.cfg	dialplan.x.applyToUserSend	0 1 (default)	Yes
site.cfg	dialplan.x.conflictMatchHandling	0 (default for Generic Profile) 1 (default for Skype Profile)	No
site.cfg	dialplan.x.digitmap.timeOut	Generic Base Profile (default) – 0 Lync Base Profile (default) – 4	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dialplan.x.digitmap	Generic Base Profile (default) - Null Lync Base Profile (default) - 4 string - max number of characters 100	Yes
site.cfg	dialplan.x.e911dialmask	Null (default) string - max number of characters 256	No
site.cfg	dialplan.x.e911dialstring	Null (default) string - max number of characters 256	No
site.cfg	dialplan.x.impossibleMatchHandling	0 (default) - Digits are sent to the call server immediately. 1 - A reorder tone is played and the call is canceled. 2 - No digits are sent to the call server until the Send or Dial key is pressed. 3 - No digits are sent to the call server until the Timeout configured by dialplan.userDial.timeOut .	Yes
site.cfg	dialplan.x.originaldigitmap	Null (default) string - max number of characters 2560	No
site.cfg	dialplan.x.removeEndOfDial	0 1 (default)	Yes
site.cfg	dialplan.x.routing.emergency.y.server.z	0 (default) 1 2 3 x, y, and z = 1 to 3	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	dialplan.x.routing.emergency.y.value	Null (default) string - max number of characters 64	Yes
site.cfg	dialplan.x.routing.server.y.address	Null (default) string - max number of characters 256	Yes
site.cfg	dialplan.x.routing.server.y.port	5060 (default) 1 to 65535	Yes
site.cfg	dialplan.x.routing.server.y.transport	DNSnaptr (default) TCPpreferred UDPOnly TLS TCPOnly	Yes

Related Links

Flexible Call Appearances on page 381

Local Contact Directory File Size Parameters

The following table lists the parameters you can configure to set the size of the local contact directory.

The maximum local directory size is limited based on the amount of flash memory in the phone and varies by phone model. Polycom recommends that you configure a provisioning server that allows uploads to ensure a back-up copy of the directory when the phone reboots or loses power.

Note that on the VVX 1500, the local directory is by default stored in the phone's non-volatile device settings and you have the option to use the phone's volatile RAM and set the maximum file size.

Local Contact Directory File Size Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	dir.local.nonVolatile.maxSize	Set the maximum file size of the local contact directory stored on the phone's non-volatile memory. VVX1500 = 100KB (default) 1 - 100KB	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`debug.cfg`	`dir.local.volatile`	0 (default) - The phone uses non-volatile memory for the local contact directory. 1 - Enables the use of volatile memory for the local contact directory.	No
`debug.cfg`	`dir.local.volatile.maxSize`	Sets the maximum file size of the local contact directory stored on the phone's volatile memory. VVX1500 = 200KB (default) 1 - 200KB	No

Related Links

User Profiles on page 270

Parameter Elements for the Local Contact Directory

The following table describes each of the parameter elements and permitted values that you can use in the local contact directory.

Local Contact Directory Parameter Elements

Element	Definition	Permitted Values
fn	The contact's first name.	UTF-8 encoded string of up to 40 bytes1
ln	The contact's last name.	UTF-8 encoded string of up to 40 bytes1
ct	Contact, Used by the phone to address a remote party in the same way that a string of digits or a SIP URL are dialed manually by the user. This element is also used to associate incoming callers with a particular directory entry. The maximum field length is 128 characters. Note: This field cannot be null or duplicated	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL

Element	Definition	Permitted Values
sd	Speed Dial Index, Associates a particular entry with a speed dial key for one-touch dialing or dialing.	VVX=Null, 1 to 9999 Polycom Trio=20
lb	The label for the contact. The label of a contact directory item is by default the label attribute of the item. If the label attribute does not exist or is Null, then the first and last names form the label. A space is added between first and last names. Note: For GENBAND, the Label element is shown as Nick Name, and is a mandatory, non-duplicate field.	UTF-8 encoded string of up to 40 bytes1
pt	Protocol, The protocol to use when placing a call to this contact.	SIP, H323, or Unspecified
rt	Ring Tone, When incoming calls match a directory entry, this field specifies the ringtone to be used.	Null, 1 to 21
dc	Divert Contact, The address to forward calls to if the Auto Divert feature is enabled.	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL

Element	Definition	Permitted Values
ad	Auto Divert, If set to 1, callers that match the directory entry are diverted to the address specified for the divert contact element. Note: If auto-divert is enabled, it has precedence over auto-reject.	0 or 1
ar	Auto Reject, If set to 1, callers that match the directory entry specified for the auto reject element are rejected. Note: If auto divert is also enabled, it has precedence over auto reject.	0 or 1
bw	Buddy Watching, If set to 1, this contact is added to the list of watched phones.	0 or 1
bb	Buddy Block, If set to 1, this contact is blocked from watching this phone.	0 or 1

Feature Activation/Deactivation Parameters

The feature parameters listed in the following table control the activation or deactivation of a feature at run time.

Feature Activation/Deactivation Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`feature.callCenterCallInformation.enable`	1 (default) - The phone displays a full-screen popup showing call information details. The popup closes after 30 seconds or you can press the **Exit** button to close it and return to the active call screen. 0 - The phone uses the active call screen and ACD call information is not available.	No
`features.cfg`	`feature.callCenterStatus.enabled`	0 (default) - Disable the status event threshold capability. 1 - Enable the status event threshold capability to display at the top of the phone screen.	No
`features.cfg`	`feature.enhancedCallDisplay.enabled`	0 (default) - The phone displays the protocol at the end of the called party identification (for example, 1234567 [SIP]). 1 - The phone displays the number only (for example, 1234567).	No
`features.cfg`	`feature.flexibleLineKey.enable`	0 (default) - Disables the Flexible Line Key feature. 1 - Enables the Flexible Line Key feature. Not available for the VVX 101, 150, 201, or 1500 business media phones.	No
`features.cfg`	`feature.nonVolatileRingerVolume.enabled`	1 (default) - User changes to the ringer volume are saved and maintained after the phone reboots. 0 - User changes to the ringer volume are reset to default after the phone reboots.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	feature.ringDownload.enabled	1 (default) - The phone downloads ringtones when starting up. 0 - The phone does not download ringtones when starting up.	Yes
features.cfg	feature.uniqueCallLabeling.enabled	0 (default) - Disable Unique Call Labeling. 1 - Enable Unique Call Labeling. Use `reg.x.line.y.label` to define unique labels.	Yes
features.cfg	feature.urlDialing.enabled	1 (default) - URL/name dialing is available from private lines, and unknown callers are identified on the display by their phone's IP address. 0 - URL/name dialing is not available.	No
features.cfg	reg.x.urlDialing.enabled	1 (default) - Enable dialing by URL for SIP registrations. 0 - Disable dialing by URL for SIP registrations.	

HTTPD Web Server Parameters

The phone contains a local Web Configuration Utility server for user and administrator features.

Note that several of these parameters can be used with Microsoft Skype for Business Server and the parameter values listed in the table Enable Web Configuration Utility have two default states: a generic default value for UC Software 5.1.0 and a different value when the phone is registered with Skype for Business Server. The following table lists the default values for both states where applicable.

The web server supports both basic and digest authentication. The authentication user name and password are not configurable for this release.

HTTPD Web Server Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	httpd.enabled	Base Profile = Generic 1 (default) - The web server is enabled. 0 - The web server is disabled. Base Profile = Skype 0 (default) - The web server is disabled. 1 - The web server is enabled.	Yes
site.cfg	httpd.cfg.enabled	Base Profile = Generic 1 (default) - The Web Configuration Utility is enabled. 0 - The Web Configuration Utility is disabled. Base Profile = Skype 0 (default) - The Web Configuration Utility is disabled. 1 - The Web Configuration Utility is enabled.	Yes
site.cfg	httpd.cfg.port	Port is 80 for HTTP servers. Take care when choosing an alternate port. 80 (default) 1 to 65535	Yes
site.cfg	httpd.cfg.secureTunnelPort	The port to use for communications when the secure tunnel is used. 443 (default) 1 to 65535	Yes
site.cfg	httpd.cfg.secureTunnelRequired	1 (default) - Access to the Web Configuration Utility is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed. 0 - Access to the Web Configuration Utility is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).	Yes

Home Screen Parameters

The following table lists parameters that configure the phone's Home screen display.

Home Screen Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`homeScreen.application.enable`	1 (default) - Enable display of the Applications icon on the phone Home screen. 0 - Enable display of the Applications icon on the phone Home screen.	No
`features.cfg`	`homeScreen.calendar.enable`	1 (default) - Enable display of the Calendar icon on the phone Home screen. 0 - Disable display of the Calendar icon on the phone Home screen.	No
`features.cfg`	`homeScreen.directories.enable`	1 (default) - Enable display of the Directories menu icon on the phone Home screen. 0 - Disable display of the Directories menu icon on the phone Home screen.	No
`features.cfg`	`homeScreen.doNotDisturb.enable`	1 (default) - VVX 0 (default) - Polycom Trio 1 - Enable display of the DND icon on the phone Home screen. 0 - Disable display of the DND icon on the phone Home screen.	No
`features.cfg`	`homeScreen.features.enable`	1 (default) - Enable display of the Features menu icon on the VVX 1500 business media phone's Home screen. 0 - Disable display of the Features menu icon on the VVX 1500 business media phone's Home screen.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	homeScreen.forward.enable	1 (default) - Enable display of the call forward icon on the phone Home screen. 0 - Disable display of the call forward icon on the phone Home screen.	No
features.cfg	homeScreen.messages.enable	1 (default) - Enable display of the Messages menu icon on the phone Home screen. 0 - Disable display of the Messages menu icon on the phone Home screen.	No
features.cfg	homeScreen.newCall.enable	1 (default) - Enable display of the New Call icon on the phone Home screen. 0 - Disable display of the New Call icon on the phone Home screen.	No
features.cfg	homeScreen.redial.enable	1 (default) - VVX 0 (default) - Polycom Trio 1 - Enable display of the Redial menu icon on the phone Home screen. 0 - Disable display of the Redial menu icon on the phone Home screen.	No
features.cfg	homeScreen.settings.enable	1 (default) - Enable display of the Settings menu icon on the phone Home screen. 0 - Disable display of the Settings menu icon on the phone Home screen.	No
features.cfg	homeScreen.status.enable	1 (default) - Enable display of the Status menu icon on the VVX 1500 business media phone's Home screen. 0 - Disable display of the Status menu icon on the VVX 1500 business media phone's Home screen.	No

Key Mapping Parameters

The following table lists parameters that enable you to change the default functions of your phone's keypad keys, a process known as remapping.

If you want to change the default function of a key, you must specify the phone model number, the key you want to change, and a new function for the key.

- For a list of products and their model codes, see System and Model Names.
- To find the key number, location of the key on each phone model, and default key functions, refer to Defining the Phone Key Layout.
- For a list of parameter values you can assign as functions to a phone key, refer Keypad Key Functions.

Caution: Polycom does not recommend remapping or changing the default functions of the keys on your phone.

Key Mapping Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`key.x.y.function.prim`	Specify a phone model, key number, and function. x can be one of the VVX 300 series, 400 series, 500 series, 600 series, or VVX1500 phones. y can be one key number.	Yes

Related Links

Defining the Phone Key Layout on page 487

Keypad Key Functions

The following table lists the functions that are available for phone keys.

Keypad Key Functions

Answer	Dialpad2	Handsfree	MyStatus	SpeedDialMenu
ArrowDown	Dialpad3	Headset	Null	Talk
ArrowLeft	Dialpad4	Hold	Offline	Video
ArrowRight	Dialpad5	Home	Redial	VolDown
ArrowUp	Dialpad6	Line2	Release	VolUp
Back	Dialpad8	Line3	Select	

BuddyStatus	Dialpad9	Line4	Setup
CallList	DialpadStar	Line5	SoftKey1
Conference	DialPound	Line6	SoftKey2
Delete	Directories	Messages	SoftKey3
Dialpad0	DoNotDisturb	Menu	SoftKey4
Dialpad1	Green	MicMute	SpeedDial

Example Custom Key Configurations

This section provides several custom key configuration examples.

Remap the Volume Up Key to Answer a Call

You can remap the volume up key.

Procedure

1. Update the configuration file as follows: `key.VVX300.6.function.prim="Answer"`
2. To remap the volume down key to launch the Settings menu on the VVX 300 using a macro:
3. Update the configuration file as follows:
   - `key.VVX300.7.function.prim="$Msetting$"`
   - `efk.efklist.1.action.string="$FSetup$"`
   - `efk.efklist.1.mname="setting"`
   - `efk.efklist.1.status="1"`

Remap the Mute Key to Forward a Call

You can remap the mute key to forward a call.

Procedure

1. Update the configuration file as follows: `key.VVX500.18.function.prim="$FDivert$"`

Remap the Transfer Key to Lock the Phone

You can remap the transfer key.

Procedure

1. Update the configuration file as follows: `key.37.function.prim="$FLockPhone$"`

Remap the Redial Key

You can remap the redial key.

Procedure

1. Update the configuration file as follows:

```
key.36.function.prim="http://vanoem02.vancouver.polycom.com:8080/
MicroBrowserTest.html"
```

Feature License Parameters

The parameters listed in the next table enable you to configure the feature licensing system.

Once the license is installed on a phone, it cannot be removed.

Feature License Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	license.polling.time	Specifies the time (using the 24-hour clock) to check if the license has expired. 02:00 (default 00:00 - 23:59	Yes

Chord Parameters

Chord-sets are the sound effect building blocks that use synthesized audio instead of sampled audio.

Most call progress and ringer sound effects are synthesized. A chord-set is a multi-frequency note with an optional on/off cadence, and can contain up to four frequency components generated simultaneously, each with its own level.

Three chord sets are supported: callProg , misc , and ringer . Each chord set has different chord names, represented by x in the following table.

For callProg , x can be one of the following chords:

dialTone, busyTone, ringback, reorder, stutter_3, callWaiting, callWaitingLong, howler, recWarning, stutterLong, intercom, callWaitingLong, precedenceCallWaiting, preemption, precedenceRingback, or spare1 to spare6.

For misc , x can be one of the following chords:

- spare1 to spare9

For ringer , x can be one of the following chords:

- ringback, originalLow, originalHigh , or spare1 to spare19

Chord Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
region.cfg	tone.chord.callProg.x.freq.y tone.chord.misc.x.freq.y tone.chord.ringer.x.freq.y	Frequency (in Hertz) for component y. Up to six chord-set components can be specified (y=1 to 6). 0-1600 0-1600 0-1600	No
region.cfg	tone.chord.callProg.x.level.y tone.chord.misc.x.level.y tone.chord.ringer.x.level.y	Level of component y in dBm0. Up to six chord-set components can be specified (y=1 to 6). -57 to 3 -57 to 3 -57 to 3	No
region.cfg	tone.chord.callProg.x.onDur tone.chord.misc.x.onDur tone.chord.ringer.x.onDur	On duration (length of time to play each component) in milliseconds. 0=infinite positive integer positive integer positive integer	No
region.cfg	tone.chord.callProg.x.offDur tone.chord.misc.x.offDur tone.chord.ringer.x.offDur	Off duration (the length of silence between each chord component) in milliseconds 0=infinite positive integer positive integer positive integer	No
region.cfg	tone.chord.callProg.x.repeat tone.chord.misc.x.repeat tone.chord.ringer.x.repeat	Number of times each ON/OFF cadence is repeated. 0=infinite positive integer positive integer positive integer	No

Message Waiting Parameters

The next table lists parameters you can use to configure the message-waiting feature, which is supported on a per-registration basis.

The maximum number of registrations (x) for each phone model is listed in the table Flexible Call Appearances under the column Registrations.

Message Waiting Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`sip-interop.cfg`	`msg.bypassInstantMessage`	0 (default) -Displays the menus Message Center and Instant Messages on pressing Messages or MSG key. 1 - Bypasses these menus and goes to voicemail.	No
`sip-interop.cfg`	`msg.mwi.x.led`	0 (default) - Red MWI LED does not flash when there are new unread messages for the selected line. 1 - The LED flashes as long as there are new unread voicemail messages for any line in which this is parameter is enabled. Also, x is an integer referring to the registration indexed by reg.x.	No

Ethernet Interface MTU Parameters

The parameters listed in this section control the Ethernet interface maximum transmission unit (MTU) on all VVX phones.

Ethernet Interface MTU Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
`site.cfg`	`net.interface.mtu`	Configures the Ethernet or Wi-Fi interface maximum transmission unit (MTU) on VVX phones and Polycom Trio system. 1496 (default) 800 - 1500 This parameter affects the LAN port and the PC port.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	net.interface.mtu6	Specifies the MTU range for IPv6. 1500 (default) 1280 - 1500	No
sip-interop.cfg	net.lldp.extenedDiscovery	Specifies the duration of time that LLDP discovery continues after sending the number of packets defined by the parameter lldpFastStartCount . 0 (default) 0 - 3600 The LLDP packets are sent every 5 seconds during this extended discovery period.	No

Presence Parameters

The next table lists parameters you can configure for the presence feature.

Note that the parameter pres.reg is the line number used to send SUBSCRIBE. If this parameter is missing, the phone uses the primary line to send SUBSCRIBE.

Presence Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	pres.idleTimeoutoffHours.enabled	1 (default) - Enables the off hours idle timeout feature. 0 - Disables the off hours idle timeout feature.	No
features.cfg	pres.idleTimeout.officeHours.enabled	1 (default) - Enables the office hours idle timeout feature 0 - Disables the office hours idle timeout feature	No

Provisioning Parameters

The parameters listed in the next table control the provisioning server system for your phones.

Provisioning Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.autoConfigUpload.enabled	1 (default) - Enables the automatic upload of configuration files from the phone or Web configuration utility to the provisioning server. 0 - Disabled the automatic upload of configuration files from the phone or Web configuration utility to the provisioning server.	No
site.cfg	prov.configUploadPath	Specifies the directory path where the phone uploads the current configuration file. Null (default) String	No
site.cfg	prov.login.lcCache.domain	The user's domain name to sign-in. Null (default) String	No
site.cfg	prov.login.lcCache.user	The user's sign-in name to login. Null (default) String	No
site.cfg	prov.login.password.encodingMode	The default encoding mode for the text in the Password field on the User Login screen. 123 (default) Alphanumeric	No
site.cfg	prov.login.userId.encodingMode	The default encoding mode for the text in the User ID field on User Login screen. Abc (default) Alphanumeric	No
region.cfg	prov.loginCredPwdFlushed.enabled	1 (default) - Resets the password field when the user logs in or logs out. 0 - Does not reset the password field when the user logs in or logs out.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	prov.startupCheck.enabled	1 (default) - The phone is provisioned on startup. 0 - The phone is not provisioned on startup.	No
site.cfg	prov.quickSetup.limitServerDetails	0 (default) - Provide all the necessary details for the given fields. 1 - Enter only the user name and password fields. Other details are taken from ztp/dhcp (option66).	No

Configuration Request Parameters

The parameters listed in the following table configure the phone's behavior when a request for restart or reconfiguration is received.

Configuration Request Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	request.delay.type	Specifies whether the phone should restart or reconfigure. call (default) - The request will be executed when there are no calls. audio - The request will be executed when there is no active audio.	Yes

General Security Parameters

The parameters listed in the next table configure security features of the phone.

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General Security Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	sec.tagSerialNo	0 (default) - The phone does not display the serial number. 1 - The phone displays the serial number through protocol signaling.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	sec.uploadDevice.privateKey	0 (default) - While generating the Certificate Signing Request from the phone, the device private key is not uploaded to provisioning server. 1 - The device private key is uploaded to provisioning server along with the CSR.	No

SRTP Parameters

As per RFC 3711, you cannot turn off authentication of RTCP.

The next table lists SRTP parameters.

SRTP Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	sec.srtp.answerWithNewKey	1 (default) - Provides a new key when answering a call. 0 - Does not provide a new key when answering the call.	No
sip-interop.cfg	sec.srtp.key.lifetime	Specifies the lifetime of the key used for the cryptographic parameter in SDP. Null (default) - 0 - The master key lifetime is not set. Positive integer minimum 1024 or power of 2 notation - The master key lifetime is set. Setting this parameter to a non-zero value may affect the performance of the phone.	Yes
sip-interop.cfg	sec.srtp.mki.enabled	0 (default) - The phone sends two encrypted attributes in the SDP, one with MKI and one without MKI when the base profile is set as Generic. 1 - The phone sends only one encrypted value without MKI when the base profile is set as Skype.	Yes
sip-interop.cfg	sec.srtp.mki.startSessionAtOne	0 (default) - The phone uses MKI value of 1. 1 - The MKI value increments for each new crypto key.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	sec.srtp.padRtpToFourByteAlignment	0 (default) - The RTP packet padding is not required when sending or receiving video. 1 - The RTP packet padding is required when sending or receiving video.	Yes
sip-interop.cfg	sec.srtp.simplifiedBestEffort	1 (default) - The SRTP is supported with Microsoft Description Protocol Version 2.0 Extensions. 0 - The SRTP is not supported with Microsoft Description Protocol Version 2.0 Extensions.	No

DHCP Parameters

Enables you to configure how the phone reacts to DHCP changes.

DHCP Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.dhcp.releaseOnLinkRecovery	Specifies whether or not a DHCP release occurs. 1 (default) - Performs a DHCP release after the loss and recovery of the network. 0 - No DHCP release occurs.	No

Domain Name System (DNS) Parameters

Allows you to set Domain Name System (DNS).

However, values set using DHCP have a higher priority, and values set using the <device/> parameter in a configuration file have a lower priority.

Domain Name System (DNS) Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.dns.server	Phone directs DNS queries to this primary server. NULL (default) IP address	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.dns.altServer	Phone directs DNS queries to this secondary server. NULL (default) IP address	Yes
site.cfg	tcpIpApp.dns.domain	Specifies the DNS domain for the phone. NULL (default) String	Yes

TCP Keep-Alive Parameters

Allows you to configure TCP keep-alive on SIP TLS connections; the phone can detect a failure quickly (in minutes) and attempt to re-register with the SIP call server (or its redundant pair).

TCP Keep-Alive Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.keepalive.tcp.idleTransmitInterval	Specifies the amount of time to wait (in seconds) before sending the keep-alive message to the call server. Range is 10 to 7200. 30 (Default) If this parameter is set to a value that is out of range, the default value is used. On VVX phones and the SoundStructure VoIP interface, specifies the number of seconds TCP waits between transmission of the last data packet and the first keep-alive message.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.keepalive.tcp.noResponseTransmitInterval	Specifies the amount of idle time between the transmission of the keep-alive packets the TCP stack waits on VVX phones and the SoundStructure VoIP interface. This applies whether or not the last keep-alive was acknowledged. If no response is received to a keep-alive message, subsequent keep-alive messages are sent to the call server at this interval (every x seconds). Range is 5 to 120.	No
site.cfg	tcpIpApp.keepalive.tcp.sip.persistentConnection.enable[1]	Specifies whether the TCP socket connection remains open or closes. 0 (Default) - The TCP socket opens a new connection when the phone tries to send any new SIP message and closes after one minute. 1 - The TCP socket connection remains open.	Yes
site.cfg	tcpIpApp.keepalive.tcp.sip.tls.enable	Specifies whether to disable or enable TCP keep-alive for SIP signaling connections. 0 (Default) - Disables TCP keep-alive for SIP signaling connections that use TLS transport. 1 - Enables TCP keep-alive for SIP signaling connections that use TLS transport.	No

File Transfer Parameters

Allows you to configure file transfers from the phone to the provisioning server.

File Transfer Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	tcpIpApp.fileTransfer.waitForLinkIfDown	Specifies whether a file transfer from the FTP server is delayed or not attempted. 1 (Default) - File transfer from the FTP server is delayed until Ethernet comes back up. 0 - File transfer from the FTP server is not attempted.	No

User Preferences Parameters

Sets phone user preferences.

User Preferences Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.25mm	Specifies whether to use a mobile phone or a PC to connect to the 2.5mm audio port on a conference phone. 1 (Default) - Mobile phone 2 - PC	No
features.cfg	up.accessibilityFeatures	Specifies whether to display accessibility features or not. 0 (Default) - Accessibility features are disabled. 1 - Screen background flashes orange for incoming calls. For VVX 1500 only.	No

Templat e	Parameter	Permitted Values	Change Causes Restart or Reboot
`features.cfg`	`up.backlight.idleIntensity`	Brightness of the LCD backlight when the phone is idle. Range is 0 to 3. 1 (Default) - Low 0 2 - Medium 3 - High VVX 300/301/310/311 = 0, 1, 2, 3 All other phones = 1, 2, 3 If this setting is higher than active backlight brightness (`onIntensity`), the active backlight brightness is used.	No
`features.cfg`	`up.backlight.onIntensity`	Brightness of the LCD backlight when the phone is active (in use). Range is 0 to 3. 3 (Default) - High 1 - Low 2 - Medium VVX 300/301/310/311 = 0, 1, 2, 3 All other phones = 1, 2, 3	No
`features.cfg`	`up.backlight.timeout`	Number of seconds to wait before the backlight dims from the active intensity to the idle intensity. Range is 5 to 60. 40 (default)	No
`features.cfg`	`up.basicSettings.networkConfigEnabled`	Specifies whether **Network Configuration** is shown or not shown under the **Basic Settings** menu. 0 (default) - **Network Configuration** is not shown under **Basic Settings**. 1 - **Basic Settings** menu shows **Network Configuration** with configurable network options for the user without administrator rights.	No
`features.cfg`	`up.DIDFormat`	NumberAndExtension (default) – Display the DID number and extension. NumberOnly – Display the DID number on the phone screen.	No

Templat e	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.cfgWarningsEnabled	Specifies whether a warning displays on a phone or not. 0 (Default) - Warning does not display. 1 - Warning is displayed on the phone if it is configured with pre-UC Software 3.3.0 parameters.	No
em.cfg	up.em.linkalivecheck.enabled	Specifies whether a host VVX phone pings expansion modules or not. 0 (Default) - Host VVX phone does not ping the expansion modules. 1 - Host VVX phone periodically sends ping packets to the expansion modules.	No
features.cfg	up.handsetModeEnabled	Enable or disable the handset port. 1 (Default) 0	No
debug.cfg	up.headsetAlwaysUseIntrinsicRinger	1 (Default) - USB headset uses the intrinsic ringer mixed with DSP ringer when the sound effect destination is the USB headset. 0	No
features.cfg	up.headsetMode	Sets the preferred audio mode of handsfree or headset. 0 (Default) - Handsfree mode is used by default instead of the handset. 1 - Headset is used as the preferred audio mode after the headset key is pressed for the first time, until the headset key is pressed again.	No
features.cfg	up.hearingAidCompatibility.enabled	Specifies whether audio Rx equalization is enabled or disabled. 0 (Default) - Audio Rx equalization is enabled. 1 - Phone audio Rx (receive) equalization is disabled for hearing aid compatibility.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.idleBrowser.enabled	Specifies if the idle browser is enabled or disabled. 0 (Default) - Idle browser is disabled. 1 - Idle browser is enabled. If the parameter up.prioritizeBackgroundMenuItem.enabled is set to 1, displays the background or the idle browser on the phone menu.	No
features.cfg	up.idleStateView	Sets the phone default view. 0 (Default) - Call/line view is the default view. 1 - Home screen is the default view.	Yes
sip-interop.cfg	up.idleTimeout	Set the number of seconds that the phone is idle for before automatically leaving a menu and showing the idle display. During a call, the phone returns to the Call screen after the idle timeout. 40 seconds (default) 0 to 65535 seconds	Yes
features.cfg	up.IdleViewPreferenceRemoteCalls	Determines when the phone displays the idle browser. 0 (Default) - Phone with only remote calls active, such as on a BLF monitored line, is treated as in the idle state and the idle browser displays. 1 - Phone with only remote calls active, such as on a BLF monitored line, is treated as in the active state and the idle browser does not display.	Yes
sip-interop.cfg	up.lineKeyCallTerminate	Specifies whether or not you can press the line key to end an active call. 0 (Default) - User cannot end an active call by pressing the line key. 1 - User can press a line key to end an active call.	No
sip-interop.cfg	up.numberFirstCID	Specifies what is displayed first on the **Caller ID** display. 0 (Default) - **Caller ID** display shows the caller's name first. 1 - Caller's phone number is shown first.	Yes

Templat e	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.numOfDisplayColumns	Sets the maximum number of columns the VVX 500/501, 600/601, or Polycom Trio solution display. Set the maximum number of columns that phones display. Range is 0 to 4. VVX 500/501 = 3 (Default) VVX 600/601 = 4 (Default) Polycom Trio=3 (Default) 0 - Phones display one column.	Yes
features.cfg	up.offHookAction.none	0 (Default) 1 - When the user lifts the handset, the phone does not seize the line and the ringer continues until the user takes further action.	Yes
features.cfg	up.OffHookIdleBrowserView.enabled	Specifies whether or not to display the idle browser on screen after the phone goes off hook. 0 (Default) - Idle browser does not display on screen after the phone goes off hook. 1 - Idle browser continues to display on screen after the phone goes off hook.	No
features.cfg	up.oneTouchDirectory	Displays the **Address Book** icon on the main menu and the **Skype for Business Directory** search option. 1 (Default) 0	No
features.cfg	up.osdIncomingCall.Enabled	Specifies whether or not to display full screen popup or OSD for incoming calls. 1 (Default) - Full screen popup or OSD for incoming calls displays. 0 - Full screen popup or OSD for incoming calls does not display.	No
sip-interop.cfg	up.prioritizeBackgroundMenuItem.enabled	User can choose whether or not the phone background should take priority over the idle browser. 1 (Default) - If up.idleBrowser.enabled is set to 1, this parameter can be set to 1 to display a Prioritize Background menu to the user.	Yes

Templat e	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	up.ringer.minimumVolume	Configure the minimum ringer volume. This parameter defines how many volume steps are accessible below the maximum level by the user. 16 (Default) - Full 16 steps of volume range are accessible. 0 - Ring volume is not adjustable by the user and the phone uses maximum ring volume. Example: Upon bootup, the volume is set to ½ the number of configured steps below the maximum (16). If the parameter is set to 8 on bootup, the ringer volume is set to 4 steps below maximum.	No
features.cfg	up.screenSaver.enabled	0 (Default) - Screen saver feature is disabled. 1 - Screen saver feature is enabled. If a USB flash drive containing images is connected to the phone, and the idle browser is not configured, a slide show cycles through the images from the USB flash drive when the screen saver feature is enabled. The images must be stored in the directory on the flash drive specified by up.pictureFrame.folder. The screen saver displays when the phone has been in the idle state for the amount of time specified by up.screenSaver.waitTime.	No
features.cfg	up.screenSaver.type	Choose the type of screen saver to display. 0 (Default) - Phone screen saver displays default images. 2 - Phone screen saver displays the idle browser. You can use this parameter with the VVX 300 and 400 series phones.	No
features.cfg	up.screenSaver.waitTime	Number of minutes that the phone waits in the idle state before the screen saver starts. Range is 1 to 9999 minutes. 15 (Default)	No
features.cfg	up.simplifiedSipCallInfo	0 (Default) - 1 - Displayed host name is trimmed for both incoming and outgoing calls and the protocol tag/information is not displayed for incoming and outgoing calls.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	up.softkey.transferTypeOption.enabled	1 (default) -The user can change the transfer type from consultative to blind and vice versa using a soft key after the user has initiated a transfer, but before completing the call to the far end. 0 - There is no option to change from consultative to blind and blind to consultative when the user is in dial prompt after pressing the Transfer soft key.	No
features.cfg	up.status.message.flash.rate	Controls the scroll rate of the status bar on VVX 250, 350, and 450 business IP phones and 300 and 400 series business media phones. Range is 2 to 8 seconds. 2 seconds (Default)	No
features.cfg	up.showDID	AllScreens (default) – Display the DID number on all the screens. None – Disable DID number on phone. LockedScreen – Display the DID number on the lock screen. StatusScreen – Display the DID number on the Status screen/Idle screen. IncomingOSD – Display the DID number on the incoming On Screen Display (OSD) screen. LockedScreenIncomingOSD – Display the DID number on the lock and incoming OSD screen. LockedAndStatusScreen – Display the DID number on the lock and Status/Idle screen. StatusScreenIncomingOSD – Display the DID number on the incoming OSD and Status/Idle screen.	No

Templat e	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	up.warningLevel	Line keys block display of the background image. All warnings are listed in the **Warnings** menu. 0 (Default) - The phone's warning icon and a pop-up message display on the phone for all warnings. 1 - Warning icon and pop-up messages are only shown for critical warnings. 2 - Phone displays a warning icon and no warning messages. For all the values, all warnings are listed in the **Warnings** menu. Access to the **Warnings** menu varies by phone model: VVX 1500 - **Menu** > **Status** > **Diagnostics** > **Warnings** VVX phones - **Settings** > **Status** > **Diagnostics** > **Warnings**	Yes
features.cfg	up.welcomeSoundEnabled	1 (Default) - Welcome sound is enabled and played each time the phone reboots. 0 - Welcome sound is disabled. To use a welcome sound you must enable the parameter up.welcomeSoundEnabled and specify a file in saf.x. The default UC Software welcome sound file is Welcome.wav.	Yes
features.cfg	up.welcomeSoundOnWarmBootEnabled	0 (Default) - Welcome sound is played when the phone powers on (cold boot), but not after it restarts or reboots (warm boot). 1 - Welcome sound plays each time the phone powers on, reboots, or restarts.	Yes
features.cfg	up.display.showFullCallerID	Phone displays the caller ID. 0 (default) – Phone displays the caller ID on the first line. 1 – Phone displays the caller ID on the second line.	No

Upgrade Parameters

Specify the URL of a custom download server and the Polycom UC Software download server when you want the phone to check when to search for software upgrades.

Upgrade Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	upgrade.custom.server.url	The URL of a custom download server. URL (default) - NULL	No
site.cfg	upgrade.plcm.server.url	The URL of the Polycom UC Software software download. URL - http://downloads.polycom.com/ voice/software/	No

Voice Parameters

The parameters listed in the following tables configure phone audio.

Voice Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.rxPacketFilter	Define a high-pass filter to improve sound intelligibility when the phone receives narrow band signals. Narrow band signals occur when a narrow band codec is in use, such as G.711mu, G.711A, G.729AB, iLBC, and some Opus and SILK variants. 0 (default) - Pass through. 1 - 300 Hz high-pass. 2 - 300 Hz high-pass with pre-emphasis. Use this value with G.729.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot

site.cfg	voice.txPacketDelay	Null (default)
		normal, Null - Audio parameters are not changed.
		low - If there are no precedence conflicts, the following changes are made:
		voice.codecPref.G722="1"
		voice.codecPref.G711Mu="2"
		voice.codecPref.G711A="3"
		voice.codecPref.<OtherCodecs>=""
		voice.audioProfile.G722.payloadSize="10"
		voice.audioProfile.G711Mu.payloadSize= "10"
		voice.audioProfile.G711A.payloadSize= "10"
		voice.aec.hs.enable="0"
		voice.ns.hs.enable="0"
			Yes

site.cfg	voice.txPacketFilter	Null (default) 0 - Tx filtering is not performed. 1 - Enables Narrowband Tx high pass filter.	Yes

Acoustic Echo Suppression (AES) Parameters

Use these parameters to control the speakerphone acoustic echo suppression (AES).

These parameters remove residual echo after AEC processing. Because AES depends on AEC, enable AES only when you also enable AEC using voice.aec.hd.enable .

Acoustic Echo Suppression Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	voice.aes.hs.enable	1 (default) - Enables the handset AES function. 0 - Disables the handset AES function.	No

Comfort Noise Parameters

Use these parameters to configure the addition and volume of comfort noise during conferences.

Comfort Noise Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
debug.cfg	voice.cn.hf.enable	1 (default) - Adds comfort noise added into the Tx path for hands-free operation. 0 - Comfort noise not added. Far end users should use this feature when they find the phone to be 'dead', as the near end user stops talking.	No
debug.cfg	voice.cn.hf.attn	35 (default) - quite loud 0 - 90 Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hf.enabled` is 1.	No
debug.cfg	voice.cn.hd.enable	0 (default) - Comfort noise is not added into the Tx path for the headset. 1 - Adds comfort noise into the Tx path for the headset. Far end users should use this feature when they find the phone to be 'dead', as the near end user stops talking.	No
debug.cfg	voice.cn.hd.attn	30 (default) - quite loud 0 - 90 Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hd.enabled` is 1.	No
debug.cfg	voice.cn.hs.enable	0 (default) - Comfort noise is not added into the Tx path for the handset. 1 - Adds comfort noise is added into the Tx path for the headset. Far end users should use this feature when they find the phone to be 'dead', as the near end user stops talking.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.cn.hs.attn	35 (default) - quite loud 0 - 90 Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when voice.cn.hs.enabled is 1.	No
site.cfg	voice.vadRxGain	Tunes VAD or CNG interoperability in a multi-vendor environment. 0 (default) -20 to +20 dB The specified gain value in dB is added to the noise level of an incoming VAD or CNG packet, when in a narrow band call. When tuning in multi-vendor environments, the existing Polycom to Polycom phone behavior can be retained by setting voice.vadTxGain = –voice.vadRxGain. This parameter is ignored for HD calls.	No
site.cfg	voice.vadTxGain	Tunes VAD or CNG interoperability in a multi-vendor environment. 0 (default) -20 to +20 dB The specified gain value in dB is added to the noise level of an incoming VAD or CNG packet, when in a narrow band call. This causes the noise level to synthesize at the local phone to change by the specified amount. When tuning in multi-vendor environments, the existing Polycom to Polycom phone behavior can be retained by setting voice.vadTxGain = –voice.vadRxGain. This parameter is ignored for HD calls.	No

Handset Parameters

The parameters listed in this section control the level of sidetone on handsets of VVX phones.

Handset Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	voice.handset.st	Adjust the handset sidetone level from the default in 1 decibel (dB) increments. 0 (default) -12 to +12	No

Headset Parameters

The parameters listed in this section control the level of sidetone on headsets connected to VVX phones.

Headset Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	voice.headset.st	Adjust the headset sidetone level from the default in 1 decibel (dB) increments. 0 (default)	No

Line Automatic Gain Control Parameters

The following parameters control audio level settings for phone handset and headset.

Line Automatic Gain Control Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	voice.lineAgc.hs.enable	0 (default) - Disables the line automatic gain control is on the handset. 1 - Enables the line automatic gain control is on the handset. This parameter is supported by the VVX 300 series, 400 series, 500 series, and 600 series business media phones and VVX 250, 350, and 450 business IP phones.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	voice.lineAgc.hf.enable	This parameter applies to the VVX 300 series, 400 series, 500 series, and 600 series business media phones and VVX 250, 350, and 450 business IP phones. 1 (default) - Enable the line automatic gain control on the handsfree speakerphone. 0 - Disable the line automatic gain control on the handsfree speakerphone.	Yes
techsupport.cfg	voice.lineAgc.hd.enable	0 (default) - Disables the line automatic gain control on the headset. 1 - Enables the line automatic gain control is on the headset. This parameter applies to the VVX 300 series, 400 series, 500 series, and 600 series business media phones and VVX 250, 350, and 450 business IP phones.	Yes

Voice Jitter Buffer Parameters

The following table lists the jitter buffer parameters for wired network interface voice traffic and push-to-talk interface voice traffic.

Voice Jitter Buffer Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.rxQoS.avgJitter	The average jitter in milliseconds for wired network interface voice traffic. 20 (default) 0 to 80 avgJitter The wired interface minimum depth will be automatically configured to adaptively handle this level of continuous jitter without packet loss.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.rxQoS.maxJitter	The average jitter in milliseconds for wired network interface voice traffic. 240 (default) 0 to 320 maxJitter The wired interface jitter buffer maximum depth will be automatically configured to handle this level of intermittent jitter without packet loss. Actual jitter above the average but below the maximum may result in delayed audio play out while the jitter buffer adapts, but no packets will be lost. Actual jitter above the maximum value will always result in packet loss. If legacy voice.audioProfile.x.jitterBuffer.* parameters are explicitly specified, they will be used to configure the jitter buffer and these voice.rxQoS parameters will be ignored.	Yes
site.cfg	voice.rxQoS.ptt.avgJitter	The average jitter in milliseconds for IP multicast voice traffic. 150 (default) 0 - 200 avgJitter The PTT/Paging interface minimum depth will be automatically configured to adaptively handle this level of continuous jitter without packet loss.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	voice.rxQoS.ptt.maxJitter	The maximum jitter in milliseconds for IP multicast voice traffic. 480 (default) 20 - 500 maxJitter The PTT/Paging interface jitter buffer maximum depth will be automatically configured to handle this level of intermittent jitter without packet loss. Actual jitter above the average but below the maximum may result in delayed audio play out while the jitter buffer adapts, but no packets will be lost. Actual jitter above the maximum value will always result in packet loss. If legacy voice.audioProfile.x.jitterBuffer.* parameters are explicitly specified, they will be used to configure the jitter buffer and these voice.rxQoS parameters will be ignored for PTT/Paging interface interfaces.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
techsupport.cfg	voice.handsfreePtt.rxdg.offset	This parameter allows a digital Rx boost for Push To Talk. 0 (default) 9 to -12 - Offsets the RxDg range of the hands-free and hands-free Push-to-Talk (PTT) by the specified number of decibels.	No
techsupport.cfg	voice.ringerPage.rxdg.offset	This parameter allows a digital Rx boost for Push To Talk. Use this parameter for handsfree paging in high noise environments. 0 (default) 9 to -12 - Raise or lower the volume of the ringer and hands-free page by the specified number of decibels.	No

Session Description Protocol (SDP) Parameters

This table describes Session Description Protocol configuration parameters.

Session Description Protocol (SDP) Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SDP.answer.useLocalPreferences	0 (default) - The phone's use of its own preference list is disabled. 1 -The phone uses its own preference list instead of the preference list in the offer when deciding which video codec to use. Note: If the H.323 call from a Polycom VVX 1500 selects a lower-quality codec (H.261) but the called device also support H.264, this parameter should be enabled to resolve the situation.	No
sip-interop.cfg	voIpProt.SDP.early.answerOrOffer	0 (default) - SDP offer or answer is not generated. 1 - SDP offer or answer is generated in a provisional reliable response and PRACK request and response. Note: An SDP offer or answer is not generated if reg.x.musicOnHold.uri is set.	No
sip-interop.cfg	voIpProt.SDP.offer.iLBC.13_33kbps.includeMode	1(default) - The phone should include the mode=30 FMTP parameter in SDP offers: If voice.codecPref.iLBC.13_33kbps is set and voice.codecPref.iLBC.15_2kbps is Null. If voice.codecPref.iLBC.13_33kbps and voice.codecPref.iLBC.15_2kbps are both set, the iLBC 13.33 kbps codec is set to a higher preference. 0 - the phone should not include the mode=30 FTMP parameter in SDP offers even if iLBC 13.33 kbps codec is being advertised.	No
sip-interop.cfg	voIpProt.SDP.useLegacyPayloadTypeNegotiation	0 (default) - RFC 3264 is followed for transmit and receive RTP payload type values. 1 - The phone transmits and receives RTP using the payload type identified by the first codec listed in the SDP of the codec negotiation answer.	No

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
sip-interop.cfg	voIpProt.SDP.offer.rtcpVideoCodecControl	This parameter determines whether or not RTCP-FB-based controls are offered in Session Description Protocol (SDP) when the phone negotiates video I-frame request methods. Even when RTCP-FB-based controls are not offered in SDP, the phone may still send and receive RTCP-FB I-frame requests during calls depending on other parameter settings. For more information about video I-frame request behavior, refer to video.forceRtcpVideoCodecControl. For an account of all parameter dependencies refer to the I-Frames section. 0 (default) - The phone does not include the SDP attribute "a=rtcp-fb". 1 - The phone includes SDP attribute "a=rtcp-fb" into offers during outbound SIP calls.	No

H.323 Protocol Parameters

The parameters listed in the next table are supported only with the Polycom VVX 500/501, 600/601, and 1500 phones.

H.323 Protocol Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
h323.cfg	voIpProt.H323.autoGateKeeperDiscovery	1 (default) - The phone will attempt to discover an H.323 gatekeeper address via the standard multi cast technique, provided that a statically configured gatekeeper address is not available. 0 - The phone will not send out any gatekeeper discovery messages.	Yes
h323.cfg	voIpProt.H323.blockFacilityOnStartH245	0 (default) - facility messages when using H.245 are not removed. 1 - facility messages when using H.245 are removed.	Yes

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
h323.cfg	voIpProt.H323.dtmfViaSignaling.enabled	1 (default) - The phone will use the H.323 signaling channel for DTMF key press transmission. 0 - The phone will not use H.323 signaling channel for DTMF key press transmission.	Yes
h323.cfg	voIpProt.H323.dtmfViaSignaling.H245alphanumericMode	1 (default) - The phone will support H.245 signaling channel alphanumeric mode DTMF transmission. 0 - The phone will not support H.245 signaling channel alphanumeric mode DTMF transmission Note: If both alphanumeric and signal modes can be used, the phone gives priority to DTMF.	Yes
h323.cfg	voIpProt.H323.dtmfViaSignaling.H245signalMode	1 (default) - The phone will support H.245 signaling channel signal mode DTMF transmission. 0 - The phone will not support H.245 signaling channel signal mode DTMF transmission.	Yes
h323.cfg	voIpProt.H323.enable	0 (default) - The H.323 protocol is not used for call routing, dial plan, DTMF, and URL dialing. 1 - The H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing.	Yes
h323.cfg	voIpProt.H323.local.port	Local port for sending and receiving H.323 signaling packets. 0 - 1720 is used for the local port but is not advertised in the H.323 signaling. 0 to 65535 - The value is used for the local port and it is advertised in the H.323 signaling.	Yes
sip-interop.cfg	voIpProt.H323.local.RAS.port	Specifies the local port value for RAS signaling. 1719 (default) 1 to 65535	Yes

Web Configuration Utility Parameters

The parameters listed specify the download location of the translated language files for the Web Configuration Utility.

Web Configuration Utility Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
site.cfg	webutility.language.plcmServerUrl	Specifies the download location of the translated language files for the Web Configuration Utility. http://downloads.polycom.com/voice/software/languages/ (default) URL	No

XML Streaming Protocol Parameters

The parameters in the following table set the XML streaming protocols for instant messaging, presence, and contact list for BroadSoft features.

XML Streaming Protocol Parameters

Template	Parameter	Permitted Values	Change Causes Restart or Reboot
features.cfg	xmpp.1.auth.domain	Specify the domain name of the XMPP server. Null (Default) Other values - UTF-8 encoded string	No
features.cfg	xmpp.1.auth.useLoginCredentials	Specifies whether or not to use the login credentials provided in the phone's Login Credentials Menu for XMPP authentication. 0 (Default) 1	No
features.cfg	xmpp.1.enable	Specifies to enable or disable the XMPP presence. 0 (Default) 1	No